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THE INTERNATIONAL TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE
(C.C.I.T.T.)

IIIrd PLENARY ASSEMBLY

GENEVA, 25 MAY — 26 JUNE 1964

BLUE BOOK

VOLUME VIII

DATA TRANSMISSION

Published by the
INTERNATIONAL TELECOMMUNICATION UNION
November 1964

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CONTENTS OF THE C.C.I.T.T. BOOKS STILL APPLICABLE FOLLOWING THE THIRD PLENARY ASSEMBLY (1964)

A. RED BOOK

Volume V — Recommendations (Series P) and Questions (Study Group XII) relative to telephone transmission performance and apparatus.

Volume V bis — Additions and amendments to Volume V following the IIIrd Plenary Assembly.

B. BLUE BOOK

Volume I
- — Minutes and reports of the IIIrd Plenary Assembly of the C.C.I.T.T.
- — Resolutions and Opinions issued by the C.C.I.T.T.
- — List of Study Groups and Working Groups for the period 1964-1968.
- — Summary Table of Questions under study in 1964-1968.
- — Recommendations (Series A) relative to the organization of the work of the C.C.I.T.T.
- — Recommendations (Series B) and Questions (Study Group VII) relative to means of expression.

Volume II
- — Recommendations (Series D) and Questions (Study Group III) relative to the lease of circuits.
- — Recommendations (Series E) and Questions (Study Group II) relative to telephone operation and tariffs.
- — Recommendations (Series F) and Questions (Study Group I) relative to telegraph operation and tariffs.

Volume III — Recommendations (Series G, H and J) and Questions (Study Groups XV, XVI and C) relative to line transmission.

Volume IV — Recommendations (Series M and N) and Questions (Study Group IV) relative to line transmission maintenance and measurements on the general telecommunication network.

Volume VI — Recommendations (Series Q) and Questions (Study Groups XI, XIII and B) relative to telephone signalling and switching.

Volume VII — Recommendations (Series R, S, T, U) and Questions (Study Groups VIII, IX, X and XIV) relative to telegraph technique.

Volume VIII — Recommendations (Series V) and Questions (Study Group A) relative to data transmission.

Volume IX
- — Recommendations (Series K) and Questions (Study Group V) relative to protection against disturbances.
- — Recommendations (Series L) and Questions (Study Group VI) relative to the protection of cable sheaths and poles.

Each volume contains extracts from contributions received dealing with the subject of the volume concerned and the interest of which is such as to warrant publication.

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RECOMMENDATION A.20

COLLABORATION WITH OTHER INTERNATIONAL ORGANIZATIONS OVER DATA TRANSMISSION

(Geneva, 1964)

The C.C.I.T.T.,

considering

that, according to Article 1 of the agreement between the United Nations and the International Telecommunication Union (Annex 6 to the International Telecommunication Convention, Geneva, 1959), the United Nations recognizes the International Telecommunication Union as the specialized agency responsible for taking such action as may be appropriate under its basic instrument for the accomplishment of the purposes set forth therein ;

that Article 4 of the International Telecommunication Convention states that the purposes of the Union are :

a) to maintain and extend international co-operation for the improvement and rational use of telecommunication of all kinds ;

b) to promote the development of technical facilities and their most efficient operation with a view to improving the efficiency of telecommunication services, increasing their usefulness and making them, so far as possible, generally available to the public ;

that Article 29 of the Convention states that, in furtherance of complete international co-ordination on matters affecting telecommunication, the Union will co-operate with international organizations having related interests and activities ;

that in the study of data transmission the C.C.I.T.T. has to collaborate with the organizations dealing with data processing and particularly the International Standardization Organization (I.S.O.) and the International Electrotechnical Commission (I.E.C.) ;

that this collaboration has to be organized in a manner which will avoid duplication of work and decisions that would be contrary to the principles set out above ;

unanimously declares the view

that international standards for data transmission should be established with the following considerations in mind :

1. clearly, it will be the responsibility of the C.C.I.T.T. to lay down standards for *transmission channels*, i.e. aspects of data transmission which require a knowledge of telecommunication networks or affect performance of these networks ;

2. the standardization of signal conversion terminal equipment (modems) is the province of the C.C.I.T.T. ; the standardization of the junction (interface) between modem and the data terminal equipment is a matter for agreement between the C.C.I.T.T. and the I.S.O. or the I.E.C. ;

3. devices designed to detect and (or) correct errors must take account of :

— the error rate tolerable to the user ;

— the line transmission conditions ;

— the code, which has to meet the exigencies of the data alphabet and the requirements of error-control (this must be such as to give an output satisfactory to the user) together with the requisite signalling (synchronism, repetition signals, etc.).

Standardization here may not come wholly within the C.C.I.T.T.'s province, but the C.C.I.T.T. has very considerable interests at stake.

4. The alphabet (Definition 31.08 in the List of Definitions) is a "table of correspondence between the written characters, together with some of the functions and the telegraph signals which represent them".

Since the I.S.O. is studying an alphabet for general (but not exclusive) use for data transmission and the C.C.I.T.T. is studying the preparation of a telegraph alphabet with greater possibilities than international alphabet No. 2, apart from a few logical functions, telegraph requirements and the needs of data transmitters coincide. The investigation under way within the Committee and the tests made by certain Administrations seem to show that adoption of a common alphabet may be possible. Hence, the C.C.I.T.T. and the I.S.O. should reach agreement on an alphabet for general use for data.

5. Coding (Definition 31.07 in the List of Definitions) is "a system of rules and conventions according to which the signals corresponding to a text should be formed, received and translated...". Hence, it consists of a transformation of the format of the signals in the alphabet for taking account of synchronous methods, and introduction of redundancy in accordance with the error-control system. This is not a field in which the C.C.I.T.T. alone may be able to decide ; however, no decision should be taken without reference to the Committee, because of the possible restrictions which transmission and switching peculiarities may impose on coding.

When the general switched network is used (telephone or telex), the error-control devices are subject to such restrictions (switching signals—reserved sequences) that it will in fact be for the C.C.I.T.T. to undertake the requisite standardizations with assistance from other bodies.

6. In all instances, the C.C.I.T.T. alone can lay down manual and automatic operating procedures for the setting-up, holding and clearing of calls for data communications when the general switched network is used, including type and form of signals to be interchanged at the interface between data-processing terminal equipment and data communication equipment.

DEFINITIONS FOR DATA TRANSMISSION

(Terms of the most frequent use in data transmission ; other definitions concerned are given in the *List of definitions of essential telecommunication terms*.)

1.

voie de transmission

Ensemble de moyens nécessaires pour assurer une transmission dans un seul sens.

Plusieurs voies de transmission peuvent emprunter un *circuit électrique* commun, comme dans les systèmes à courants porteurs ; dans ce cas, chaque voie de transmission est caractérisée par une bande particulière de fréquences qui lui est réservée.

channel

A means of one-way transmission.

Several channels may share a common path as in carrier systems ; in this case each channel is allotted a particular frequency band which is reserved to it.

2.

voie de communication

circuit (de télécommunication)

Ensemble de deux *voies de transmission* associées pour assurer, entre les mêmes points, une transmission dans les deux sens.

Les deux voies de transmission associées peuvent être symétriques (c'est-à-dire offrir aux utilisateurs les mêmes possibilités pour les deux sens de la transmission) ou être asymétriques dans le cas contraire.

Exemple de voie de communication symétrique : les deux voies conjuguées formant une voie de télégraphie harmonique normale.

Exemple de voie de communication asymétrique : pour la transmission des données, voie permettant la rapidité de 1200 bauds pour un sens de la transmission, associée à une voie ne permettant que la rapidité de 100 bauds pour l'autre sens de la transmission.

Remarque. — La voie de communication comprend les équipements de conversion de signaux (dans le cas de transmission de données).

(telecommunication) circuit

A means of both-way communication between two points, comprising associated "go" and "return" *channels*.

The two associated channels may be symmetrical (that is to say, they may offer users the same possibilities in either direction of transmission) or, on the other hand, asymmetrical.

Example of a symmetrical circuit : the two channels together making one standardized voice-frequency telegraph circuit.

Example of an asymmetrical circuit : for data transmission, a channel offering a rate of 1200 bauds in one direction, associated with a channel offering only 100 bauds in the other direction.

Note. — The circuit includes the signal conversion equipment in the case of data transmission.

3.

installation terminale pour transmission de données

Ensemble constitué par : l'équipement terminal de données, par l'équipement de conversion des signaux et, éventuellement, par l'équipement intermédiaire.

Remarque. — Dans certains cas, l'équipement terminal de données peut être directement connecté à une machine de traitement de données ou en faire partie.

terminal installation for data transmission

Installation comprising : the data terminal equipment, the signal-conversion equipment, and any intermediate equipment.

Note. — In some instances, the data terminal equipment may be connected directly to a data-processing machine or may be a part of it.

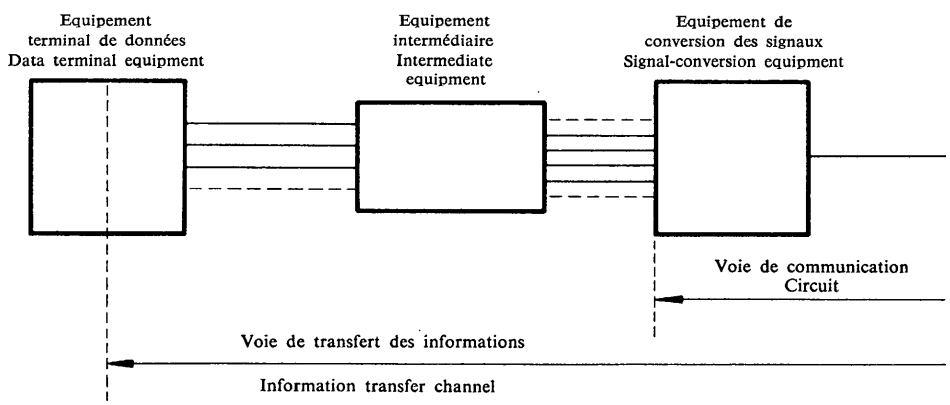


FIGURE 1

4.

source de données

Equipement qui fournit les signaux de données à transmettre ; il peut aussi recevoir des signaux de protection contre les erreurs.

data source

The equipment which supplies data signals to be transmitted ; it may also accept error-control signals.

5.

collecteur de données

Equipement qui reçoit les signaux de données après leur transmission ; il peut aussi contrôler ces signaux et produire des signaux pour la protection contre les erreurs.

data sink

The equipment which accepts data signals after transmission ; it may also check these signals and originate error-control signals.

6.

équipement terminal de données

Equipement formé par :
— la source de données, ou
— le collecteur de données, ou
— par l'ensemble de la source et du collecteur de données.

data terminal equipment

Equipment comprising :
— the data source,
— the data sink,
— or both.

7.

équipement intermédiaire

Equipement auxiliaire qui peut être intercalé entre l'équipement terminal des données et l'équipement de conversion des signaux, en vue d'assurer certaines fonctions additionnelles avant la modulation ou après la démodulation.

Remarque. — De part et d'autre d'un équipement intermédiaire tous les circuits et les signaux échangés sur ces circuits doivent être conformes aux normes établies pour les signaux de jonction.

intermediate equipment

Auxiliary equipment which may be inserted between the data terminal equipment and the signal-conversion equipment to perform certain additional functions before modulation or after demodulation.

Note. — All input and output circuits and signals of the intermediate equipment must conform to the established standards for the interface.

8.

équipement de conversion de signaux

Partie de l'installation terminale appartenant à la voie de communication de données, comprenant au moins un modulateur ou un démodulateur et assurant :

— la modulation en correspondance avec les signaux à transmettre

— et/ou la démodulation des signaux reçus.

Remarque. — L'équipement de conversion de signaux est utilisé pour la transmission des données comprenant l'information, les signaux de service, les répétitions, etc. ; il peut comprendre, par exemple, des générateurs de rythme et des régénérateurs de signaux, mais il ne comprend pas, par exemple, les équipements pour la protection contre les erreurs.

signal-conversion equipment

That part of the terminal installation belonging to the data channel, comprising at least one modulator or one demodulator and providing :

— modulation according to the signals to be transmitted

— and/or demodulation of the signals received.

Note. — The signal-conversion equipment is used for transmission of data, including information, service signals, repetition, etc. ; it may, for example, comprise clocks and signal regenerators but not error-control equipment.

9.

modem

Contraction pour « modulateur-démodulateur ».

Ce terme peut être utilisé lorsque le modulateur et le démodulateur sont associés dans un même équipement de conversion de signaux.

modem

A contraction of "modulator-demodulator".

The term may be used when the modulator and the demodulator are associated in the same signal-conversion equipment.

10.

jonction

Concept pour spécifier l'interconnexion entre deux équipements affectés à des fonctions différentes.

Cette spécification porte sur le type, le nombre et le rôle des circuits d'interconnexion ainsi que sur le type et la forme des signaux échangés sur ces circuits.

interface

A concept involving the specification of the interconnection between two equipments having different functions.

The specification includes the type, quantity and function of the interconnecting circuits and the type and form of signals to be interchanged via those circuits.

11.

mémoire

Dispositif dans lequel les données peuvent être enregistrées : elles peuvent y être conservées ; il est également possible de les y prélever le cas échéant.

storage register
store
memory (deprecated)

A device into which data can be inserted, in which they can be retained and from which they can be obtained when desired.

12.

transfert d'information

Résultat final d'une transmission de données d'un équipement terminal de données (source) à un autre équipement terminal de données (collecteur).

Exemples :

— copie à distance d'une bande magnétique ;

— reproduction à distance du contenu de cartes perforées sur une bande magnétique.

information transfer

The final result of a data transmission from a data terminal equipment (source) to another data terminal equipment (sink).

For example :
— remote reproduction of a magnetic tape ;
— remote translation from punched cards to magnetic tape.

13.

voie de transfert des informations

Ensemble des milieux et organes intervenant pour le transfert des informations, dans un sens déterminé, entre deux points extrêmes de données.

Une voie de transfert des informations comprend le modulateur et le démodulateur, l'équipement pour la protection contre les erreurs, quel que soit son emplacement, et également la voie de retour, quand il en est une.

information channel

The transmission media and intervening equipment involved in the transfer of information in a given direction between two terminals.

An information channel includes the modulator and demodulator, and any error-control equipment irrespective of its location, as well as the backward channel when provided.

14.

voie d'aller

Voie de transmission de données dont le sens de transmission coïncide avec le sens du transfert de l'information auquel elle participe.

forward channel

A data transmission channel in which the direction of transmission coincides with that in which information is being transferred.

15.

voie de retour

Voie de transmission de données utilisée pour des signaux de supervision et/ou des signaux de protection contre les erreurs et associée à la voie d'aller, mais dont le sens de transmission est opposé à celui du transfert d'information auquel elle participe.

Remarque. — Dans le cas de transferts simultanés d'information dans les deux directions, cette définition doit s'appliquer en fonction de la source de données considérée.

backward channel

A data transmission channel used for supervisory and/or error-control signals and associated with the forward channel, but with a direction of transmission opposite to that in which information is being transferred.

Note. — In case of simultaneous transfer of information in both directions, this definition applies with respect to the data source under consideration.

16.

caractère

Lettre, chiffre, signe de ponctuation ou autre, entrant dans la constitution d'un message.

En dehors de ces caractères, il peut en exister d'autres correspondant à des symboles spéciaux ou à la commande de certaines fonctions.

character

Letter, figure, number, punctuation or other sign contained in a message.

Beside such characters, there may be other characters for special symbols and for some control functions.

17.

alphabet (télégraphique ou de données)

Tableau de correspondance entre un ensemble conventionnel de caractères et les signaux qui représentent ces caractères.

alphabet (telegraph or data)

A table of correspondence between an agreed set of characters and the signals which represent them.

18.

signal de caractère

Sématème correspondant à un caractère.

character signal

A set of signal elements corresponding to a character.

19.

code (télégraphique ou de données)

Répertoire des règles et des conventions selon lesquelles doivent être formés, émis, reçus et traités les signaux télégraphiques intervenant dans un message ou les signaux de données intervenant dans les blocs.

code (telegraph or data)

A system of rules and conventions according to which the telegraph signals forming a message or the data signal forming a block should be formed, transmitted and processed.

20.

code redondant

Code utilisant un plus grand nombre d'éléments de signaux qu'il n'est nécessaire pour représenter l'information intrinsèque.

Exemple :

— Un code à cinq moments utilisant tous les caractères de l'alphabet télégraphique international n° 2 n'est pas redondant.

— Un code à cinq moments n'utilisant que les chiffres de l'alphabet télégraphique international n° 2 est redondant.

— Un code à sept moments n'utilisant que des signaux composés avec quatre éléments A et trois éléments Z est redondant.

redundant code

A code using more signal elements than necessary to represent the intrinsic information.

For example :

— A five-unit code using all the characters of International Telegraph Alphabet No. 2 is not redundant.

— A five-unit code using only the figures in International Telegraph Alphabet No. 2 is redundant.
— A seven-unit code using only signals made of four "space" and three "mark" is redundant.

21.

symbole binaire
chiffre binaire

Un des deux symboles servant à représenter des données avec une notation binaire, par exemple, 0 ou 1, A ou Z, FERMÉ ou OUVERT, etc.

binary digit

One of the two digits in the representation of data in binary notation, i.e. 0 or 1, A or Z, ON or OFF, etc.

22.

symbole n-aire
chiffre n-aire

Un des n symboles servant à représenter des données avec une notation de base n.

n-ary digit

One of the n digits in the representation of data in n-ary notation.

23.

signal de données

Signal qui représente un ensemble de symboles utilisés pour transmettre des informations et/ou des commandes de fonctions de service et qui peut comprendre des symboles de contrôle.

data signal

Signal representing a set of digits used to convey information and/or service functions and which may include check digits.

24.

bit
binon

1) Abréviation pour chiffre binaire.

2) L'unité de quantité d'information, à savoir la quantité d'information qui résulte du choix entre deux possibilités qui s'excluent l'une l'autre et qui ont chacune une probabilité de réalisation égale à ½.

Remarque. — Le terme « bit » ne doit pas être utilisé à la place d'élément unitaire, ou de moment. dans les transmissions de données bivalentes.

bit

1) An abbreviation for binary digit.

2) The unit of selective information, i.e. the amount of information derived from knowledge of the occurrence of one of two equiprobable, exclusive and exhaustive events.

Note. — The term "bit" is not to be used in two-condition data transmission instead of "unit element".

25.

bloc

Groupe de bits (binons), ou de symboles *n*-aires, transmis comme un tout auquel est généralement appliquée une méthode de codage aux fins de protection contre les erreurs.

block

A group of bits, or *n*-ary digits, transmitted as a unit over which an encoding procedure is generally applied for error-control purposes.

26.

signal de début de bloc

Signal qui identifie le début d'un bloc.

start-of-block signal
synch (Am)

Signal which identifies the start of a block.

27.

signal de fin de bloc

Signal qui identifie la fin d'un bloc.

end-of-block signal

Signal which identifies the end of a block.

28.

bits (binons) d'information

Les bits (binons) qui sont produits par la source de données et qui ne sont pas utilisés par le système de transmission de données pour la protection contre les erreurs.

information bits

Those bits which are generated by the data source and which are not used for error-control by the data transmission system.

29.

bit (binon) de contrôle

Bit associé à un caractère ou à un bloc en vue de contrôler l'absence d'erreur au sein de ce caractère ou de ce bloc.

check bit

A bit associated with a character or block for the purpose of checking the absence of error within the character or block.

30.

modulation télégraphique

Variation dans le temps d'une ou de plusieurs caractéristiques quantifiées d'une onde électromagnétique, d'un courant alternatif ou d'un courant continu suivant les signaux télégraphiques ou de données à transmettre.

telegraph modulation

Variation in time of one or more quantized characteristics of an electromagnetic wave, alternating current or direct current according to the telegraph or data signals to be transmitted.

31.

état significatif d'une modulation (télégraphique)

Etat du sémateur (ou de l'organe approprié) correspondant à chacune des valeurs quantifiées de la caractéristique (ou des caractéristiques) choisies pour former la modulation télégraphique.

Exemple :

Dans une modulation bivalente, il y a deux états significatifs, désignés en général par état A, état Z (état 0, état 1).

significant condition of a modulation

Condition assumed by the appropriate device corresponding to the quantized value (or values) of the characteristic (or characteristics) chosen to form the telegraph modulation.

Example :

In a two-condition modulation (or binary modulation) there are two significant conditions, generally designated A and Z (or 0 and 1).

32.

valence (d'une modulation ou d'une restitution)

Nombre des états significatifs distincts employés dans une modulation (ou dans une restitution) pour caractériser les éléments de signal à transmettre ou reçus.

number of significant conditions (of a modulation or a restitution)

Number of distinct significant conditions employed in a modulation (or restitution) to characterize the signal elements to be transmitted or received.

33.

instants significatifs

Instants auxquels débutent les états significatifs pris successivement par le sémateur.

Chacun d'eux est déterminé dès que le sémateur prend l'état significatif utilisable pour un enregistrement ou un traitement ultérieur.

significant instants

The instants at which the successive significant conditions recognized by the appropriate device of the modulation begin.

Each of these instants is determined as soon as the appropriate device takes up the significant condition usable for a recording or a processing.

34.

intervalle significatif

Intervalle de temps compris entre deux instants significatifs successifs.

significant interval

Time interval between two consecutive significant instants.

35.

intervalle unitaire
[intervalle élémentaire]

Intervalle de temps tel que les durées théoriques des intervalles significatifs d'une modulation ou d'une restitution télégraphique sont toutes des multiples entiers de cet intervalle.

unit interval
signal element (Am)

The interval of time such that the theoretical durations of the significant intervals of a telegraph modulation (or restitution) are whole multiples of this interval.

36.

intervalle minimum

Durée de l'intervalle significatif le plus court dans un système utilisant un code tel que les durées théoriques des intervalles significatifs ne sont pas toutes des multiples d'un intervalle unitaire.

minimum interval

The duration of the shortest significant interval in a system using a code such that the theoretical durations of the significant intervals are not all multiples of a unit interval.

37.

moment
élément unitaire

Elément de signal de caractère de durée égale à l'intervalle unitaire.

unit element

Character signal element having a duration equal to the unit interval.

38.

rapidité de modulation
[vitesse télégraphique]

Inverse de l'intervalle unitaire évalué en secondes. (Cette rapidité s'exprime en *bauds*.)

modulation rate
[telegraph speed]

Reciprocal of the unit interval measured in seconds. (This rate is expressed in *bauds*.)

39.

baud

Unité de rapidité de modulation. Elle correspond à une rapidité d'un intervalle unitaire par seconde.

Exemple : Si la durée de l'intervalle unitaire est de 20 millisecondes, la rapidité de modulation est de 50 bauds.

baud

The unit of modulation rate. It corresponds to a rate of one unit interval per second.

Example : If the duration of the unit interval is 20 milliseconds, the modulation rate is 50 bauds.

40.

transmission synchrone
transmission isochrone

Procédé de transmission tel que, entre deux instants significatifs quelconques, il y a toujours un nombre entier d'intervalles unitaires.

synchronous transmission

A transmission process such that between any two significant instants there is always an integral number of unit intervals.

41.

transmission asynchrone

Procédé de transmission tel que, entre deux instants significatifs d'un même groupe*, il y a toujours un nombre entier d'intervalles unitaires. Entre deux instants significatifs placés dans des groupes différents, il n'y a pas toujours un nombre entier d'intervalles unitaires.

* En transmission de données, ce groupe est un bloc ou un caractère ; en télégraphie, ce groupe est un caractère.

asynchronous transmission

A transmission process such that between any two significant instants in the same group*, there is always an integral number of unit intervals. Between two significant instants located in different groups, there is not always an integral number of unit intervals.

* In data transmission this group is a block or a character ; in telegraphy this group is a character.

42.

transmission arythmique

Transmission asynchrone où chaque groupe d'éléments de code correspondant à un signal de caractère est précédé d'un signal de départ qui sert à préparer l'appareil récepteur pour la réception et l'enregistrement d'un caractère, et est suivi d'un signal d'arrêt qui sert à mettre au repos l'appareil récepteur pour le préparer à recevoir le caractère suivant.

start-stop transmission

Asynchronous transmission in which each group of code elements corresponding to a character signal is preceded by a start signal which serves to prepare the receiving mechanism for the reception and registration of a character, and is followed by a stop signal which serves to bring the receiving mechanism to rest in preparation for the reception of the next character.

43.

transmission série

Transmission en des intervalles de temps successifs des éléments qui composent un même signal télégraphique ou de données.

Remarque. — Les éléments peuvent être transmis les uns à la suite des autres (sans interruption) ou être séparés, pourvu qu'ils ne soient pas transmis simultanément.

Exemples :

— transmission des éléments de signaux selon l'alphabet télégraphique international n° 2 au moyen d'un téléimprimeur normal ;
— transmission télégraphique par une voie dérivée en temps.

serial transmission

Transmission at successive intervals of signal elements constituting the same telegraph or data signal.

Note. — The sequential elements may be transmitted with or without interruption, provided that they are not transmitted simultaneously.

For example :

— transmission of signal elements by a standard teleprinter, in accordance with International Telegraph Alphabet No. 2 ;
— telegraph transmission by a time divided channel.

44.

transmission parallèle

Transmission simultanée d'un certain nombre d'éléments de signal composant un même signal télégraphique ou de données.

Exemple : Utilisation d'un code selon lequel chaque signal est caractérisé par une combinaison de trois fréquences choisies parmi 12 que l'on émet simultanément dans la voie.

parallel transmission

The simultaneous transmission of a certain number of signal elements constituting the same telegraph or data signal.

For example : Use of a code according to which each signal is characterized by a combination of three out of 12 frequencies simultaneously transmitted over the channel.

45.

débit binaire

Dans un système de communication où le nombre de voies en parallèle est m, le débit binaire est la somme définie par

$$\sum_{i=1}^{i=m} \frac{1}{T_i} \log_2 n_i$$

T_i étant la durée de l'intervalle minimum pour la voie i (exprimée en secondes) et n_i étant le nombre d'états significatifs de la modulation sur la voie i.

(Definitions)

Le débit binaire s'exprime en bits par seconde.

Remarque 1. — Pour une voie unique (transmission série) le débit binaire est $\frac{1}{T} \log_2 n$; si la modulation est bivalente ($n = 2$) le débit binaire est $\frac{1}{T}$.

Remarque 2. — Pour une transmission parallèle où le nombre d'états significatifs et l'intervalle minimum sont les mêmes sur chaque voie le débit binaire est $m \frac{1}{T} \log_2 n$ $\left(\frac{m}{T}$ en cas de modulation bivalente$\right)$.

data signalling rate

It is given by

$$\sum_{i=1}^{i=m} \frac{1}{T_i} \log_2 n_i$$

where m is the number of parallel channels, T_i is the minimum interval for the ith channel expressed in seconds, n_i is the number of significant conditions of the modulation in the ith channel.

Data signalling rate is expressed in bits per second.

Note 1. — For a single channel (serial transmission) it reduces to $\frac{1}{T} \log_2 n$; with a two-condition modulation ($n = 2$), it is $\frac{1}{T}$.

Note 2. — For a parallel transmission with equal minimum intervals and equal number of significant conditions on each channel, it is $m \frac{1}{T} \log_2 n$ $\left(\frac{m}{T}$ in case of a two-condition modulation$\right)$.

46.

rapidité de transfert de données

Nombre moyen de bits, caractères ou blocs transférés dans un système de transmission de données entre deux équipements correspondants pendant l'unité de temps.

Elle s'exprime en bits, caractères ou blocs par seconde (minute ou heure).

Remarque. — Les équipements correspondants doivent être précisés : modems ou équipements intermédiaires ou source et collecteur.

data transfer rate

The average number of bits, characters or blocks per unit time passing between corresponding equipments in a data transmission system.

It is expressed in terms of bits, characters or blocks per second, minute or hour.

Note. — Corresponding equipments should be indicated : modems or intermediate equipments or source and sink.

47.

bit (binon) erroné

Bit (binon) qui n'est pas conforme à celui qui aurait dû être reçu.

erroneous bit

A bit which is not in accordance with that which should have been received.

48.

erreur simple

Bit (binon) erroné, précédé et suivi d'au moins un bit (binon) non erroné.

single error

An erroneous bit, preceded and followed by at least one correct bit.

49.

erreur double, triple, ...

Groupe de deux, trois,...bits (binons) erronés consécutifs, précédé et suivi d'au moins un bit (binon) non erroné.

double, triple, ... error

A group of two, three,...consecutive erroneous bits, preceded and followed by at least one correct bit.

50.

paquet d'erreurs

Une série de bits (binons) dans laquelle le nombre de bits (binons) corrects séparant deux bits (binons) erronés successifs est toujours inférieur à un nombre donné (X).

(Definitions)

Remarque. — Le dernier bit (binon) erroné d'un paquet et le premier bit (binon) erroné du paquet suivant sont donc séparés par X bits (binons) non erronés ou plus.
Il convient de préciser le nombre X en décrivant un paquet d'erreurs.

error burst

A group of bits in which two successive erroneous bits are always separated by less than a given number (X) of correct bits.

Note. — The last erroneous bit in a burst and the first erroneous bit in the following burst are accordingly separated by X correct bits or more.
The number X should be specified when describing an error burst.

51.

bloc erroné

Bloc présentant un ou plusieurs bits erronés.

erroneous block

A block in which there are one or more erroneous bits.

52.

taux d'erreurs sur les bits (binons)
taux d'erreurs sur les éléments
taux d'erreurs sur les caractères
taux d'erreurs sur les blocs

Rapport du nombre de bits (binons), éléments, caractères ou blocs reçus de façon erronée au nombre de bits, éléments, caractères ou blocs émis.

bit error rate
element error rate
character error rate
block error rate

The ratio of the number of bits, elements, characters of blocks incorrectly received to the total number of bits, elements, characters, or blocks sent.

53.

contrôle par caractère

Système de protection contre les erreurs basé sur la vérification de certaines règles prédéterminées de composition des caractères.

character check

A system of error-control based on the check that some preset rules for the formation of characters are observed.

54.

contrôle par bloc

Système de protection contre les erreurs basé sur la vérification de certaines règles prédéterminées de composition des blocs.

block check

A system of error-control based on the check that some preset rules for the formation of blocks are observed.

55.

code détecteur d'erreurs

Code selon lequel chaque signal télégraphique ou de données est conforme à des règles précises de composition telles que tout écart de ces règles dans les signaux reçus peut être automatiquement détecté. Un tel code exige un plus grand nombre d'éléments de signal qu'il n'est nécessaire pour transmettre l'information elle-même.

error-detecting code

A code in which each telegraph or data signal conforms to specific rules of construction, so that departures from this construction in the received signals can be automatically detected. Such codes require more signal elements than are necessary to convey the fundamental information.

56.

système détecteur d'erreurs sans répétition

Système utilisant un code détecteur d'erreurs ou un détecteur de la qualité d'un signal de données et conçu de telle sorte que tout signal reconnu erroné est, ou bien :

a) supprimé des données transmises au collecteur de données dans certains cas avec l'indication qu'une telle suppression a eu lieu, ou bien
b) transmis au collecteur de données en même temps qu'une indication que ce signal a été reconnu comme erroné.

error-detecting system

A system employing an error-detecting code or a data signal quality detector and so arranged that any signal detected as being in error is :

a) either deleted from the data delivered to the data sink, in some cases with an indication that such deletion has taken place,
b) or delivered to the data sink together with an indication that it has been detected as being in error.

57.

RQ automatique
système détecteur d'erreurs avec demande de répétition

Système utilisant un code détecteur d'erreurs et conçu de telle sorte qu'un signal reconnu erroné provoque automatiquement une demande de répétition de ce signal.

error detecting and feedback system
decision feedback system
request repeat system
ARQ system

A system employing an error-detecting code and so arranged that a signal detected as being in error automatically initiates a request for retransmission of the signal detected as being in error.

58.

code correcteur d'erreurs

Code selon lequel chaque signal télégraphique ou de données est conforme à des règles précises de composition telles que tout écart de ces règles dans les signaux reçus peut être automatiquement détecté et permet la correction automatique à l'extrémité réceptrice de certaines ou de toutes les erreurs.

Un tel code exige un plus grand nombre d'éléments de signal qu'il n'est nécessaire pour transmettre l'information elle-même.

error-correcting code

A code in which each telegraph or data signal conforms to specific rules of construction so that departures from this construction in the received signals can be automatically detected, and permits the automatic correction, at the receiving terminal, of some or all of the errors.

Such codes require more signal elements than are necessary to convey the basic information.

59.

système correcteur d'erreurs

Système utilisant un code correcteur d'erreurs et conçu de telle sorte que, parmi les signaux reconnus erronés, certains ou tous sont automatiquement corrigés à l'extrémité réceptrice avant d'être envoyés au collecteur de données ou au récepteur télégraphique.

error-correcting system

A system employing an error-correcting code and so arranged that some or all signals detected as being in error are automatically corrected at the receiving terminal before delivery to the data sink or to the telegraph receiver.

60.

taux d'erreurs résiduelles

Rapport du nombre des bits, éléments unitaires, caractères, ou blocs incorrectement reçus mais non détectés ou non corrigés par l'équipement de protection contre les erreurs, au nombre total des bits (binons), éléments unitaires, caractères, ou blocs émis.

residual error-rate
undetected error-rate

The ratio of the number of bits, unit elements, characters, blocks incorrectly received but undetected or uncorrected by the error-control equipment, to the total number of bits, unit elements, characters, blocks sent.

61.

contrôle par retour de l'information

Méthode visant à contrôler l'exactitude de la transmission des données selon laquelle les données reçues sont retournées vers l'extrémité émettrice afin de les comparer avec les données originales conservées en mémoire dans ce but.

loop checking
message feedback
information feedback

A method of checking the accuracy of transmission of data in which the received data are returned to the sending end for comparison with the original data, which are stored there for this purpose.

62.

système correcteur d'erreurs par retour de l'information

Système utilisant le contrôle par retour de l'information avec répétition du groupe erroné par l'extrémité émettrice.

information feedback system

An error-control system using message feedback with repetition of the erroneous group from the sending station.

63.

contrôle de parité (ou d'imparité)

Contrôle vérifiant si, dans un groupe de chiffres binaires, le nombre des symboles « 1 » est pair (contrôle de parité) ou impair (contrôle d'imparité).

Remarque. — Dans certains cas ce contrôle peut porter sur le nombre des symboles « 0 ».

even parity check (odd parity check)

Check which tests whether the number of digits in a group of binary digits is even (even parity check) or odd (odd parity check).

Note. — In some cases the control may be done on the number of digits "0".

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SERIES V RECOMMENDATIONS

Recommendations concerning data transmission

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RECOMMENDATION V.1

EQUIVALENCE BETWEEN BINARY NOTATION SYMBOLS AND THE SIGNIFICANT CONDITIONS OF A TWO-CONDITION CODE

(New Delhi, 1960, amended at Geneva, 1964)

Binary numbering expresses numbers by means of two digits normally represented by the symbols 0 and 1. Transmission channels are especially well suited to the transmission of signals by a modulation (or a semation) having two significant conditions (two-condition modulation). These two significant conditions are sometimes called "space" and "mark" or "start" and "stop", or they may be called condition A or condition Z. [1]

It is very useful to make the two conditions of a two-condition modulation correspond to the binary digits 0 and 1. Such equivalence will facilitate the transmission of numbers resulting from binary calculation, the conversion of codes for binary numbers and of codes for decimal numbers, maintenance operations and relations between transmission personnel and the personnel in charge of data-processing machines.

At first sight, it does not seem to matter whether the symbol 0 corresponds in transmission to condition A or condition Z, the symbol 1 then corresponding to condition Z or condition A or vice versa.

In telegraphy however, when a telegraphic communication is set up and the sending of signals is stopped (called the idle condition of the line), the signal sent over the line consists of condition Z throughout the suspension of transmission.

It is logical (and for certain V.F. telegraph systems also essential) to use the same rule in data transmission. During the "idle periods" of transmission, condition Z should be applied to the circuit input.

Data transmission on a circuit is often controlled by perforated tape. On perforated tapes used for telegraphy, condition Z is represented by perforation. When binary numbers are represented by means of perforations, it is customary to represent the symbol 1 by a perforation. It is therefore logical to make this symbol 1 correspond to condition Z.

For these reasons, the C.C.I.T.T.

unanimously declares the following view :

1. In transmitting data by two-condition code, in which the digits are formed using binary notation, the symbol 1 of the binary notation will be equivalent to condition Z of the modulation, and the symbol 0 of the binary notation will be equivalent to condition A of the modulation.

2. During periods when there is no signal sent to the input of the circuit, the circuit input condition is condition Z.

3. If perforation is used, one perforation corresponds to one unit interval under condition Z.

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[1] Definitions of condition A and condition Z : *List of Definitions*—Part I—No. 31.38.

4. In accordance with C.C.I.T.T. Recommendation R.31 (*Blue Book*, Volume VII), the sending of symbol 1 (condition Z) corresponds to the tone being sent on a channel using amplitude modulation.

5. In accordance with C.C.I.T.T. Recommendation R.35 (*Blue Book*, Volume VII), when frequency modulation is used, the sending of symbol 0 corresponds to the higher frequency, while the sending of symbol 1 corresponds to the lower frequency.

6. a) for phase modulation with reference phase :

the symbol 1 corresponds to a phase equal to the reference phase,
the symbol 0 corresponds to a phase opposed to the reference phase.

b) for differential phase modulation :

the symbol 1 corresponds to a no-phase inversion from previous element,
the symbol 0 corresponds to a phase inversion from previous element.

Summary table of equivalence

	Digit 0 "Start" signal in start-stop code Line available condition in telex switching "Space" element of start-stop code Condition A	Digit 1 "Stop" signal in start-stop code Line idle condition in telex switching "Mark" element of start-stop code Condition Z
Amplitude modulation	Tone-off	Tone-on
Frequency modulation	High frequency	Low frequency
Phase modulation with reference phase	Opposite phase to the reference phase	Reference phase
Differential phase modulation	Inversion of the phase	No-phase inversion
Perforations	No perforation	Perforation

Note. — The standardization described in this Recommendation is general and applies to any two-condition transmission, whether over telegraph-type circuits or over circuits of the telephone type, making use of electromechanical or electronic devices.

RECOMMENDATION V.2

POWER LEVELS FOR DATA TRANSMISSION OVER TELEPHONE LINES

(New Delhi, 1960, amended at Geneva, 1964)
(Recommendation H.41, Volume III of the Blue Book*)*

The objectives in specifying data signal levels are as follows :

a) to ensure satisfactory transmission and to permit co-ordination with devices such as signalling receivers or echo suppressors, the data signal levels on international circuits should be controlled as closely as possible ;

b) to ensure correct performance of multichannel carrier systems from the point of view of loading and noise, the mean power of data circuits should not differ much from the conventional value of channel loading (—15 dbm0 for each direction of transmission : see note below). This conventional value makes allowance for a reasonable proportion (under 5%) of the channels in a multichannel system being used for non-speech applications at fixed power levels at about —10 dbm0 simultaneously in both directions of transmission.
If the proportion of non-speech applications (including data) does not exceed the above figure of 5% then the mean power of —10 dbm0 simultaneously in both directions of transmission would be allowable for data transmission also.
However, assuming an appreciably higher (e.g. 10 to 20%) proportion of non-speech circuits (due to the development of data transmission) on an international carrier system, a reduction of this power by 3 db might be reasonable. In this way the sum of the mean powers in both directions of transmission in a duplex (i.e. transmitting tones in both directions simultaneously) system would be —10 dbm0 (i.e. —13 dbm0 for each direction). The power transmitted on the channel of a simplex (i.e. transmitting in one direction only) system or on either channel of a half duplex (i.e. transmitting in opposite directions consecutively) would be —10 dbm0 (assuming that there were no echoes) ;

Note. — The distribution of long-term mean power among the channels in a multichannel carrier telephone system (conventional mean value : of —15 dbm0), probably has a standard deviation in the neighbourhood of 4 db (*Blue Book*, Volume III, Annex 6).

c) it is probable that Administrations will wish to fix specific values for the signal power level of data modulators either at the subscriber's line terminals or at the local exchanges. The relation between these values and the power levels on international circuits depends on the particular national transmission plan ; in any case, a wide range of losses among the possible connections between the subscriber and the input to international circuits must be expected ;

d) considerations a) to c) suggest that specification of the maximum data signal level only is not the most useful form. One alternative proposal would be to specify the nominal power at the input to the international circuit. The nominal power would be the statistically estimated mean power obtained from measurement on many data transmission circuits.

For these reasons, the C.C.I.T.T.

unanimously declares the following view :

A. Data transmission over leased telephone circuits (private wires) set up on carrier systems

1. The maximum power output of the subscriber's apparatus into the line shall not exceed 1 mW.

2. For systems transmitting tones continuously, for example frequency modulation systems, the maximum power level at the zero relative level point shall be −1.15 Nm0 (−10 dbm0). When transmission of data is discontinued for any appreciable time, the power level should preferably be reduced to −2.3 Nm0 (−20 dbm0) or lower.

3. For systems not transmitting tones continuously, for example amplitude-modulation systems, higher levels up to −0.69 Nm0 (−6 dbm0) at the zero relative level point may be used provided that the sum of the mean powers during the busy hour on both directions of transmission does not exceed 64 microwatts (corresponding to a mean level, of −1.73 Nm0 (−15 dbm0) on each direction of transmission simultaneously). Further the level of any tones above 2400 c/s should not be so high as to cause interference on adjacent channels on carrier-telephone systems.

Note 1. — In suggesting these limits, the C.C.I.T.T. has in mind that the recommended maximum level of −5 db referred to the zero relative level point for private circuits for alternate telephony and telegraphy may no longer be acceptable having regard to the recommendation that "to avoid overloading carrier systems, the mean power should be limited to 32 μW if such systems are subject to considerable extension".

Note 2. — The proposed limit of −10 db for continuous tone systems is in line with the existing Recommendation H.31 (T.11) of the C.C.I.T.T. for frequency-modulation-phototelegraph transmissions.

Note 3. — It is not possible to give any firm estimate of the proportion of international circuits which will at any time be carrying data transmissions. If the proportion should reach a high level, the provisional limits now proposed would need to be reconsidered.

B. Data transmission over the switched telephone system

The maximum power output of the subscriber's equipment into the line shall not exceed 1 mW at any frequency.

In systems continuously transmitting tone, such as frequency or phase-modulation systems, the power level of the subscriber's equipment should be adjusted to make allowance for loss between his equipment and the point of entry to an international circuit ; so that

the corresponding nominal level of the signal at the international circuit input shall not exceed −1.15 Nm0 (−10 dbm0) (simplex systems) or −1.5 Nm0 (−13 dbm0) (duplex systems).

In systems not transmitting tones continuously, such as amplitude-modulation systems and multifrequency systems, higher levels may be used, provided always that the mean power of all the signals at the international circuit input during any one hour in both directions of transmission does not exceed 64 microwatts (representing a mean level of 1.73 Nm0 (−15 dbm0) in each direction of transmission simultaneously).

Furthermore, the frequency level in carrier telephone systems which are part of a circuit should not be so high that it might cause interference in adjacent channels. Recommendation G.224 (*Blue Book*, Volume III) could be referred to with a view to providing adequate levels.

Note 1. — In practice, it is no easy matter to assess the loss between a subscriber's equipment and the international circuit, so that this part of the present Recommendation should be taken as providing general planning guidance. As mean level at the international circuit input, the mean figure obtained from measurement or calculation (on numerous transmission data) may be adopted.

Note 2. — In switched connections the loss between subscribers' telephones may be high : 30 to 40 db. The level of the signals received will then be very low, and these signals may suffer disturbance from the dialling pulses sent over other circuits. Hence the transmission level should be as high as possible.

If there is likely to be a heavy demand for data-transmission international connections over the switched network, some Administrations might want to provide special four-wire subscribers' lines. If so, the levels to be used might be those proposed for leased circuits.

RECOMMENDATION V.10

USE OF THE TELEX NETWORK FOR DATA TRANSMISSION AT THE MODULATION RATE OF 50 BAUDS

(Geneva, 1964)

The telex network is well adapted for the economical transmission of data at fairly slow speeds, for the equipment required for binary transmission of data by telex stations, over and above the normal equipment, is relatively simple.

But some limits have to be imposed on data transmission codes used in the telex network because of :

— the need to make sure that telex calls will not be abruptly released ;

— exaggerated distortion which may be introduced by amplitude-modulation voice-frequency telegraph systems when an excessively long-duration "start (A)" modulation element appears in a signal ;

— the fact that in some networks there is regenerative repetition of start-stop signals which can be handled only as if they were constructed like five-unit start-stop information signals ;

— the possibility that certain long-distance calls are established over synchronous systems which can handle only five-unit start-stop signals.

The limitation due to regenerative repeaters and synchronous systems imposes the use of a five-unit start-stop code for information hence the first part of the Recommendation (dealing with the most general procedure) ; this deals with data transmission with a 5-unit code on start-stop systems.

But in certain circumstances alphabets with more than five units can be used for data transmission, whence the second part of the Recommendation.

For these reasons, the C.C.I.T.T.

unanimously declares the following view :

I. Part I — Data transmission with a five-unit code on start-stop systems

(See note)

Telex calls for data transmission may be set up in the international telex network, subject to the following provisions :

I.a) The call shall be set up between the caller and the called subscriber in accordance with the procedure recommended by the C.C.I.T.T. for the setting-up of a telex call and its supervision by exchange of answer-back codes (C.C.I.T.T. Recommendations F.60 —Volume II and U.1—Volume VII, *Blue Book*).

I.b) When one of the subscribers concerned wishes to introduce data transmission equipment into the connection, he shall transmit the sequence SSSS (or "") of signals 19 from Alphabet No. 2 (signal for transfer to data).

Upon reception of this sequence of signals, the data transmission or reception equipment, as the case may be, shall be connected to the line. This change-over to the data position may be effected :

a) manually at both terminals,

b) automatically at both terminals,

c) manually at one terminal and automatically at the other.

In case a), in order to avoid any misunderstanding between the stations concerned, it is recommended that the operator at the station which will transmit data, before sending the sequence of four combinations No. 19, checks that the receiving station is :

i) attended

ii) ready to change-over to data transmission.

I.c) The sequence of four combinations No. 19 will make ineffective, where necessary :

— devices which might conceivably emit signals disturbing to data transmissions, in particular, the answer-back or possibly the delay signal used in connection with error-correcting synchronous radio systems (Recommendation U.22—*Blue Book*, Volume VII).

— devices which might be falsely operated by data signals, such as devices for operator-recall (Recommendation U.21—*Blue Book*, Volume VII).

I.d) Data transmission should be made by means of a start-stop code formed according to the structure of Alphabet No. 2. Users should be left free to decide how combinations should be allocated to the various components of the alphabet (of course, Alphabet No. 2 may itself be used).

I.e) When error-control is necessary, one of the following methods of error-control may be used :

— return of information to the transmitting station (information feedback system) ;

— block transmission with check characters at the end of the block ;

— character-by-character transmission with check bits (in the case of five-unit signals with redundancy).

I.f) Unless the exception stated in paragraph g) is employed at the end of the data transmission, the telex clearing signal described in Recommendation U.1 shall be emitted ; this will cause the call to be cleared down and the terminal equipment to return to the telex position, and will cause the devices which might have been rendered inoperative on certain special circuits (see I.c) to go back to normal. This clearing signal must set off the clear-confirmation (see Recommendation U.1).

Note. — Users may expect that some combinations No. 32, possibly followed by other combinations, may be received before the connection is cleared.

I.g) Should the data user feel that, after the data transmission, it might be well to go back to teleprinter working in order to have a telex connection instead of sending the clearing signal as mentioned in I.f), he may use an appropriate signal to control the return of the connection to teleprinter operation. This return should be accompanied by rendering the answer-back device effective again. Clearing shall take place after the end of the telex call, as in telex working. Be it observed, however, that during the teleprinter working following data transmission, facilities of delay signal and of operator-recall remain inoperative until the clearing of the call.

I.h) The signals transmitted by the data transmission devices must meet the exigencies set forth in items 1, 2, 3 and 5 of Recommendation S.3 (*Blue Book*, Volume VII). The receiving equipment of the data-reception devices must meet the exigencies set forth in items 1, 2, 4 and 6 of Recommendation S.3.

II. Part II — Data transmission with codes different from the start-stop code of International Alphabet No. 2

(See note at the end)

Between telex networks which can take signals different from those of the five-unit start-stop code (that is to say, when telex calls between such networks do not call for regenerative repeaters, or for certain synchronous systems which would clash with them), by agreement between the Administrations concerned, data transmission with data transmission alphabets using these signals may be made, subject to the following :

II.a) Application of the procedure described under I.a).

II.b) Application of the procedure described under I.b).

II.c) Application of the procedure described under I.c).

II.d) Use of a code with a modulation rate of 50 bauds should avoid composition of signals having more than seven consecutive elements of start polarity.

Note. — This limit is imposed to avoid clearing the connection unexpectedly in the exchanges as well as not to introduce excessive distortion on AM voice-frequency telegraph channels.

Data may be transmitted by start-stop, or isochronously.

II.e) When error control is necessary, one of the following methods of error control may be used :

— return of information to the transmission station (information feedback system) ;

— block transmission with check characters at the end of the block ;

— character-by-character protection by means of a parity check or a constant ratio code, for example the seven-unit code standardized in Recommendation S.13 (*Blue Book*, Volume VII).

In all cases restrictions given in the above paragraph II.d) should be taken into consideration.

II.f) Application of the procedure described under I.f).

II.g) Application of the procedure described under I.g.).

Note. — Telex data transmission calls may be set up in the international telex network, subject to the provisions in Part 1 for the purpose of transmitting messages with alphabets having more than five units. Such a service can be achieved by regrouping the alphabets into a five-unit start-stop form. The regrouping would require complementary converters at the sending and receiving terminals.

RECOMMENDATION V.21

200-BAUD DATA TRANSMISSION ON TELEPHONE CIRCUITS

(Geneva, 1964)

A system of data transmission at a low modulation rate, such that data could be transmitted over a telephone circuit operated alternatively for telephone calls and data transmissions, using simple input/output equipment and easy operating procedures, would be economical.

The modulation rate must be such as to allow the use of current type of data sources and sinks, especially electromechanical devices.

The system for data transmission will be duplex, either for simultaneous two-way data transmission, or for the transmission of signals sent in the backward direction for error-control purposes. The transmission must be such that use can be made of normal telephone circuits, and this applies both to the bandwidth available and to the restrictions imposed by signalling in the telephone networks.

The two correspondents are brought into contact by a telephone call, and the circuit is put into the data-transmission position manually, by agreement between the operators. The equipment, designed for use on connections set up by switching in the general public network, can obviously be used on leased lines.

For these reasons, the C.C.I.T.T.

unanimously declares the following view :

1. Data transmission may take place at low modulation rates on telephone calls set up on switched telephone circuits (or on leased telephone circuits).

The user himself undertakes the manual transition from telephony to data transmission, and vice versa [1].

2. The communication circuit for data transmission is a duplex circuit whereby data transmission in both directions simultaneously is possible at 200 bauds or less.

The modulation is a binary modulation obtained by frequency shift.

3. For channel No. 1, the nominal mean frequency is 1080 c/s.

For channel No. 2, it is 1750 c/s.

The frequency deviation is $\pm$ 100 c/s. In each channel, the higher characteristic frequency corresponds to the symbol 0.

The characteristic frequencies [2] as measured at the modulator output must not differ by more than $\pm$ 6 c/s from the nominal figures.

A maximum drift of frequency of $\pm$ 6 c/s is assumed for the line. Hence the demodulation equipment must tolerate drifts of $\pm$ 12 c/s between the frequencies received and their nominal values.

4. Data may be transmitted by synchronous or asynchronous procedures. With synchronous transmission, the modem will not have to provide the signals which would be necessary to maintain synchronism when transmission is not proceeding.

5. There is no urgent requirement to equip the modem for transmission of signals designed to disable the echo suppressors. It will be for the user to decide whether, in view of the connections he makes with this system, he will have to request that the modem be equipped with facilities for disabling echo suppressors. The international characteristics of the disabling signal for echo suppressors are still under study; with agreement of Administrations or recognized private operating Agencies concerned the disabling signal may provisionally have the following characteristics :

transmitted frequency = 2100 $\pm$ 15 c/s

transmission level = that of a characteristic frequency for data transmission (see under 6)

the disabling tone to last at least 300 ms

the tolerable interruptions by the data signal to last not more than 100 ms (after an interruption in frequency transmission of more than 100 ms, the disabling signal will have to be sent again).

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[1] Automatic transfer to data transmission, and vice versa, is the subject of an investigation (see Question 1/A—Y).

[2] Table showing the nominal characteristic frequencies.

Note : Channel No. 1 (F_A = 1180 c/s and F_Z = 980 c/s)
Channel No. 2 (F_A = 1850 c/s and F_Z = 1650 c/s).

6. The maximum power output of the subscriber's equipment into the line shall not exceed 1 mW at any frequency.

The power level of the subscriber's equipment should be adjusted to make allowance for loss between his equipment and the point of entry to an international circuit; so that the corresponding nominal level of the signal at the international circuit input shall not exceed −1.5 Nm0 (−13 dbm0). (See Recommendation V.2, Section B).

7. a) When both channels are used for simultaneous both-way data transmission, channel No. 1 is used for transmission of the caller's data (i.e. the person making the telephone call) towards the called station, while channel No. 2 is used for transmission in the other direction.

b) When one channel is used for data transmission and the other is used for transmission of check signals, service signals, etc., only, it is channel No. 1 which is used for transmission from the calling to the called station regardless of the direction in which the data is transmitted.

c) The procedure for the assignment of the channels described under a) and b) above applies in the case of the general service of data transmission, making it possible to transmit data or check signal, service signal etc. bilaterally between any two subscribers. In special cases which do not come under this rule, the procedure of assignment of the channels is determined by the prior agreement between the correspondents, bearing in mind the requirement proper to each service.

8. When the two stations are in a position to transmit data, any interruption of frequency for more than 100 ms in a channel sets off an alarm at the station situated at the receiving end.

Thus:

— simultaneous interruption of the two channels for more than 100 ms sets off the alarm in both stations;

— one station can inform the other that it is no longer able to receive its signals (it suffices if it stops sending the frequency for more than 100 ms over the channel for which it acts as the transmitter station).

RECOMMENDATION V.22

STANDARDIZATION OF MODULATION RATES AND DATA-SIGNALLING RATES FOR SYNCHRONOUS DATA TRANSMISSION IN THE GENERAL SWITCHED TELEPHONE NETWORK [1]

(Geneva, 1964)

1. Data transmission by international communications carried on the general switched telephone network, using a synchronous transmission procedure (see Note 3) will be done with two-condition modulation and serial transmission.

[1] See Notes 1 and 2.

2. The modulation rates on the data transmission channel will be :

1200 or 600 bauds.

The user will choose between the two rates, in accordance with the facilities afforded by the communication.

3. The data-signalling rate (see Note 4) at the interface of a modem will be 1200 bits per second (when the modulation rate is 1200 bauds in the transmission channel) or 600 bits per second (when the modulation rate is 600 bauds). This is on the assumption that transmission on the interface of the modem is done in serial, with two-condition modulation, and the modem does not itself add or subtract any other bits.

4. Provisionally, the tolerances in modulation rate and data-signalling rate shall be

at 1200 bauds (or 1200 bits per second) : $\pm$ 0.5%,

at 600 bauds (or 600 bits per second) : $\pm$ 0.5%.

Note 1. — For data transmission at 200 bauds, see Recommendation V.21.

Note 2. — Questions concerning the parallel transmission of multi-condition serial modulations and their modulation rates are still being studied (see Question 1/A—Q).

Note 3. — Synchronous transmission
A transmission process such that between any two significant instants there is always an integral number of unit intervals.

Note 4. — Data-signalling rate
It is given by

$$\sum_{i=1}^{i=m} \frac{1}{T_i} \log_2 n_i$$

where m is the number of parallel channels, T_i is the minimum interval for the i th channel expressed in seconds, n_i is the number of significant conditions of the modulation in the i th channel.
Data signalling rate is expressed in bits per second :

a) For a single channel (serial transmission) it reduces to $\frac{1}{T} \log_2 n$; with a two-condition modulation ($n = 2$), it is $\frac{1}{T}$;

b) For a parallel transmission with equal minimum intervals and equal number of significant conditions on each channel, it is $m \frac{1}{T} \log_2 n \left(\frac{m}{T}\right.$ in case of a two-condition modulation$\left.\right)$.

RECOMMENDATION V.23

600/1200-BAUD STANDARDIZED MODEM FOR USE IN THE GENERAL SWITCHED TELEPHONE NETWORK

(Geneva, 1964)

1. The principal characteristics recommended for a modem to transmit data at medium speed in the general switched telephone network are as follows :

— use of modulation rates up to 600/1200 bauds on the communication channel (see Recommendation V.22) ;

— frequency modulation with synchronous or asynchronous mode of transmission ;

— inclusion of a backward channel at modulation rates up to 75 bauds for error control, use of this channel being optional.

2. *Modulation rates and characteristic frequencies for the forward data-transmission channel*

	F_0	F_Z (symbol 1, Mark)	F_A (symbol 0, Space)
Mode 1 : up to 600 bauds	1500 c/s	1300 c/s	1700 c/s
Mode 2 : up to 1200 bauds	1700 c/s	1300 c/s	2100 c/s

It is understood that the modem would be used in mode 1 when the presence of long loaded cables and/or the presence on some connections of signalling receivers operating close to 2000 c/s would prevent satisfactory transmission in mode 2. The modem could be used in mode 2 on suitable connections.

3. *Tolerances on the characteristic frequencies for the forward channel*

It should be possible with all rates of modulation to permit a tolerance, at the transmitter of $\pm$ 10 c/s on both the F_A and F_Z frequencies. This tolerance should be considered as a limit.

Acceptance of these tolerances would give a tolerance of $\pm$ 10 c/s for the mean-frequency $F_0 = \frac{F_A + F_Z}{2}$.

The tolerance on the frequency difference $F_A - F_Z$ with regard to the nominal value would be $\pm$ 20 c/s.

A maximum frequency drift of $\pm$ 6 c/s has been assumed in the connection between the modems which might consist of several carrier circuits connected in tandem. This would make the tolerances on the mark and space frequencies at the receiving modem $\pm$16 c/s.

4. *Modulation rate and characteristic frequencies for the backward channel*

The modulation rate and characteristic frequencies for the backward channel are as follows :

	F_Z (symbol 1, Mark)	F_A (symbol 0, Space)
Modulation rate up to 75 bauds	390 c/s	450 c/s

In the absence of any signal on the backward channel interface, the Condition Z signal is to be transmitted.

5. *Tolerances on the characteristic frequencies of the backward channel*

As the backward channel is a VF telegraph-type channel, the frequency tolerances should be as recommended in Recommendation R.35 (*Blue Book*, Volume VII) for frequency-shift voice-frequency telegraphy.

The $\pm$ 6 c/s frequency drift in the connection between the modems postulated in 3 above would produce additional distortion in the backward channel. This should be taken into account in the design.

6. *Division of power between the forward and backward channels*

Considering the following table which shows the relative levels of power for total power remaining equal to 0 dbm :

Forward channel level (dbm)	*Backward channel level* (dbm)
0	$-\infty$
— 1	— 7
— 2	— 4
— 3	— 3

Equal division of power between the forward and backward channels could be recommended provisionally.

7. The following information is provided to assist equipment manufacturers:
   a) The nominal range of attenuations in subscriber-to-subscriber connections is from 5 to 30 db at the reference frequency (800 or 1000 c/s), assuming up to 35 db attenuation at the recommended mean frequency (F_0) of the forward channel.
   b) A convenient range of sensitivity at the mean frequency F_0 for data receivers has been found to be −40 to 0 dbm for the forward channel at the subscribers' terminals.
   c) The data modem should have no adjustment for send level or receive sensitivity under the control of the operator.

8. *Time constants of the carrier detector*

   (This point remains for further study; Question 1/A—M.)

9. *Equipment for the disablement of echo suppressors*

   (See Recommendation V.21—Paragraph 5)

10. *Inclusion of a clock in the modem*

There is no provision in the present specifications for the standardized modem for the inclusion of a clock in the modem.

RECOMMENDATION V.24

STANDARDS OF THE TYPE AND FORM OF SIGNALS TO BE EXCHANGED AT AN INTERFACE BETWEEN DATA-PROCESSING TERMINAL EQUIPMENT[1] AND DATA COMMUNICATION EQUIPMENT

(Geneva, 1964)

I. Scope

I-1 These standards are applicable to the interconnection of data-processing terminal equipment[1] and data communication equipment. They define a method of exchanging control signals and binary serialized data signals between these two classes of equipment.

I-2 An interface is postulated and the functions of the interconnecting circuits crossing this interface are defined. This interface is characterized by being intended for the transfer of a serialized stream of binary signal elements on a single circuit, the timing or periodicity of these elements being uniform or regularized by conventions or standards.

[1] For the purposes only of these standards, the term "Data-processing terminal equipment" has been used in place of "Data terminal equipment", in order to emphasize the distinction between the two sides of the interface.

I–3 These standards are applicable for data transmission when used in conjunction with electronic terminal equipment, and for a wide range of operating rates, extending probably from zero to an upper limit which is approximately 10 000 to 30 000 bits per second. There is no implication that the rates in the two directions of transmission are necessarily the same.

I–4 These standards apply to all classes of data communication, both synchronous and asynchronous, including the following:

a) Leased line service, either two-wire or four-wire. Consideration is given to multipoint operation.

b) Switched network service, either two-wire or four-wire. Some consideration is given to automatic answering of calls, but it should be noted that this standard may not include all the interchange circuits that may be required for establishing the connection.

c) It also applies equally to both sides of separate intermediate equipments which provide other functions required by the communications system and which may be inserted between the data-processing terminal equipment and the data communication equipment. Such separate equipments might comprise error control equipment, and in this case, identical standards would apply on both sides of the error control equipment.

I–5 These standards are applicable where the length of the interchange circuits between data-processing terminal equipment and data communication equipment is not more than about 15 metres (50 feet). (A precise definition is given in Section IV-9 of these standards.)

I–6 The interchange circuits defined are those required for efficient operation of data transmission systems. It is not intended to limit all systems to those circuits enumerated. By mutual arrangement, any other signals may be exchanged. It is strongly recommended for such optional circuits that, whenever feasible, signals be employed which conform to the type and form defined in these standards.

I–7 The interchange circuits are illustrated in the table at the end of the recommendation. In any one type of practical equipment a selection would be made from these circuits, as appropriate.

I–8 The data communication equipment (the modem, for instance) may include transmitting and receiving signal converters. The term "signal converter" refers to the equipment which changes the data signal into a form suitable to the transmission medium, or the reverse. In general, separate signal converters are provided for transmission and for reception. A data set may include signal converters and control circuits. Other functions, such as pulse regeneration and error control, may or may not be provided. Equipment to provide these other functions may be included in the data-processing terminal equipment or in the data communication equipment, or it may be separate equipment, as described in paragraph I-4 c). It is not intended that, when other functions are provided within the data-processing terminal equipment or data communication equipment, this interface standard apply other than to the interchange circuits between the two classes of equipment.

II. Interchange circuits

II–1 The interchange circuits regarded as more important are given in this section.

II–1–1 *Ground or Earth*

Circuit 1 — Protective Ground or Earth

This conductor shall be electrically bonded to the machine or equipment frame. It may be further connected to external grounds as required by applicable regulations.

Circuit 2 — Signal Ground or Common Return

This conductor establishes the common reference potential for all interchange circuits except the Protective Ground or Earth (Circuit 1). Within the data communication equipment this circuit shall be brought to one point, and it shall be possible to connect this point to Protective Ground or Earth by means of a wire strap inside the equipment.

II–1–2 *Data signals*

Circuit 3 — Transmitted Data

Direction : TO data communication equipment.

Signals on this circuit are originated by the data-processing terminal equipment and are connected to the transmitting signal converter for transmission to remote data-processing terminal equipment.

The data-processing terminal equipment shall hold Circuit 3 in marking condition during any time interval between characters or words or at such other times when no signals are to be transmitted. Data-processing terminal equipment designed for Receive-Only service shall hold this circuit in the marking condition at all times.

The marking or spacing signal condition shall be held for the total duration of each signal element.

Circuit 4 — Received Data

Direction : FROM data communication equipment.

Signals on this circuit are originated by the receiving signal converter in response to data signals received from remote data-processing terminal equipment.

In half-duplex service, when a data set is transmitting, it shall hold marking condition on Circuit 4 so long as its Circuit 5 (Request to Send) is in the ON condition. Alternatively, in half-duplex service, the Received-Data circuit may be used to monitor transmitted signals (e.g. for local copy).

A data set equipped for Transmit-Only service shall hold Circuit 4 in the marking condition at all times.

The marking or spacing signal condition shall be held for the total duration of each signal element.

II–1–3 *Control signals*

Circuit 5 — Request to Send

Direction : TO data communication equipment.

Where relevant, a wire strap shall be provided within the data communication equipment which, when connected, shall cause Circuit 5 to have the following functions :

Signals on this circuit are generated by the data-processing terminal equipment to condition the local data set to transmit.

For example, if the data set contains a modulator, the carrier signal shall be transmitted during the ON condition of Circuit 5.

The ON condition is maintained whenever the data-processing terminal equipment has information ready for transmission or being transmitted.

Note. — The signal converter shall transmit all data on Circuit 3 (Transmitted Data) while the ON condition is maintained on Circuit 5, Circuit 6 (Ready for Sending), and Circuit 7 (Data Set Ready).

In half-duplex service, the OFF condition shall hold the data set in the Receive-Data condition and the ON condition shall hold the data set in the Transmit-Data condition. The above conditions shall be established without regard to signals on Circuits 3 (Transmitted Data) and 4 (Received Data).

Data-processing terminal equipment designed for Receive-Only service shall hold Circuit 5 in the OFF condition at all times.

Data processing terminal equipment designed for either Transmit Only or full-duplex service may hold Circuit 5 in the ON condition at all times. Similarly, data communication equipment used for Transmit-Only or full-duplex service may be arranged to be in transmit condition at all times, regardless of the signal condition on Circuit 5.

On a multipoint communication channel which may successively carry data signals, transmitted by several data communication equipment stations, Circuit 5 shall be used by each data-processing terminal equipment to condition its local data set to transmit.

In the duplex service, the wire strap inside the data transmission equipment will not be connected.

When the wire strap is removed, Circuit 5 shall have the same functions as above, except that this circuit does not control the Receive-Data functions.

Note. — The Data Receiver Cut Off circuit (Circuit 24) controls the conditions of the receive data functions, and this circuit may be combined with Circuit 5 (Request to Send) within the data-processing terminal equipment.

Circuit 6 — Ready for Sending

Direction : FROM data communication equipment.

Signals on this circuit are generated by the data communication equipment to indicate that it is prepared to transmit data. The ON condition is a response to the ON condition on Circuit 5 (Request to Send), delayed as may be appropriate to the data communication equipment for establishing a communication channel to a remote data-processing terminal. When Circuit 5 is turned OFF, Circuit 6 shall also be turned OFF.

In Receive-Only service, the data communication equipment shall hold this Circuit 6 OFF at all times.

In Transmit-Only or full-duplex service, if the data communication equipment is arranged to be in transmit condition at all times, then Circuit 6 shall be held in the ON condition at all times, except perhaps in cases where error control arrangements require control of transmission.

Circuit 7 — Data Set Ready

Direction : FROM data communication equipment.

The signals on this circuit are generated by the local data set to indicate that it is ready to operate (receive or transmit).

The OFF condition shall be used to indicate either :

a) any abnormal or test condition which disables or impairs any normal function associated with the class of service being furnished ;

b) that the communication channel is switched to an alternate means of communication (e.g. alternate voice telephone) ;

c) that the local data set is not connected to a communication channel (i.e. the data set is "on hook").

The ON condition shall appear at all other times.

This circuit shall be used only to indicate the status of the local data set. The ON condition shall not be interpreted either as an indication that a communication channel has been established to a remote station or the status of any remote station or equipment.

II–2 Additional interchange circuits may be utilized in some systems.

II–2–1 *Control Signals*

Circuit 24 — Data Receiver Cut-Off

Direction : TO data communication equipment.

This circuit is only used when the wire strap of Circuit 5 is removed.

The ON condition on this circuit shall prevent the received line signals from reaching the data set detector, thus ensuring a constant mark condition on the Received Data Circuit.

The OFF condition permits received line signals to be transferred to the data-processing terminal equipment on Circuit 4 (Received Data).

In half-duplex service, this circuit may be combined within the data-processing terminal equipment with Circuit 5 (Request to Send).

Circuit 8 — Connect Data Set to Line

Direction : TO data communication equipment.

Signals on this circuit are used to control switching of the signal converter to the communication channel. The ON condition causes the signal converter to be connected to the communication channel. However, if the station is equipped only for call origination by means external to this interface (e.g. manually or an automatic call origination unit), then the ON condition serves only to maintain the connection established by these external means. When the station is equipped for automatic answering of received calls, connection to the line may be arranged to occur only in response to a calling signal.

The OFF condition removes the signal converter from the communication channel, for such reasons as :

a) freeing the line for alternate use (e.g. voice or use by other terminal stations) ;

b) permitting use of the data-processing terminal equipment for an alternate function ;

c) terminating a call (i.e. going "on hook").

The OFF condition shall not disable the operation of Circuit 25 (Calling Indicator).

Circuit 25 — Calling Indicator

Direction : FROM data communication equipment.

Signals on this circuit indicate that a calling signal is being received from a remote station. This circuit may be required for automatic answering of received calls.

The ON condition indicates that a calling signal is being received. The OFF condition shall be maintained at all other times.

Operation of this circuit shall not be disabled by the OFF condition on Circuit 8 (Connect Data Set to Line).

Circuit 9 — Data Carrier Detector

Direction : FROM data communication equipment.

Signals on this circuit are used to provide an indication that the data carrier is being received. When the data carrier is lost because the transmitting signal converter is turned OFF or because of a fault condition, the OFF condition follows after an appropriate guard time delay.

In half-duplex service where the signal converter is arranged for local copy (see Circuit 4, Received Data) this Circuit 9 may respond to carrier signals from either the local or remote transmitting signal converter.

The ON condition indicates reception of the data carrier. The OFF condition provides an indication of the end of present transmission activity or a fault condition.

Circuit 10 — Data Signal Quality Detector

Direction : FROM data communication equipment.

Signals on this circuit are used to indicate whether or not there is a high probability of an error.

An ON condition is maintained when there is no reason to believe that an error has occurred.

An OFF condition indicates there is a high probability of an error. It may, in some instances, be used to call automatically for the retransmission of the previously transmitted data signal. Preferably the response of this circuit shall be such that a transition from an ON to an OFF condition occurs before the next signal element.

Circuit 11 — Data-signalling Rate Selector (data-processing terminal equipment source)

Direction : TO data communication equipment.

Either Circuit 11 or Circuit 12 may be used, but not both.

Signals from the data-processing terminal equipment on this circuit are used to select between the two rates in the case of dual rate synchronous modems or the two ranges of rates in the case of dual range asynchronous modems.

An ON condition shall select the higher signalling rate or range of rates.

Circuit 12 — Data-signalling Rate Selector (data communication equipment source)

Direction : FROM communication equipment.

Either Circuit 11 or Circuit 12 may be used, but not both.

Signals from the data communication equipment on this circuit are used to select one of two rates or ranges of rates in the data-processing terminal equipment to coincide with the rate or range of rates in use in a dual rate synchronous or dual range asynchronous modem.

An ON condition shall select the higher signalling rate or range of rates.

Circuit 26 — Select Transmit Frequency

Direction : TO data communication equipment.

Signals on this circuit are generated by the data-processing terminal equipment to provide a selection of the frequency of the data set transmitter.

An ON condition shall select the higher transmitter frequency.

Circuit 27 — Select Receive Frequency

Direction : TO data communication equipment.

Signals on this circuit are generated by the data-processing terminal equipment to provide selection of the frequency at the data set receiver.

An ON condition shall select the lower receiver frequency.

II–2–2 *Timing Signals*

Circuit 13 — Transmitter Signal Element Timing (data-processing terminal equipment source)

Direction : TO data communication equipment.

Either Circuit 13 or Circuit 14 may be used, but not both.

Signals on this circuit are used to provide the transmitting signal converter with signal element timing information.

The waveform shall nominally be ON and OFF for equal periods of time and a transition from ON to OFF shall nominally indicate the centre of each signal element on Circuit 3 (Transmitted Data).

Circuit 14 — Transmitter Signal Element Timing (data communication equipment source)

Direction : FROM data communication equipment.

Either Circuit 13 or Circuit 14 may be used, but not both.

Signals on this circuit are used to provide the data-processing terminal equipment with signal element timing information.

The waveform shall nominally be ON and OFF for equal periods of time. The data-processing terminal equipment shall provide a data signal on Circuit 3 (Transmitted Data) in which the transitions between signal elements nominally occur at the time of the transitions from OFF to ON condition of the signal on this Circuit 14.

Circuit 28 — Receiver Signal Element Timing (data-processing terminal equipment source)

Direction : TO data communication equipment.

Either Circuit 28 or Circuit 15 may be used, but not both.

Signals on this circuit are used to provide the receiving signal converter with signal element timing information.

The waveform shall nominally be ON and OFF for equal periods of time, and a transition from ON to OFF shall nominally indicate the centre of each signal element on Circuit 4 (Received Data).

Circuit 15 — Receiver Signal Element Timing (data communication equipment source)

Direction : FROM data communication equipment.

Either Circuit 28 or Circuit 15 may be used, but not both.

Signals on this circuit are used to provide the data-processing terminal equipment with signal element timing information.

The waveform shall nominally be ON and OFF for equal periods of time, and the transition from ON to OFF condition shall nominally indicate the centre of each signal element on Circuit 4 (Received Data).

Circuit 16 — Transmitter Block Timing

Direction : TO data communication equipment.

Signals on this circuit are used to provide the transmitting signal converter with block timing information. It may be used in those systems in which this information is transmitted by some other means than the normal data circuit.

The waveform shall be ON during one signal element only within each block. The particular signal element will be determined by system requirements. It shall be OFF at all other times.

Circuit 17 — Receiver Block Timing

Direction : FROM data communication equipment.

Signals on this circuit are used to provide the data-processing terminal equipment with block timing information. It may be used in those systems in which this information is transmitted by some other means than the normal data channel.

The waveform shall be ON during only one signal element within each block. The particular signal element will be determined by system requirements. It should be OFF at all other times.

II–3 A further group of interchange circuits are provided for some systems which use feedback error control, or other supervisory channels.

II–3–1 *Data signals*

Circuit 18 — Transmitted Supervisory Channel Data

Direction : TO data communication equipment.

This circuit is equivalent to Circuit 3 (Transmitted Data) except that the modulation rate need not be the same.

Supervisory information transmitted over this circuit may be used, for example, to notify the data transmitting terminal whether it should continue transmission or repeat certain parts of the data message. This information may be obtained from an analysis of the received data on Circuit 4 of the same terminal.

Circuit 19 — Received Supervisory Channel Data

Direction : FROM data communication equipment.

This circuit is equivalent to Circuit 4 (Received Data) except that the modulation rate need not be the same.

Supervisory information received over this circuit may be used, for example, to notify the data transmitting terminal whether it should continue transmission or repeat certain parts of the data message. This information thereby controls the flow of transmitted data on Circuit 3 in the same terminal.

II–3–2 *Control signals*

Circuit 20 — Transmit Supervisory Channel Carrier

Direction : TO data communication equipment.

This circuit is equivalent to Circuit 5 (Request to Send), except that, instead of requesting the establishment of a data channel, it requests the establishment of a supervisory channel, such as Circuits 18 and 19.

Circuit 29 — Supervisory Receiver Cut-Off

Direction : TO data communication equipment.

This circuit is equivalent to Circuit 24 (Data Receiver Cut-Off), except that, instead of controlling the Data Receiver, it controls the Supervisory Receiver.

Circuit 21 — Supervisory Channel Ready

Direction : FROM data communication equipment.

This circuit is equivalent to Circuit 6 (Ready for Sending), except that, instead of indicating the availability of a data channel, it indicates the availability of a supervisory channel, such as Circuits 18 and 19.

Circuit 22 — Supervisory Channel Carrier Detector

Direction : FROM data communication equipment.

This circuit is equivalent to Circuit 9 (Data Carrier Detector), except that, instead of indicating the proper reception of a data channel carrier, it indicates the proper reception of a supervisory channel carrier.

Circuit 23 — Supervisory Channel Signal Quality Detector

Direction : FROM data communication equipment.

This circuit is equivalent to circuit 10 (Data Signal Quality Detector) except that, instead of monitoring the signal quality of a data channel, it monitors the signal quality of a supervisory channel.

Circuit 24 — See before Circuit 8 in Section II–2–1

Circuit 25 — See after Circuit 8

Circuits 26 and 27 — See after Circuit 12

Circuit 28 — See after Circuit 14

II–4 *Other interchange circuits* not mentioned here may be provided to perform special functions. These additional circuits shall conform to the electrical specifications in Section 4.

III. Line of demarcation

The interface is the definition of the circuits at the interchange point of a connection which occurs at a plug connector, between the data communication equipment and the data-processing terminal equipment.

The following interchange circuits shall be uniquely allocated to pins on this connector :

Circuits Nos. 1, 2, 3, 4, 5, 6, 7, 8 and 9.

IV. Electrical signal characteristics

IV–1 The maximum open-circuit voltage to ground (either Frame Ground or Signal Ground) on any interchange circuit shall not exceed 25 volts, and the maximum short-circuit current flow between any two conductors (including grounds) shall not exceed one-half ampere.

IV–2 Any circuitry used to generate a signal voltage on an interchange circuit shall be so designed that no damage will be caused by either an open circuit condition or a short circuit to either Protective Ground (Circuit 1) or Signal Ground (Circuit 2). Any circuitry used to receive signals from an interchange circuit shall be designed for continuous operation with any input signal within the maximum voltage limits specified in Section IV–1.

IV–3 [1] For Circuit 3 (Transmitted Data) and Circuit 4 (Received Data) the signal shall be considered in the "marking" condition when the voltage on the circuit is more negative than minus 3 volts with respect to Signal Ground, and the signal shall be considered in the "spacing" condition when the voltage is more positive than plus 3 volts with respect to Signal Ground. During transmission of data, the "marking" condition shall be used to denote the binary state "ONE" (e.g., hole punched in paper tape) and the "spacing" condition shall be used to denote the binary state "ZERO". Note that "marking" is the normal condition on a data circuit when no signals are present.

Summary of data circuit interface terms

Binary state :	ONE	ZERO
Signal condition :	Marking	Spacing
Voltage :	Negative	Positive
Paper tape :	Perforation	No perforation

[1] In certain countries, in case of direct connection to d.c. telegraph-type circuits only, the line voltage polarities in Section IV-3 may be reversed.

For all control circuits shown in Section II, the control function shall be considered ON when voltage on the circuit is more positive than plus 3 volts with respect to Signal Ground, and shall be considered OFF when the voltage on the circuit is more negative than minus 3 volts with respect to Signal Ground.

Summary of Control Circuit Interface Terms

Control function :	OFF	ON
Voltage :	Negative	Positive

Note. — These conventions for the polarity of control signals at this interface should not be regarded as necessarily applying also to the signals on the transmission line or to the control signals at other interfaces within data-processing equipments.

IV–4 In some applications, it may be necessary for fail-safe operation to detect either the power-off condition in the equipment connected across the interface or the disconnection of the interconnecting cable.

The following circuits shall be required to indicate the OFF condition in such circumstances :

a) Request to send (Circuit 5) ;

b) Data set ready (Circuit 7) ;

c) connect Data set on Line (Circuit 8) ;

d) transmit Supervisory Channel Carrier (Circuit 20).

The power-off source impedance at the sending end of these circuits shall not be less than 300 ohms, measured at an applied voltage not greater than plus or minus 2 volts referenced to Circuit 2 (Signal Ground or Common Return).

IV–5 The operation of the circuitry that receives data and control signals from an interchange circuit shall be dependent only on the signal voltage, as specified in Section IV–3, and should, therefore, be insensitive to the rise time, fall time, presence of signal overshoot, etc. The design of this circuitry shall minimize the effects of any circuit time constants which would delay the circuit response, thus introducing time distortion in the signals.

IV–6 The terminating impedance at the receiving end of an interchange circuit shall have a d.c. resistance of not less than 3000 ohms or more than 7000 ohms, and the voltage in open-circuited condition shall not exceed 2 volts. The effective shunt capacitance at the receiving end of a signal interchange circuit, measured at the interchange point and including any connecting cable, shall not exceed 2500 picofarads.

IV–7 The source impedance at the sending end of an interchange circuit is not specified, except for Circuits 5, 7, 8 and 20.

IV–8 For both data circuits and timing signal circuits (circuits 3, 4, 13, 14, 28 and 15), neither the rise time nor the fall time, through the range (+3 V to −3 V) in which the signal condition is not defined, shall exceed 3 per cent of the nominal duration of a signal element. The circuitry used to generate a signal voltage on an interchange circuit shall meet this specification with any receiving termination which complies with Section IV–6.

IV–9 *Equivalent circuit*

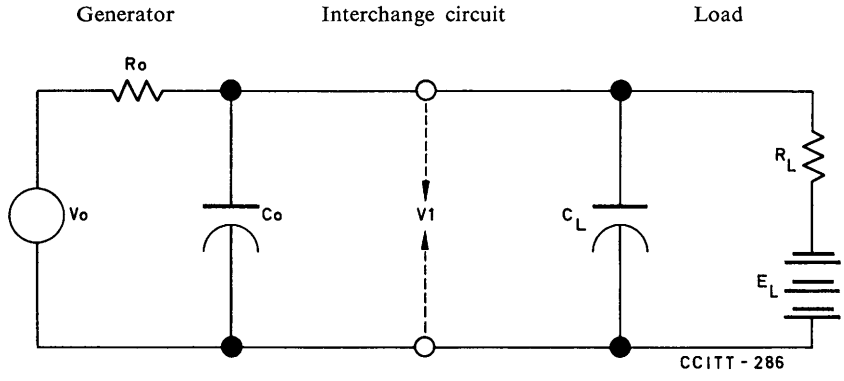


FIGURE 2. — **Circuit 2 : Signal Ground or Common Return**

V_0 = is the open-circuit generator voltage.
V_1 = is the interface voltage.
R_0 = is the generator internal impedance.
R_L = is the load impedance.
E_L = is the load open-circuit voltage (bias).
C_L = is the total effective load capacitance, measured at the interchange point and including any cable.
C_0 = is the capitance associated with the generator, including any cable to the interchange point.

The following conditions shall apply :

1) V_0 shall not be greater then $\pm$ 25 volts.

2) When the interchange circuit is short-circuited, the circuit shall not be damaged and the short-circuit current shall not be greater than one-half ampere.

3) R_0 is not otherwise specified, except that, when R_L is 3000 ohms and E_L is zero, V_1 shall not be less than $\pm$ 5 volts. Good design practice indicates that R_0 should be made small.

Note. — The nominal service voltage at this point is 3 volts ; the 5-volt specification applies only to this condition 3).

4) E_L shall not be greater than $\pm$ 2 volts.

5) R_L shall not be less than 3000 ohms or more than 7000 ohms.

6) C_L shall not be greater than 2500 picofarads.

7) C_0 is not specified, but must be considered in determining the ability of the generator to comply with the rise times and fall times specified in Section IV–8.

8) The circuitry that receives signals from an interchange circuit shall recognize the binary signal when V_1 equals $\pm$ 3 volts. (See Section V–3). The signal condition is not defined when V_1 is in the range between +3 and —3 volts.

IV–10 In order to avoid inducing voltage surges on interchange circuits, signals from interchange circuits should not be used to drive inductive devices, such as relay coils.

IV–11 The Signal Ground (or Common Return) potential may be offset (biased) from protective ground (Earth) by up to ± 5 volts if required, and if the wire strap is not present.

V. Recommendations and Explanatory Notes

V–1 The data are to be serialized by the data-processing terminal equipment so that the design of the data communication equipment may be independent of the character length and code used by the data-processing terminal equipment.

V–2 The control circuits at the interface are arranged to permit the alternate use of a higher class of communication service, as follows :

a) data-processing terminal equipment designed for transmit only service may also use either half-duplex or full-duplex service ;

b) data-processing terminal equipment designed for receive-only service may also use either half-duplex or full-duplex service ;

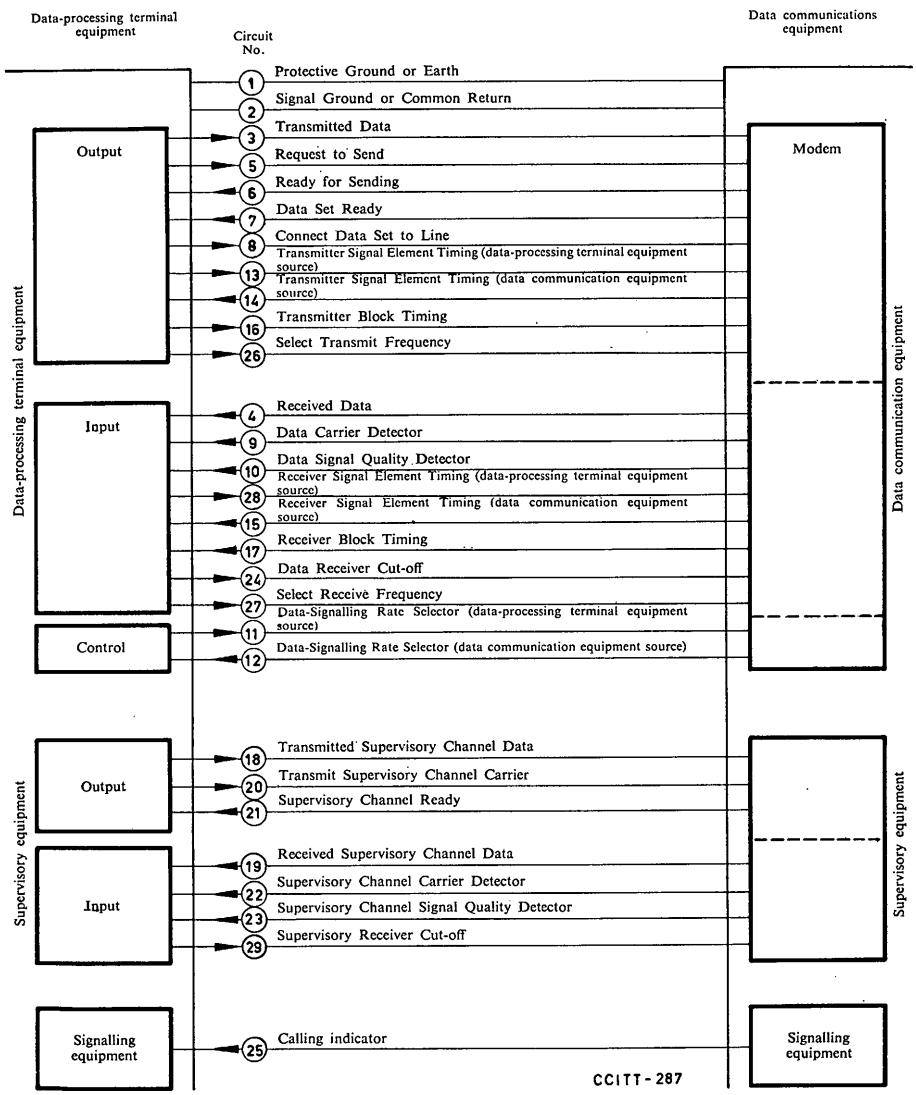
c) data-processing terminal equipment designed for half-duplex service may also use full-duplex service.

The substitution of a higher class of communication service does not require a change in the operation of the control circuits.

V–3 The electrical specifications are intended to provide a two-volt margin in rejecting noise introduced either on signal interchange circuits or by a difference in reference ground potential across the interface. This margin of safety should be maintained in all interchange circuitry.

V–4 Interchange signals should have approximately rectangular waveforms.

V–5 It should be noted that, on some types of communication channels, it may not be economically feasible to provide a "mark hold" on Circuit 4 (Received Data) as specified in Section II–1–2. It may, therefore, be desirable for the operation of the data-processing terminal equipment to be independent of this specification for "mark hold".



Circuits Nos. 1 to 9 require a unique pin assignment in the connector

FIGURE 3. — **Functional diagram**

QUESTION ENTRUSTED TO SPECIAL STUDY GROUP A: DATA TRANSMISSION

Chairman : Mr. J. RHODES (United Kingdom)

Vice-Chairman : Mr. H. E. VAUGHAN (United States of America — American Telephone and Telegraph Co.)

Question 1/A — Data transmission

(continuation of Question 43 of S.G. 1 and 8, 1957-1960, amended at Geneva, 1964)

What general characteristics should be standardized to permit international data transmission ?

ANNEX TO QUESTION 1/A

Study programme for further study of the Question:
Points for special study and recommendations on the tests to be made

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B	Supplement to the vocabulary for data transmission		60
C	Alphabet for telegraphy and data transmission	See Question 6/I = 8/VIII (to be studied by Working Party on the "New Telegraph Alphabet")	60
D	Quality of service of a telegraph circuit at 50 bauds, 100 bauds, 200 bauds, or more		61
E	Use of the telex network		61
F	Automatic originating and answering of calls in the telex network		62
G	Control of errors attributable to electro-mechanical equipment		63
H	Data transmission up to 200 bauds in switched telegraph networks		63
I	Error control in the telegraph network		64
J	Use of telephone-type circuit for the simultaneous transmission of data and V.F. telegraph signals		64
K	Duplex data transmission at 200 bauds, alternating with telephony		64

Point	Brief description	Remarks	Page
L	Test for data transmission systems on telephone-type circuits	Results to be sent for information of Study Group IV	64
M	Complement to standardization of the modem at 600/1200 bauds		67
N	Standardization of data-signalling rates		69
O	Modem for modulation rates above 1200 bauds		70
P	Temporary use of non-standardized modems over the general switched network		70
Q	Parallel transmission		71
R	Error control in the general telephone network		71
S	Measurement of phase distortion between subscribers		71
T	Subscriber-to-subscriber loss		71
U	Specification of frequency-loss characteristics phase distortion and impulsive noise limits for circuits leased for data transmission and measuring techniques to check these specifications *Note :* The study will have to include data transmission with or without error control	Study Group IV to be informed in respect of measuring techniques	72
V	Specification of impulsive noise limits for data transmission over switched telephone circuits. *Note :* The study will have to include data transmission with or without error control		72
W	Maintenance methods	Study Group IV to be kept informed	72
X	Problems raised by echo suppressors		73
Y	Automatic originating and answering of calls in the telephone network		74
Z	Use of wide-band circuits		75

Question 1/A - point B. — Supplement to the vocabulary for data transmission

(continuation of point O of the study programme 1961-1964)

This study should be pursued, bearing in mind :

— definitions already established and shown at the beginning of this Volume,

— definitions included in the *List of definitions of essential telecommunication terms*—2nd edition, 1961, and in the First Supplement to this List.

Question 1/A - point C. — Alphabet for telegraphy and data transmission

(continuation of points C and D of the study programme 1961-1964)

(Question 6/I of Study Group I and 8/VIII of Study Group VIII : Question to be studied by Joint Working Party on the "New Telegraph Alphabet".)

1. Study of the requirements which the telegraph service may be called upon to meet in future.

(Question 1/A)

2. Study of a telegraph alphabet which would meet these needs.

3. Study of an international alphabet for telegraphy and data transmission.

Note 1. — This alphabet will be for general (though not exclusive) use in data transmission and it will be usable by Administrations and private operating Agencies where needs cannot be met by Alphabet No. 2.

Note 2. — The study of an alphabet for data transmission by telephone subscribers is included in this study, since only one alphabet has to be proposed.

Note 3. — The study should also include the character structure (bit-order of transmission, etc.) for transmission of the new alphabet in both synchronous and asynchronous operation.

For this study, see Supplements Nos. 2 and 3 of this Volume.

Question 1/A - point D. — Quality of service of a telegraph circuit at 50 bauds, 100 bauds, 200 bauds, or more

(continuation of points G and H of the study programme 1961-1964)

For standardized 50-baud telegraph circuits, the following results which were obtained from the study during the period 1961-1964 could be considered as a reply to this question : *Quality of service of a standardized 50-baud telegraph circuit* (voice-frequency telegraph system with channels spaced at 120 c/s).

Most probable error rate

a) *In point-to-point service*
on elements : 1 to 2 10^{-5}
on characters : 1 to 8 10^{-5}

b) *With switching*
on elements : 1 to 2 10^{-5}
on characters : 4 to 5 10^{-5}

Most probable time interval without error : one hour approximately.

Bursts of errors (regarded as constituted by errors separated by less than ten non-erroneous elements) :

isolated errors : 50 to 60 %
bursts with two errors : 10 to 20 %
bursts with three errors : 3 to 10 %
bursts with four errors : 2 to 6 %

Time intervals in which a "start" polarity is received for more than 300 milliseconds should be considered as "out of service".

Periods in which the circuit may be considered to be "out of service" should be ignored when the test results are evaluated.

For telegraph circuits at 100 bauds or 200 bauds, the results already known are given in Supplements Nos. 5 and 6 of this Volume.

For the telex network, see Supplement No. 4 of this Volume.

Question 1/A - point E. — Use of the telex network

(continuation of point I of the study programme 1961-1964)

Further study of the use of the telex network, bearing in mind Recommendation V.10 : use of the international telex network for data transmission at a modulation rate of 50 bauds :

1. According to Recommendation V.10, I.b) and II.b) on reception of the sequence of four combinations No. 19 transmitted by either subscriber, the data transmission equipment will be connected to the line.

The transfer to data operation may be effected :

a) manually at both terminals ;

b) automatically at both terminals ; or

c) manually at one terminal and automatically at the other.

In all cases it may be advisable if a signal "Invitation to begin data transmission" is sent from the station which will receive the data to the station which transmits the data after the transfer to data of the former one.

In addition, in case a) it may be advisable in order to avoid any misunderstanding between the stations concerned to exchange a signal "Ready to change-over to data transmission".

How should these signals be composed and how should they operate ?

2. Examination of the possibilities of using codes with more than five units on the telex network.

The six-unit code was of some attraction because of the special requirements of some equipment. Considering the fact that combination No. 32 of the normal five-unit alphabet seemed to be suitable for use without any restriction, at least in the future, the only dangerous combination of the six-unit code could be the combination with six units of start polarity ; measurements have shown, however, that this combination could also be used without restriction, provided that the stop pulse of the start-stop signal was equal to at least two units.

Doubts were expressed about the interest of a solution of this type, owing to the reduction of the practical efficiency in signals transmitted at 50 bauds with lengthened-cycle signals.

It was pointed out that this prolongation of the stop polarity might be applicable only in the case of a special six-unit start combination.

The transmission of higher-unit codes (seven and eight units) will not be universally possible without the use of additional equipment, for example, an error-detecting and correcting equipment, or a special code regrouper that inserts (for transmission purposes only) an element of stop polarity between the 3rd and 4th information elements of each character (this element being removed at the receiving terminal before the character is passed on to the processing equipment).

3. Consideration of the various aspects of the problem of connecting data-transmission equipment to the telex network, according as to whether this equipment is connected :

a) between the teleprinter and the control unit ;

b) between the control unit and the telegraph exchange.

4. If the operator should make a mistake, or if some automatic device should fail, or should there be a disturbance on the line, or for some other reason, one of the two terminals may be permanently in the data-handling position, while the other is prepared to deal with telex, not only during the change-over to data, but also during data transmission.

It would be well if things could be put right without interrupting the telex call and, if possible, automatically.

Can some way out of this difficulty be devised ?

Note. — See Supplements Nos. 7, 8 and 9 of this Volume.

Question 1/A - point F. — Automatic originating and answering of calls in the telex network

Is there a requirement to provide, in association with the telex network, a system of data transmission with fully automatic originating and answering of calls ? If so, what means would

be necessary to guarantee that the appropriate connection is established and to prevent unacceptable connection in operation sources, e.g. barring ?

In any case, the devices replacing the teleprinter on a subscriber's telex line should send a signal which conforms to the one prescribed in Recommendation U.1 (Volume VII of the *Blue Book*) for the call-connected signal.

The need for an answer-back code simulator seems to meet with general approval.

However, two special problems may lead to different solutions :

a) the question of whether the releasing of the answer-back code takes place :

— automatically at the time that the receiving equipment is engaged ;

— following the receipt of any telegraph modulation ;

— only after receipt of the "Who are you" signal, which presupposes a receiving unit with a storage device.

b) The question of the composition of this code. This composition may be made in such a way that use is made of the code signals :

— to introduce the sequence SSSS (or ""),

— to identify data transmission equipment from which information is required,

— to reject unwanted calls.

Question 1/A - point G. — Control of errors attributable to electro-mechanical equipment

Certain experiments have shown that the errors attributable to electro-mechanical terminal equipment connected to the telecommunication network may be as numerous as those in an unprotected circuit. Would it be possible to arrange for the error-checking systems to afford some protection against errors in electro-mechanical equipment ? If so, what characteristics should be recommended for such a system ?

It emerged from the discussions that, with the present methods for detecting errors, it is generally possible to correct almost all errors due to transmission proper. There remain the errors caused by operating defects in the terminal electro-mechanical equipment. Although the percentage of such errors is low, it nevertheless constitutes an important fraction of the remaining uncorrected errors.

It is possible that the addition of a simple procedure, such as a parity check, combined with the conventional detection methods, might appreciably reduce these undetected errors.

Studies would be made to this end by various international organizations such as the I.S.O., and it would be desirable for contacts to be kept with these organizations in this respect.

As the alphabets generally used are not redundant, it may be necessary to introduce additional elements in the signals actually sent. Even with this solution, it seems difficult to incorporate, in the error-detection loop, all the electro-mechanical apparatus involved at the communication terminals. Also, consideration might be given to a method which would consist of dividing the correction circuit into three sections, i.e. terminal equipment, line, terminal equipment.

Note. — See Supplement No. 18 of this Volume.

Question 1/A - point H. — Data transmission up to 200 bauds in switched telegraph networks

Is there need for a low-speed (i.e., modulation rate up to 200 bauds) duplex data transmission system utilizing telegraph-type circuits over public switched telegraph networks in national and international services ? If so, what characteristics should be specified ?

Note. — See Supplements Nos. 9, 10 and 11 of this Volume.

Question 1/A - point I. — Error control in the telegraph network

Consideration of a form of data transmission error control for the telegraph networks.

Note 1. — Special allowance will have to be made for intervention of ARQ equipment, and for the presence or absence of regenerative repeaters.

Note 2. — See Supplements Nos. 7, 8, 9, 12, 13, 14, 15, 16 and 17 of this Volume.

Question 1/A - point J. — Use of telephone-type circuit for the simultaneous transmission of data and V.F. telegraph signals

For the most efficient use to be made of an intercontinental telephone-type circuit, it is desirable simultaneously to carry data signals at rates of 600, 1200 bauds or more and to use voice-frequency telegraph signals at 50 bauds or above.

What should be:

a) the frequency allocations on such a circuit when the C.C.I.T.T. 600/1200-baud standard modem is used?

b) the characteristics of the separating filters?

c) the power level of the data signal?

Note 1. — It is recognized that where, by agreement between Administrations and recognized private operating Agencies, the C.C.I.T.T. standard modem is not used, other frequency allocations may be permitted.

Note 2. — See Supplement No. 19 of this Volume.

Question 1/A - point K. — Duplex data transmission at 200 bauds, alternating with telephony

Continuation of the study of duplex data transmission at 200 bauds, alternating with telephony, bearing in mind Recommendation V.21.

In particular,

a) Is it advisable to provide for automatic transfer from telephony to data transmission when a telephone circuit may be used either for telephony or for data transmission at 200 bauds?

If so, what signalling should be recommended for such automatic control?

b) Asynchronous data-signalling rate.

c) Tape-to-tape transmission.

d) Teleprinter-to-teleprinter transmission.

e) Selection of the necessary interchange circuits from Recommendation V.24 (Interface Standard).

f) Standardization of echo suppressors (see Question 1/A - point X).

Question 1/A - point L. — Tests for data transmission systems on telephone-type circuits

(continuation of point L of the study programme 1961-1964)

It would be difficult and probably undesirable to establish a rigid test specification, but nevertheless some guiding principles can be given for future tests.

1. *Preliminary laboratory tests*

The recommendations given below apply to systems which have already been subjected to the normal laboratory tests applied to new transmission systems.

2. *Arrangement of the test connection*

It is possible to set up the test connection on a point-to-point basis or in the form of a loop. The former arrangement most closely resembles the practical case but there are serious difficulties involved in transporting equipment and personnel.

A satisfactory form of loop testing can be achieved by connecting the transmitting apparatus to a distant point by means of a high-quality circuit producing no noise or attenuation in the test conditions. The return circuit to the receiving apparatus can be set up in a variety of ways as detailed later.

3. *Modulation rate*

It is recommended that a modulation rate be selected from those which are preferred or permitted by Study Group A.

Study Group A expressed the opinion that two preferred modulation rates can be stated, namely 1200 bauds and 600 bauds, for use over telephone connections established on the switched network.

Study Group A would also like tests to be made at other rates, namely 200, 1800, 2400 and 3000 bauds.

On point-to-point circuits, the same preferred rates (1200 bauds and 600 bauds) have been expressed, and tests asked for modulation rates of 1800, 2400, 3000 bauds and 600×2^n bauds for the other rates (or for data signalling rates of 1800, 2400, 3000, 600×2^n bits per second).

(See point N of the Study Programme.)

4. *Signal level*

It is recommended that the tests should be made at the signal level which is proposed in Recommendation V.2. Additional tests should be carried out at levels 6 db above and below the level proposed. In the event that no particular value of signal level is recommended, systems should be tested on the basis of a signal level of −10 dbm0 with auxiliary tests at levels 6 db above and below this value. Tests at 6 db above the recommended levels should be made only if they do not cause interference with other circuits.

5. *Sampling of test circuits*

The system performance tests should be carried out on a selection of samples totalling not less than the quantities shown in the following table:

Modulation rate (or data-signalling rate)	Leased circuits	Switched circuits
600	2×10^7 bits	4×10^7 bits
1200	4×10^7 bits	8×10^7 bits
1800 and above	8×10^7 bits	2×10^8 bits

(Question 1/A)

For the leased circuits, the samples chosen should be representative of the facilities likely to be frequently used in practice.

For the switched circuits, the tests should include at least 25 different connections which are representative of the facilities likely to be experienced in practice.

The tests should be restricted to the business hours of normal working days and the samples should be representative both of the working day and the different attenuation conditions which may be experienced in practice.

Tests should also be made on long and complicated circuits.

No special maintenance should be carried out on circuits.

6. *Error record*

It was agreed that there are two possible forms in which the error statistics might be recorded :

i) A complete fault statement including the exact position of all faults. Such raw material could be used for a variety of statistical studies necessary for the over-all design of a data transmission system.

ii) A simplified fault record indicating totals for a number of predetermined parameters which could be easily obtained during the testing process.

It was agreed that the complete fault statement was preferable for general assessment and essential for the evaluation of the error detection part of the system. The simplified record will be valuable for the comparison of different data transmission systems and line facilities.

7. *Presentation of results*

It is recommended that for all tests the following statistics should be recorded :

i) block-error rate for given block sizes ;

ii) number of erroneous bits within erroneous blocks ;

iii) error burst length (a burst being regarded as constituted by errors separated by 10 or more error-free elements) ;

iv) structure of error bursts or error distribution ;

v) average element-error rate.

8. *Information concerning connections used for test*

It is desirable that characteristics of the connection used for data tests should be obtained by monitoring and by measurement. For example, if the transmission of data causes any unwanted operation of signalling equipment with a consequential adverse effect on the stability of the connection, this should be recorded. Similarly it should be recorded if the reception of the test message is adversely influenced by warning tones, metering signals, etc. The presence and extent of switch noise, dial impulses and random impulse noise should be noted. The level of the signals at the receiving apparatus, the level of white noise and, if possible, noise bursts, attenuation distortion and phase distortion, should also be recorded.

9. *Interruption of the connection used for the test*

It was agreed that in the event of the connection used for the test being interrupted for a period exceeding 300 milliseconds, it should be considered that the connection was "out of service"

during such an interval ; such periods are to be ignored when the test results are evaluated, but will be indicated in the test record (duration, time of occurrence, and, if possible, the cause).

10. *System description*

It is recommended that the performance results should be accompanied by a brief description of the basic characteristics of the data transmission system undergoing test. In addition, a short explanation of the method of testing used would be advisable.

11. The results of this test are of interest for Study Group IV (maintenance on the general telecommunication network), which is to be kept informed by the C.C.I.T.T. Secretariat.

Note. — See Supplements Nos. 20 to 41 of this Volume.

Question 1/A - point M. — Complement to standardization of the modem at 600/1200 bauds

The following points should be examined :

a) automatic selection between the rates of 600 and 1200 bauds ;

b) distribution of power between the "forward" and "backward" channels (see Recommendation V.23) ;

c) equalization in the modem of phase distortion in calls (see contribution COM Sp. A — No. 71 by the United Kingdom published in Supplement No. 75) ;

d) co-ordination between Recommendations V.23 on the modem and V.24 on the interface.

Recommendation V.23 for 600/1200 bauds FM modem involves the use of several interface circuits although these were not specified explicitly. Recommendation V.24 lists the interface circuits which seem to be required for this modem for various operating modes. Some of these circuits need to have characteristics, such as operating and releasing times, specified. Where this has been considered, the following table has been annotated appropriately. Absence of a note cannot be taken as indicating that special characteristics are not required. Attention should be given in future studies to completing the specification in respect of time delays, cross-connections, etc., on interface circuits.

e) tolerance on modulation rates of 600 and 1200 bauds (see Recommendation V.22) ;

f) time constant of the phase detector.

f 1) The "pick-up" and "hang-over" times of the carrier detector circuit should be specified, as the range of values were proposed for these two functions :

Pick-up times : from 2 to 100 ms
Hang-over times : from 2 to 30 ms.

f 2) In half-duplex systems, the "turn-around" time of the system is prolonged by the sum of the "pick-up" and "hang-over" times. In the case of short messages, a considerable loss of efficiency would result if these times were not short with respect to the message length (which could be as short as 200 ms). As a longer time constant, if required, could easily be incorporated in the data terminal equipment or the modem, the most universal solution is considered to be the one with the shortest time constant. Also, with a short time constant the carrier fail detector can be used as some kind of disturbance detector.

Interface circuit		Has time delay, etc., to be specified?	Required for			
			One-way data [1]		Half-duplex data [1]	Duplex [2] data/ supervision
No.	Name		Send	Receive		
1	Protective ground or earth	—	Not covered in present 600/1200 modem specification			
2	Signal ground or common return	—				
3	Transmitted data	—	Yes	No	Yes	Yes
4	Received data	—	No	Yes	Yes	Yes
5	Request to send	Possibly	Yes	No	Yes	Yes
6	Ready for sending	Yes	Yes	No	Yes	Yes
7	Data set ready	Yes	Yes	Yes	Yes	Yes
8	Connect data set to line	Yes	Yes	Yes	Yes	Yes
9	Data carrier detector	Yes	No	Yes	Yes	Yes
10	Data signal quality detector	Possibly	No	Not covered in present 600/1200 modem specification		
11	Data-signalling rate selector	—	Yes	Yes	Yes	Yes
12	Data-signalling rate selector	—	Not covered in present 600/1200 modem specification			
13	Transmitter signal element timing					
14	Transmitter signal element timing					
15	Receiver signal element timing					
16	Transmitter block timing					
17	Receiver block timing					
18	Transmitted supervisory channel data	—	No	No	No	Yes
19	Received supervisory channel data	—	No	No	No	Yes
20	Transmit supervisory channel carrier	Possibly	No	No	No	Yes
21	Supervisory channel ready	Yes	No	No	No	Yes
22	Supervisory channel carrier detector	Yes	No	No	No	Yes
23	Supervisory channel signal quality detector	Possibly	No	No	No	Not covered in present 600/1200 modem spec.
24	Data receiver cut-off	Yes	Not covered in present 600/1200 modem specification			
25	Calling indicator	Possibly				
26	Select transmit frequency	No				
27	Select receive frequency	No				
28	Receiver signal element timing	—				
29	Supervisory receiver cut-off	Yes				

[1] Using 600/1200 data channel only.
[2] Using 600/1200 data channel and the optional low-speed backward channel.

Note. — Full-duplex data transmission would be provided by the equivalent of two, oppositely arranged, independent data-sending and data-receiving systems using the appropriate interface circuits given above.

f 3) In favour of medium times (10 ms) for both pick-up and hang-over times, it should be noted that :

i) a *short* "hang-over" time can lead to frequent and unnecessary "drop-out" on short interruptions, which in any case will probably be catered for by the error-correction systems ;

ii) a *long* "hang-over" time can cause unnecessary delays in initiating alarms and corrective action, and may lead to undetectable errors arising from long disconnections of the line.

f 4) A reasonable compromise between these limiting conditions seems to be about 10 ms, bearing in mind likely error-correction systems.

Similar reasoning in respect of i) false resumption and ii) unnecessary delay in resumption, would indicate that "pick-up" time should be of like duration.

Note. — See Supplements Nos. 42, 43 and 44 of this Volume.

Question 1/A - point N. — Standardization of data-signalling rates

a) It is suggested that the range should be as follows *for a synchronous transmission* :

Preferred range	*Permitted range*	
600		
1200		
	1800 [1]	2000 [2]
2400		
	3000 [1]	
	3600 [1]	
4800		

In determining this range, the Study Group had in mind the need to restrict the number of data-signalling rates (and hence modem designs required), yet at the same time to allow the best use to be made of technical progress in both modem development and improvement in the telephone plant. It is considered that a geometric progression in standard rates provides the most satisfactory basis of development.

b) *For asynchronous transmissions :*

Two distinct viewpoints can be identified :

i) the communication authorities are anxious to define recommended data-signalling rates for asynchronous operation :

a) to facilitate circuit testing by restricting the number of different test modulation rates (although it is recognized that this need could be satisfied by standardizing test rates, as distinct from actual operating rates) ;

---

[1] Not preferred rate, if the next higher preferred rate can be obtained with the same reliability over the same facilities.

[2] It was noted that there is a substantial operation at 2000 bits/s. While this rate is not preferred, it is recognized that usage will continue until the facilities for the next higher preferred rate (2400 bits/s) are available.

b) to facilitate inter-communication between users of data-transmission equipment of different manufacture ;

ii) the data-processing industry requires the facility to send data at any rate up to the maximum usable on the circuit, where the maximum rates themselves could correspond to a recommended range, since :

a) at the present stage of development no clear range of speeds has emerged from the wider range of speed now in common use which would provide the basis for standardization ;

b) a free choice of data-signalling rates permits considerable flexibility.

One solution to this problem is for the C.C.I.T.T. to make recommendations for maximum data-signalling rates and to allow any rate to be used up to these maxima. On the other hand, it was felt preferable to establish a range of fixed rates to facilitate interconnection of terminals and to provide a proper framework for the future development of suitable terminal devices.

For data-signalling rates less than 1200 bits per second, the Study Group would like a standard rate of 600 bits per second for reasons similar to those given for synchronous operation.

For data-signalling rates greater than 1200 bits per second, it is suggested that the same rates should be recommended as for synchronous operation on the switched network.

Note. — See Supplement No. 27 of this Volume.

Question 1/A - point O. — Modem for modulation rates above 1200 bauds

Which data-transmission characteristics should be standardized for modulation rates in excess of 1200 bauds, and which figures should be recommended for them ? (Synchronous and start-stop methods of transmission should be considered.)

a) for switched calls in the general telephone network ?

b) for leased point-to-point circuits ?

For the rates in question, consideration should be given to :

— the type of modulation,

— synchronization.

Note. — See Supplements Nos. 20, 27, 35, 45 to 53 of this Volume.

Question 1/A - point P. — Temporary use of non-standardized modems over the general switched network

Modems using other modulation techniques are operating over telephone leased lines at data-signalling rates higher than 1200 bits per second.

In case of a breakdown of the leased line, and whenever it is not possible to establish quickly an emergency leased line, one can accept, in order to maintain service to the user, the temporary use of such modems over the switched network, provided that these modems have also been specifically designed to operate at 600 or 1200 bits per second.

Question 1/A - point Q. — Parallel transmission

If there is a requirement for data transmission on switched telephone circuits making use of parallel transmission, which characteristics should be standardized, and which figures should be recommended for these characteristics, if these systems are to be compatible in international service? (Consideration should also be given to intercontinental working.)

Note. — See Supplements Nos. 54 to 60 of this Volume.

Question 1/A - point R. — Error control in the general telephone network

Which characteristics should be recommended for systems of error control in data transmission on general telephone network at:

a) 200 bauds

b) 600/1200 bauds

c) higher modulation rates?

Note. — See Supplements Nos. 34, 40, 61 to 70 of this Volume.

Question 1/A - point S. — Measurement of phase distortion between subscribers

Measurement of phase distortion between subscriber and subscriber is desirable, and Administrations are encouraged to report any such tests that are made.

It would be very helpful to the designers of data modems to have statistical information on over-all subscriber-to-subscriber phase distortion. The information received gives little assistance on this point, with the exception of the results of the U.S.A. contribution recorded in the *Red Book*, Volume VII, page 439 ff. Because of the magnitude of the task, the C.C.I.T.T. hesitates to ask for a programme of tests. The results of any test the Administrations are able to make would be a help on the future study of this problem. In the absence of any test results on this topic, designers can only be referred to the data on phototelegraphy contained in the *Red Book*, Volume VII, on pages 104-105 and 328-335. Additionally, Administrations might provide estimates of any circuit routing of special interest.

On the other hand, the IIIrd Plenary Assembly of the C.C.I.T.T. has adopted Recommendation G.114 (Volume III of the *Blue Book*) on the propagation time between subscribers and Recommendation G.133 (published also in Volume III of the *Blue Book*) on limits of the phase distortion on a chain of international circuits.

Note. — See Supplements Nos. 71, 72, 73, 75 and 78 of this Volume.

Question 1/A - point T. — Subscriber-to-subscriber loss

Information already available in the contributions suggested that a nominal range of losses in subscriber-to-subscriber connections might be 5 to 30 db at the reference frequency (800 or 1000 c/s), perhaps assuming up to 35 db loss at the carrier and characteristic frequencies recommended. Studies and, where necessary, tests should be made from which statistical conclusions could be drawn for national, international and intercontinental connections. These might be

expressed in terms of the loss at the reference frequency, plus the slope over the frequency band of interest, and would form the basis of a study on equalization.

Note. — See Supplements Nos. 73 and 75 of this Volume.

Question 1/A - point U. — Specification of frequency-loss characteristics, phase distortion and impulsive noise limits for circuits leased for data transmission; measuring techniques to check these specifications

Note 1. — The study will have to include data transmission with or without error control.

Note 2. — Information useful for this study is given in the following Recommendations (Volume III, *Blue Book*): G.101, G.123, G.124, G.125, G.132, G.133, G.141, G.151, G.152, G.153.

Note 3. — Study Group IV will be informed in respect of measuring techniques.

Note 4. — See Supplements Nos. 20, 24, 27, 29, 31, 35 to 39, 71, 72 and 74 to 80 of this Volume.

Question 1/A - point V. — Specification of impulsive noise limits for data transmission over switched telephone circuits

Note 1. — The study will have to include data transmission with or without error control.

Note 2. — This is Question 7/C of Special Study Group C and Question 8/XI of Study Group XI.

Note 3. — See Supplements Nos. 35, 74, 77, 79, 80, 81 and 82 of this Volume.

Question 1/A - point W. — Maintenance methods

What binary-digital maintenance testing methods can be recommended for data communication over telephone-type circuits?

Points to be considered:

1. *Isochronous or start-stop transmissions*

2. *Modulation rates*
   a) fixed rates (bearing in mind the rates of 200, 600 and 1200 bauds);
   b) variable rates up to a limit to be decided on.

3. *Type of test signal*
   a) reversals and asymmetrical signals;
   b) sequence pattern and number of elements either quasi-random, or adjustable to any desired combination, or fixed combination.

4. *Type of modulation with a view to maintenance tests*
   a) either according to all preferred types and characteristics of modulation;
   b) or according to one (or more) standardized kind of modulation with standard characteristics for maintenance testing purposes.

5. *Synchronization at the receiving end*

6. *Telegraph distortion measurement*
   a) bias (and fortuitous) distortion;
   b) total distortion;
   c) duration of observation.

7. *Measurement of the number of errors or error bursts*
   a) preferred type of test signal (see 3);
   b) method of comparison for error discrimination;
   c) duration of test.

8. *Methods of making loop-around tests and other sectionalized tests*
   Suitable testing points and loop-around arrangements would have to be provided.

9. *Simplified test procedures which would locate the majority of troubles*

10. Similar dispositions to those of 1 to 7 above for the *backward channel*, operated at low modulation rates.

Note 1. — Study Group IV will be informed in respect of measuring techniques.

Note 2. — See Supplements Nos. 81, 83 and 84 of this Volume.

Question 1/A - point X. — Problems raised by echo suppressors

(See Recommendation V.21, point 5)

1. The suggested frequency of the transmitted signal for disablement of echo suppressors, viz. 2100 $\pm$ 15 c/s, did not take into account possible frequency shift in the connection. A figure of at least 2100 $\pm$ 21 c/s at the echo suppressor should be used for this purpose.

2. It is desirable to have a reasonably wide band for the disablement of echo suppressors so as to provide not only for the standardized 600/1200-baud modem but also for possible future systems. The need for a minimum band of 2100 $\pm$ 21 c/s has already been mentioned under 1 above. A.T. & T. practice in this respect is a conditioning band of 2000-2250 c/s.

Attention was drawn by the Italian Administration to the possibility of interference by the proposed echo-suppressor disablement tone with a 1 V.F. signalling receiver operating at 2040 c/s on certain circuits.

3. The proposals on the length of the disablement signal (F_d) are summarized below:

3.1 It was agreed that it would be possible to precede each data transmission with a disablement tone signal at least 300 ms in length. It was understood that this length of signal would be sufficient to disable any echo suppressors appearing on the circuit.

3.2 It was agreed that, in the case of simplex data transmission, it would be possible to limit the break between the transmission of the echo-suppressor disabling tone and the transmission of data signal to less than 100 ms.

3.3 It was however emphasized that, in the case of half-duplex working, the break between 1) the cessation of transmission at the transmitting end A, and 2) the arrival of a signal at an echo suppressor situated at end A from the "turned round" transmitter at end B, would be equal to the time of propagation from A to B plus the time of propagation from B to A plus the "turn-around" time at modem B.

It was estimated that a propagation time of 100 ms could reasonably be expected on long circuits and a possible "turn-around" time would be 50 ms. This would mean a total signal break of 250 ms at one echo suppressor.

The problems of operating data transmission systems on connections having long propagation time, especially half-duplex systems where a quick turn-around is desirable, could, however, only be studied superficially. It is clear that further detailed study is desired, preferably with co-operation from specialists in world-wide transmission. It seems that the data-processing equipment may have to obey certain rules.

4. It was confirmed that the assumed level for the disabling signal, viz. -12 ± 6 dbm0, was reasonable.

Note 1. — The question on disablement of echo suppressors is also studied from the viewpoint of telephone switching: Questions 2/XI and 3/XI of Study Group XI.

Note 2. — See Supplement No. 85 of this Volume.

Question 1/A - point Y. — Automatic originating and answering of calls in the telephone network

Characteristics to be standardized for fully automatic originating and answering of calls from or to data terminal equipment:

1. Specifications for an interface between the telephone network and data terminal equipment;

2. Recognition of the various signals which originate on the telephone network (ringing current, dial tone, busy tone, etc.) and their conversion, as required, to suitable interface signals;

3. Conversion, as required, of the interface signals originating in the data terminal equipment into suitable signals to control the switching and supervisory operations of the telephone network.

A start on this work has been made by the inclusion in the current interface specification of the CALLING INDICATOR lead (Recommendation V.24). This will be used for automatic answering. Equipment on the communicators side of this interface will recognize the ringing current from the exchange and convert it into appropriate signals on this interface lead.

(Question 1/A)

Some Administrations feel that, between the trip of ringing and the activation of the CALLING INDICATOR lead, there should be a time delay during which a pre-recorded voice identification is transmitted. Others think that there is no need for this pre-recorded voice identification of the called station. This is illustrative of some of the problems to be studied.

Note. — See Supplements Nos. 86 to 90 of this Volume.

Question 1/A - point Z. — Use of wide-band circuits

1. There will be a need for data transmission at higher modulation rates than can be obtained over a telephone type circuit and, for this purpose, the use of the group (48 kc/s) and supergroup (240 kc/s) bandwidths has to be considered. Within these bandwidths there exist C.C.I.T.T. standardized pilots.

A proposal has been made to divide the group bandwidth into two bands each approximately 24 kc/s wide and the supergroup bandwidth into two bands each approximately 96 kc/s wide, so as to permit the retention of the existing pilots. Nevertheless it is thought that there will be a requirement for data-transmission systems which will occupy the whole (or a large part) of the group and supergroup bandwidths.

2. Therefore, it would be well to study the possibility when such bands are used for data transmission of:

a) the removal of the pilots from the group and supergroup bands;

b) changing the frequency of the pilots to a position near the edge of the bands.

3. An alternative proposal is the possibility of transmitting data using the full bandwidth of the group and supergroup bands without removing the pilots — i.e. by the provision of narrow band filters to avoid mutual interference between the pilot and the data signal.

It is suggested that this possibility should be jointly studied by Special Study Group A and Study Groups IV and XV. (Question 14/XV of Study Group XV).

4. Special Study Group A also requests a study on the quality and stability of the group and supergroup bandwidths, namely:

a) what would be the attenuation/frequency and delay/frequency characteristics?

b) what would be the level of noise and of any spurious signals that may exist?

c) what would be the level of an unmodulated data carrier signal that may be allowed?

d) what variation of loss, with time, may be expected?

The above problems should be considered for:

a) the case of a point-to-point group or supergroup;

b) the case where a network of such bands may be switched at one or more points.

5. In addition to the use of group and supergroup bands, there may be other ways of providing channels capable of much higher data-signalling rates than can be obtained over a telephone-type circuit.

Study Group XV is studying under Question 33/XV the possibility of transmission systems using pulse-code modulation ; it would be useful if Study Group XV would take into consideration Special Study Group A interest in the possibility of using these transmission systems for data transmission as well as for speech.

For instance, the possibility of direct insertion of the data signal into the quantized signal path should be foreseen.

6. For the higher data-signalling rates used on wider bandwidths, it is suggested that the technical evaluation of these circuits for data transmission and the determination of standard rates should be the subject of further study by Study Group XV, and Special Study Group A. The views of I.S.O. as regards compatibility of input/output equipment for these standard rates will be sought. Where the relevant technical factors will allow, the Study Group feels that the geometric series with factor 2 based on 600 bits per second should be continued in view of the resulting simplification in multiplexing operations.

Note. — See Supplements Nos. 91 to 94 of this Volume.

SUPPLEMENTS TO RECOMMENDATIONS AND QUESTIONS CONCERNING DATA TRANSMISSIONS

SUPPLEMENTS TO RECOMMENDATIONS AND QUESTIONS CONCERNING DATA TRANSMISSION

Contributions received during 1961-1964 which are published on account of their special interest

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2	Working Party on "New alphabet"	Report of October 1964	C	88
3	F. R. of Germany	Five-unit alphabet with protected figures	C	103
4	F. R. of Germany	Tests at 50 bauds over the telex network	D	104
5	N.T.T.	Data transmission over voice-frequency telegraph channels at 200 bauds	D	111
6	F. R. of Germany	Measurement on data transmissions over the telex network at 200 bauds, 1200 bauds and 2400 bauds	D	113
7	F. R. of Germany	A data-transmission system over the telex network	E, I	120
8	Chile Telephone Co.	Protection for the data transmission over the telex network	E, I	122
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10	Canada	180/200-baud telegraph-type switching service for data transmission	H	141
11	F. F. of Germany	200-baud duplex system over switched telegraph network	H	143
12	Special Study Group A	Report on error-control system on the telex network	I	145
13	France	Protection against errors by information feedback	I	152
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15	K.D.D.	Send-back error detection method	I	158
16	United Kingdom	Loop-error detection	I	164
17	Netherlands	Data transmission over isochronous systems and error control	I	169
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19	American Cable and Radio Co.	Combined operation of a 1200-baud data channel and 50-baud telegraph channels in a single voice-baud	J	174
20	Sweden	Data-transmission test over leased circuits	L, O, U	177
21	Sweden	Data-transmission test over switched circuits	L	209
22	Chile Telephone Co.	Measurements on switched and leased telephone lines at 250, 800 and 1000 bauds	L	215
23	United Kingdom	Test for data-transmission systems on telephone-type circuits	L	226
24	A.T.T.	Referential documents for the use of the telephone network for data transmission	L, U	236
25	N. V. Philips T.I.	Data-transmission tests on leased telephone circuits	L	236
26	France	Data-transmission tests on telephone circuits in the general switched network	L	242
27	France	Data-transmission tests on the switched telephone network at the data-signalling rates higher than 1200 bits/second	L, N, O, U	247
28	F. R. of Germany	Measuring the error statistics of switched telephone lines (800 bauds)	L	265
29	F. R. of Germany	Data-transmission tests on the German telephone network, in co-operation with I.B.M. Germany	L, U	278
30	F. R. of Germany	Results of a test at 1200 bauds over switched telephone circuits	L	305
31	F. R. of Germany	Report on tests by Felten and Guilleaume (1200 bauds)	L, U	306
32	Netherlands	Tests of data-transmission systems via telephone-type circuits	L	321
33	Denmark	Data-transmission tests on switched telephone circuits	L	322
34	I.B.M. (W.T.E.C.)	Contribution on data transmission	L, R	328
35	N.T.T.	Data-transmission tests over telephone circuits	L, O, U, V	340
36	N.T.T.	Data-transmission tests over point-to-point telephone circuits on coaxial cables	L, U	355

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37	I.B.M. (W.T.E.C.)	Data-transmission test on a multipoint telephone network	L, U	361
38	N.T.T.	Data-transmission tests over point-to-point telephone circuits on radio-relay links	L, U	377
39	M.I.T. (U.S.A.)	Transmission testing on forward-propagation tropospheric scatter circuits	L, U	395
40	Chile Telephone Co.	Error measurements at 250 bauds	L, R	418
41	Special Study Group A	Report on test results over switched telephone circuits	L	426
42	United Kingdom	Compatibility of modems and the telephone signalling systems	M	430
43	Chairman of Study Group XI	Compatibility of data-transmission methods and telephone signalling systems	M	433
44	Sweden	Advantages and disadvantages of the various forms of modulation	M	437
45	Special Study Group A	Data transmission at modulation rates above 1200 bauds	O	440
46	I.B.M. (United States)	Phase reversal system for 600- to 2400-baud operation	O	444
47	I.B.M. (United States)	A vestigial sideband phase reversal modem	O	453
48	United States	An AM vestigial sideband data-transmission set	O	463
49	Bell Telephone Laboratories	Serial transmission at 2000 and 2400 bits per second	O	470
50	M.I.T. (United States)	Sequential decoding for high-rate data transmission	O	480
51	Siemens & Halske	Modulation methods	O	484
52	Telefunken	Quaternary phase modulation	O	489
53	Lenkurt Electric Co.	System with duobinary coding	O	492
54	United Kingdom	The need for a parallel data transmission system on switched telephone circuits	Q	501
55	Siemens & Halske	Modems for parallel transmission over telephone circuits	Q	503
56	Standard Electrik Lorenz	Parallel transmission on switched telephone circuits	Q	505
57	A.T.T.	Low-speed parallel data sets	Q	509
58	A.T.T.	Parallel data-transmission up to 75 characters per second	Q	513

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59	Chile Telephone Co.	Parallel mode data transmission	Q	523
60	Special Study Group A	Report on parallel transmission of data	Q	529
61	Special Study Group A	Report on methods of error control	R	535
62	I.B.M. (United States)	Study of error control systems	R	545
63	F. R. of Germany	Block checking by means of a feedback shift register	R	552
64	F. R. of Germany	Efficiency of error-detection arrangements	R	554
65	Chile Telephone Co.	Error-correction arrangements	R	570
66	Sweden	Control of errors for data transmission on switched telephone connections	R	577
67	United Kingdom	Control of errors for data transmission on switched telephone connections	R	578
68	Chile Telephone Co.	Medium speed data-transmission system on the telephone network	R	584
69	F. R. of Germany	Control of errors	R	589
70	Olivetti	Block-error detection	R	592
71	Philips T.I.	Phase distortion	S, U	599
72	Study Group III of the C.C.I.R.	Measurements of path time delay differences on radio links	S, U	600
73	United Kingdom	Phase distortion and losses between subscribers	S, T	603
74	A.T.T.	Specifications for lines used for data transmission	U, V	608
75	United Kingdom	Line characteristics for data transmission	S, T, U	610
76	United Kingdom	Effects of transmission time delay and switching discontinuities	U	613
77	Special Study Group A	Noise limits for data transmission	U, V	616
78	L. M. Ericsson	Influence of some transmission defects on binary data systems	S, U	617
79	Sweden	Effect of white noise on data transmission	U, V	640
80	Australia	Noise measurement	U, V	649
81	M.I.T. (United States)	Standard performance tests for data modems	V, W	650

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83	Bell Telephone Laboratories (U.S.A.)	Digital data-transmission test sets	W	666
84	United Kingdom	Binary-digital testing methods	W	672
85	A.T.T.	Disablement of echo suppressors	X	674
86	United Kingdom	Automatic originating and answering of calls in data transmission	Y	689
87	F. R. of Germany	Fully automatic originating and answering of calls	Y	692
88	France	Data transmission equipment with automatic response	Y	693
89	Philips T.I.	Data transmission via the public telephone network	Y	697
90	A.T.T.	Automatic calling programme for switched data communication services	Y	700
91	Western Union	Broad-band switching service	Z	711
92	United Kingdom	Use of circuits of very large bandwidth	Z	713
93	Chile Telephone Co.	Broad-band circuits (48 kc/s)	Z	715
94	A.T.T.	Data transmission systems at 40.8 kilobits per second	Z	717

Supplement No. 1

UNITED KINGDOM. — *(Extract from contribution COM Sp.A/No. 5 — July 1961)*

UNITED KINGDOM POST OFFICE DATA TRANSMISSION SURVEY

1. *Introduction*

1.1 From the outset it was apparent that the lack of general information about line-transmission possibilities, likely facilities and techniques in the computing field would, to a considerable extent, frustrate efforts to obtain sound information about users' requirements. Furthermore, a market research, which normally relies on a sample of potential users, appeared to be somewhat premature, because of the difficulties of establishing the eventual market for computing machinery and of making enquiries about a semi-technical subject of other than informed organizations.

1.2 Accordingly, it was decided, as a first stage, to make contact with computer users and manufacturers, etc., in order to engender interest in the subject and, as a second stage, to issue a data-transmission survey to all known computer users and potential users in order to obtain information of likely requirements in a limited but informed field.

2. *Distribution*

330 survey forms, together with suitable covering letters and copies of "Facilities for Data Transmission", were issued. The addressee included banks, insurance, heavy and light engineering, chemical and electrical manufacturing, the distributive trades, oil companies, transport, computer manufacturers (in their capacity as potential users), and appropriate Government Departments. These addressee and the main answers are analysed by type of interest in Table 1.

3. *General comments*

3.1 The response to the survey (238 replies—72%, see Table 2) was good and represented a high level of interest in the subject. This interest has been emphasized by the number of enquiries concerning the use of line facilities which have been, and continue to be, made during distribution and subsequent to the return of the survey forms. These enquiries are, in the main, of a general, unspecific character, relating to forward planning of computer installations. It seems reasonably certain, however, from their nature, that many will result in the use of lines for data transmission within about five years.

3.2 Of the 238 replies to the survey (see Table 3), 52 (22%) gave an unqualified "No" to the question "Will line facilities be required for data transmission ?". The remainder, 186 (78%), were divided between 111 (47%) expressing the opinion that "line facilities would be required",

and 75 (31%) that it was "too early to say". However, with the latter, there is a presumption that many of these will ultimately say "Yes". This 31% generally qualified their replies by stating that further information would become available by a certain time—see Table 3, Note 2.

3.3 Of the 111 replies agreeing that line facilities were likely to be required, 26 (23%) stated that the requirements were definite; the remainder—85 (77%)—were tentative. An analysis of dates of initial requirements is given in Table 4.

4. *Transmission rate*

4.1 Table 5 gives details of the character transmission rates required in the 111 cases mentioned in para. 3.3. 125 requirements are listed, because some projects called for more than one rate. It was impossible to convert the figures quoted to baud speeds owing to the lack of answers to a question concerning the code to be used and lack of information about the error-control redundancy introduced by the terminal transmission apparatus that will ultimately be employed in particular cases.

4.2 There are, however, two fairly clear areas of requirement, viz., for telex or private telegraph circuits (say) up to 10 (quoted) characters per second—32%—and for telephone networks or private circuits, say, 25-200 (quoted) characters per second—33%. These services would apparently meet 65% of the requirements. The even division between telegraph and speech circuits is not thought to be too significant because, in discussion with interested firms, etc., there have been many marked expressions of preference for the higher-speed services. Ultimately, however, much will depend on relative cost, the volumes of data and the time available in which to effect transmission—questions to which no satisfactory answers are yet available.

4.3 Few survey forms were completed sufficiently to enable a full and detailed pattern of users' requirements to be constructed. Accordingly, when compiling Table 6 showing the type of service likely to be required, viz., telegraph, speech, network or private circuit, it has been necessary to take into account traffic volumes and intentions, where stated, supplemented by information obtained during discussions, etc.

5. *Error rates*

5.1 The minimum acceptable error rates required from the transmission channel and terminal apparatus are summarized in Table 7 in respect of the 111 cases for which facilities may be required. It is noteworthy that these figures appear reasonable and are probably obtainable by means of suitable error-control systems.

5.2 The replies to a question about users' preferences concerning error detection or correction were not very helpful. In most cases, error detection appeared to satisfy the requirements, but the majority expressed preference for correction if the increased cost was not too great.

6. *Duplex transmission*

From Table 8 it will be seen that 26 (23%) replies stated that simultaneous data transmission in both directions would be required. However, bearing in mind that the terminal equipments

currently undergoing examination utilize the return channel for supervisory purposes, this figure probably has no particular significance other than as regards the provision of additional channels. 78 replies (70%) stated that a duplex transmission facility was not required.

7. *Method of data reception*

98 (89%) of the answers—see Table 9—envisage that data would be received at the computing centre on to paper tape, magnetic tape or punched cards. In only six cases (5%) was it visualized that direct connection between line and computer might be required and in two of these cases this was only a possibility. As might be expected, three of the six replies involved computer interrogation systems for airline seat reservation purposes.

8. *Conclusion*

8.1 The analyses in this paper have demonstrated that, on present evidence, there is a mixed demand which, in terms of transmission rate, may possibly be satisfied by existing telegraph and speech services. Acceptable error rates appear capable of attainment by means of suitable terminal equipment, and error correction is generally preferred, if obtainable at reasonable cost. But, failing this, error detection would be acceptable in most cases.

8.2 This paper gives the main conclusions to be drawn from the PO Data-Transmission Survey. It would, of course, be possible to relate the information in the accompanying tables in a variety of ways. Most of the remaining questions of significance were either not answered or the answers were so vague, indefinite or varied as to suggest that analyses would be difficult and unlikely to lead to results in keeping with the effort needed to obtain them. Accordingly, there are no plans for further analysis of the survey at present.

Data-transmission survey

TABLE 1

Classification by industry of survey forms distributed

	Order, using standard industrial classification	Reply not received	"Are line facilities required?"				Total forms distributed a+b+c+d+e	Percentage distribution of cols. d + e
			"No"	"Too early to say"	"Yes"			
					Tentative	Definite		
		a	b	c	d	e		
VI	Engineering, shipbuilding and electrical goods manufacturing	11	2	9	10	9	41	17
IV	Chemicals and allied trades — (manufacturing and preparation)	3	2	7	8	3	23	10
VII	Motor vehicles and aircraft manufacturers	8	2	2	9	1	22	9
XIII	Food, drink and tobacco manufacture and preparation	17	7	10	9	—	43	8
XXI	Insurance, banking, finance	3	8	7	5	4	27	8
XXII	Public administration and national defence	4	9	7	6	3	29	7
XIX	Transport, communications	12	3	1	7	1	24	7
V	Metal manufacturers	4	2	4	6	—	16	5
XVII	Civil engineering, building and contracting	1	1	3	2	3	10	4
III	Cement, brick, pottery and glassware producers	1	—	3	4	—	8	4
XVI	Miscellaneous manufacturing industries (rubber, plastic goods, etc.)	2	—	1	3	—	6	3
XVII	Gas, electricity and water (public utilities)	6	—	7	3	—	16	3
XX	Distributive trades (including merchants, brokers, dealers)	4	3	4	2	1	14	3
X	Textile manufacturers	—	—	1	1	1	3	2
XII	Clothing manufacturers	—	1	1	2	—	4	2
XV	Paper manufacturing, printing and publishing	2	5	—	2	—	9	2
XXIII	Professional services	4	3	—	2	—	9	2
XXIV	Entertainment, sports, hôtels	5	—	1	2	—	8	2
VIII	Metal goods manufacturers	5	2	6	1	—	14	1
IX	Precision instruments	—	2	—	1	—	3	1
XIV	Furniture manufacture	—	—	1	—	—	1	—
	Totals	92	52	75	85	26	330	100

TABLE 2

General

Survey forms completed	238	72%
,, ,, not returned	92	28%
,, ,, issued	330	100%

TABLE 3

Line transmission requirements

Organizations requiring line transmission	111	47%
Organizations not requiring line transmission	52	22%
"Too early to say"	75	31%
	238	100%

Note 1. — Of the 111 replies stating that line facilities may be required, 26 (23%) were definite requirements; the remainder—85 (77%)—were tentative only.

Note 2. — The 75 replies stating "Too early to say" also mentioned that more definite information might be available as follows:

Within one year	16	21%
,, two years	17	23%
,, three ,,	10	13%
,, five ,,	5	7%
No period given	27	36%
	75	100%

TABLE 4

Date initial requirement expected to mature

These answers related to the 111 replies that line facilities may be required:

Required by 1961	24	(16)	21%
,, ,, 1962	17	(6)	15%
,, ,, 1963	13	(1)	12%
,, ,, 1964	4		4%
,, ,, 1965	3		3%
,, ,, 1966-68	3	(1)	3%
Do not know	42	(2)	38%
Existing private circuits to be used	5	(4)	4%
	111	(26)	100%

Note. — The bracketed figures refer to the 26 definite projects; see Table 3, note 1.

TABLE 5

Transmission rate

6-7 characters per second	34	(6)	27%
10 ,, ,, ,,	6	(4)	5%
20 ,, ,, ,,	4	(1)	3%
25-50 ,, ,, ,,	11	(1)	9%
50-100 ,, ,, ,,	22	(7)	18%
200 ,, ,, ,,	4	(2)	3%
300 ,, ,, ,,	1		1%
400 ,, ,, ,,	1		1%
500-2500 ,, ,, ,,	3	(1)	2%
Dependent on relative costs	4		3%
No speed quoted	35		28%
	125	(26)	100%

Note 1. — In Table 5, 125 replies are quoted in respect of 111 cases, because some organizations visualize using more than one rate—e.g., in the case of a service bureau, small volumes of data might be routed over telex and larger volumes over the public telephone network.

Note 2. — The bracketed figures refer to the 26 definite projects.

TABLE 6

Assessment of type of circuit

Telex	12	(3)	9%
Public telephone network	19	(7)	15%
Private circuits, Tariff H	18	(5)	14%
,, ,, ,, D or E	25	(9)	20%
Broad-band circuits	3	(1)	2%
Not possible to assess	51	(5)	40%
	128	(30)	100%

Note 1. — This assessment is based on traffic volume (where stated), and stated intentions supplemented by personal knowledge obtained during discussions, etc.

Note 2. — 128 requirements are quoted in respect of 111 cases, because some appear to justify the use of more than one service.

Note 3. — The bracketed figures refer to the services likely to be used for the 26 definite projects.

TABLE 7

Over-all minimum acceptable error rate (excluding breakdowns)

1 character in 100 000	36	(8)	33%
1 ,, ,, 250 000	3	(2)	3%
1 ,, ,, 500 000	19	(4)	17%
1 ,, ,, 1 000 000	28	(8)	25%
1 ,, ,, 10 000 000	8	(1)	7%
No figure quoted	17	(3)	15%
	111	(26)	100%

Note. — The bracketed figures refer to the 26 definite projects.

TABLE 8

Simultaneous transmissions in both directions

Facility required	25	23%
Facility not required	78	70%
Question not answered	8	7%
	111	100%

TABLE 9

Method of data reception

On paper tape, magnetic tape or punched cards	98	89%
Direct into computer	6	5%
Question not answered	7	6%
	111	100%

Supplement No. 2

JOINT WORKING PARTY ON NEW ALPHABET. — *(Extract from report GM ALP/No. 6 — October 1964)*

REPORT ON THE MEETING OF THE WORKING PARTY ON THE NEW ALPHABET HELD IN GENEVA FROM 6 TO 9 OCTOBER 1964

SECTION I — PRINCIPLES FOR THE STUDY OF A NEW ALPHABET

The Working Party noted that the new telegraph alphabet must first of all be capable of meeting the requirements of private users on leased circuits and of users of data transmission by means of connections set up by switching on the general telegraph network or the telex network.

This alphabet for data transmission will be used to the exclusion of any other alphabet that might be better adapted to the special needs.

The new alphabet is not intended to replace Alphabet No. 2. It is a supplementary alphabet for the use of those who might not be satisfied with the more limited possibilities of Alphabet No. 2.

It is not to be expected that this alphabet will be used for the general public service before a very long time. The general opinion is that International Alphabet No. 2 is satisfactory for the present requirements of the public telegraph service and, in most cases, for the telex service.

A study of the anticipated requirements in the distant future for the general service was started in 1957. After the original requirements had been compressed in an attempt to reach agreement with I.S.O., a C.C.I.T.T. draft was prepared for the information of the IIIrd Plenary Assembly in June 1964 (document AP III/29). The IIIrd Plenary Assembly approved the method hitherto adopted and confirmed that the study should be continued on the same lines (document AP III/122).

SECTION II — I.S.O. DRAFT 7-BIT CODE

II.1 General

The Working Party based its discussions on the third draft 7-bit code prepared by I.S.O. Technical Committee 97 at its meeting in New York in May 1964. The final text of this preliminary draft is given in Annex 1 to this report. The Working Party considered that this I.S.O. draft provided a good basis for study.

It therefore began by studying the alphabet and code proposed by I.S.O., without at that stage considering protection against errors or start-stop operation : the code thus reduced to its units of pure information will henceforward be known as the "naked code".

II.2 Study of the "naked" code

II.2.1 *Switching signals for the general network*

The question arose as to whether combinations in this code should not be reserved for control signals in switching operations on the general network.

The Working Party's reply is in the negative for the following reasons :

i) these combinations would be ineffective when switching is made on the telephone network ;

ii) they would be useless in point-to-point or multipoint line working ;

iii) in a switched telegraph network, the present technique uses signals called "permanent space" and "permanent mark" signals (or combinations of these "space" and "mark" signals) to control switching operations, the word "permanent" signifying that the length of a constant polarity signal of this type clearly exceeds the length of the cycle of a code signal.

The transmission of these signals, which are easily emitted without using a semator (or generator of coded signals), presents no problem when transmission is by a start-stop code over the whole chain of circuits from modem to modem, as the risk that a clear signal (permanent space signal) will be simulated by the code signals is eliminated in principle by the presence of a "stop" element with "mark" polarity at the end of the cycle of each code signal.

There is a difficulty, on the other hand, in case of synchronous transmission, where two problems may arise :

First case. — Synchronous transmission proceeds from modem to modem (e.g. transmission of data blocks) ; if connection has been established by switching, a series of successive "all space" combinations (combination 0/0) will simulate the clearing signal ; inversely, and always in the case of switched connection, a series of "all mark" combinations (combination 7/15) will simulate the condition "circuit held, but idle" ("permanent stop").

These difficulties would be overcome by using an odd parity control on the "mark" elements in the naked code, a "mark" unit being added to the 0/0 signal and a "space" unit to the 7/15 signal.

Second case. — Transmission is start-stop at the beginning and end of the connection, but the connection (point-to-point) to be established by switching must use an intermediate synchronous system. This system removes the start element and the stop element from the start-stop codes. From that moment, there is no means of distinguishing the "permanent space" and "permanent

mark" signals from the 0/0 and 7/15 combinations, respectively. For a synchronous system, which would obviously have the advantage of simplicity, and in which the seven-unit combination of the new "naked" code would be transmitted without modifications, the following two possibilities only would exist :

— the synchronous system would treat these 0/0 and 7/15 combinations as true start-stop signals, in which case all permanent polarity input periods would at the output end be transformed into a series of signals, thus necessitating special arrangements, in addition to other drawbacks, where error correction devices are in use at both ends ;

— the synchronous transmission system would consider such combinations as corresponding only to a permanent polarity state at the input end, and at the output end would systematically reproduce that permanent polarity throughout the length of the appropriate cycles. This is equivalent to admitting that the synchronous section of the connection abolishes the two start-stop signals for the combinations under consideration, even if they occur at the input end. As in the previous case, but for quite a different reason, special measures would then have to be adopted to leave out of account signals thus cancelled particularly when error-correction devices are put into operation in users' installations. The need for such measures derives from the 7/15 "delete" combination in particular, which there is a danger of finding frequently on punched tapes.

In communications operated with protective devices against transmission errors from end to end it is necessary that the transmission channel, including the sections of any synchronous channels, should restore to the receiving end the same number of signals, no more and no less, as were originally emitted.

It should be noted that, where the naked code is used on a channel section operated with a synchronous system which itself includes an error-correction device by automatic repetition, use of the signal RQ (request to repeat) may pose a similar problem.

It must also be remembered that communications operated by cypher code should be able to use all possible combinations in the alphabet.

To avoid these difficulties, all existing synchronous systems used in start-stop communications, whether they themselves are protected or not, are arranged so as to reproduce the permanent space (or mark) polarities of the start-stop communication by means of alpha and beta combinations, which are obtained from the particular code of these systems. Like these, future synchronous systems will have to be designed to supply equivalent combinations which are not subject to confusion with any combination in the alphabet used by terminal equipment.

The Working Party therefore considered that the use of two combinations of the seven-unit code to facilitate transmission of the two permanent polarity conditions on synchronous channel sections was inadvisable.

It was, however, borne in mind that the permanent space polarity could cause the 0/0 combination to be received if no special precautions were taken in the start-stop apparatus receiving that polarity (case of a clearing signal). The "all space" (0/0) NUL combination must therefore be left unallocated for the present.

II.2.2 *Diacritic signs*

The allocation of combinations for the five diacritic signs previously requested by the C.C.I.T.T. was considerably changed in the third I.S.O. draft 7-bit code. The I.S.O. representative explained the reasons for these changes to the members of the Working Party. These are based mainly

on the fact that three non-diacritic signs (i.e. with normal printing progression) can also be used to accent capital or small letters ; when such signs are used as diacritic signs and are printed above, to ensure that they are suitably placed by reference to the letter, the signal "BS" (Backspace) will be used, as provided for in the alphabet.

These signs are the apostrophe, upwards arrow (mathematical sign) and quotation marks, which become respectively the acute accent, the circumflex accent and the diaeresis when used as diacritic signs.

Certain delegates did not particularly favour the method of printing above with backspace for diacritic signs, and thought that the "Backspace" mechanism was more complicated than that of the forward stop with diacritic signs. The Working Party, however, admits that apparatus using the 7-bit code, with all its possibilities, will have to be provided with a backspace.

Should the use of purely diacritic signs be retained, discussion shows that two methods are used by different countries to accent letters :

i) the letter is first printed, then the diacritic sign. This generally means that diacritic signs are inserted in the printing mechanism, one step back, since there has been a step forward after the letter to be accented has been printed.

ii) First the accent is printed, without any forward movement, then the letter is placed underneath.

These two methods involve two types of apparatus which produce printing which does not match up and, even if a single apparatus could be designed, there might be many faults in accentuation due to acquired habits. This is therefore another reason for giving favourable consideration to the I.S.O. proposal. In the case of accentuation by using signs which are not pure diacritics —i.e. in the normal printing position—this incompatibility is avoided by using top printing. The signal "BS" (Backspace) need only be transmitted between the letter and the accent or between the accent and the letter, at the choice of the operator, to have in any case a correct printing of the accented letter.

Finally, the control procedures conforming to the I.S.O. proposal were adopted, with the following observations :

In view of their position in the code table, the signal 2/2 (quotation marks) and the signal 2/7 (apostrophe), if they are used with the signal 0/8 (backspace), can provide diaeresis and acute accents for all international communications, without taking any account of special agreements concerning signals reserved for national uses. On the other hand, as the other signals can produce the grave accent, the circumflex accent and the tilde are indicated only as being characters preferred for international use, but which may be possibly replaced by characters for national use (by notification—see under II.2.6).

This arrangement cannot of course constitute an obstacle to the use of combinations reserved for national use in printing with one signal for an accented capital or small letter such as the letter ä, ö and ü for instance, in the Germanic languages. In the internal traffic of the countries concerned, this allocation would facilitate operations by avoiding the transmission of a sequence of three signals, including the signal BS (Backspace).

II.2.3 *Special case of the sign "tilde" (~)*

The I.S.O. suggested that the circumflex accent should also represent the tilde. Following discussion, during which it was confirmed that some countries needed both signs, the Working Group preferred the two signs to be kept separate. Box 5/12 of the code table, made available by the abandonment of the second currency sign CS-2 (see II.2.5), would be allotted to the tilde.

II.2.4 *Carriage return and line feed*

The possibility envisaged by the I.S.O. of choosing between use of the two signals CR (Carriage Return) and LF (Line Feed) or of one signal NL (New Line) has once more been rejected.

The two-signal method is more elastic, since it makes possible corrections by top printing if required, and also underlining from the beginning of the line. More important still, the suggested variant would create an incompatible situation by making communication impossible between apparatus of different designs. Communication with stations of the operating Administrations in particular would be impossible for apparatus working with the signal NL.

It follows that note (I) of the I.S.O. Table should be deleted, use of the signal NL being inadmissible.

II.2.5 *Currency signs*

The I.S.O. draft provided for two currency signs : CS-1 (square 2/4) and CS-2 (square 5/12). It was thought to use these signs for the written dollar ($) and pound (£) signs. However, all that is really needed by those who use data-processing systems is a sign showing that the following (or preceding) figures refer to an agreed currency unit, specified elsewhere if necessary. For communications between computers, or between a computer and a teleprinter, a general currency sign CS is therefore sufficient.

Box 2/4 is reserved for the CS signal, the graphical sign for which is ¤. The sign CS-2 is abandoned.

II.2.6 *Signals reserved for national use*

The C.C.I.T.T. had requested five possibilities in capital letters and the corresponding five possibilities in small letters. The I.S.O. draft reduced these possibilities.

The Working Group proposes the following solution :

i) Transfer the grave accent from box 5/15 to box 4/0 and the underlining sign (_) from box 4/0 to box 5/15 ;

ii) allocate boxes 7/11, 7/12, 7/13 and 7/14 for national use only, as provided in the I.S.O. draft ;

iii) the following boxes are, for the normal international service, given the following "preferential" allocations :

— box 4/0 = grave accent (`)
— box 5/11 = left square bracket ([)
— box 5/12 = tilde (~) (Ref. II.2.3)
— box 5/13 = right square bracket (])
— box 5/14 = circumflex accent (^)
— box 6/0 = commercial at (@)

However, if an Administration wishes to use one of these last six combinations with another meaning, and for its national service only, it may do so, but must notify the General Secretariat of the Union of its decision to allot the signal for national use. The use of this signal for international relations with that country will not then be allowed, except by special arrangement.

II.2.7 *Duodecimal numbering*

The possibility of an alternative use of the combinations in boxes 3/10 for : or 10, and 3/11 for ; or 11 is not being retained. These boxes are used solely for the colon (:) and semi-colon (;).

II.2.8 *Tabulating signs*

It is made clear that the manner in which the signals for :

— vertical tabulation (VT) in box 0/11,

— horizontal tabulation (HT) in box 0/9,

— form feed (FF) in box 0/12,

have to act on the receiving apparatus will have to be a matter for prior agreement between the users.

II.2.9 *"Backspace" signal*

It is specified that the use of this signal is limited to printing. It cannot control a backspace on intermediate recording devices (perforated tape, magnetic tape, etc.).

II.2.10 *"Shift out, Shift in, Escape, Start of Special Sequence, and Data Link Escape" signals*

These are the signals in boxes 0/14(SO), 0/15(SI), 1/11(ESC), 1/10(SS) and 1/0(DLE).

The description of these signals' functions in the I.S.O. draft lacks precision and is sometimes contradictory, the reason being that the standardization of them by I.S.O. is not finished. For data transmission these five functions will be needed but the role allotted to each has yet to be settled.

The Working Party notes that, for the transmission of messages, similar needs to those envisaged by I.S.O. may arise in the future. For instance, the combinations "Shift out" and "Shift in" might be used, as proposed by the Nippon Telegraph and Telephone Public Corporation, to provide a shift-lock giving the possibility of transmitting an alphabet with non-Latin characters (by agreement between the Administrations concerned or between correspondents). Such a possibility might be accepted even on an international communication by prior agreement between the interested parties.

Thus the Working Party agrees to the use of boxes 0/14 and 0/15 for the signals "Shift out" and "Shift in" respectively. The description of these functions is the one given by I.S.O., but without the explanation given for their use combined with special sequences.

The uses of boxes 1/11, 1/10 and 1/0 have been reserved ; their assignment will be decided later by I.S.O. It will have to be possible with the assignment to obtain an inversion which, on one or two columns of the code, will introduce supplementary signs to be standardized with a view to their use by the telecommunication services.

II.2.11 *"Enquiry" signal*

The details supplied by I.S.O. on the definition of the "Enquiry" function (box 0/5) show that this signal is sent as a request for a response from a remote station. The response may include the name of that station.

For operation with switching on the general network, the Working Party considers that the sending of the answer-back code (abbreviated name which may be followed by information on the type of equipment) must be capable of remote control, at least at the beginning and end of a connection. A "Who are you ?" signal is therefore indispensable.

The Working Party considers that the assignment of the signal 0/5 for the remote control of the answer-back code must be indicated with priority. The use of this signal to obtain other responses during an established connection may be allowed, in operating conditions to be defined, for example by combining this signal with the "Escape" signal.

To sum up, the Working Party proposes :

— that the signal corresponding to box 0/5 should signify "Who are you ?" ;

— that this control signal should be defined as follows: "This signal normally causes a response in the receiving equipment. The response includes the identification of the station and the type of equipment." "During a communication set up by switching, or on a specialized network, the meaning of this signal may be changed."

II.2.12 *Signals for the control of auxiliary devices*

As proposed by I.S.O., the signals DC-1, DC-2, DC-3 and DC-4 (boxes 1/1, 1/2, 1/3 and 1/4) will be allotted to the controls of auxiliary devices. A more thorough standardization of the allotment and use of these signals remains to be studied. However, the allotment of the signal DC-4 (box 1/4) to "Stop", as defined in the I.S.O. draft, has already been agreed.

II.2.13 *"Synchronous Idle" signal*

The definition of this signal will be clarified by adding at the end: ". . . between the terminal equipments of the connection" (to avoid the risk of confusion with the maintenance of synchronism between intermediate synchronous equipment on the communication channel).

II.3 Conclusions concerning the naked code

The Working Party is in agreement with the assignments made, apart from those considered in II.2.

The naked code, as amended by the Working Party, is given in Annex 2. The boxes relating to signals on the use of which further information is required show a reference number in brackets. The reference numbers (1) to (9) relate to general notes; reference numbers (10) onward relate to specific notes included in the description of functional characteristics.

SECTION III — BIT ORDER OF TRANSMISSION

Within each binary combination, the bits are identified by b_7, b_6 ... b_1, b_7 being the highest order or most significant bit, and b_1 the lowest order or least significant bit.

If desired, these may be given a numerical significance in the binary system, thus:

Bit identification:	b_7	b_6	b_5	b_4	b_3	b_2	b_1
Significance:	64	32	16	8	4	2	1

Any signal in the naked code table may be identified by its bit pattern, by its column and row numbers, or by its representation in the telegraph two-condition code. For example, the position containing the figure 1 may be identified:

— by its bit-pattern, i.e. 011 0001

— by its column and row numbers, i.e. 3/1

— by its representation in the telegraph two-condition code, i.e., according to Recommendation V.1:

AZZAAAZ
SMMSSSM (M = Mark S = Space)
TRRTTTR (R = Repos T = Travail)

provided the bits are transmitted, in serial transmission, in descending order (high order bit first), or, on the contrary, by:

ZAAAZZA
MSSSMMS (M = Mark S = Space)
RTTTRRT (R = Repos T = Travail)

when bits are transmitted, in serial transmission, in ascending order (low order bit first).

For the telecommunication services, the unit element which occurs first in serial transmission will inevitably be numbered one. When transmission is by low order bit first, the units will thus have the same number as the corresponding bits, unit No. 1 denoting bit b_1, unit No. 2 bit b_2, and so on. When serial transmission is by high order bit first, unit No. 1 may correspond to bit b_7 (or to bit b_8 when a parity bit code is used), unit No. 2 to bit b_6, and so on. In the latter case, difficulties may be expected to arise between telecommunication operators, who will refer to the order number of signal units, and data processors, who will refer to the order number of the bits.

The telecommunication services are therefore in favour of transmitting low order bit first in serial transmission. There is no doubt however that, if transmission by high order bit first offered considerable advantages for processing, a way of getting round the difficulty would have to be devised.

So far as the I.S.O. is concerned, Sub-Committee 6 of Technical Committee 97 passed the following Resolution SC6/53 at its meeting in New York in May 1964 :

"ISO/TC/97/SC6 wishes to express a *preference* for transmitting high order bit first when transmitting the proposed ISO code(s) over serial-serial data-transmission systems" (see Section IV of the present report). "ISO/TC/97/SC6 recommends that a character parity bit (if used) be transmitted last."

Sub-Committee 2 of Technical Committee 97, on the other hand, recommended the use of perforated tape, which requires transmission by *low* order bit first.

Technical Committee 97 decided to forward Resolution SC6/53 to the C.C.I.T.T. and to leave the final decision in this matter to that body.

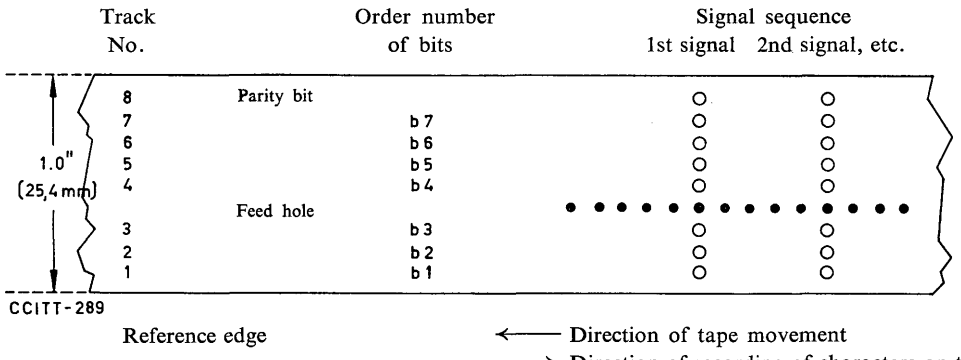
The E.C.M.A. also has not taken a definitive position.

In these circumstances, the Working Party decided by a very large majority in favour of transmitting low order bit first.

Section IV — Layout of the alphabet on perforated tape

With the adoption of serial transmission by low order bit first, the layout on perforated tape as proposed by Sub-Committee 2 of I.S.O. Technical Committee 97 is acceptable to the C.C.I.T.T.

This layout is on the following lines :



The other proposals of the I.S.O. for standardizing the perforated tape to be used for a 7-bit alphabet are shown in Annex 3. The question of the parity check is considered in Section V of the present report.

SECTION V — PROTECTION AGAINST ERRORS

It does not seem feasible to adopt a standard method of protection against errors, as the procedure depends upon the degree of protection required, which varies considerably with the nature of the traffic.

Nevertheless, the protection procedure known as the "parity check", while it cannot be recommended exclusively, will probably be widely used either singly or in conjunction with other methods.

It would be advisable to standardize the parity check method to ensure that, although it is not obligatory, it is applied in a certain way.

In this connection, the Working Party agrees with the I.S.O. that the eighth parity bit should be transmitted after the seven bits containing the actual information, i.e. at the end of the "naked" combination.

The I.S.O. has proposed that the even parity check system should be adopted for the 7-bit alphabet. In choosing the "even" or "odd" parity check, due consideration should be given to the implications for start-stop and synchronous transmission as well as compatibility with switching signals and the use of the 0/0 and 7/15 combinations.

The Working Party therefore believes that this matter should be the subject of a special study.

Notes about the 7-bit table

① The controls CR and LF are intended for printer equipment which requires separate combinations to return the carriage and to feed a line.

For equipment which uses a single control for combined carriage return and line feed operation, the function FE_2 will have the meaning of "New Line" (NL). This requires agreement between the sender and the recipient of the data.

② If 10 and 11 as single characters are needed (for example, for Sterling currency subdivision), they should take the place of the "colon" (:) and the "semi-colon" (;) respectively. These substitutions require agreement between the sender and the recipient of the data.

③ "Reserved for national use." These positions are primarily intended for alphabetic extensions. If they are not required for that purpose, they may be used for symbols and a recommended choice is shown between parentheses in some cases.

④ The graphical representation and the meaning of the two currency signs CS_1 (position 2/4) and CS_2 (position 5/12) may vary according to national use.

⑥ The "number sign" (#) in position 2/3 may have an alternate graphical representation (No.).

⑦ It is acceptable to represent the "tilde" (~) by the "circumflex accent" (^) for international interchange of information in Spanish and Portuguese languages. In Spanish and Portuguese speaking countries, the "tilde" may replace the "circumflex accent" in position 5/14.

⑧ The graphics in positions 2/2, 2/7, 5/14 have respectively the significance of "quotation mark", "apostrophe" and "upwards arrow"; however, these characters take on the significance of "diaeresis", "acute accent" and "circumflex accent" diacritical signs when they follow the "Backspace" character.

The symbol shown in position 5/15 did not receive any denomination other than "grave accent" which corresponds to its use as a diacritical sign.

ANNEX 1

7-bit set table

Coded character sets for information processing interchange

Bits b_7	b_6	b_5	Column →	0	0	0	0	1	1	1	1
				0	0	1	1	0	0	1	1
				0	1	0	1	0	1	0	1
b_4	b_3	b_2	b_1	Row ↓ 0	1	2	3	4	5	6	7
0	0	0	0	0	(TC_0) Null	(TC_7) DLE	Space	0	_	P	(@) ③	p
0	0	0	1	1	(TC_1) SOH	DC_1	!	1	A	Q	a	q
0	0	1	0	2	(TC_2) STX	DC_2	" ⑧	2	B	R	b	r
0	0	1	1	3	(TC_3) ETX	DC_3	# ⑥	3	C	S	c	s
0	1	0	0	4	(TC_4) EOT	DC_4 (Stop)	CS_1 ④	4	D	T	d	t
0	1	0	1	5	(TC_5) Enq	(TC_8) Nack	%	5	E	U	e	u
0	1	1	0	6	(TC_6) Ack	(TC_9) Sync	&	6	F	V	f	v
0	1	1	1	7	Bell	(TC_{10}) ETB	' ⑧	7	G	W	g	w
1	0	0	0	8	FE_0 (BS)	Cncl	(	8	H	X	h	x
1	0	0	1	9	FE_1 (HT)	EM	)	9	I	Y	i	y
1	0	1	0	10	FE_2 (LF) ①	SS	*	: ②	J	Z	j	z
1	0	1	1	11	FE_3 (VT)	Esc	+	; ②	K	([) ③	k	③
1	1	0	0	12	FE_4 (FF)	IS_4 (FS)	,	<	L	(CS_2) ④ ③	l	③
1	1	0	1	13	FE_5 (CR) ①	IS_3 (GS)	–	=	M	(]) ③	m	③
1	1	1	0	14	SO	IS_2 (RS)	.	>	N	^ ⑦ ⑧	n	③
1	1	1	1	15	SI	IS_1 (US)	/	?	O	` ⑧	o	Delete

See notes 1 to 8 on page 96.

ANNEX 2

Proposed seven-bit alphabet based on the third draft I.S.O. proposal as amended by the C.C.I.T.T. Working Party on the New Alphabet

Bits b_7	b_6	b_5	b_4	b_3	b_2	b_1	Ligne / Row	Colonne / Column 0	1	2	3	4	5	6	7
			0	0	0	0	0	(1)	(2)	SPACE ESPACE (27)	(0)	` (4) (5)	P	@ (4)	p
			0	0	0	1	1	SOH (26)	DC1 (6)	!	1	A	Q	a	q
			0	0	1	0	2	STX (29)	DC2 (6)	" (5)	2	B	R	b	r
			0	0	1	1	3	ETX (19)	DC3 (6)	# (34)	3	C	S	c	s
			0	1	0	0	4	EOT (17)	STOP (28)	¤ (35)	4	D	T	d	t
			0	1	0	1	5	WRU (33)	NACK (23)	%	5	E	U	e	u
			0	1	1	0	6	ACK (10)	SYNC (30)	&	6	F	V	f	v
			0	1	1	1	7	BELL SONNERIE (11)	ETB (18)	' (5)	7	G	W	g	w
			1	0	0	0	8	BS (12)	CNCL (13)	(	8	H	X	h	x
			1	0	0	1	9	HT (21)	EM (16)	)	9	I	Y	i	y
			1	0	1	0	10	LF (22)	(2)	*	:	J	Z	j	z
			1	0	1	1	11	VT (32)	(2)	+	;	K	[(4)	k	(3)
			1	1	0	0	12	FF (20)	FS (31)	,	<	L	~ (4) (5)	l	(3)
			1	1	0	1	13	CR (14)	GS (31)	−	=	M	] (4)	m	(3)
			1	1	1	0	14	SO (25)	RS (31)	.	>	N	^ (4) (5)	n	(3)
			1	1	1	1	15	SI (24)	US (31)	/	?	O	— (36)	o	Oblitération Deleted (15)

CCITT-291

GENERAL NOTES ON THE 7-BIT CODE

(1) Combination not to be used.

(2) Reserved combination the use of which will be decided later (proposals to be submitted by I.S.O.).

(3) Reserved for national use. Use of this combination in international service is not allowed.

(4) The signal shown in this case corresponds to the normal international use. However, an Administration may decide to use it with another meaning and for its national use only ; in this case, that Administration will have to notify to the General Secretariat of the Union its decision to take this combination for its national use. The use of this combination will then not be allowed in the international relations with that country.

(5) The graphics in positions 2/2, 2/7, 5/14 have respectively the significance of "quotation mark", "apostrophe" and "upwards arrow" ; however, these characters take on the significance of "diaeresis", "acute accent" and "circumflex accent" diacritical signs when they follow the "Backspace" signal (0/8). The symbol shown in position 4/0 is the "grave accent", diacritical sign ; the symbol in position 5/12 is the tilde, diacritical sign.

(6) Signal reserved for the control and particularly the putting into service of an auxiliary device associated with a data-processing or telecommunication system. A more precise standardization of the allotment and use of this signal is under study.

Description of certain functions or signs

(10) Ack — *Acknowledge*—A signal that is generated at a receiving device to indicate to the sending device that data have been received successfully.

(11) Bell — A signal for use when there is a need to call for human attention ; it may control alarm or attention devices.

(12) BS — *Backspace*—A layout signal whose sole function is to control the movement of the printing position one printing space backward on the same printing line.

(13) Cncl — *Cancel*—A signal used to indicate that the data with which it is associated are in error or are to be disregarded.

(14) CR — *Carriage Return*—A layout signal which controls the movement of the printing position to the first printing position on the same printing line.

(15) Delete — *Delete*—This signal is used to erase or obliterate locally erroneous or unwanted characters in punched tape, or for remote control of the progress of the tape.

(16) EM — *End of Medium*—A control signal which may be used to identify the physical end of the medium, or the end of the used, or wanted, portion of information recorded on a medium. The position of this character does not necessarily correspond to the physical end of the medium.

(17) EOT — *End of Transmission*—A signal used to indicate the conclusion of the transmission of one or more texts (every one of which must be terminated by the ETX character).

(18) ETB — *End of Transmission Block*—A signal used to indicate the end of a block of data for transmission purposes. ETB is used for blocking data where the block structure has no necessary relationship with the processing format.

(19) ETX — *End of Text*—A signal used to terminate a sequence of characters started with an STX character and transmitted as an entity.

(20) FF — *Form Feed*—A layout signal which controls the movement of the printing position to the first predetermined printing line on the next form ; this signal is used only by agreement between correspondents.

FS — *File Separator* (see US).

GS — *Group Separator* (see US).

(21) HT — *Horizontal Tabulation*—A layout signal which controls the movement of the printing position to the next in a series of predetermined positions along the printing line ; this signal is used only by agreement between correspondents.

(22) LF — *Line Feed*—A layout signal which controls the movement of the printing position to the next printing line.

(23) Nack — *Negative Acknowledge*—A signal transmitted by a receiver as a negative response to the sender.

RS — *Record Separator* (see US for definition).

(24) SI — *Shift-in*—The shift-in signal means that the code combinations which follow shall be interpreted according to the standard code table.

(25) SO — *Shift-out*—The shift-out signal means that the code combinations which follow shall be interpreted as outside of the standard code table until a shift-in character is reached ; this signal is used only by agreement between correspondents.

(26) SOH — *Start of Heading*—A signal used at the beginning of a sequence of characters which comprise a machine-sensible address or routing information. Such a sequence is referred to as the "Heading".
An STX character has the effect of terminating a heading sequence.

(27) Space — *Space*—A normally non-printing graphic character used to separate words. It is also a layout character which controls the movement of the printing position, one printing position forwards.

(28) Stop — *Stop*—This device control signal is used to interrupt or turn off ancillary devices.

(29) STX — *Start of Text*—A signal which precedes a sequence of characters that is to be treated as an entity and entirely transmitted through to the ultimate destination. Such a sequence is referred to as "Text". The STX character may be used to terminate a sequence of characters started by a SOH character.

(30) Sync — *Synchronous Idle*—A signal used by a synchronous transmission system in the absence of any other character (idle condition) to provide a signal from which synchronism may be achieved or retained between the terminal equipments of the connection.

(31) US — *Unit Separator—Unit Separator (US), Record Separator (RS), Group Separator (GS)* ; *File Separator (FS)*, are information separators which have a hierarchical relationship (in the order above) with each other, where the unit separator (US) is the least powerful of the set. The information separators follow the path to which they apply.

A "*unit*" cannot include a "file", "group", or "record".

A "*record*" may include (and if it includes, it includes completely) a variable number (none, one or more) of "units". A "record" cannot include a "file" or "group".

A "*group*" may include (and if it includes, it includes completely) a variable number (none, one or more) of "records" and/or "units". A "group" cannot include a "file".

A "*file*" may include (and if it includes, it includes completely) a variable number (none, one or more) of "groups", "records", and/or units".

(32) VT — *Vertical Tabulation*—A layout signal which controls the movement of the printing position to the next in a series of predetermined printing lines ; this signal is used only by agreement between correspondents.

(33) WRU — *Who are you ?*—Signal which causes a response in the receiving equipment ; the response includes the station identification and the type of equipment.

(34) # — This sign denotes Number ; it may have a second graphic representation : No.

(35) ¤ — Currency sign : it indicates that the figures to which it refers represent monetary values (with a unit to be specified, if necessary, by the correspondents).

(36) _ — Sign : underline.

ANNEX 3

Representation of 7-bit character codes on 1″ perforated tape (draft I.S.O. proposal)
(Source : ISO/TC/97/SC/2)

1. *Scope*

This draft proposal covers the following points :

1.1 Width of tape

1.2 Number of tracks

1.3 Reference edge

1.4 Track numbering

1.5 Allocation of bits to track

1.6 Parity control

1.7 Direction of movement and character sequence

1.8 Trailers and leaders

2.1 Width of tape

Only tape with a nominal width of 1.0 in. or 25.4 mm is covered in this draft proposal.

2.2 Number of tracks

2.3 Reference edge

The reference edge shall be that with *three* data tracks between it and the feed track.

2.4 Track numbering

The data tracks shall be numbered consecutively from one to eight. Track number one shall be the track closest to the reference edge so that the feed-hole track lies between tracks three and four.

2.5 Allocation of bits to track

2.5.1 The perforations shall be arranged in up to eight longitudinal tracks on the basis that each track is assigned to its corresponding bit number with one extra for parity.

Note. — When the 6-bit code is represented on the tape, track must remain unperforated.

2.5.2 A hole in a data track shall represent binary one ; the absence of a hole shall represent binary nought.

2.6 Parity control

2.6.1 A parity check bit for each character shall be part of the perforated tape code.

2.6.2 The parity bit shall be placed on track 8 for the 7-bit code and track 7 for the 6-bit code.

2.6.3 The parity bit shall be chosen so as to provide for even parity for the 7-bit code and odd parity for the 6-bit code.

2.7 Direction of movement and character sequence

2.7.1 The direction of movement of tape shall be from right to left as shown in the diagram.

2.7.2 The character sequence shall be from left to right as shown on the diagram.

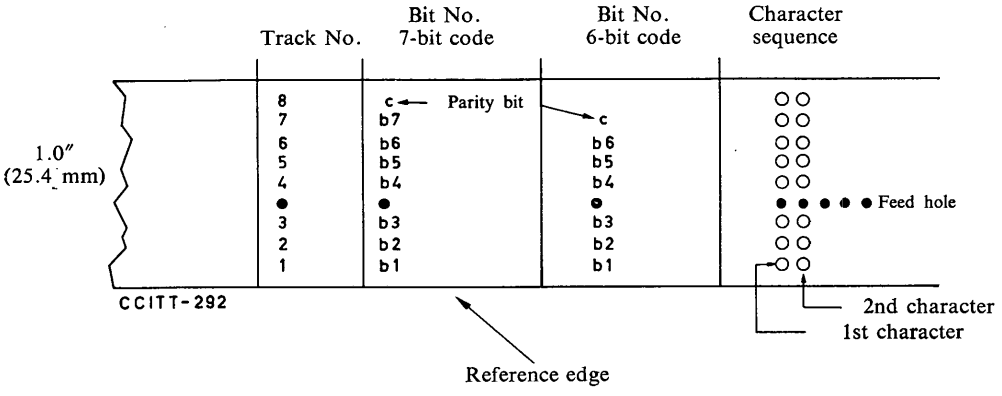
2.8 Trailers and leaders

All leaders and trailers of perforated tapes shall be perforated with feed holes, and, when using the 6-bit code, holes along track 6 also. This has the effect that these leaders and trailers carry the Null character and correct parity is maintained.

3. *Diagrammatic representation*

Points 2.1, 2.2, 2.3, 2.4, 2.5, 2.6 and 2.7 are illustrated in the diagram below.

DIAGRAM



←—— Direction of tape movement (right to left)

——→ Direction of recording of characters on tape (left to right)

Supplement No. 3

FEDERAL REPUBLIC OF GERMANY. — *(Extract from contribution COM Sp. A/No. 14 — September 1961)*

EXAMPLE OF A 5-UNIT ALPHABET WITH PROTECTED FIGURES

Elements 1 2 3 4 5	C.C.I.T.T. Alphabet No. 2		Alphabet for the protection of figures	
	Letters	Figures	Letters	Figures
1 1 0 0 0	A	—	M	.
1 0 0 1 1	B	?	P	0
0 1 1 1 0	C	:	Y	6
1 0 0 1 0	D	Answer back	D	Answer back
1 0 0 0 0	E	3	A	—
1 0 1 1 0	F		U	7
0 1 0 1 1	G		R	4
0 0 1 0 1	H		H	
0 1 1 0 0	I	8	J	Aud. signal
1 1 0 1 0	J	Aud. signal	I	8
1 1 1 1 0	K	(	K	(
0 1 0 0 1	L	)	L	)
0 0 1 1 1	M	.	E	3
0 0 1 1 0	N	,	N	,
0 0 0 1 1	O	9	F	
0 1 1 0 1	P	0	T	5
1 1 1 0 1	Q	1	C	:
0 1 0 1 0	R	4	G	
1 0 1 0 0	S	'	S	'
0 0 0 0 1	T	5	Z	+
1 1 1 0 0	U	7	O	9
0 1 1 1 1	V	=	V	=
1 1 0 0 1	W	2	W	2
1 0 1 1 1	X	/	X	/
1 0 1 0 1	Y	6	Q	1
1 0 0 0 1	Z	+	B	?
0 0 0 1 0	CAR. RET.	CAR. RET.	CAR. RET.	CAR. RET.
0 1 0 0 0	LINE FEED	LINE FEED	LINE FEED	LINE FEED
1 1 1 1 1	LETRS	LETRS	LETRS	LETRS
1 1 0 1 1	FIG	FIG	FIG	FIG
0 0 1 0 0	SPACE	SPACE	SPACE	SPACE
0 0 0 0 0	NOT USED	NOT USED	NOT USED	NOT USED

In the alphabet proposed the 32 combinations are associated with the conventional figure/letter pairs of the International Alphabet No. 2 in such a way that the 10 digits are characterized by three mark and two space elements. The plus and minus signs as well as the function signals LINE FEED, CAR. RET., and SPACE are represented by one mark and four space elements, and the LETRS function signal by five mark elements. This assignment scheme makes it possible to detect transmission errors of these 16 odd number of mark elements. This check is not performed for the corresponding letters.

The assignment of letters to the combinations of this alphabet has been made in a way that the allocation of types according to the International Telegraph Alphabet No. 2 is preserved. If a teletypewriter has to be modified because of a change-over to the alphabet with protected figures the types need not be replaced ; thus the modification effort remains low.

Supplement No. 4

FEDERAL REPUBLIC OF GERMANY. — *(Extract from contribution COM Sp. A/No. 15 — September 1961)*

TESTS CONDUCTED IN THE GERMAN TELEX NETWORK WITH 50-BAUD START-STOP TRANSMISSION

1. *Test set-up*

Two groups of measurements were performed in order to test the German telex network under different conditions.

1.1 *Test transmissions from various subscriber sets to an evaluation centre*

In order to get representative results about the quality of connections in the German telex network, the outdoor personnel of the German Bundespost was asked to transmit a test message from a total of 3007 subscriber sets to the evaluation centre at the Fernmeldetechnisches Zentralamt. Every one of the outdoor officials concerned was provided with a perforated plastic tape containing 1000 characters, i.e. the C.C.I.T.T. test message LETRS, S, CAR. RET., LINE FEED, Q, FIGS, SPACE, 9, repeated several times. This message was chosen because it contains combination No. 29 relatively frequently. With an error in the start element of a character and a subsequent running through of the receiving machine, the machine falls into step again at the latest with the occurrence of this combination.

The transmission on each connection lasted 2.5 min. Perforated tape transmitters provided by the personnel were used ; these transmitters were adjusted for minimum transmission distortion and minimum speed tolerance.

The test connections were subdivided into four categories, depending on the number of V.F telegraph sections they comprised. For each category the number of the necessary test connections

Category	Number and kind of V.F telegraph systems		Number of connections
A (3 sections)	3 AM		1355
B (4 sections)	3 AM	1 FM	804
C (5 sections)	5 AM		745
D (7 sections)	5 AM	2 FM	103

was determined in such a way that the ratio of these amounts is about equivalent to the distribution of the actual connections in the German telex network comprising three, four, five and seven V.F telegraph sections.

Connections of the single categories occur within the German telex network as follows :

Category A : Local exchange — central exchange — central exchange — local exchange

Category B : Subexchange — local exchange — central exchange — central exchange — local exchange

Category C : Subexchange — local exchange — central exchange — central exchange — local exchange — subexchange

Category D : Subscriber for toll service only — subexchange — local exchange — central exchange — central exchange — local exchange — subexchange — subscriber for toll service only.

1.2 *Test transmissions from the evaluation centre into loop connections established by dialling*

In order to enable these tests, cross connections were established within all central exchanges, i.e. there was the possibility to cascade a variable amount of connections between central exchanges by means of dialling the appropriate number in the Fernmeldetechnisches Zentralamt. Tests were conducted over five, six and eight V.F telegraph sections. The tests over five V.F telegraph sections were furthermore intended to check whether the error rate depends also on the length of the sections. All V.F telegraph systems applied amplitude modulation only. The test message again consisted of the same sequence as described in 1.1, the total length, however, was 23 000 characters and more.

2. *Evaluation of the punched test tapes*

In the evaluation centre of the Fernmeldetechnisches Zentralamt the incoming lines were terminated with printing reperforators. Before operation started they were adjusted to a receiving tolerance of at least 38%. The perforated tape transmitters used for the tests described in 1.2 had a maximum transmitting tolerance of 3%.

Evaluation of the received tapes was performed in a comparing device for perforated tapes, which consisted mainly of a transmitter with two heads and the necessary relay, counter, and printer units. The received tape (head 2) was compared element by element with the original tape (head 1), errored mark or space elements being indicated in the counter unit.

The comparison of received text showed that sometimes one or more characters got lost. This points to longer interruptions which, however, are not yet recognized by the system. The German telegraph switching system TW 39 clears a connection only if the start polarity lasts for 600 ms $\pm$ 30%. An average character contains normally 3.5 elements with stop polarity ; therefore an interruption lasting for the total length of one character affects on average only these 3.5 elements. Therefore in the error statistics an interruption of up to 150 ms was counted as 3.5 erroneous elements, an interruption of up to 300 ms as 7 erroneous elements. According to paragraph G of the programme annexed to Question 1/A an interruption of more than 300 ms was disregarded.

All series of errors concerning an errored start element and subsequently to a wrong start of the receiving machine have been counted, because this kind of errors is typical for the start-stop system.

The number of elements transmitted was derived by multiplying the number of characters with the factor 7, because the stop element, lasting 1.5 time slots, is regarded as one element only by the receiving reperforator.

We have used the following definition : An error burst is considered to be terminated when the last errored element of the burst is followed by at least ten correct elements.

TABLE 1

Number of sections of V.F.T.	Number of connections	Number of characters transmitted	Number of faulty characters	Average error rate of telegraph communication	Number of elements transmitted (div. 3×7)	Number of elements in error	Number of sequential errors [1] with x following immediately disturbed elements changed to position						Number of disconnections which caused no printing of characters			Sum of elements changed to position		Number of elements in error div. 7 + div. 14 × 3.5 + div. 15 × 7	Number of transpositions	Average error rate of elements div. 19 / div. 4	Number of error bursts	Number of faulty elements in error bursts
							x = 1		x = 2		x = 3		x = 4		x > 5	< 150 ms	< 300 ms	More than 300 ms						
							Z	A	Z	A	Z	A	Z	A					Z	A					
1	2	3	4	5	6	7	8		9		10		11		13	14	15	16	17	18	19	20	21	22	23
3	1355	1 355 000	38	$2.804\times10^{-5} = 1:35\,600$	9.48×10^{6}	54	14	12	12	10	4	2	—	—	0	5	1	—	30	24	78	10	$8.28\times10^{-6} = 1:120\,000$	13	37 = 68 %
4	804	804 000	44	$5.47\times10^{-5} = 1:18\,300$	5.63×10^{6}	71	22	24	9	13	—	3	—	—	0	1	—	—	31	40	74	9	$13.2\times10^{-6} = 1:75\,700$	9	50 = 70 %
5	745	745 000	50	$6.71\times10^{-5} = 1:14\,900$	5.22×10^{6}	81	22	17	10	12	5	7	2	6	0	6	—	—	39	42	102	14	$19.5\times10^{-6} = 1:51\,300$	14	65 = 80 %
7	103	103 000	20	$19.4\times10^{-5} = 1:5\,150$	0.721×10^{6}	23	3	14	5	1	—	—	—	—	0	1	—	—	8	15	26	1	$36.8\times10^{-6} = 1:27\,200$	2	6 = 26 %
5	108	1 003 694	—	—	7.03×10^{6}	92	54	11	7	11	5	4	—	—	0	4	1	1	66	26	113	13	$16.1\times10^{-6} = 1:62\,200$	13	37 = 40 %
5	108	1 045 518	—	—	7.32×10^{6}	219	55	54	36	30	17	19	3	5	0	7	1	1	111	108	250	36	$34.3\times10^{-6} = 1:29\,200$	21	112 = 51 %
6	65	628 883	89	$14.1\times10^{-5} = 1:7\,100$	4.40×10^{6}	160	49	27	22	18	8	13	4	4	0	1	1	2	94	66	170	28	$27.1\times10^{-6} = 1:36\,900$	25	124 = 77 %
8	12	203 715	28	$13.7\times10^{-5} = 1:7\,230$	1.426×10^{6}	42	8	5	5	7	5	7	—	—	0	3	—	1	21	21	52	8	$37.0\times10^{-6} = 1:27\,000$	9	37 = 88 %

[1] A sequential error is defined so that the disturbed elements are following immediately one by one. The sequence is finished at the last disturbed element.

3. *Test results*

The results of the tests described in 1.1 and 1.2 are shown in the attached tables and figures. The following relations were found :

The number of errored characters and elements strongly depends upon the number of the V.F telegraph sections cascaded, eight V.F telegraph sections cause an error rate approximately five times greater than three V.F telegraph sections (Figs. 1 and 2). The number of falsified space elements approximately equals the quality of errored mark elements ; interruptions lasting longer occur relatively seldom. Furthermore, it is apparent that an increased number of V.F telegraph sections leads to an increased length of sequences with consecutively falsified elements ; also the occurrence of falsified elements within the error bursts rises, whilst the percentage of single element errors outside the error bursts declines (Table 1).

The main result of the tests concerned with the investigation of the influence of the length of the V.F telegraph sections was that the probability of errors and even the number of interruptions increases with the length. This leads to the conclusion that part of the faults is caused by the work of the maintenance staff on the transmission equipment (Table 1).

The structure of the error bursts may be described as follows :

The quantity of element errors contained in an error burst increases with the number of V.F telegraph sections ; the length of the error burst also depends to a certain extent upon the number of V.F telegraph sections (Table 2).

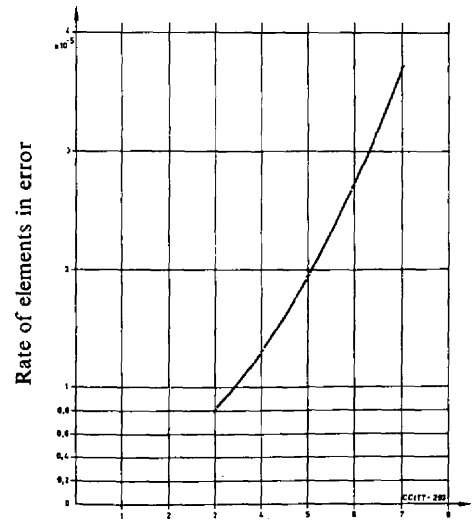


FIGURE 1

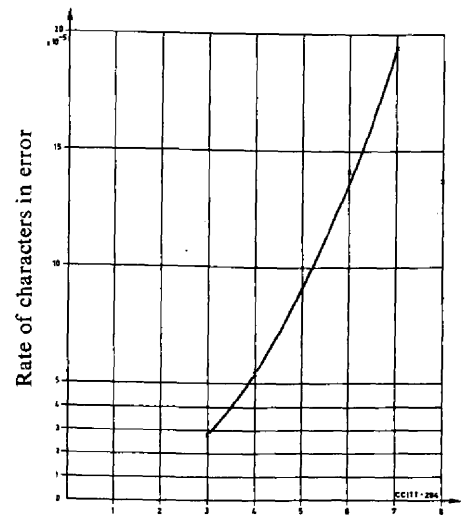


FIGURE 2

TABLE 2

Distribution of the errored elements within the error bursts and of the lengths of the error bursts

a) *3-VF telegraph sections*

Errored elements/burst	2	3	4	5	6	7	8	9	10
Number of bursts	5	3	2	2	1	—	—	—	—
Percentage of bursts	38	23	15.5	15.5	8	—	—	—	—

Length of bursts	0 to 5 elements	6 to 10 elements	11 to 15 elements	16 to 20 elements
Number of bursts	9	3	1	—
Percentage of bursts	69	23	8	—

b) *4-VF telegraph sections*

Errored elements/burst	2	3	4	5	6	7	8	9	10	18
Number of bursts	3	2	—	1	1	—	—	1	—	1
Percentage of bursts	34	22	—	11	11	—	—	11	—	11

Length of burst	0 to 5 elements	6 to 10 elements	11 to 15 elements	16 to 20 elements	25 to 30 elements	45 to 50 elements
Number of bursts	5	—	1	1	1	1
Percentage of bursts	56	—	11	11	11	11

c) *5-VF telegraph sections*

Errored elements/burst	2	3	4	5	6	7	8	9	10	22
Number of bursts	6	3	2	1	—	—	—	1	—	1
Percentage of bursts	43	22	14	7	—	—	—	7	—	7

Length of bursts	0 to 5 elements	6 to 10 elements	21 to 25 elements
Number of bursts	12	1	1
Percentage of bursts	86	7	7

d) *6-VF telegraph sections*

Errored elements/burst	2	3	4	5	6	7	9	11	14
Number of bursts	9	2	3	2	3	2	1	1	2
Percentage of bursts	36	8	12	8	12	8	4	4	8

Length of bursts	0 to 5 elements	6 to 10 elements	11 to 15 elements	16 to 20 elements
Number of bursts	16	6	3	—
Percentage of bursts	64	24	12	—

e) *7-VF telegraph sections*

Errored elements/burst	2	3	4	5	6	7	8	9	10
Number of bursts	1	—	1	—	—	—	—	—	—
Percentage of bursts	50	—	50	—	—	—	—	—	—

Length of bursts	0 to 5 elements	6 to 10 elements
Number of bursts	2	—
Percentage of bursts	100	—

f) *8-VF telegraph sections*

Errored elements/burst	2	3	4	5	6	7	8	10	11	12
Number of bursts	2	4	1	1	—	—	—	—	—	1
Percentage of bursts	22	45	11	11	—	—	—	—	—	11

Length of bursts	0 to 5 elements	6 to 10 elements	11 to 15 elements
Number of bursts	13	—	1
Percentage of bursts	93	—	7

TABLE 3

Distribution of the errored elements within the blocks

a) *3-VF telegraph sections*

Block length : 500 elements

Number of errored elements/block	0	1	2	3	4	5	6	11
Number of blocks	18 376	11	5	3	6	2	1	1
Percentage of blocks	99.84				0.16			

Block length : 1000 elements

Number of errored elements/block	0	1	2	3	4	5	6	11
Number of blocks	9 174	9	6	3	6	2	1	1
Percentage of blocks	99.7				0.3			

Block length : 5000 elements

Number of errored elements/block	0	1	2	3	4	5	6	11
Number of blocks	1 819	7	7	3	6	2	1	1
Percentage of blocks	98.54				1.46			

b) *4-VF telegraph sections*

Block length : 500 elements

Number of errored elements/block	0	1	2	3	4	5	18	22
Number of blocks	11 146	19	2	1	1	1	1	1
Percentage of blocks	99.77				0.23			

Block length : 1000 elements

Number of errored elements/block	0	1	2	3	4	5	18	22
Number of blocks	5 567	19	2	1	1	1	1	1
Percentage of blocks	99.53				0.47			

Block length : 5000 elements

Number of errored elements/block	0	1	2	3	4	5	18	22
Number of blocks	1 091	19	2	1	1	1	1	1
Percentage of blocks	97.67				2.33			

c) *5-VF telegraph sections*

Block length : 500 elements

Number of errored elements/block	0	1	2	3	4	5	6	7	16	22
Number of blocks	10 319	12	8	—	5	1	—	2	1	1
Percentage of blocks	99.71					0.29				

Block length : 1000 elements

Number of errored elements/block	0	1	2	3	4	5	6	7	8	16	22
Number of blocks	5 137	12	4	2	5	—	1	1	1	1	1
Percentage of blocks	99.46					0.54					

Block length : 5000 elements

Number of errored elements/block	0	1	2	3	4	5	6	7	8	16	22
Number of blocks	974	12	4	1	4	—	1	2	1	1	1
Percentage of blocks	97.3					2.7					

d) *7-VF telegraph sections*

Block length : 500 elements

Number of errored elements/block	0	1	2	3	4
Number of blocks	1 260	6	5	1	2
Percentage of blocks	98.9		1.1		

Block length : 1000 elements

Number of errored elements/block	0	1	2	3	4
Number of blocks	626	5	5	2	2
Percentage of blocks	97.8		2.2		

Block length : 5000 elements

Number of errored elements/block	0	1	2	3	4	5
Number of blocks	117	3	3	2	3	
Percentage of blocks	91.4		8.6			

It is important to know the distribution of element errors at different block lengths when determining a block checking method for error protection. Consequently the probability of errors was determined for block lengths of 500, 1000 and 5000 elements. It was found that the probability of errored blocks rises in an approximately linear fashion with the block length (Table 3, Fig. 3). Here again the dependence upon the number of VF telegraph sections can be seen.

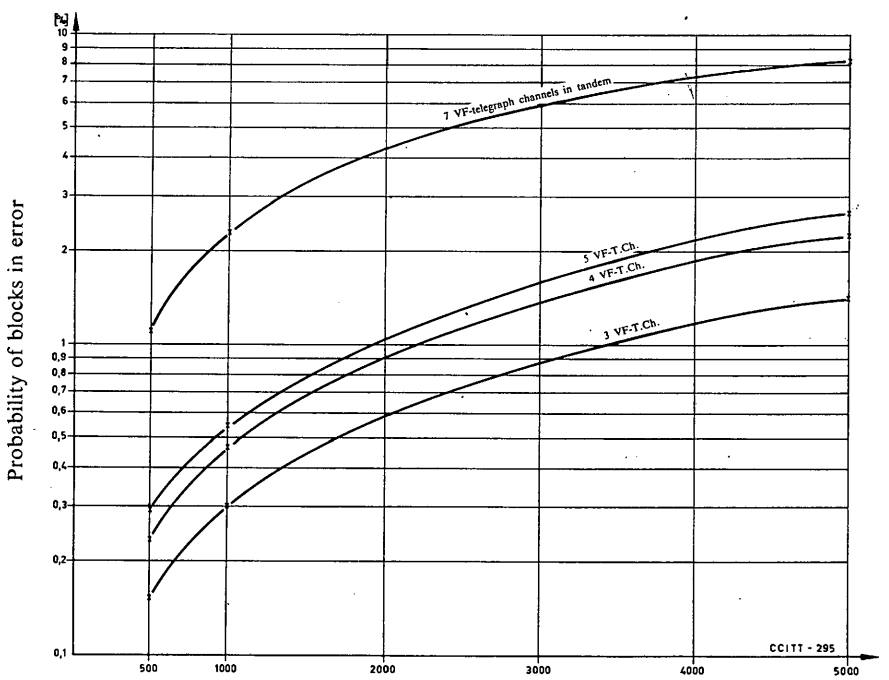


FIGURE 3

Supplement No. 5

NIPPON TELEGRAPH AND TELEPHONE PUBLIC CORPORATION. — *(Extract from contribution COM Sp. A/No. 72 — August 1963)*

DATA TRANSMISSION OVER VOICE-FREQUENCY TELEGRAPH CHANNELS AT 200 BAUDS

1. *Introduction*

Data transmission at low speed—i.e. modulation rates up to 200 bauds—is considered to be practical over voice-frequency telegraph circuits.

To provide the 200-baud data transmission, the 360-cycle spaced channels can be obtained by a multiple of the 120-cycle spacing as used for the 50-baud voice-frequency telegraph systems.

This paper describes the results of tests to confirm the feasibility of 200-baud data transmission over existing telegraph facilities.

The frequency allocation of the existing voice-frequency telegraph systems in Japan, and the arrangement of carrier and shift frequencies for the modem applied for the tests are shown in Figure 1.

Carrier and shift frequencies for the modem are selected on the same frequency band as the baseband of the voice-frequency telegraph system G2, so that other data channels can be transmitted on G1, G3 and G4 subgroups by group modulation systems.

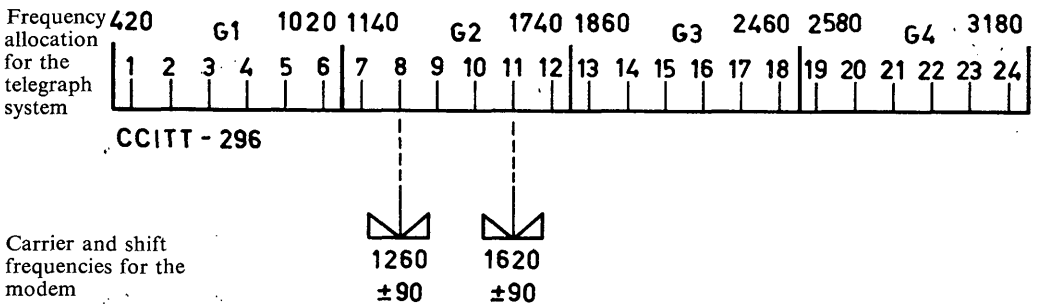


FIGURE 1. — **Frequency arrangement**

2. *Test equipment*

As shown in Figure 2, the test installation comprises a modem described above and a data-transmission test equipment in both of the stations.

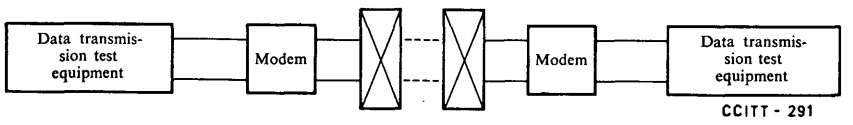


FIGURE 2

All test circuits were four-wire circuits throughout and did not include local lines.

The bit format was set for repetitive patterns of 128 bits or 16 letters (each letter comprises 8 bits). A block length was set at 256 bits. The pattern for the tests is given in Table 2.

TABLE 2

Pattern for the tests

	1	2	3	4	5	6	7	8
A	0	0	1	1	1	1	1	1
B	0	1	1	1	0	0	1	1
C	0	1	0	0	1	1	0	1
D	0	1	0	1	0	1	0	1
E	0	1	0	0	1	1	1	1
F	0	0	1	1	0	1	1	1
G	0	0	0	0	1	1	0	1
H	0	0	1	1	1	1	1	1
I	0	0	0	1	0	1	1	1
J	0	0	0	0	1	0	0	1
K	0	0	0	0	1	1	1	1
L	0	0	1	0	1	1	1	1
M	0	0	1	1	0	0	0	1
N	0	0	1	1	1	0	0	1
O	0	1	1	0	1	0	1	1
P	0	0	0	0	1	0	0	1

0 : position " A "
1 : position " Z "

3. *Test results*

Table 3 shows the test results. The influence of interruption for a period longer than 300 milliseconds are excluded from the results.

TABLE 3

Route	Type of circuit	Number of bits transmitted	Average element error rate	Average letter error rate	Average block error rate	Number of interruption times
Tokyo-Osaka	Coaxial	9.84×10^{6}	1.01×10^{-7}	8.10×10^{-7}	2.60×10^{-5}	2
Tokyo-Sapporo	Cable carrier system	1.15×10^{7}	5.21×10^{-6}	2.64×10^{-5}	3.78×10^{-4}	0
Tokyo-Fukuoka	SHF (4 GC)	1.15×10^{7}	7.82×10^{-7}	4.86×10^{-6}	1.56×10^{-6}	2

Supplement No. 6

FEDERAL REPUBLIC OF GERMANY. — *(Contribution COM Sp. A/No. 84 — September 1963)*

MEASUREMENTS ON DATA TRANSMISSIONS OVER THE TELEX NETWORK, AT 200 BAUDS, 1200 BAUDS AND 2400 BAUDS

A. Measurements on 200-baud telegraph circuits. Measurements in part of the German telex network modified for 200-baud start-stop transmission

1. *Test arrangement*

1.1 *Test transmissions from the analysing centre of the Fernmeldetechnisches Zentralamt via dialled loop connections*

For the performance of these series of measurements special arrangements were made in the central exchanges Frankfurt, Hamburg and Munich, allowing the connection in tandem of special circuits for transmission at 200 bauds between the central exchanges by dialling the respective prefix of these central exchanges. The measurements were made over two and three V.F. telegraph sections with 5 and 7 selector stages. The total length of the connections amounted to 1090 km and 2060 km respectively. Use was exclusively made of frequency-modulated V.F. telegraph systems with 480-c/s spacing.

The C.C.I.T.T. text of Recommendation R.51, viz. letters, S, carriage return, line feed, Q, figures, space, 9 with double-stop element was used as test text. It contains combination No. 29 so that in case of a falsification of a start element of a character the receiving equipment can synchronize itself again after having received at the latest this combination.

1.2 *Investigations into usability of the conventional mechanical telegraph relays for higher modulation rates (200 bauds)*

A great number of measurements and trials indicated that the armature chatters experienced to a greater extent with increasing modulation rates can be eliminated when treating the armature friction areas with a special lubricant. A newly rated spark quench is used to reduce the wear of the contacts.

2. *Analysing of the perforated test tape*

In the analysing centre of the Fernmeldetechnisches Zentralamt the incoming circuit was terminated with a reperforator. The receiving margin of the connected electronic series-parallel converter amounted to ± 48%. Use was made of an electronic-type transmitter, the transmitting distorsion of which was less than 0.5%.

A perforated-tape comparison arrangement mainly consisting of a double track transmitter with associated relay and counter panel was used for analysing the received tape.

The analysis showed that one or several characters were sometimes missing in the received text. The reason for the above was found to be disturbances of the synchronization of the receiving start-stop machine or short-term interruptions, etc.

The series of errors occurring in the case of a falsification of a start element due to the wrong start of the receiving arrangement were all counted since this type of error is typical for the start-stop method.

Analysing of error groups was done on the basis that a group is considered as finished when the last error of a group is followed by at least 10 correct characters.

3. *Test results*

The results gained from these series of measurements are shown in the attached tables and diagrams. The following correlations are found to exist.

The number of erroneous characters depends upon the number of V.F. telegraph sections in tandem. The diagram showing the statistical occurrence of the errors indicates that data transmission can be carried out with good results. The error rate observed during previous tests is even somewhat lower than in the 50-baud telex network.

This is evidently to be attributed to the fact that part of the short-time faults falsify the same number of characters with both 50-baud and 200-baud transmissions. In the period between two consecutive errors the higher modulation rate results in the error-free transmission of a greater number of characters.

ANNEX 1

Number of V.F. telegraph sections	Number of transmitted characters	Number of erroneous characters	Average error rate	Number of error sequences [1] with x erroneous characters following each other immediately												Number of disconnections resulting in the non-printing of characters			Number of false clearings
				$x = 1$	$x = 2$	$x = 3$	$x = 4$	$x = 5$	$x = 6$	$x = 7$	$x = 8$	$x = 9$	$x = 10$	$x = 11$	$x = 12$	< 40 ms	< 80 ms	< 120 ms	
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
2	8 046 367	201	1 : 40 000 = 2.50×10^{-5}	144	2	1	1	2	—	—	1	1	1	—	—	1	1	2	6
3	3 936 849	115	1 : 34 200 = 2.925×10^{-5}	61	2	2	1	3	—	—	—	—	1	—	1	—	—	1	6

[1] An error sequence is defined by the fact that the erroneous characters are following each other immediately.

ANNEX 2

Distribution of erroneous characters within the error groups and the length of error groups [1]

Two-V.F. telegraph sections

Erroneous characters/group	1	2	3	4	5	6	7	8	9	10	12
Number of groups	145	4	3	—	2	—	—	1	1	—	1
Percentage of groups	91.6	2.8	2.1	—	1.4	—	—	0.7	0.7	—	0.7

Length of groups	0 to 5 characters	6 to 10 characters	11 to 15 characters	16 to 20 characters
Number of groups	154	2	1	—
Percentage of groups	97.9	1.4	0.7	—

Three-V.F. telegraph sections

Erroneous characters/group	1	2	3	4	5	6	7	8	9	10	12
Number of groups	61	2	1	1	3	1	—	—	—	1	1
Percentage of groups	85	3	1.5	1.5	4.5	1.5	—	—	—	1.5	1.5

Length of groups	0 to 5 characters	6 to 10 characters	11 to 15 characters	16 to 20 characters
Number of groups	68	2	1	—
Percentage of groups	95.5	3	1.5	

B. Test transmissions at 1200 and 2400 bauds according to the telegraph principle

The previous test transmissions of the German Administration carried out at 200 bauds according to the telegraph principle indicated that, from the point of view of the transmission technique, a higher-speed data transmission with d.c. keying on the subscriber lines and sectional re-keying of the V.F. telegraph channels on the long-distance circuits should be practicable with a low error rate. The distribution of the signal energy over the spectrum found to exist in the case of the two most essential kinds of telegraph transmission — viz. d.c. and a.c. — shows that d.c. transmission will probably allow the application of a higher power level than the a.c. transmission. On the other hand the noise power in the frequency range of the d.c. signal is higher than in the frequency range of the a.c. signal. Within the exchanges, however, the application of high telegraph voltages will prove advantageous from the point of view of transmission technique.

Therefore, a test programme was prepared in co-operation with the German telecommunications industry, according to which the following measurements had to be carried out:

a) Test transmissions with 1200/2400-baud-d.c. telegraphy on local lines.

b) Test transmissions at 1200 bauds over part of the German telex network specially prepared for this purpose.

Details of the test conditions and results are stated below.

---

[1] An error group is considered to be over when the last error of a group is followed by at least 10 correct characters.

a) *Test transmissions with 1200/2400-baud-d.c. telegraphy on local lines*

The measurements were carried out in the local network of Munich on a 14-km and a 22.7-km circuit (4 km with 0.6 mm wires, the rest with 0.8 mm wires). Electronic relays balanced to earth were used at the sending and receiving ends (pair No. 1 of a star quad). The weighted noise voltage was measured at the sending station on pair No. 2 of the same star quad (terminated with 600 ohms). A capaciter artificially reduced the near-end crosstalk attenuation at 800 c/s to 7 nepers (61 db). In several cases a smoothing filter was inserted. At the receiving end the line was equalized in a simple way by means of a RC circuit. The over-all attenuation, measured at 1600 c/s, amounted to

−1.6 neper (14 db) on the 14-km circuit and

−2.5 nepers (22 db) on the 22.7-km circuit.

The results of these measurements are compiled in Tables 1 and 2.

TABLE 1

Noise and telegraph distortion

1. Modulation rate (bauds) . . .	1200	1200	2400	3500
2. Smoothing filter	yes	no	no	no
3. Transmitting voltage	900	400	400	400 mV
4. Psophometric voltage (near-end crosstalk)				
reversals 1 : 1	0.54	0.30	0.53	0.48 mV
C.C.I.T.T. text	0.35	0.21	0.33	0.39 mV
5. Distortion (C.C.I.T.T. text)				
14-km circuit	—	1.2	2.5	3.5 %
22.7-km circuit				
without equalizer.	1.2	1.5	7.5	17 %
with equalizer	—	1.5	4	7 %

TABLE 2

Error rates

1. Modulation rate (bauds).	1200	1200	2400	2400	2400
2. Smoothing filter	yes	no	no	no	no
3. Circuit	22.7	22.7	14	14	22.7 km
4. Transmitting voltage . .	900	400	270	400	400 mV
5. 8-bit characters	26	16	82	97	64×10^6
6. Erroneous characters . .	0	5	38	7	15
7. Error rate.	0	3.1	4.6	0.7	2.3×10^{-7}

b) *Test transmissions at 1200 bauds over part of the German telex network specially prepared for this purpose*

As the test series for data transmission over local lines at 1200 and 2400 bauds using d.c. keying had shown highly satisfactory results, the tests were extended on connections likely to be found in a data network. For this purpose Munich was chosen as an analysing centre. Normal four-wire subscriber lines of the Munich telex exchange were fitted with electronic relays. Furthermore, 1200-baud V.F. telegraph circuits were established between Munich, Hamburg, and Frankfurt, and the corresponding telex trunk relay sets were equally fitted with electronic relays. Otherwise, use was made of the normal installations of the German telex network. Especially, within the exchanges the keying voltages over selector contacts remained unaltered.

In the exchanges of Frankfurt and Hamburg arrangements were made to facilitate the loop connections Munich-Hamburg-Frankfurt-Munich and Munich Hamburg-Munich where by seven or else five selector stages of the telex network were used. Two series of tests were carried through. The connections consisted of the following individual sections, connected in tandem :

1. Subscriber line — two V.F. telegraph sections — five selector stages — subscriber line ;

2. Subscriber line — three V.F. telegraph sections — seven selector stages — subscriber line.

The following levels were used for the tests :

On the subscriber line : ± 450 mV ; d.c.

On the V.F. telegraph sections : −1.15 neper = −10 db (250 mV) referred to the zero relative level point 1900 ± 400 c/s.

The subscriber lines were operated with soft keying in order to ensure that between pairs of the same quad the admitted psophotometric voltage of 0.35 mV was not exceeded, even in case of a crosstalk attenuation of 7 nepers at 800 c/s.

Methods of measurement

Use was made of the same methods of measurement practised in the case of d.c. transmission tests over local lines. The working principle is as follows : at 1200 bauds a telegraph signal generator transmits to the line the character "e" (C.C.I.T.T. code No. 2 : ZAAAA) and then, alternatively, stop polarity (Z) for the duration of one character (approx. 7 ms). On the receiving end a shift register is set for the same character. After the arrival of a start element on the receiving end the character for which the shift is set is compared with the arriving one. If they do not coincide, an impulse is given on an electronic counter a slowly moving paper recorder.

A change of polarity during the 7 ms stop polarity may be regarded as a start element and may be counted as a character error. Therefore it can be used as a character for error evaluation.

Test results

Period of measurement : 15 days, during busy hours.

Quantity of characters transmitted : 5×10^7.

Duration of a connection : 56 minites (approx. 0.5×20^6 characters).

Error probability of characters in two V.F. telegraph sections : approx. 2×10^{-6}.
in three V.F. telegraph sections : approx. 4×10^{-6}.

Error-free interval between two consecutive errors : generally between 1 and 2 hours.

The distribution of errors is shown in the annexed diagram.

Remarks

Thus far the measurements have shown that a data network built up according to the pattern of the telex network, *the error rate will remain by two-or-three-tenth powers under the error rate in the switched telephone network*. Furthermore, it has again become clear that, in the case of modulation rates of over 50 bauds, a more favourable error rate is to be expected than in the case of a rate of 50 bauds. Because of the structure of the noise impulses in most cases only one character at a time is disturbed, and at a great modulation rate more correct characters can be transmitted in the meantime.

The modulators and demodulators used in this test had a relatively high isochronous distortion resulting for the total connection in an isochronous telegraph distortion of

— approximately 25% in case of two sections,

— and approximately 30% in case of three sections.

As in a switched network working according to the telegraph principle, the possibility exists that there are several modulation sections connected in tandem, the telegraph distortion caused by the modulators and demodulators should be kept as small as possible, at least smaller than in the case of the test modem used. By this method better results with regard to the total distortion might be expected than those measured in this test.

C. Conclusion

From the test results obtained until now the conclusion may be drawn that good transmission results may be expected from a network built up according to the telegraph principle and provided with electronic relays, using V.F. installations without any further subdivision of the carrier frequency channel. In such a network it should be possible to keep the error rate by a tenth power or two-tenth powers lower than in case of data transmission over the switched telephone network.

Supplement No. 7

FEDERAL REPUBLIC OF GERMANY. — *(Extract from contribution COM Sp. A/No. 14 — September 1961)*

A LOW-SPEED DATA TRANSMISSION SYSTEM OPERATED IN A HALF-DUPLEX TELEX NETWORK (THIS DATA TRANSMISSION SYSTEM MAY ALSO BE USED WITH SLIGHT MODIFICATIONS ON OTHER TYPES OF LINES AND NETWORKS)

1. Connections will be built up and the answer-back unit used in the same manner as recommended by the C.C.I.T.T. for the telex service; ARQ links will be excluded for the time being. Ordinary intelligence will be transmitted in the form of characters according to the International Telegraph Alphabet No. 2.

2. Data can likewise be transmitted on the basis of the International Telegraph Alphabet No. 2.

Note. — The use of additional alphabets and codes is not precluded; these would have to be studied separately.

3. Each item of intelligence will be arranged in blocks of arbitrary length.

4. Procedure of data transmission with the checking signal verified at the receiving station.

Preliminary Note. — The procedure is laid down in such a manner that both manual and automatic operation are possible, and that data originating from different sources—e.g., teleprinter keyboard, perforated tape with and without checking signal—can be handled equally well.

Data transmission, initiated by LINE FEED and CAR. RET., always starts on a new line. A sequence of four combinations No. 19 is used as a signal marking the beginning of data transmission. In order to avoid undesired repetitions or omissions of blocks, the blocks are cyclically numbered (in this paper designated with A, B, C). Each block is requested by the receiver with the number assigned to the block. Thus a compelled system results in leaving no doubt which block has just been transmitted and which one has to follow next (see diagram at the end of this Supplement). The block ends with LINE FEED and an end-of-block signal. A checking signal made up of three characters follows. This checking signal is acknowledged after verification at the receiving station by a request for the next block, or for the last block which has just been transmitted. If necessary, the end of the data transmission is indicated by a different signal.

4.a) Before starting data transmission, code combinations No. 28 (LINE FEED) and No. 27 (CAR. RET.) are transmitted.

4.b) The beginning of data transmission is marked by a sequence of four combinations No. 19 (SSSS or "").

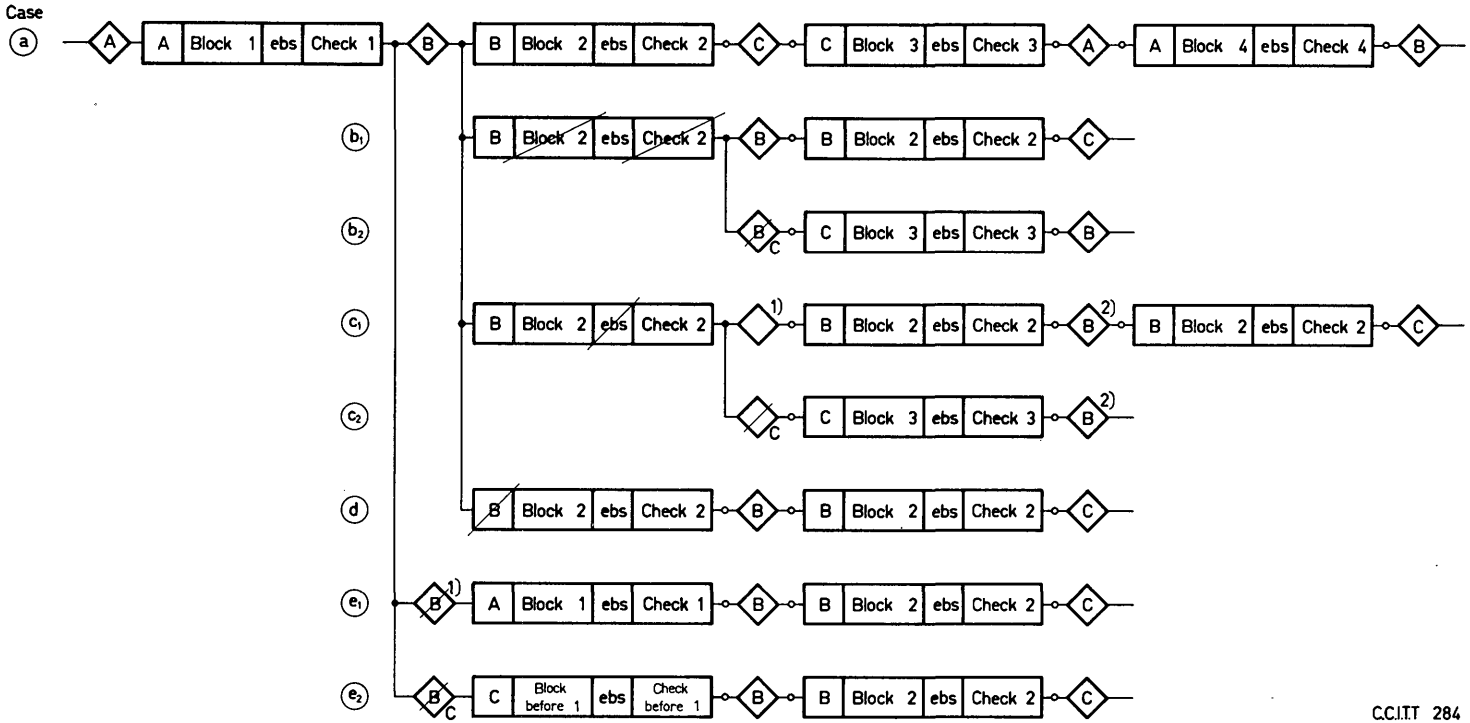
4.c) After this signal the receiver asks for the first block by transmitting the cyclical number "A".

4.d) Thereafter the first block with the cyclical number "A" is transmitted. The position of the cyclical number has yet to be determined.

4.e) At the end of the information of each block, combination No. 28 (LINE FEED) is transmitted.

4.f) Next comes the end-of-block signal consisting, e.g., of combinations Nos. 29, 27, 29, 27 (LETRS, CAR. RET., LETRS, CAR. RET.).

4.g) The end-of-block signal is followed by the checking signal. Counting commences with the first space element in each block; it is stopped after the last combination of the end-of-



A — Signal from the transmitting station.

A — Signal from the receiving station.

/ — Disturbed signal.

B/C — Signal B changed into signal C.

ebs — End-of-block signal.

1) No acknowledgement signal received, therefore repetition of the same block after some time.

2) Because ebs is missing, the next block is considered to belong to the preceding one, i.e. check gives wrong result.

Note. — A storage capacity of two blocks will be required in the cases B2 and C2, where, for example, signal B or no signal at all is interpreted as signal C. Alarm may be given instead.

Possible cases of a half-duplex system with sequence numbers for block designation and acknowledgement
(Numbers not included in the check)

block signal. The 12 bits of the checking signal are converted into the form of three printable characters. This signal is transmitted over the line and compared with a locally produced checking signal at the receiving end. If both signals agree, an acknowledgement signal consisting of the cyclical number "B" is returned to the transmitter, causing there the transmission of the next block (cyclical number "B") and so on.

4.h) If the two signals fail to agree, block "A" is requested once again. Some special cases not described here are shown in the diagram.

4.i) Change-over to normal telex transmission will be marked by another sequence of characters, e.g. by a sequence of four combinations No. 12 of the International Telegraph Alphabet No. 2.

Supplement No. 8

CHILE TELEPHONE Co. — *(Extract from contribution COM Sp. A/No. 7 — July 1961)*

A. Protection for a low-speed data transmission system operated on a full-duplex telex network or a leased telegraph circuit

(Manual or automatic operation — all alphabets)

1. The principle of protected data transmission is that of checking the message by means of information feedback and correcting errors by means of repeated transmission. The data signals are transmitted over the forward path of the duplex circuit to the receiving station, where these signals are repeated and sent back over the return path of the circuit. At the sending station the returned signals are compared with stored information corresponding to the signals originally transmitted. Transmission and subsequent comparison of the data continue until an error is detected in the returned signals, when normal transmission is interrupted. A cancel signal is transmitted followed by a repetition of the character detected as being in error, thereupon normal transmission of data continues.

2. A telex call will be set up between the calling and called subscribers in accordance with the procedure recommended by the C.C.I.T.T. for setting up and checking a telex call by the interchange of answer-back signals.

3. If one of the subscribers wishes to introduce data transmission equipment on the connection, he will send the Alphabet No. 2 signal sequence to cause the transmitting or receiving data equipment to be brought into circuit at either the sending or the receiving end (the signal No. 19 repeated four times).

If the connection involves the use of a radio channel with an automatic error detection device transmitting delay signals (see C.C.I.T.T. Recommendation U.22, Volume VII of the *Blue Book*), the sending of the delay signals should be suppressed by sending the HHHH sequence (in start-stop Code No. 2). This sequence should be sent before the sequence that brings the data transmission equipment into circuit.

4.

. . . . . .

5. Upon receipt of the Switch-to-Data signal at the receiving station, the incoming signals in addition to being passed on to the data receiving equipment should also be repeated and transmitted back to the sending end.

6. Procedure for protected data transmission.

Preliminary Note. — The procedure should be laid down in such a manner that both manual and automatic operation are possible, and that data originating from different sources (e.g. teleprinter keyboard or perforated tape) can be handled equally well.

7. Data can be transmitted on the basis of the start-stop five-unit code (International Telegraph Code No. 2).

The system is also suitable for other alphabets after the characters have been regrouped to form five-unit combinations.

For some applications it will be possible to send start-stop combinations with more than five information elements and the system can be modified for such application by increasing the storage necessary.

8. For applications using International Telegraph Code No. 2 data transmission should commence with code combination No. 28 (LINE FEED).

9. When the error detection equipment is connected at the sending station, the data signals are sent out to line, and simultaneously fed into a temporary store.

10. The store should have sufficient capacity to allow for the transmission delay around the looped circuit between the original transmission of any one character and the return of that same character. The maximum delay to be catered for is to be specified later.

11. The returned signals are compared with the corresponding stored signals, and whilst the two sets of signals agree the transmission of data continues. When the two signals fail to agree futher data transmission is interrupted.

12. *Correction procedure with manual control*

When the two signals being compared fail to agree, data transmission is interrupted and an alarm is given.

A cancel signal is sent from the sending station and upon the return of the cancel signal the last N characters that were transmitted before the alarm are transmitted a second time ; followed directly by a continuation of the data message. The value of N is determined by the maximum transmission delay around the looped circuit mentioned in paragraph 10.

Note 1. — If transmission is by manual operation then the repeated characters could be manually repeated by the operator. Where the transmission employs a perforated tape, the operator should manually reset the tape by the appropriate number of characters. In the case of short messages it might be preferable to reset the tape back to the beginning of the then current message.

Note 2. — For many messages a Start-of-Message signal such as ZCZC, and an End-of-Message signal such as NNNN, may be valuable. The use of such signals involves no special requirements, and in the case of short messages the cancel signal need not be sent, since the incomplete message containing the error will be prefixed by a Start-of-Message signal but will not be suffixed by an End-of-Message signal, whereas correct messages, including the repeated ones, will always have both Start-of-Message and End-of-Message signals.

13. *Correction procedure with automatic control*

Where data transmission is from perforated tape, it will generally be preferable that the correction of an error by repetition is fully automatic.

Note. — Such automatic repetition should not preclude data transmission by manual operation.

When the returned signals are compared with the corresponding stored signals and the two signals fail to agree, further data transmission is prevented, and a cancel signal is sent.

Upon the return of the cancel signal to the sending station the stored information is transmitted character by character, repeating the last *N* data characters which were transmitted before the error was detected.

Note. — The cancel signal may be disturbed during its transmission forward, thus preventing the receiving station from recognizing it as a cancel signal. However, an alarm will be given upon its return to the sending station, and further transmission will be prevented.

Alternatively, the disturbance to the cancel signal may occur during its transmission back to the sending station, in which case, although the signal would be recognized correctly at the receiving station, an alarm will still be given at the transmitting station, and further transmission will be prevented.

There is the third possibility that the cancel signal, having been disturbed during its forward transmission, is disturbed a second time during its backward transmission such that the first error is corrected. In this case the receiving station will not recognize the cancel signal, but the sending station will assume that this signal has been recognized correctly. The sending station will repeat the section in error but this will not be recognized as such at the receiving station. However, the probability of this form of double disturbance occurring is so small that it may safely be neglected.

14. As the stored information is retransmitted it is at the same time re-inserted back into the store ready for checking once more against possible errors during repeated transmission.

Should an error occur in the repeated characters the procedure under item 13, paragraphs 2 and 3, and item 14, paragraph 1, is repeated.

When no error occurs in the repeated characters then the data transmission and automatic checking continue normally.

When the cancel signal is transmitted but not received back correctly an alarm should be given to bring in an operator.

15. *Composition of cancel signal*

15.1 It is essential that the cancel signal or any sequence in the data message should not clear down the connection or otherwise interfere with the telegraph switching when operating on the telex network. It is also essential that, if the cancel signal is simulated by groups of characters in the message, this should not cause the receiving terminal to falsely reject some of the data.

15.2 *Hazards of the telex network :* The telex network is essentially a 50-baud network and is intended for the transmission of 7½ unit start/stop coded information. It is a switched network and therefore signals, other than those of the message, have to be passed before and after the message to enable the switching equipment to set up and release the required connection.

Release of connection : The signal used to release the switching equipment and break down the connection is a continuous space condition lasting for at least 300 ms. With the majority of switching equipments even an interrupted space condition will release the connection, provided that the ratio of spacing/marking times is large enough and that the condition is of sufficient duration. With normal telegraph transmission the critical spacing/marking ratio is never reached unless character combination No. 32 (all start) is repeated consecutively. The C.C.I.T.T. has recognized this fact and has allocated no special significance to this character combination in the Alphabet No. 2 and it has stipulated that no message should contain more than two such characters transmitted consecutively.

Regenerative repeaters : The C.C.I.T.T. has proposed the use of regenerative repeaters to reduce the degree of distortion in the telegraph signals, to reduce the number of errors in the received information, and/or to enable longer connections to be established. The regenerative repeaters accept the distorted telegraph signals, re-shape and re-time these signals before passing them on as clean start/stop characters. Thus any circuit which contains a regenerative repeater can only pass 7½ (or 7) unit start/stop characters.

ARQ links : Increasing use is being made of radio circuits for the transmission of telegraph signals, and these are rapidly being incorporated into the telex network. Because of the high probability of errors occurring on the radio link, automatic error-correcting equipment is fitted to the radio telegraph channels. This equipment is generally known as ARQ.

Even with the use of the ARQ equipment the probability of an error is still far above that which is acceptable for data transmission. Thus the proposed error-detection and correction system for data transmission should, if possible, operate over an ARQ link. There are many features of the ARQ system that present difficulties to the transmission of protected data, but in so far as the cancel signal is concerned the only limitation is that the characters must be 7½-unit start/stop.

15.3 *Hazards of private wire circuits :* On a private wire, or leased, circuit there will be no switching, hence the possibility of the message or the cancel signal clearing down the connection or otherwise interfering with switching operations does not arise. Similarly, there will be a negligible chance of a regenerative repeater being in circuit, and therefore the data would not be restricted on this account to 7½-unit transmission.

ARQ links : If a radio link is included in the leased circuit connection there is a strong possibility of ARQ equipment being included also. If this is so, then, as for the telex connections, data must be transmitted as 7½-unit start/stop characters.

15.4 *Simulation by message :* Obviously, if possible, the cancel signal should not be simulated by groups of characters within the data message. If such a thing occurred the receiving terminal would falsely identify this group of data characters as a cancel signal, whereupon not only would these signals be rejected as being not part of the message but so also would the immediately previous group of characters.

15.5 *Solutions :*

Interruption of message sequence : It is possible by checking the message during transmission to interrupt the sequence of the message in such a way that any simulated cancel signal combination is broken and is never transmitted as a complete cancel signal sequence.

When the check on the data message indicates that all but the last character of the cancel signal sequence have been transmitted, further transmission of data is prevented. A true cancel signal is inserted followed by the repetition of the immediately previous N characters. From then on transmission of data continues normally. Thus the message is so split that initially all but the last character of the cancel signal sequence are transmitted, and subsequently all but the first character of the cancel signal sequence are transmitted. This requires the cancel signal combination to contain $(N + 2)$ characters ; where N is the number of characters stored for repeated transmission upon detection of an error.

In practice the value of N will have to be at least four and therefore the cancel signal would have to be at least six characters.

Cancel signal :	XXXYYY
Message :	CBXAXXXYYYPQRS
Actual transmission :	CBXAXXXYY XXXYYY XXYYYPQRS

Note. — Characters underlined are eliminated when processing the received message.

It will be seen that this procedure provides a complete guard against a false cancel signal appearing in the received message. It does, however, require at least six characters in the cancel signal combination.

Introduction of deliberate errors : It is again possible by checking the message during transmission to deliberately introduce an error into the transmitted message in such a way that a repetition is automatically incurred, causing any simulated cancel sequence within the message to be broken and prevent its recognition as a cancel signal at the receiving terminal.

When the check on the data message indicates that all but the last character of the cancel signal sequence have been transmitted, an error is deliberately introduced into the next character to be transmitted. However, the correct character information is fed into the store for subsequent comparison with the returned signals. Upon the error being detected the normal correction procedure is automatically performed, the cancel signal is sent followed by a repetition of the characters held in the store. Provided that the first character to be repeated is always the character in error then the cancel signal sequence in the message is broken and will not be falsely recognized as a cancel at the receiving terminal.

Cancel signal :	XYYY
Message :	CBXAXYYYPQRS
Actual transmission :	CBXAXYYØPQR XYYY YPQRS

Note. — Characters underlined are eliminated when processing the received message. Character deliberately transmitted in error shown as Ø.

It will be apparent that in this way the cancel signal could be made as short as only two characters. However, this would cause a great number of repeated transmissions and considerable advantage is to be gained by using more characters. With a four-character combination the probability of a repetition occurring due to the composition of the message is approximately once every 30 000 characters.

15.6 *Composition of cancel signal :* An important factor in the composition of the cancel signal is that it must be composed of at least two different character combinations. If only one type of character is repeated, say, four times, then it may well happen that the cancel signal immediately follows one or more of the same characters in the message. This will cause the receiving terminal to recognize the cancel signal sequence one or more characters before the true cancel signal ; thus rejecting the correct characters as being in error.

Cancel signal :	XXXX
Message :	UTABCDXEFG
Actual transmission :	UTAβCDX XXXX BCDEFG

Note. — Characters underlined are eliminated when processing the received message. Character received in error shown β.

Little is to be gained by the use of more than two different character combinations in the cancel signal. It is therefore proposed that the cancel signal be made up of four characters, the first and second of which should be of one character combination and third and fourth characters each be of a second character combination, i.e. XXYY.

15.7 *Proposed characters :* There is some advantage if the cancel signal could be printed out to enable a printed copy of the received information to be made on a teleprinter. However, when considering data transmission it is hardly likely that the code used would be suitable for printing on a normal teleprinter. The majority of data will be received on a punched tape for feeding into a processor, therefore it is somewhat immaterial whether or not the cancel signal can be printed out. Where keyboard transmission is used it is unlikely that automatic correction will be used and hence some printable cancel signal could be prearranged.

With regard to automatic correction equipment the cancel signal should be kept as simple as possible to avoid undue complication and added cost to the equipment. The simplest characters to generate and to identify are combination Nos. 29 and 32.

15.8 *Proposed cancel signal :* The proposed cancel signal is therefore character combination No. 29 repeated twice followed by character combination No. 32 repeated twice.

Letters, Letters, All space, All space.

16. *Clear-down procedure*

At the completion of data transmission, an End-of-Data signal comprising a code combination repeated four times should be sent. This signal should not be sent for a period of, say, five seconds after the transmission of the message has ceased. The combination to be used is of more importance in respect to the half-duplex system and there is no reason why same sequence should not be used in both cases. This signal should revert the station back to its normal telegraph text condition, disconnecting all data-transmission equipment.

In addition to the procedure in the above paragraph, the data-transmission equipment should be switched out of circuit and the station revert to its normal telex condition upon the clear-down of the telex connection.

17. *Loop delay*

The sending station will not be aware of the loop delay which will occur on each connection and it is possible that the element store may be filled before the return of the first character. In such an event, the reading of the message will be delayed until the store can accept another character. During this interruption an alarm shall be provided.

B. Specification for a low-speed data transmission system operating over a half-duplex telex network or a leased telegraph network

(Automatic operation, all alphabets)

1. With this system the message will be transmitted in the form of signals which are not in accordance with International Telegraph Alphabet No. 2 except for the establishment of the connection.

2. The system is suitable for 6-unit, 7-unit and 8-unit codes which have been regrouped to form 5-unit combinations.

For some applications it will be possible to send start/stop combinations with more than five information elements and the system can be modified for such application.

3. A telex call will be set up between the calling and the called subscribers in accordance with the procedure recommended by the C.C.I.T.T. for setting up and checking a telex call by the interchange of answer-back signals.

4. If one of the subscribers wishes to introduce data transmission equipment on the connection he will send an Alphabet No. 2 signal sequence to cause the transmitting or receiving data equipment to be brought into circuit at either the sending or the receiving end.

If the connection involves the use of a radio channel with an automatic error-detection device transmitting delay signals (see C.C.I.T.T. Recommendation U.22), the sending of the delay signals should be suppressed by sending the HHHH sequence (in start/stop code No. 2). This sequence should be sent before the sequence that brings the data transmission equipment into circuit.

5. The messages will be grouped into blocks of predetermined lengths, say, 250 data elements.

6. The decision as to the correctness of the block received is made at the sending station from a comparison of a checking signal received from line and a locally generated check signal.

7. The backward signals comprise only the checking signals. Any alteration to these signals are detected like errors in the text of the message and result in the block concerned being repeated.

8. The checking signal will be obtained at both terminals by counting the start polarity pulses. The start and stop elements should be excluded from this count.

9. A series of three sequence number signals should be employed cyclically to precede each block in turn. If the check following the transmission of a block fails to agree, the sending station does not continue with the next sequence number and block, but sends a cancel signal comprising a single character. (See explanatory charts).

10. The receiving station, on recognizing the cancel signal, sends back the normal checking signal which has to be agreed by the normal procedure before the block is repeated.

11. Should line disturbance cause the cancel signal to be changed into the next sequence number, the receiving station will not return an agree signal at once and, in the absence of response, the cancellation signal will be repeated until an acknowledgement is obtained. There should be a pause of not less than 600 ms between repetitions.

12. Before commencing the message the sending station should transmit the cancel signal, which should be acknowledged by a checking signal.

13. The receiving station is therefore so arranged as to examine the first combination of each new transmission because this first combination always constitutes a signal and not part of the message.

14. In order to make the cancel signal as different as possible from the sequence number the sending station should be so arranged as to send the inverse of the next sequence number. The receiving station is aware of the next sequence number and examines for this or its inverse simultaneously.

15. In the event that the receiving station receives a cancel signal followed by characters or the next sequence number followed by an incomplete block it should send no check signal but prepare to receive a cancel signal.

16. The backward checking signal will consist of several characters which will be determined by the error-detecting codes.

17. In the event that the sending station is unable to release a new block in a period exceeding, say, five seconds, an alarm should be given.

18. Before reverting from secure to normal transmission, steps should be taken at the sending station to discontinue the transmission of the message.

19. If the receiving station finds that message transmission has ceased for a period of, say, four seconds, it will prepare to accept the "transfer to normal" signal which should be acknowledged. During the reception of a message this signal must not be accepted.

20. The "transfer to normal" signal should comprise a combination repeated four times.

21. If either the sending or receiving station receives the start-polarity sustained long enough to terminate the connection, it will—in closing down—restore the transmission conditions from "secure" to "normal".

22. In order to simplify the task of the sending and the receiving stations it is proposed that the signals used for the initiation and the termination of the "secure" condition should be given by the transmission four times in sequence using two of the combinations assigned for sequence signals. Suitable combinations might be :

Sequence signal 1 combination 19-(S) (') MSMSS
,, ,, 2 ,, 4-(D) (WRU) MSSMS
,, ,, 3 ,, 26-(Z) (+) MSSSM

Repeat signal (replacing sequence signal No. 1) 7 (G) (a) SMSMM
,, ,, (,, ,, ,, No. 2) 16 (P) (c) SMMSM
,, ,, (,, ,, ,, No. 3) 3 (C) (:) SMMMS

Transfer to "secure" transmission combination No. 3 four times.
Restoration to normal transmission combination No. 19 four times.

Note. — There are many other possibilities.

Charts illustrating the use of backward redundancy and sequence number signals for a telex network using alphabets other than the International Alphabet No. 2

Forward signals (F)	*Backward signals (B)*
Sequence number signals S1, S2, S3	Redundancy checking signals ch1, ch2, ch3, etc.
Cancel signals ca1, ca2, ca3	Cancel checking signals chca1, chca2, chca3.

Note. — After sending the first block prefixed by S1 the sending station checks the redundancy checking signals. If these are correct it sends the second block prefixed by S2 ; if they are incorrect it sends the ca1 signal to tell the receiving station to cancel the previous block, which is being sent again.

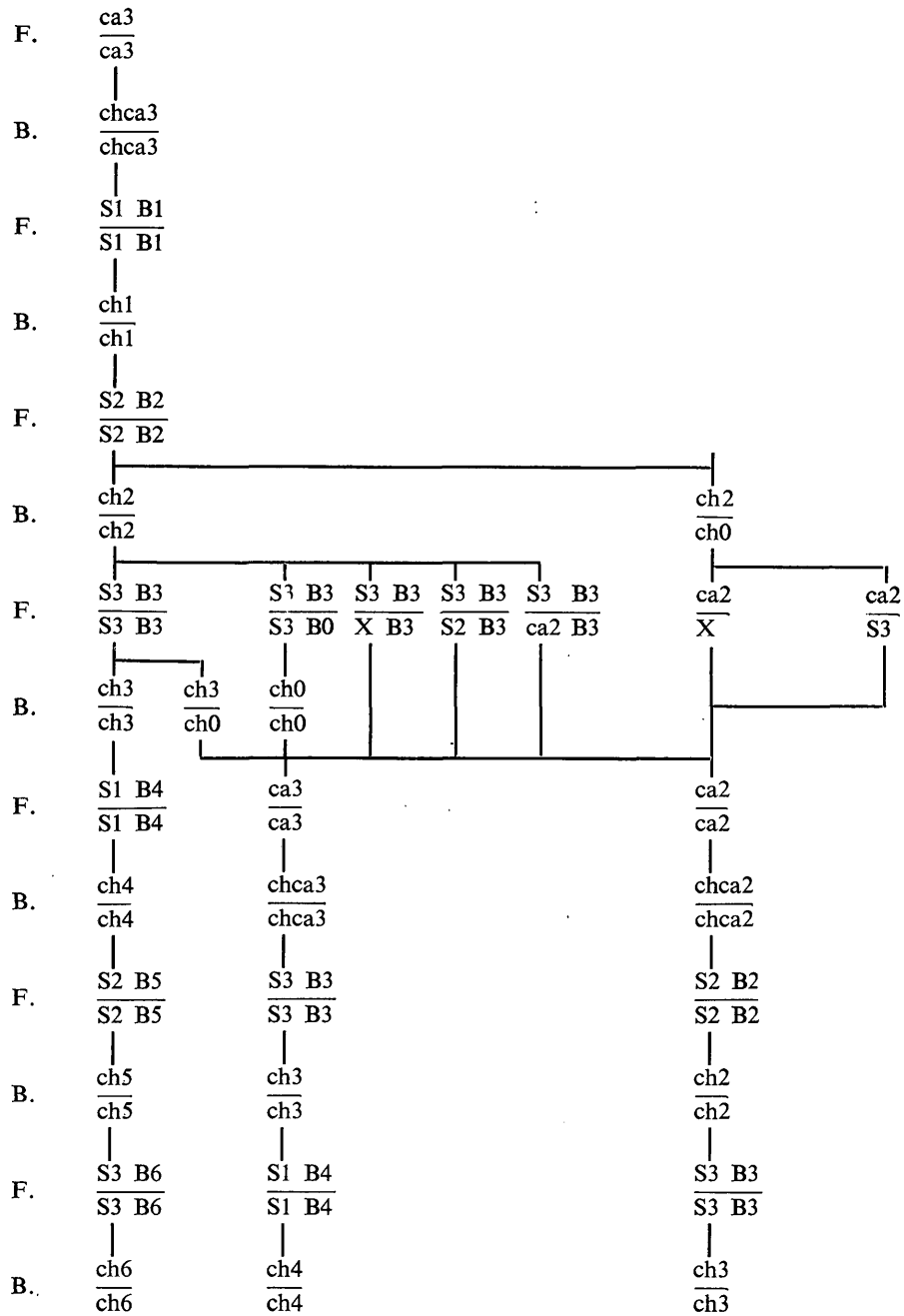
Other symbols used in the charts :

B1, B2, B3, B4, etc.	Blocks in order
B0	A block received with an error
X	A signal which is neither S1-3 nor ca1-3
ca3 ----------	Signal transmitted
$\overline{\text{ca3}}$ ----------	Signal received
S (alarm)	Alarm at sending station
R (alarm)	Alarm at receiving station.

In order to reduce the chances of error, the combinations used for ca1, ca2 and ca3 are related to those chosen for S1, S2 and S3. After sending an S1 block, the sending station will follow with either S2 block or a cancellation of S1 block, i.e. ca1. These two combinations are therefore given different polarities for each element ; S3 is reversed to ca2 and S1 is reversed to ca3.

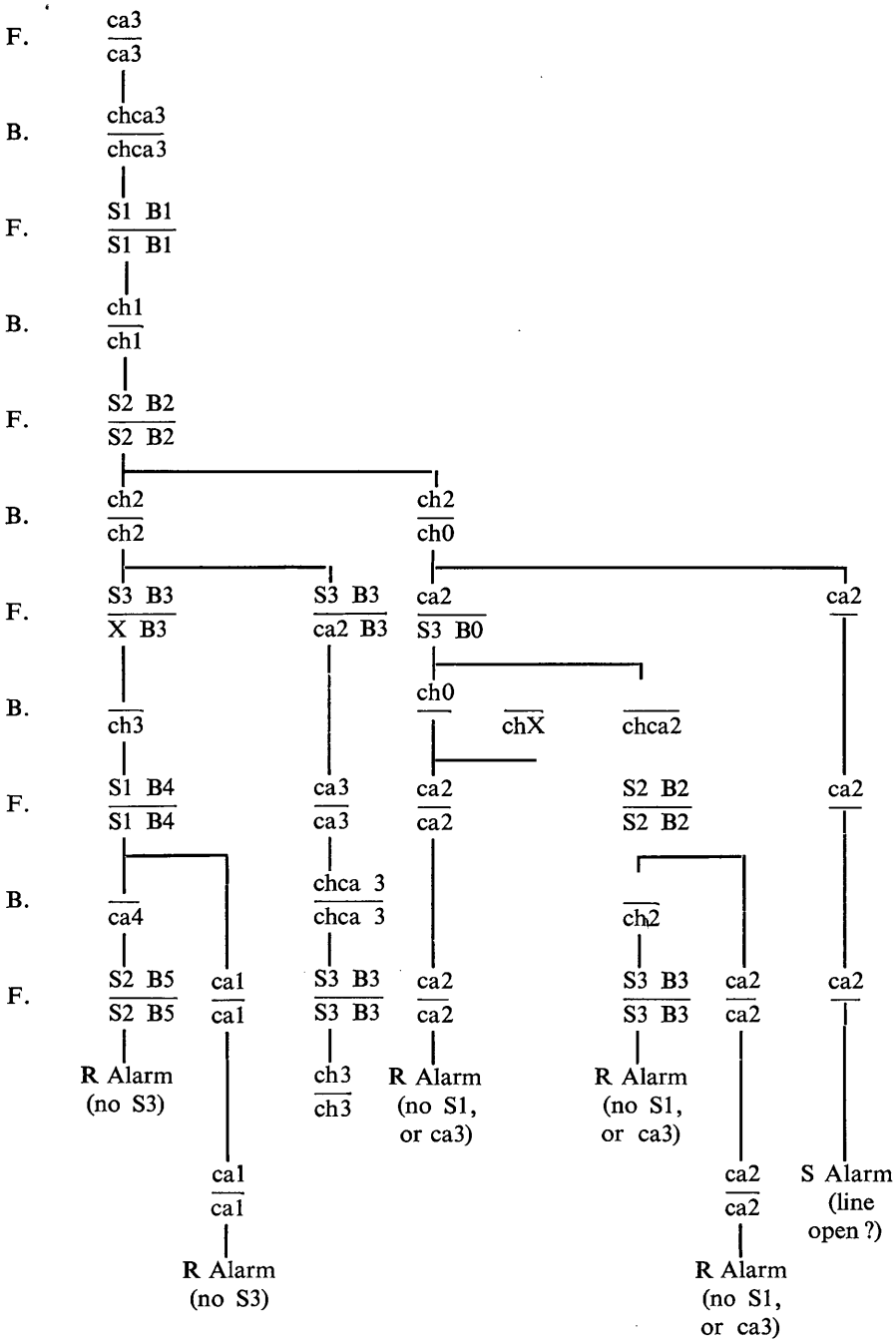
After receiving block 1, the receiving station sends a checking signal such as ch1 and then awaits S2 or ca1. Should the next block be unrecognizable, it awaits ca1 or ca2 because S2 will not be sent a second time until ca2 has been acknowledged.

SHEET 1 — NORMAL OPERATION



(Suppl. 8)

SHEET 2 — ABNORMAL OPERATION



(Suppl. 8)

C. Specification for a low-speed data transmission system operating over a half-duplex network or a leased telegraph circuit

(Manual or auto operation—Alphabet No. 2)

1. With this system all intelligence should be transmitted in the form of signals according to International Telegraph Alphabet No. 2.

2. A telex call should be set up between the calling and called subscriber in accordance with the procedure recommended by the C.C.I.T.T. for setting up and checking a telex call by the interchange of answer-back signals.

3. If one of the subscribers wishes to introduce data-transmission equipment on the connection, he should send an Alphabet No. 2 signal sequence to cause the transmitting or receiving data equipment to be brought into circuit at either the sending or the receiving end.

Operation over radio channel

If the connection involves the use of a radio channel with an automatic error-detection device transmitting delay signals (see C.C.I.T.T. Recommendation U.22), the sending of the delay signals should be suppressed by sending the HHHH sequence (in start-stop code No. 2). This sequence should be sent before the sequence that brings the data-transmission equipment into circuit.

4. The message should be grouped into blocks of arbitrary length which should not exceed, say, three lines of text.

The end-of-block signal should be composed of code combinations Nos. 29, 27, 29, 27 (Letters, Car. Ret., Letters, Car. Ret.).

Note. — Fixed length blocks could be used by arrangement with or without end-of-block signals.

5. A series of three sequence number signals should be employed cyclically to indicate the next block required.

6. A checking signal should be obtained at both terminals by counting the start-polarity pulses. The start and stop elements should be excluded from this count.

7. After the end-of-block signal has been transmitted the checking signal calculated from the count of the start-polarity pulses should be transmitted to the receiving station in the form of several characters. (See explanatory charts hereinafter.)

8. A check should be made at the receiving station and if there is agreement the next sequence signal should be sent back as a single character combination.

9. If the sending station receives a disturbed sequence No. signal it should repeat the end-of-block signal which should cause the receiving station to re-send the sequence No. signal for the next block required.

10. If there is no response within, say, 600 ms of the termination of the block the sending station should repeat the end-of-block and checking signals.

11. *Clear-down procedure*

At the completion of data transmission, an end-of-data signal comprising a code combination repeated four times should be sent. The combination to be used is of more importance in respect to the half-duplex system for other alphabets. With manual operation it is necessary for the

receiving station to be continually looking for the signal. This signal should revert the station back to its normal telegraph text condition, disconnecting all data-transmission equipment.

In addition to the procedure in the above paragraph, the data-transmission equipment should be switched out of circuit and the station revert to its normal telex condition upon the clear-down of the telex connection.

12. *Operational proposals*

The system is suitable for manual or automatic operation and for data originating from keyboard or tape with or without prepared checking signals.

12.1 The first block should be preceded by "line feed" and an "end-of-block" signal which should be acknowledged by the first sequence number signal.

12.2 Each block may commence with the text or with one or more "letters" combinations.

12.3 "Line feed" should be sent immediately before the "end-of-block" signal.

Charts illustrating the use of sequence number signals for decision feedback in a telex network using the International Alphabet No. 2

Forward signals (F)	*Backward signals (B)*
End-of-block signals — EOB	Sequence number signals S1, S2, S3.
Redundancy checking signals — ch1, ch2, etc.	

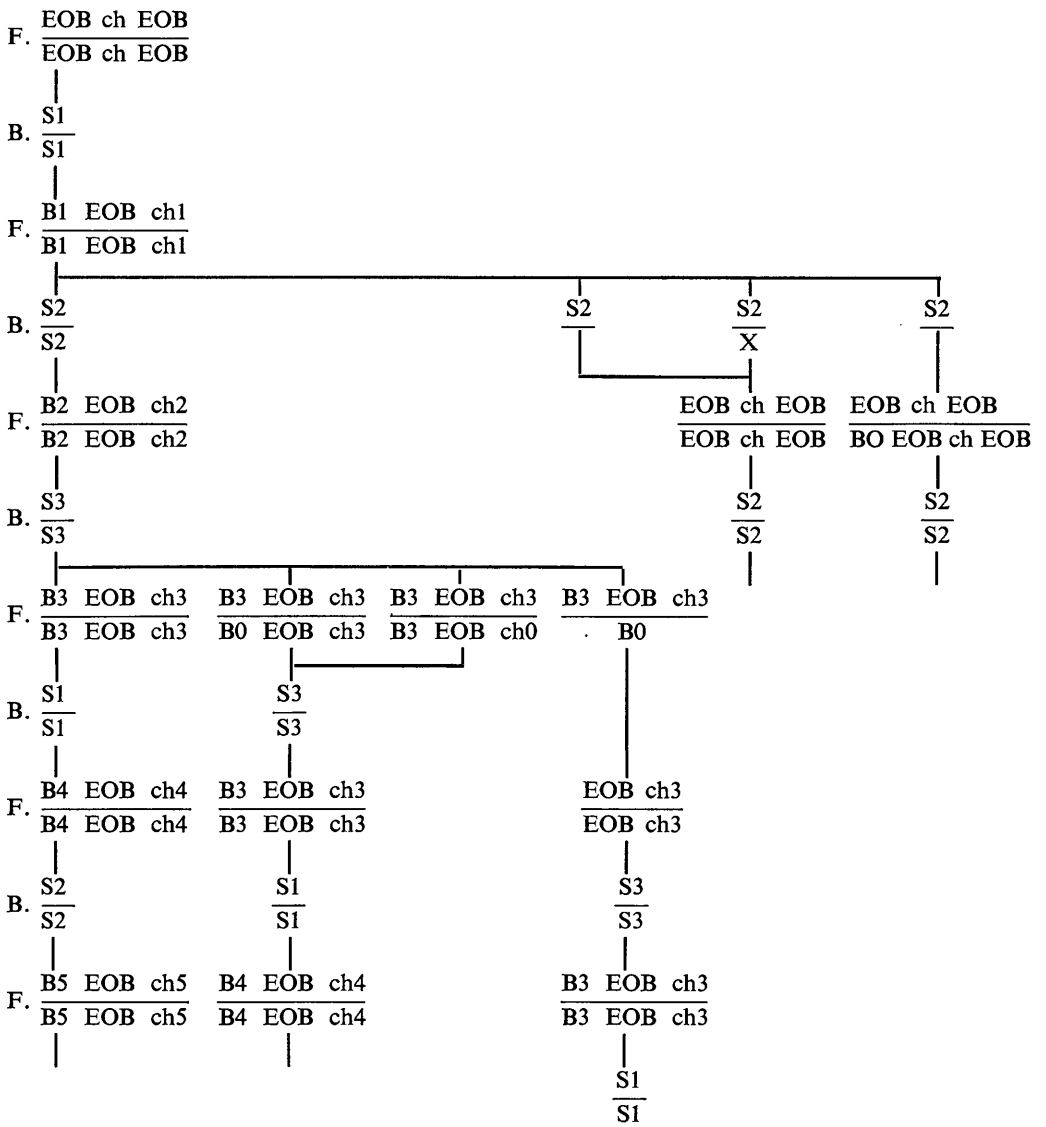
Note. — After sending the first block terminated by EOB and the redundancy check signal the receiving station compares the check signals. If these are correct the receiving station sends the next sequence number. If these are incorrect the receiving station repeats the sequence number of the block just received.

Other symbols used in charts:

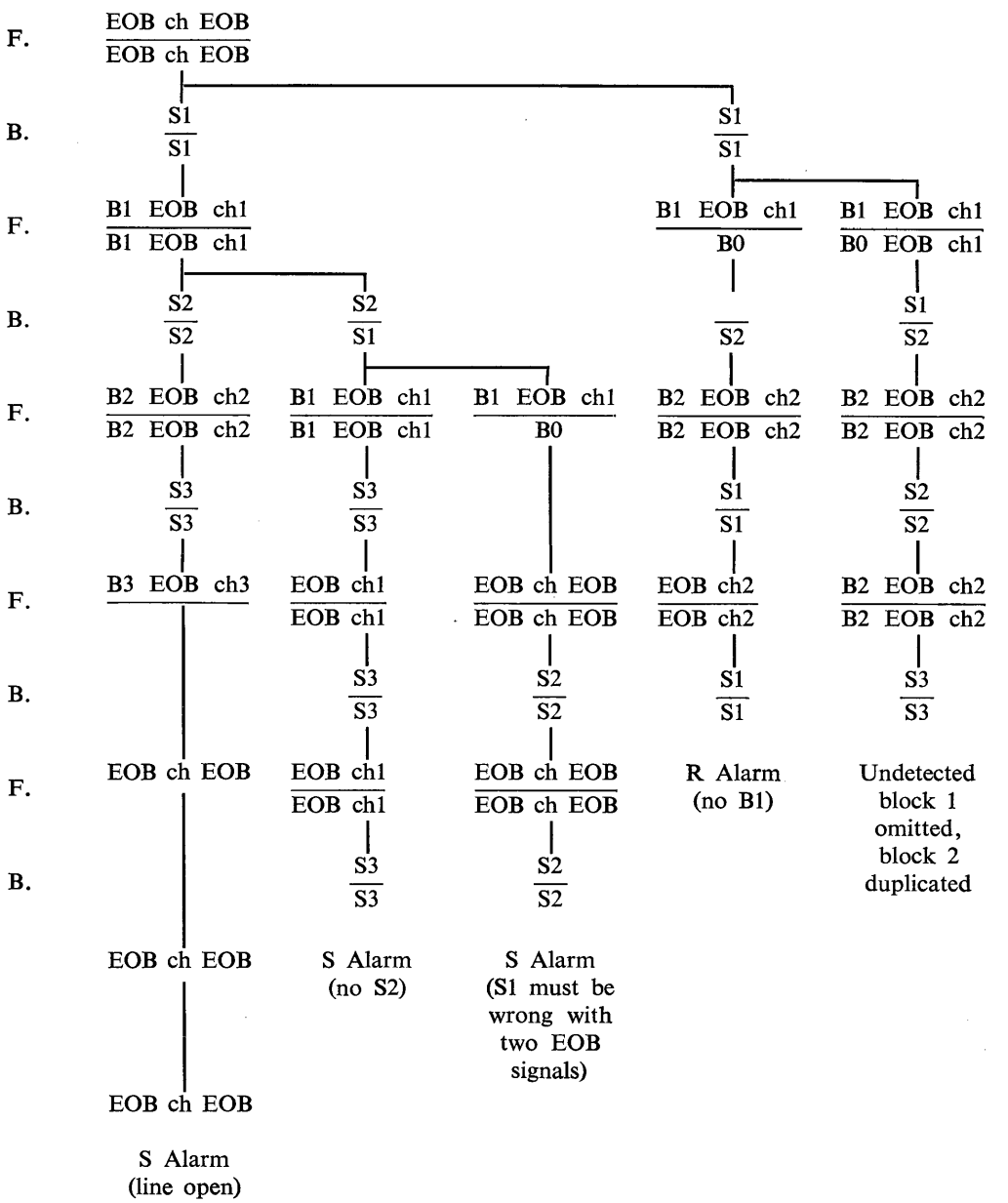
B1, B2, B3, B4	— Blocks in order
B0	— A block received with an error
X	— A backward signal other than either of the serial numbers expected
S1	— Signal transmitted
$\overline{S1}$	— Signal received
S (Alarm)	— Alarm at sending station
R (Alarm)	— Alarm at receiving station.

The last example on Sheet 2 shows a case which is liable to occur in which block 1 is omitted and block 2 is duplicated. This case can normally be alarmed if the redundancy counters commence counting with the sequence number. The possibility of compensating errors causing the redundancy to check correctly, in spite of the error, cannot be dismissed.

SHEET 1 — NORMAL OPERATION



SHEET 2 — ABNORMAL OPERATION



Supplement No. 9

OLIVETTI. — *(Extract from contribution COM Sp. A/No. 88 — October 1963)*

DESCRIPTION OF THE OLIVETTI DATA-TRANSMISSION SYSTEM FOR TELEGRAPH CIRCUITS AT THE MODULATION RATES FROM 50 TO 200 BAUDS

1. *Introduction*

Olivetti, Ivrea, Italy, has made a Low-speed Data-transmission System with variable length block division and automatic procedure for complete error correction.

2. *Flexibility of the system*

The Olivetti system can be used on actual telegraph circuits—i.e. on either leased lines or switched networks and in both cases with half-duplex circuits ; in particular, it can be used on the international telex network.

The links which use regenerative repeaters or include trunks with ARQ systems impose some limitations and in particular the use of a five-unit alphabet.

In all other cases the transmission can be made with any code, and the original alphabet and format of data can be maintained so that the transmitted data can be processed exactly as the locally originated data.

Provisions are taken when necessary in order to avoid the long series of start elements capable of producing an unwanted break of the connection.

The description covers the transmission of data recorded on punched tape, with any alphabet ; nevertheless, some considerations which follow are applicable also in the case of data recorded on other media as well as in the case of a processor directly connected to the line.

3. *Composition of the system*

The block diagram of the system is shown in Figure 1, in the case of use on the telex network.

At both sides a switching box (D) is inserted between the teleprinter (A) and the line unit (B) which form the normal subscriber set.

The data-transmission equipments are connected to this switching box.

This arrangement does not require any modification in the normal subscriber set.

The data-transmission equipments are :

a) send side : an automatic photoelectric bidirectional tape reader, with two independent reading heads for error control.
An electronic send unit (G) and a circuit (E) for the recognition of the sequence SSSS.

b) receive side : an electronic receive unit (H), and a circuit (E) for the recognition of the sequence SSSS.
A tape reperforator (I), with monitoring signals for error control.

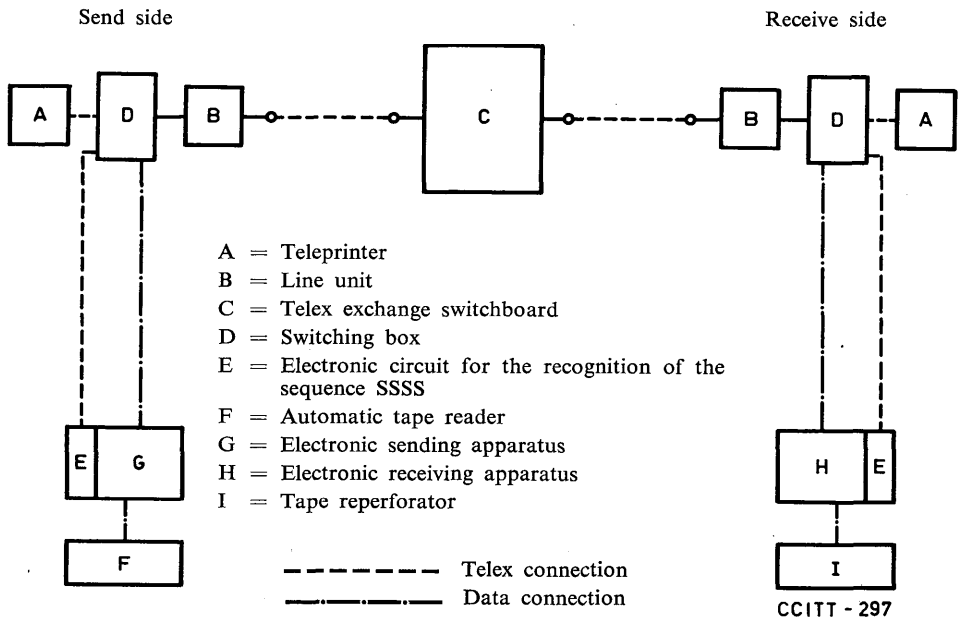


FIGURE 1. — **Block diagram of "Olivetti Data-transmission System" on telex circuits**

4. *Performance*

a) *Error protection*

The use of a cyclic code for generation of the redundancy bits gives an efficient protection against all kinds of error.

b) *Automatic procedure of operation*

The system uses the procedure described in the Annex to the present Supplement ; such a procedure gives the possibility of automatic operation as follows :

b.1) *Start of operation*

Only one operator at either of the two terminals is required to start the automatic operation of the system.

b.2) *Automatic error correction*

The system does not require the intervention of the operator for the correction of both the line and local errors.

b.3) *Automatic return to telex at the end of the transmission*

When all the data have been transmitted, the system switches automatically back to telex after a given time, without need of any special code.

b.4) *Permanent irregular condition*

In the case of a permanent irregular condition the data transmission is stalled and the return to telex is obtained automatically as explained above.

b.5) *Intentional interruption of data transmission*

When an urgent telex message is to be sent, it is sufficient to stop the data transmission in order to obtain the return to telex.

b.6) *Resuming of transmission after an interruption*

After an intentional or unwanted interruption, the transmission of data can be resumed from the point of interruption, without danger of missing or doubling any block.

c) *Transmission at speeds of more than 50 bauds*

When allowed by the line circuits involved, it is possible that at the moment of switching from telex to data the modulation rate changes automatically from 50 bauds to another rate up to 200 bauds.

ANNEX

A procedure for low-speed data transmission with automatic retransmission of erroneous blocks

Note. — On the left side of the page reference is made to the lines of the attached block diagram.

1. *Establishing the telex connection*

According to the normal procedure.

2. *Switching to data*

(a) a) At one terminal of the connection, the operator sends the sequence SSSS : as soon
(b) as the sequence is completed, switching to data is automatically accomplished at both terminals.
A return signal A is transmitted by the receive toward the send terminal. As soon as this signal is acknowledged, the tape reader starts transmission of the first block of data.

(c) b) After the transmission of signal A, the receive terminal waits for the first block, which must arrive within a fixed delay r, that depends on propagation time. A delay longer than r is interpreted as if A has not been received correctly by the other terminal, hence A is repeated.

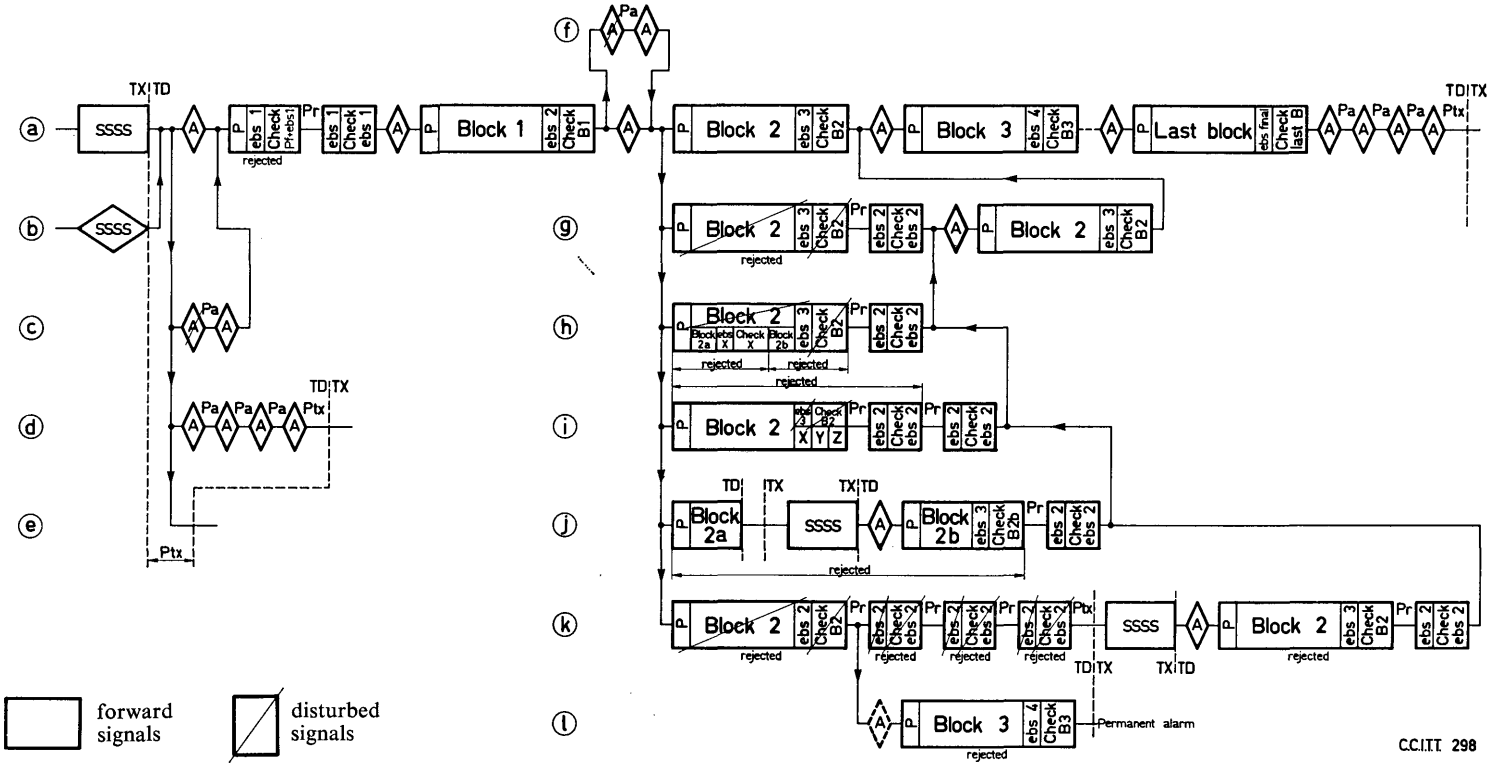
3. *Switching back to telex after an unsuccessful attempt at transmission*

(d) a) If the sequence SSSS is not received correctly by the send terminal, or if this terminal is not ready for transmission, its switching to data does not take place and signal A has no effect. This signal is repeated four times, after which a stop polarity follows. A timing circuit causes the receive terminal to switch back to telex, after the stop polarity has lasted for a period p_{tx}.

(e) b) If the sequence SSSS is emitted by the send terminal but for some reason the receive terminal does not switch to data, no signal A is sent on the line and the send terminal switches back to telex after an interval p_{tx}.

4. *Transmission of data*

(a) a) As soon as signal A is acknowledged the reader starts to read the data step by step. The tape stops when the end-of-block character is encountered.
Each block is headed by a prefix code P and followed by the redundancy codes. A block may be formed by a single end-of-block code : in some instances this happens to be the case of the first block.



Block diagram — Automatic data-transmission procedure, with rejection and request of repetition of the erroneous blocks

At the beginning of transmission, the first block is always rejected, even if it does not contain errors, to give assurance that the correct sequence of the blocks is maintained in case a transmission is resumed after an interruption. The procedure of repetition is the same as the one which will be described in the case of erroneous blocks.

At the receive side, the prefix and the redundancy codes are not perforated. The end-of-block code is stored in a buffer during the check of the redundancy : if there is no error, the end-of-block code is perforated ; if there are errors the error code is perforated in lieu of the end-of-block code.

If there is no error, a signal A is sent to the transmit side. When this signal is received, the tape reader reads the following block.

(f) b) If signal A is not received correctly, it is repeated after an interval p_a.

(g) c) If the block contains errors, signal A is not emitted. In the absence of signal A, after a sufficiently long duration p_r in the stop polarity, the reader backs up one block to the end-of-block code which precedes the rejected block. This code is sent without prefix and followed by the redundancy codes, and forms a very short block, which permits a rapid check to determine if the condition which caused the repetition request is transitory or permanent.

If the repeated end-of-block code is received correctly, it is perforated on the tape ; a signal A is sent to the transmit terminal and the rejected block is entirely repeated with its prefix and redundancy codes.

The error correction procedure is the same when the useful information or the prefix or redundancy are affected by an error. Errors in redundancy can be intentionally introduced when there is an error in the tape reading. In case of perforation, the block is equally rejected.

(h) d) In case an error transforms a code into the end-of-block code, the following characters are interpreted as redundancy codes (obviously wrong), and the first part of the block as well as the remaining part are rejected. The only difference from the previous case is that now the wrong block appears as two erroneous blocks.

(i) e) If the end-of-block code is transformed by an error into any other code, the perforation is not stopped and both this code and the redundancy codes that follow are considered as data and are perforated.

After an interval p_r of stop polarity, which follows the block, the reader goes back and sends the end-of-block code that precedes the erroneous block. The redundancy will indicate an error and further repetition of the same end-of-block code will take place.

(j) f) Whenever the connection breaks during the transmission of a block, the transmission can be resumed correctly from the point at which it was interrupted, through the rejection of the first block transmitted (always made).

(k) g) In the case of permanent faulty conditions, after four unsuccessful attempts at transmission, switching back to telex is done automatically at both ends when stop polarity has remained steady for a fixed period p_{tx}. When the normal conditions have been restored, the transmission can be easily resumed from the right point without any special procedure or readjustment of the tape.

h) If, after a rejected block, a line disturbance simulates a signal A, thus causing the improper transmission of another block, the presence of the unwanted prefix P that heads this block makes the equipment come to a permanent stop and an alarm situation is signalled. In this case, the intervention of the operator is necessary. Such an event, however, is extremely improbable.

5. *Switching back to telex at the end of data*

(a) At the end of data, always marked by an end-of-block code, after emission of signal A repeated four times a long stop polarity produces a switch back to telex.

Supplement No. 10

CANADA (DEPARTMENT OF TRANSPORT) — *(Contribution COM Sp. A/No. 47 — January 1963)*

180-200-BAUD TELEGRAPH-TYPE SWITCHING SERVICE FOR DATA TRANSMISSION

There is a definite need for a 180-200-baud duplex data transmission system using telegraph type circuits over public switched networks, to cater for the large quantity of existing business machines which are capable of delivering and reproducing data in card or tape format at rates up to 20 characters per second. It is also to be noted that suppliers of teleprinter equipments are conscious of this need and have made available readers and punches operating at this speed, with development work well advanced to produce printed documents.

The existing potential market can be realized if a public switched network of suitable characteristics is offered at a satisfactory tariff level. It is our view that a binary switched system will most effectively meet these requirements. The characteristics of such a system are :

1. Transmission of digital information by d.c. methods over balanced loop.
2. Use of electronic relays with low current and voltage characteristics.
3. The integration of 180-200 bauds into the telex switched networks.
4. No restrictions on codes or speed up to the maximum permissible in the system.

Loops

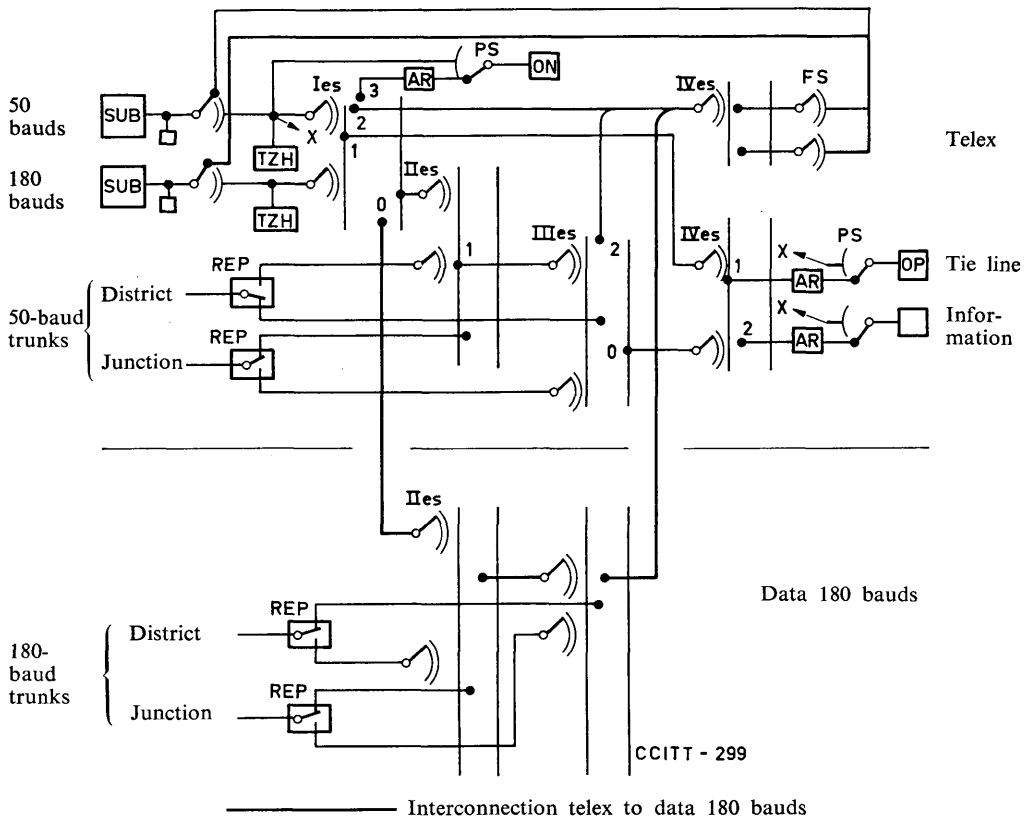
To avoid interference from data into voice, or conversely, the loops must be operated in a balanced arrangement with voltages and currents regulated to the extent necessary to meet this requirement. Equipment has been designed and tested over loops which indicate that speeds considerably in excess of 180-200 bauds are practicable.

Relays

Electronic relays are presently available which are capable of meeting the requirement for loop and trunk transmission. It is believed that a serious investigation into the requirement for such relays will produce equipment with the same effective characteristics but at considerable reduction in costs.

Integration with existing switched networks

The development of such service will be very considerably expanded if the 180-200-baud services are integrated with the existing telex network. A circuit arrangement providing for integration of 180-200 bauds with standard dial telex switching is shown on the sketch below. It will be noted that this arrangement provides for common usage of portions of the existing telex network and permits equipping a customer with 50- and 180-200-baud services. The ability of a customer to use the same loop and associated time zone meters for transmission of telex or medium-speed data transmission offers a very significant advantage to the customer.



Integration plan telex and medium-speed switching

Codes

The various classes of codes which are presently used in customer services make it necessary that the medium-speed data transmission services be permitted for any code combination or speed within the limitations of the speed of the system.

Considerable work is going forward with a view to standardizing a code arrangement for all data and teletypewriter services. If such standardization is successfully completed then the use of regenerative repeater equipment could be considered. Pending standardization, regenerative repeaters would not be possible and the system provided would be limited by the distortion characteristics of permissible sections which could be operated in tandem.

Modems

The characteristics of modem units for trunk and customer operation would employ frequency division methods and be designed for frequency modulation (frequency-shift key). The standardization of channel frequencies within a carrier telephone channel should conform as nearly as practicable to frequency-division methods used in telex : i.e. 360-cycle spacing, or it could be made to conform to the 340- or 300-cycle spacing currently in use on the American continent. A choice of these frequencies would provide from 5 to 7 trunks per voice circuit.

Supplement No. 11

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 78 — August 1963)*

200-BAUD DUPLEX DATA TRANSMISSION OVER SWITCHED TELEGRAPH NETWORKS

There is the method of telegraph switching offering itself for the transmission of digital information and justifying a thorough study because of its efficiency. The transmission of data according to the principles of telegraphy requires either the modification of an existing telex network or the setting-up of a new network specially designed for the needs of data subscribers. Considering that the operational demands of data subscribers are not the same as those of telephone subscribers, a modified telex network or, if appropriate, a new special network would be advantageous in so far as it could more readily be adapted to special demands (broadcast technique, fastest establishment of calls, intermediate storage technique in the exchange, etc.). According to its principles, the telegraph technique offers also the following advantages:

— the adverse effect of contact faults with regard to the error rate is considerably reduced by the high keying voltage existing in the exchange;

— operation with unattended subscriber stations does not require the application of expensive techniques at subscriber stations;

— identification of subscribers can be performed in a simple way.

Considering that such properties are inherent in a network operated according to the telegraph exchange principles, it is advantageous also to make provisions for data transmission over a modified telex network or a special network and to draw up relevant recommendations. The following refers to some technical features which would be of interest with regard to such a network.

1. *Speed*

In the light of experience gained in several countries during the last few years, it can be taken for granted that in practice there is much traffic which cannot be handled satisfactorily with 50-baud operation but does not, on the other hand, need the speed available when using a full channel of 300-3400 c/s. Several proposals suggest an upper limit of about 200 bauds for low-speed data transmission, which seems to be a useful compromise for many tasks to be performed in practice. As far as intercontinental transmission paths are concerned, the relatively great difference in charges against circuits with high transmission speed may be a considerable incentive for users with smaller amounts of data. From the technical point of view this limit is, *inter alia*, warranted by the fact that it allows the use of mechanical terminal equipment being relatively simple and of moderate price. Investigations made at mechanical telegraph relays indicate that up to 200 bauds they have a high reliability, a long life and minor distortions. Test results on

the above will be submitted by the German Administration. From the viewpoint of transmission technique a speed of 200 bauds seems to be a reasonable limit for low-speed data transmission, since up to this speed subdivision of the speech channels into V.F. telegraph channels can still be considered as suitable and economical.

2. *General comments*

From the technical point of view, a telegraph-type switched network operated at 200 bauds consists of three groups of facilities :

a) subscriber lines,

b) exchange equipment,

c) transmission channels.

In general, the transmission channels will be the characteristic feature of this network. The use of subscriber lines and exchange equipment of other networks (e.g. of the 50-baud telex network) should not be excluded from the very beginning ; however, this question will have to be answered separately for each case according to the existing technical conditions. In the interest of an optimum flexibility all equipment should be able to transmit at any speed below a specified upper limit, as is common practice in the existing telex network.

3. *Subscriber lines*

Test transmissions on subscriber lines up to 23 km indicated that double-current keying allows the bridging of distances occurring in practice in a reliable way and with minor distortions. This applies to the speed range of 200-2400 bauds, where electronic relays are required for speeds higher than 200 bauds and where, because of crosstalk, reduced telegraph voltages and, if required, soft keying have to be used. Hence, a full-duplex connection of the subscriber station by means of a four-wire circuit is proposed for 200-baud operation. A two-wire connection with half-duplex operation should, however, not be excluded. In this case, use had to be made of "counter-voltage keying".

4. *Exchange equipment and procedures for the establishment of connections*

Signalling will depend on whether use is made of a modified telex network with teleprinter machines at subscriber stations or a special network is set up.

In the first case, signalling will be the same as that employed in the national telex network.

In the second case, neutral signalling should be applied, which allows a fast dialling method and where no use is made of printed signals.

In the case of unattended stations, there must be the possibility of an irreproachable identification of the subscribers taking part in a call. This could be effected in two ways :

a) the calling subscriber asks for the answer-back of the called subscriber and verifies it upon receipt ;

b) the calling subscriber transmits a code-group, exclusively assigned to the respective called subscriber. Verification is done by the called subscriber. If the transmitted code-group is not the one assigned to the called station, the call is cleared-down by the latter.

A decadic pulse information is also proposed for the answer-back according to a) and the code according to b).

If, upon establishment of a call, changes of speed are to be made at the subscriber equipment, supervision of the impeccable performance of the change-over should not entail any difficulty. Upon the clearing-down of the call, whether on purpose or inadvertently, all terminals should automatically go back to a defined state.

5. *Transmission channels*

The standardization of 200-baud transmission channels has already been the subject of studies of Study Group IX. Two frequency schemes were drawn up at the 1963 meeting of Study Group IX. The 480-c/s channel scheme should be proposed—especially for world-wide links—with regard to a high-quality transmission, and to the great many sections connected in tandem.

The use of code-restricted synchronous transmission links and regenerative repeaters in this network should be excluded from the very beginning to allow a service being as flexible as possible.

6. *Possible application of several speed ranges*

A network set up according to the above guiding lines may be used for slow and medium speed data transmission. In the latter case broader channels and electronic relays will be required additionally.

According to the investigations of the German Administration, made over two and three sections with several selector stages, impeccable operation will be possible up to at least 1200 bauds, even if using conventional exchange equipment. In this connection the error rate is not expected to be below that experienced with the 50-baud telex network.

Supplement No. 12

SPECIAL STUDY GROUP A — *(Extract from report COM Sp. A/No. 34 — October 1961)*

REPORT OF STUDY GROUP A ON ERROR-CONTROL SYSTEMS FOR USE ON THE INTERNATIONAL TELEX NETWORK

1. *Terms of reference*

Study Group A has to formulate the points requiring standardization for error-control systems capable of transmission of data over international telex circuits which may include start-stop regenerative repeaters and synchronous section but not including ARQ radio.

Two basic procedures were to be assumed :

a) information feedback systems over a full-duplex circuit, and

b) block transmission systems, for use over a half-duplex or full-duplex circuit.

2. *Information feedback systems — full-duplex circuit*

a) *Points requiring standardization for a full-duplex system with automatic error correction*

The following features require standardization :

i) Setting up of the circuit would be done under normal telex procedure. Then the data transmission equipment, which may employ any five-unit alphabet, shall be switched into circuit. Such teleprinter equipment which could lead to false operation should be switched out.

ii) It must be possible for either the "calling" or the "called station" to originate the "switch to data" signal. A signal taken from the data alphabet may be required to start transmission of data from either the "calling" or the "called" station. For security against unauthorized access it may be necessary to provide special checks which should not be standardized but left to the user to arrange.

iii) To safeguard the users against false clear-down it is necessary, for the present, to limit the successive transmissions of the character containing 120 ms of start polarity, followed by the stop element, to not more than two such characters. It is possible for a data transmission system to provide a pause, of the duration of one character at least, in transmission after the occurrence of such a character. This would overcome the restriction on the user in respect of the alphabet.

iv) For automatic correction of errors a "cancel signal" must be sent to the receiving terminal to indicate that the "check-back" is faulty and a repetition is about to occur. The signal sequence SSFF has been suggested. Would this sequence be satisfactory ?

v) The number of characters to be stored at the sending end to provide the "check-back" must be specified. Most circuits would operate satisfactorily with storage of four characters ; very long circuits may require six characters, but where time-division systems are encountered storage would have to accommodate the additional delay of such systems. However, the size of the character storage may be restricted to four characters by withholding further transmission until characters are returned. This will reduce the efficiency of the path on long circuits. The method to accommodate connections with different delay times is described in Annex 1. The number of characters to be stored at the receiving end must be increased by the number of characters comprising the "cancel signal".

b) *Points for consideration*

vi) Means must be provided to overcome simulation of the "cancel signal" by the data.

vii) The variation of loop propagation time from one connection to another, and that occurring during transmission over synchronous systems, should be accommodated automatically (see Annex 1).

viii) In general the requirement is for "clean copy" but there may be cases where something less is acceptable.

ix) Where necessary, the receiving terminal must provide regeneration of the signal. This may add 10 ms delay. Such regeneration may be inherent in the terminal equipment.

x) It cannot be assumed that the cancellation sequence will always arrive at the receiving terminal without error and the sending terminal can only check whether this sequence returns correctly ; when an error is detected it may have originated in either the forward or the backward channel. It is essential that the sending terminal should, at least, give an alarm when the cancellation sequence does not return correctly.
It is also necessary to ensure that the four characters which are cancelled at the receiving terminal commence with the first character which fails to check correctly at the sending terminal. It is therefore necessary to ensure that the sending terminal transmits exactly three characters after the one which did not check correctly and to make sure that three characters are received back before the cancellation sequence. If these characters do not return it is necessary to take similar action to that for the case where the cancellation sequence does not return correctly.

xi) For certain applications the teleprinter may be unnecessary even for setting up the circuit.

c) *Points requiring standardization for a full-duplex system with automatic error detection*

For some applications error detection only may be sufficient; this system is simple, cheap and provides the same high rate of detection of errors.

The features for standardization are:

i) "Switch to data" signal—See 2. a) i).

ii) Number of characters to be stored—Four characters stored at the sending end only would be satisfactory for most circuits—see 2. a) v).

iii) Since the error is detected at the sending terminal, how should the receiving terminal be warned, and how should the message be corrected? A simple routine for indicating an error on the tape may be sufficient, transmission having been stopped when the error was detected.

iv) The variation in loop propagation time from one connection to another should be accommodated automatically (see Annex 1).

v) If the use of the system were limited to calls having not more than four V.F. links in the loop, regeneration would not be necessary.

vi) In general an automatic correction system terminal equipment should be compatible in working to an error-detection system terminal. Obviously the correction would be invalidated but the operation of detection should be effective.

d) *Points requiring standardization for a full-duplex system with visual or mechanical error detection*

i) For some applications it may be sufficient to make a "local record" by means of "message feedback" and to make a subsequent check that the message has been correctly transmitted.

ii) If the remote station is required to change its mode of operation for this service (e.g. to insert a regenerator for the retransmission of the message), the signal for switching in the special equipment must be specified.

iii) As 2. a) ii).

iv) As 2. a) iii).

v) As 2. c) v).

vi) The receiving station must provide for the return of the signals with regeneration.

3. a) *Points requiring standardization for the block check system, for half-duplex or full duplex with automatic correction*

i) Setting up — As for 2. a) i).

ii) Origin of "switch to data" — As for 2. a) ii).

iii) To safeguard the users against false clear-down it is necessary, for the present, to limit the succession of characters containing 120 ms of start polarity, followed by the stop element to not more than two such characters (i.e. combination No. 32 in Alphabet No. 2). It is possible for a data-transmission system to provide a pause of, at least, one character in transmission after the occurrence of such a character. This would overcome the restriction on the user in respect of the alphabet.

iv) Fixed block length — The block length must be specified. This may have to be related to a punched card. For this it is suggested that a block length of about 500 bits is suitable. For other forms of data, block lengths of 240 bits or 120 bits are considered suitable when using five-, six- and eight-unit codes, whereas for five- and seven-unit codes block lengths of 245 bits or 140 bits are considered suitable, when regrouping a more than a five-unit code into a five-unit code is employed.

v) Each fixed block must be provided (where required) with "null" characters to complete the transmission of each block.

vi) Variable block length — At the end of each block an "end-of-block" signal must be provided, e.g. as described in Supplement No. 7. There it has been proposed to use a sequence of four characters which are, in Alphabet No. 2, the combinations Nos. 29, 27, 29, 27 (Letrs, Car. Ret., Letrs, Car. Ret.).
Any alphabets in which this sequence is sufficiently unlikely to appear in the message may be used without special measures. Other alphabets require special supervision of the message and special handling.

vii) Redundancy should be transmitted in the forward direction ; this redundancy may be as described in Annex 2. It is the same for fixed and variable block lengths. It comprises 12 redundancy bits which may be grouped into three "check" characters and is designed to provide a reduction of the rate of undetected erroneous blocks of approximately 4000 : 1. This ratio does not depend appreciably on the length of the block in the range referred to above.

viii) Repetition procedure — For the case of fixed block lengths a suitable procedure is shown in Annex 2 ; for the case of variable block lengths in Supplement No. 7.

ix) If error detection only is required, the procedure can be simplified considerably, since in this case each block has to be transmitted together with its "check" characters only. The check may be carried out immediately by the receiving equipment or by the data-processing equipment.

Note. — The redundancy recommended for block transmission provides a very powerful means for detecting errors during transmission and does not necessitate very much equipment. This coding arrangement gives adequate protection for a message which contains no redundancy.

If the message contains a considerable amount of protection (exceeding a parity bit per character) and it is required to retain this protection in the message, it may be questioned whether it is still necessary to employ the full redundancy recommended for normal messages. It needs to be considered whether in the interests of economy and simplicity it is desirable to indicate a very simple code which could be expected to detect and correct a sufficient proportion of the errors to ensure that the over-all service could be provided with a quality approximately equivalent to that provided with the coding arrangements proposed for the messages with no redundancy. Such a simplified code would be interesting only if it permitted a significant reduction in the amount of equipment needed for error-correction purposes.

This question cannot be resolved at the present time but might justify study.

ANNEX 1

A method whereby loop transmission delays can be accommodated with a minimum of storage, together with automatic phasing for checkers

It is believed that the vast majority of telex connections will have a loop propagation delay time of less than 450 ms ; this is equivalent to three character periods of 150 ms each. Thus the storage required for automatic error correction on such connections would be four characters.

On those connections with longer loop propagation delay times additional storage would appear to be necessary.

However, considerable economies can be made in the quantity and cost of the equipment by designing the equipment to cater for efficient transmission over the majority of possible telex connections, and allowing the efficiency to be reduced over the minority of longer connections (by efficiency is meant the length of time required to transmit a given number of characters, and it is not meant to refer to the accuracy with which the data will be propagated). Such a system can be designed as follows :

The data characters as they are sent to line, at the transmitting terminal, are inserted into a four-character store. They are inserted cyclically in such a way that the first character is always put into the first section of the store, the second character into the second section, and so on. As each character is returned around the looped circuit it is checked with the stored characters ; the first returned character is compared with that in the first section of the store, and so on. Whilst returned characters are correct the related sections of the store are automatically cleared, thus allowing the insertion of subsequent characters. The fifth character sent is put into the first section of the store, the sixth character into the second section, etc. Providing that the transmission delay time for the transmission of a character around the looped circuit is less than approximately 450 ms, there will always be a section of the store free for the insertion of another character.

When for any reason, such as a very long loop transmission delay time, the store becomes full and no section is free to accept a new character, further transmission should be interrupted until such time as when a character is received back to clear a section of the store.

Should a returned character contain an error, such that it does not check correctly with the stored character, the section of the store containing this particular stored character should not be cleared. In this way the transmission of data will continue, normally, until all sections of the store are full, when further transmission will be interrupted.

Thus loop-transmission delays greater than 450 ms can be accommodated with a minimum of equipment. Such large loop delays though possible are likely to be in the minority.

It will also be seen that with the above method automatic phasing is achieved between the stored characters and those returned around the loop.

ANNEX 2

Proposal for a data transmission system for the telex network (fixed block length)

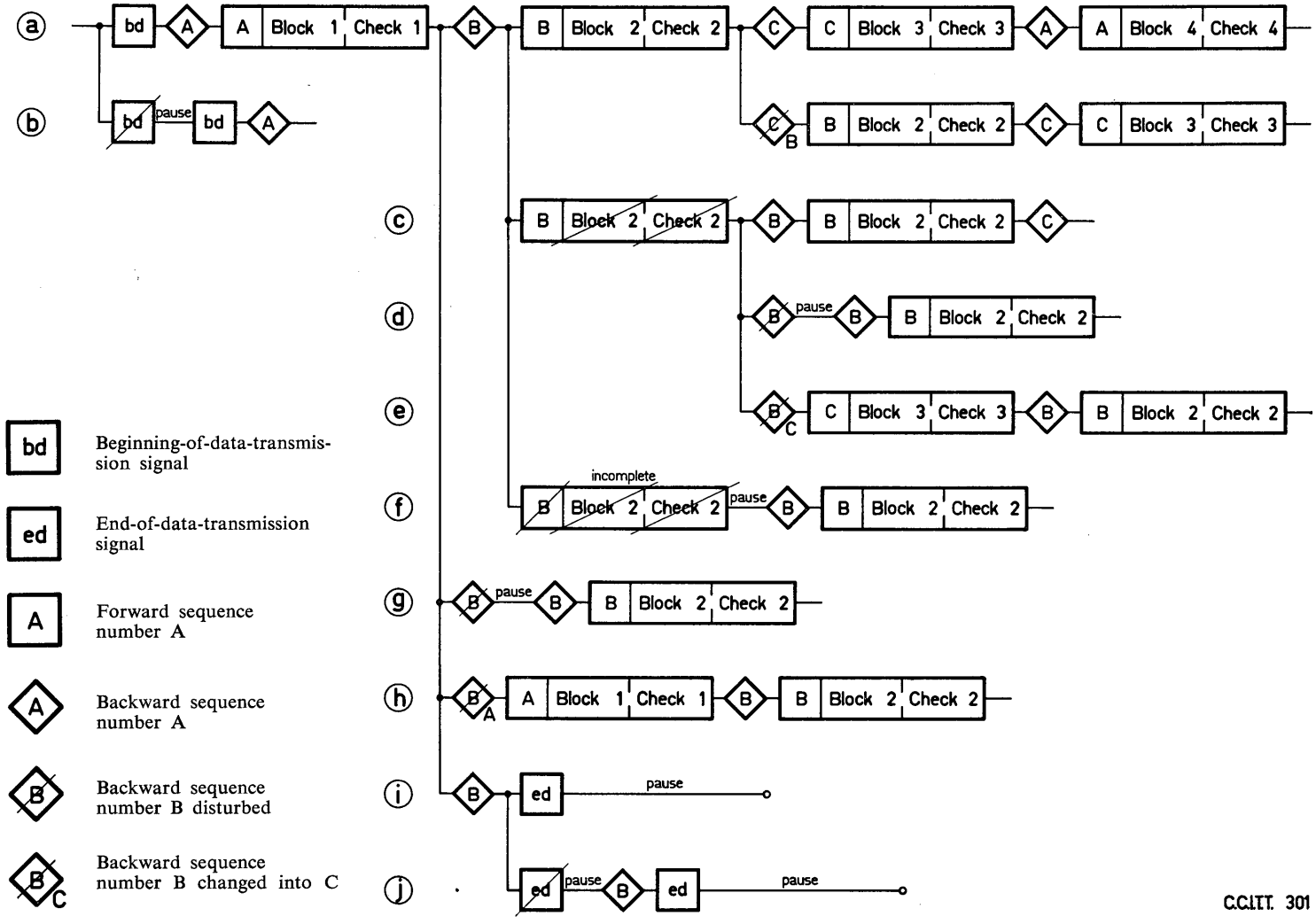
1. The telex connection will be set up as recommended by the C.C.I.T.T., answer-back signals will be interchanged, normal text will be sent using the International Telegraph Alphabet No. 2.

2. Data transmission will be announced by a special switch-to-data signal.

3. As soon as the receiving station is prepared to accept data, it calls for the first block by transmitting backwards the sequence number A, represented e.g. by the letter F (combination No. 6) (see the diagram line a).

4. After having received the first sequence number A correctly, the sending station will proceed to transmit the same sequence number A followed immediately by the first block with a fixed number of bits and its check 1. (The generation of the check signal is described later.) It is proposed that the sequence number be not included in the check.

5. If the received check 1 is in accordance with check 1 generated locally, the receiving station will call for the next block by sending backwards the sequence number B, represented e.g. by letter G (combination No. 7).



Half-duplex system fixed block length

6. This procedure continues with sequence No. C (e.g. letter H, combination No. 8) and so on using the same sequence numbers A, B and C over and over again. In this way, a certain compelled system is built up with the effect that unwanted repetitions or omissions of blocks will be detected.

7. The diagram lines *b* to *j* show the procedures in the different possible cases of disturbances.

— *line b :* (left) If the "switch-to-data" signal is not answered, it is repeated after a pause of a fixed duration, e.g. 1.5 s min. and 2.5 s max.

— *line b :* (right) If by chance the backward sequence No. C is changed into B, block 2 is repeated although not having been called for. Block 2 is rejected and block 3 called for once more.

— *line c :* If block 2 or check 2 is disturbed, it will not agree with check 2, generated at the receiving end, and block 2 will be called for once more by sequence No. B.
If the forward sequence number in front of the block is disturbed and therefore does not agree with the expected one, the whole block is disregarded. A repetition of the block is called for.

— *line d :* If the backward sequence number is disturbed and hence no block is sent out, the sequence number is repeated after a pause of fixed duration (see above).

— *line e :* If backward sequence No. B is changed into C, the called repetition of block 2 has not been followed. Therefore block 2 is called for once more.
If the sending terminal stores 2 blocks, block 2 will be repeated. If it is equipped for one block only, an alarm will be given.

— *line f :* In the case of any interruption of the line, it may appear that the block or the check remains incomplete. Then, after a pause (see above), backward sequence No. B will be repeated.

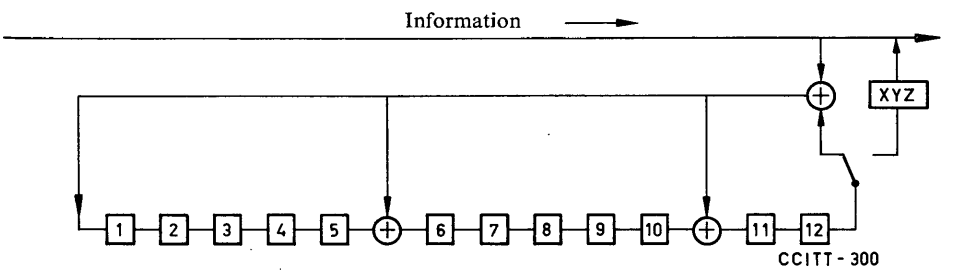
— *lines g and h :* Similar to lines *d* and *b* respectively.

— *line i :* At the end of the data transmission an "end-of-data-transmission signal" if required may be sent followed by a pause of e.g. 1.5 to 2.5 s. This pause is essential to distinguish between the real end of data transmission and the case that a simulated end-of-data-transmission signal appears by chance in a block.

— *line j :* It describes the case of a disturbed end-of-data-transmission signal. The same procedure applies if the pause is disturbed.

Generation of the check signal

1. For the generation of a check signal for data blocks a cyclic code is recommended.



Feedback shift register

The figure shows an example using a feed-back shift register with 12 cells. At the sending station as well as at the receiving station a shift register of the same form is connected to the information path. The shift pulse is in synchronism with the bit rate of the information flow. It is advisable to count the "zeros" of the data message (instead of the "ones", which is usual). That means that all data bits have to be inverted before being fed into the shift register (not indicated in the figure).

2. Counting begins with the first bit after the sequence number and ends with the last bit of the block (of predetermined length).

3. At the sending station after the end of the block the shift register is read out and its content of 12 bits is used to produce three characters, which are appended to the block. It is suggested to group the 12 bits into three five-unit combinations in such a way that the first four elements of each character are used for the redundancy bits.

Note. — The fifth elements of these three characters can be used, e.g., to produce printable combinations in Alphabet No. 2 by making the fifth elements to be of opposite polarity to the first elements.

4. The idle position of the shift register may be "all zeros" but preloaded with one "one" before the data are fed into the register.

5. The described cyclic shift register is designed to have the following features for block length as referred to :

— Reduction factor $1 : 2^k = 1 : 4096$, i.e. in the average only one block out of 4096 erroneous blocks is not detected.

— Hamming distance $d = 4$, i.e. all erroneous blocks containing less than four erroneous bits are detected.

— Resistant against error bursts not longer than 12 bits (a burst being defined as the sum of all bits in a block between the first and the last erroneous bit, both included).

— Low equipment expenses.

Supplement No. 13

FRANCE — *(Contribution COM Sp. A/No. 77 — August 1963)*

PROTECTION AGAINST ERRORS BY INFORMATION FEEDBACK

The method of protection against errors by information feedback whereby the signals received by the receiving station are sent back via the return channel and compared with the signals transmitted was considered very interesting by Special Study Group A.

The French Administration has studied various solutions based on the application of this principle. From these studies it has emerged that the solutions can be much more simple (and consequently much cheaper) if it is accepted as sufficient to detect and signal the errors rather than to require also the automatic correction of these errors.

Detection alone without automatic correction may be of interest in cases where, in view of the nature of the message to be transmitted, it is easy to correct errors and this correction can be left to the initiative of operators, or also in cases where there is only a very slight probability that an error will appear. The system then provides certain knowledge that there is no error in a message. If, on the other hand, the correction of errors is not evident and there is a fairly strong probability that a message is wrong, the interest of the system is obviously limited : to acquire accurate information it would then be necessary either to stop transmission each time an error is detected—manual intervention being necessary to cause the last characters to be repeated and transmission to be resumed—or to have whole messages repeated. Such procedures would risk placing a heavy burden on operation and might even paralyse it.

If it is desired to correct errors automatically, each time an error is detected it must be ensured by automatic means that the last characters from the wrong character onwards are repeated. The application of this principle may in turn lead to more or less complex solutions. It may be considered enough to register a signal at the receiving end, indicating that there has been an error, and to ensure that it is followed by the repetition of a certain number of characters. The gross message received is not therefore the same as the original message, but it can be used to reconstitute a message that is the same as the original one by eliminating error signals and the repeated characters preceding them. This operation may be effected either manually by an operator or automatically by a machine which proceeds, for example, to read a perforated tape containing the gross message, interprets the error signals, identifies the repeated characters and prints or perforates on the basis of this tape a message that is the same as the original.

Finally, it is possible to try to design a system which gives directly a message that is the same as the original by eliminating all trace of errors that may have occurred and of the repetitions required by them. We shall call this the "clean copy". Such a system is bound to be more complex, especially as regards the receiving terminals which will have to include memories enabling the perforation or printing of characters to be postponed long enough for it to be certain that these characters have not given rise to the signalling of errors. On the other hand, in the systems referred to above, the receiving terminal can be fairly simple since its role is limited to registering the characters received and repeating them on the return channel.

Summing up, we shall contemplate below the following three cases in increasing order of complexity :

A : Detection only

B : Automatic repetition

C : Clean copy.

The most important standard to be defined for these systems, which conditions the very design of the system, is the procedure which should be used to enable the sending terminal which has detected an error to warn the receiving terminal thereof.

It could be considered best to use a special character for this purpose (for example, the 32nd combination of the five-unit code) or a special sequence such as the "SSFF" sequence, or the sequence "E space E space E" which is already used in normal telegraph operation. The basic difficulty of such a solution is that in the case of data transmission in the form of signals which are different in meaning from those of Alphabet No. 2, it may happen that the error signal is simulated by data and this would involve false error signallings. It would then be necessary to study the data code to ensure that such simulation is out of the question.

Another, quite different, solution which has been considered in France and which does not involve this difficulty consists in utilizing a break in the regularity of the transmission cadence

—i.e. prolongation of the stop polarity separating two characters—to indicate that there has been an error.

This solution obviously assumes that the transmission is made on the basis of an automatic reading, which will practically always be the case ; but it assumes above all that the propagation time on the circuit is more or less constant and not subject to sudden variations ; in particular, it is quite clear that this method cannot be used on connections equipped with synchronous systems (owing to "idle conditions" on the synchronous channel), or on radio circuits with ARQ systems. In spite of these limits, this method nevertheless is of great interest owing to the complete freedom which it leaves to the data code.

As regards choice of the duration of prolongation of the stop polarity which indicates the presence of an error, it appears desirable for this duration to be fairly short, for example 20 ms, the total duration of the stop pulse then being 30 + 20 = 50 ms.

Other points lending themselves to standardization

1. *Regeneration*

The regeneration of signals on the receiving side before retransmission should be absolutely insisted upon, for the following reasons :

On the one hand, the transmission circuits of the world telegraph network have been designed to ensure that the distortion introduced by the circuits when two stations are connected up remains within admissible limits, but they have in no wise been designed to ensure that the accumulation of the distortions contributed by the go and return paths likewise remains within admissible limits. The absence of regeneration on the receiving side therefore risks incorrectly multiplying the error signals.

On the other hand, it should be noted that regeneration before retransmission involves no difficulties and hardly increases the cost of apparatus, since regeneration is in any case essential for the translation of signals.

From this point of view, one of the objections that have been made to the protection system by information feedback is that errors due to electromechanical apparatus are not eliminated. But it has been demonstrated that an overwhelming majority of faults noted in teleprinter or reperforating apparatus is due to the devices for the reception and recording of information units arriving in series rather than to the printing or perforating systems themselves, which are nearly always controlled by the conversion of units in parallel. The above objection practically disappears provided that the sub-assembly constituting the regenerator input is used as reception and recording device.

2. *Number of characters to be stored at the sending end*

This number depends essentially on the circuit propagation time and it does not concern the sending end in cases A and B. The best thing is for it to be determined at the sending end either by an operator or by an automatic process as a function of the characteristics of the connection used. It therefore seems neither necessary nor desirable to standardize it.

3. *Possible variations in propagation time*

The method whereby transmission is interrupted until a character is returned if the memories are saturated is interesting. It is not, however, compatible with the use of a prolongation of the stop pulse to signal the error.

4. *Connection of error-correcting equipment*

The general method for connecting data-transmission equipment is applicable to the present case, which raises no special difficulty.

5. *Interval between the start of cycles*

This interval should always be the same on the return channel as on the go channel so that it will not be necessary to use memories on the receiving side (in cases A and B) and so that the number of characters stored at the sending end may remain constant.

This arrangement also provides some defence against disturbances which could affect error signalling, by checking that the prolonged stop polarity which characterizes the error has been correctly retransmitted on the return channel. If this is not so, a special signal can then be sent.

6. *Compatibility of different systems*

In cases A and B, the role of the reception terminals is to record and retransmit the characters received. They may therefore be identical. This means that the same receiver can operate with a transmitter that only detects or with a transmitter providing automatic correction. However, it will be necessary for the receiving end to know which case is involved in order to interpret the message received.

Conversely, in cases B and C, it is the transmission terminals which have the same function (sending and memorizing of characters, comparison after retransmission, error signalling and repetition) ; they may therefore be the same ; this means that a transmitter of this type can operate with a detection or automatic correction receiver or with a receiver giving a clean copy.

All this assumes that the same principle is applied for signalling errors. A transmitter which signals errors by prolonging the stop and a receiver designed to signal errors by means of a character sequence, or vice versa, are obviously incompatible.

In conclusion, this question of error signalling and operating method is of primary importance if it is desired to introduce world-wide use of the telex network for data transmissions. In the absence of a single standard, the risk may arise of having various incompatible systems trying to co-exist on this network.

Supplement No. 14

SAGEM — *(Contribution COM Sp. A/No. 82 — August 1963)*

DATA TRANSMISSION ON DUPLEX 50-BAUD TELEGRAPH CHANNELS WITH INFORMATION FEEDBACK

I. General

The principle of information feedback systems can be applied in many ways. If we want no more than error detection, and make do with the minimum of automatic operations, the problem is a simple one. But the matter becomes much more complicated if we wish to cut down the part played by the operators and to obtain, notwithstanding, automatic error-correction at the incoming end.

In what follows, we shall consider two possibilities which have been examined in France. As far as complexity is concerned, it might be said that they are at opposite poles. The first makes use of telegraph equipment with hardly any change. The second makes use of a special data-transmission equipment with automatic error correction, self-sufficient in point-to-point communication, and usable in a switched network when equipped with the requisite switching organs or when supplementing a subscriber's telex equipment.

II. Simple telegraph connection with information feedback

1. *Diagram*

In its very simplest form, such a connection includes :

a) *at the transmitting station*

A transmitter connected to the forward channel, and a receiver connected to the backward one ;

b) *at the receiving station*

A receiver connected to the forward channel, a regenerative transmitter connected to the backward channel, and an SSSS-sequence detecting and local switching equipment.

If, now, we assume that information feedback may not be immediate and automatic and that text corrections may be left to the operators' initiative, we shall be put to very little outlay while the system will prove, nevertheless, effective for a user who has to use Telegraph Alphabet No. 2, chiefly for the transmission of ordinary messages.

Some precautions will be necessary if actual data are to be sent over such a connection, that is to say, sequences of combinations with meanings other than those in Code No. 2—sequences in which, furthermore, operators cannot intervene either to include such operating combinations as "carriage-return" or "line-feed", or to avoid certain combinations such as "Secondary of D".

If page-printing is to be used for reception—even for reception of data—although it would seem better to use the tape from an associated perforator, we shall have either to use tape-machines or introduce automatic operation of "carriage-return" and "line-feed" in the page-printing apparatus.

A random appearance of combination No. 4 (D) of Code No. 2 in the figure-shift in a signal sequence should normally transmit the answer-back signals. Hence, as soon as the sequence SSSS appears, this particular function should be blocked in the receiving equipment.

2. *Standards to be defined*

A full degree of freedom, compatible with ordinary co-operation, can be left to users. We consider that only essential standards would have to be imposed :

a) *Choice of a switching sequence*

Hitherto the sequence chosen has been SSSS.

b) *Regenerator required at the incoming end before retransmission over the return channel*

Unless this precaution is taken, distortion will accumulate on both channels and might become prohibitive. In any event, it would multiply faults.

c) *Compulsory blocking of the answer-back unit by the sequence SSSS*

The introduction of a compulsory "carriage-return" and "line-feed" is desirable only if there is a page-printing apparatus available to receive data and if these signals are not included in the data themselves.

d) *Error signalling*

Could not the error-signalling sequence, for simplicity's sake, be the same as in classical telegraphy (E space E space E space....) ?

e) *Automatic suppression by the sequence SSSS*

Of any transmission of a delay signal in both directions of transmission of a radio channel with error correction, the introduction of such a signal meaning that a systematic error is introduced into the data transmission.

III. Special data-transmission equipment

1. *General principles*

The special equipment studied in France is designed to supplement the telegraph positions and to provide data transmission with automatic error correction.

The equipment considered may be described as follows :

a) Transmission is fully automatic, the data being punched beforehand. After contact has been made in the classical manner by the teleprinters, the data-transmission equipment, as soon as it is started, automatically repeats the sequence which puts the machine at the other end in a position to receive data. The actual data transmission begins only when these sequences are correctly returned to the transmitter, after regeneration at the receiving station.

b) The number of combinations stored, both at the transmitting station (to await the return of the modulation) and at the receiving station (to await the outcome of the comparison performed at the transmitter station) is variable, depending on propagation time, and adapts itself automatically to this time.

Experience has in fact shown that these storage devices represent a major part of the logical equipment of the machine, and also a considerable proportion of the total cost. Now, many specialized connections have to operate over fairly short distances (district branch-offices and suburban factories towards head offices, for example). Hence it seems proper that such equipment should embody just enough storage elements to cope with the number of connections expected from each user, thus cutting down expenditure to a minimum (and also transmission time, in the event of repetition).

c) When propagation time is such that the normal effect of distortion suffices to render this number of storage elements indeterminate, when, for example, in the case we are considering, it is an almost exact multiple (to within $\pm$ 10 milliseconds) of the duration of a cycle, an automatic device provides sufficient additional propagation time into the channel to abolish this indeterminacy.

d) Propagation time is measured by observing the time which, at the transmitting station, separates despatch of a signal on the forward channel and its receipt on the backward channel. So that the receiving station may measure this propagation time, this signal is made up of a continuous stop polarity which the transmitting station sends on the forward circuit, further to the last stop element of the last SSSS sequence, which it interrupts as soon as it comes back to it over the backward channel.

e) To ensure the validity of this signalling and to render it effective only if it contains no error, the transmitting station adds, automatically, an additional SSSS sequence.

The data transmission begins only if this signal sequence returns free of error to the transmitter. Otherwise (as, for example, when an unwanted start pulse appears in the continuous stop signal), the transmission stops and an alarm warns the transmitting station operator that he must begin again with the operations of starting.

f) No particular sequence has been provided to enable the transmitter to tell the receiver that an error has been observed. Clearly, such a sequence could only be taken from the data alphabet. Now in certain instances data will have to be coded. As a result, there will always

be some probability that such a sequence will be unexpectedly found in the line. The same uncertainty exists if we wish to transmit data drawn up in accordance with six-, seven- or eight-unit codes, after five-unit transcoding.

We have preferred to make use of the fact that transmission is start-stop and automatic to signal errors by a systematic prolongation of the stop elements, which leaves all the facilities afforded by the data alphabet intact. When the transmitter notices a mistake, the stop element of the combination currently being transmitted at the time is prolonged by 20 milliseconds. The transmitter reverts to transmission straightaway, from the erroneous combination put right by it.

g) To guard against a possible alteration in this error signalling, the transmitter automatically checks that the signal has indeed been returned to it, at the proper place, by the receiving correspondent. Otherwise, there will be a full stoppage of transmission, and an alarm will warn the transmitting operator that he must begin again from the combination previously recognized to be defective. At the receiver, arrangements are made for a special mark to show the combination which (in these circumstances) has to be eliminated from the sequence of punched data.

h) To guard against a possible alteration in the start elements, the transmitter checks to see that each of them is returned in its proper place by the receiving correspondent. Should one be altered, the transmitter acknowledges that it has received a faulty signal instead of a start element. The process which follows is identical to that described in the preceding paragraph, about the stoppage of transmission and the setting-off of an alarm.

i) To ensure that no more than two No. 32 combinations of International Code No. 2 are transmitted at random to line, a special device automatically transforms any third No. 32 combination into a No. 29 combination (letter shift) on transmission, while it is stored in the memory unchanged.

This systematically erroneous combination is then corrected by the error-correction device and is erased from the sequence of the reconstituted No. 32 combinations.

In this fashion, it becomes possible to transmit a sequence of No. 32 combinations of any desired length. These are restored in their entirety at reception in an uninterrupted sequence, while in transmission, in the channel, a No. 29 combination is automatically inserted after two consecutive No. 32 combinations, when a third No. 32 combination occurs.

j) Any continuous stop polarity transmitted to line for more than about one second will automatically put the corresponding stations back into the telegraph position.

Supplement No. 15

KOKUSAI DENSHIN DENWA — *(Contribution COM Sp. A/No. 57 — May 1963)*

SEND-BACK ERROR-DETECTION METHOD IN AN AUTOMATIC MESSAGE SWITCHING SYSTEM

1. *Commercial messages and send-back error-detection system*

If there is a return circuit along which to pass the send-back signals between the transmitting and receiving points, an application of the send-back error-detection system, in principle, is highly feasible. In the telex or the commercial telegraph exchange network with the answer-

back system, return circuits already in idle condition may be available. So error detection is possible through the active use of this idle return circuit.

There still are many problems that remain to be solved as to whether or not the send-back error-detection system should be adopted for the commercial telegraph exchange network. But it could be said that the introduction of the system to provide high-quality telegraph transmission network will be a significant service for the subscribers, when it does not entail a considerable increase in the equipment investment.

Generally, it would be justifiable to state that a domestic telegraph exchange network which meets the following requirements will warrant the adoption of the send-back error-detection system within a given economical framework :

a) the network is of circuit-switching type ;

b) it comprises a two-way duplex circuit in which the idle return circuit is available ;

c) the introduction of error-detection capability into the transmitting and receiving equipment involves no marked increase in cost.

In the case of the international or intercontinental commercial telegraph, on the other hand, difficulties are encountered in the application of the send-back system on an international scale, for in most cases, with special exceptions such as gentex, the transmitting and the receiving channels are constructed of two unilateral two-wire circuits independent of each other, even if the exchange networks are constructed of a two-way duplex circuit.

However, in the case of short-wave radio channels protected by the ARQ, the system connecting the international trunk circuit with local exchange network by a message relay equipment with send-back error-detection capability, which will protect the international circuit by ARQ system and the local network alone by send-back system, assumes a greater significance.

Figure 1 shows an example of the telegraph exchange network which incorporates the send-back error-detection system. This is an automatic message-handling system which will make use of the all-electronic telegraph exchange and the computer being developed by the K.D.D. The

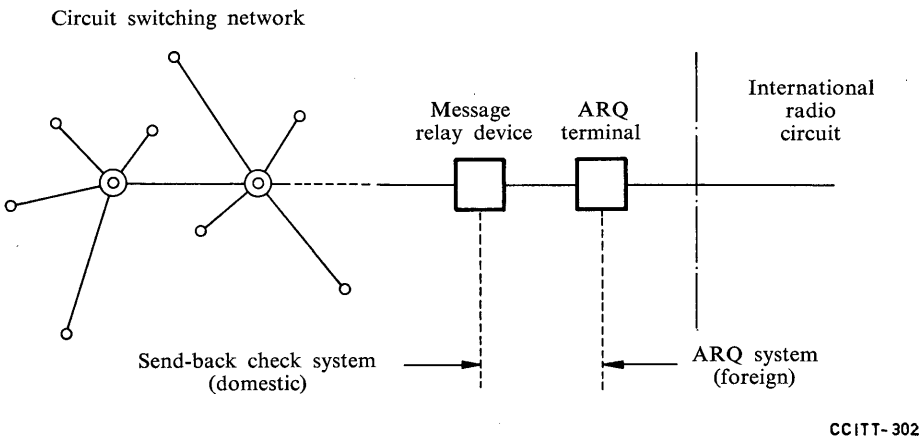


FIGURE 1. — **Application of send-back error-detection system for international message switching network**

system handles and processes commercial messages in the central office and accomplishes the relay-exchange functions all automatically. The system, incorporating the answer-back and send-back check systems for the domestic network, is composed of perfect two-way duplex circuits and is of circuit-switching type. It is connected to foreign channels via message relay equipments of magnetic tape recording type. This message relay equipment functions as subscriber of the domestic exchange network. As a relay equipment, it is so constructed as to accomplish the function of a message storage equipment for the procession under the control of the electronic computer.

Constructing the system in this manner makes it possible to protect the domestic network by the send-back method beside keeping the radio circuit free from mutilations by ARQ. In the case of an almost fully automatic system requiring no operator attendance at the receiving end, such as the K.D.D.-developed system, the return check system acquires still greater significance.

2. *Send-back system*

As the send-back error-detection is a means intended to ensure that the same message as the original has been received at the receiving end without any errors, the signals sent back should be of the final condition as received and recorded.

However, any attempt to materialize this in the most ideal form involves a complex equipment and considerably high cost. For this reason, in some cases, it would be more advantageous to send it back from an intermediate stage of the receiver than after it has attained the final receiving condition, depending on the relative considerations of the required quality of transmission, effects of error detection and the rate of errors which develop in various sections. As is evident from the above, there are many factors that make it difficult to determine which method to adopt for the send-back system.

The telegraph signal receiving equipments generally used (including relay equipment) are:

a) printer;

b) reperforator;

c) magnetic tape recording system.

Relationships between the actually used equipment to receive the signals and the stage from which send-back signals are made available may be classified conceptionally into the three categories A, B and C as shown in Figure 2.

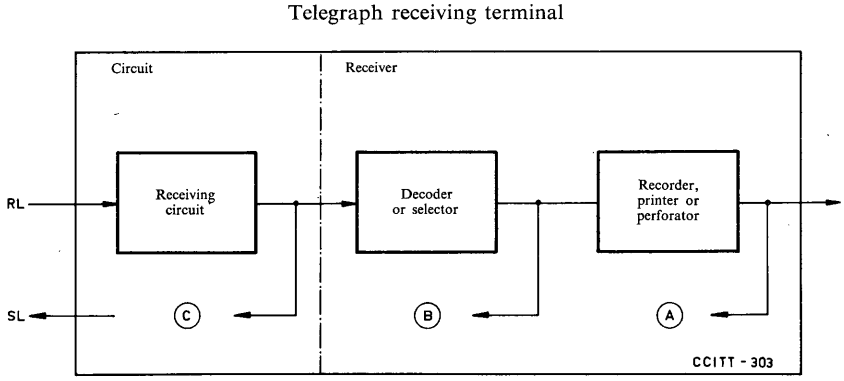


FIGURE 2. — **A signal receiver and the send-back points**

TABLE 1

Methods of send-back

Stage / Device	(A)	(B)	(C)
Printer (receiver)	Printed letters are read by the print reader and are then reconverted into telegraph signals and sent back	Mechanical or electro-mechanical condition in stages somewhere between the decoding or selecting and the printing or the reperforating process is reconverted into telegraph signals and sent back	Signal currents that flow through the selector magnet circuit are directly sent back as telegraph signals
Reperforator (receiver)	Perforated codes are read by the tape reader and are then reconverted into telegraph signals and sent back		
Magnetic recorder (message relay device)	Signals recorded on magnetic tape or drum are read out and are then reconverted into telegraph signals and sent back	Code currents that flow through the recording head after being decoded into parallel signals are reconverted into serial signals and sent back	Signals that flow through the receiving circuit are directly sent back

Category A is the method to send back the final condition under which the message is perfectly received and recorded through printing, reperforation or magnetic recording. Category C is the method which calls for the sending back of signals from the point where the signals are fed to the receiver. Category B, on the other hand, is the method which calls for the sending back of signals from somewhere between the two stages—that is, between the stage where received signals are selected, decoded or converted and the stage of their printing, reperforation or magnetic recording.

Relationships between the stage from which the send-back signals are picked up and the devices such as the printer, reperforator and the magnetic recorder are shown in Table 1.

It is by no means an easy thing to determine which method of the three, each with its own advantages, is the best in practical application.

For example, it is necessary to follow method A in cases where it is desirable to make a thoroughgoing return check. However, in cases where the printer is in use, it becomes necessary to use a print reader, i.e., a character recognition machine which is still in the laboratory stage at present. It is also expected that it will cost a great deal. For this reason a check by method A is hardly feasible.

In cases where the reperforator or the magnetic tape recording system is in use, on the other hand, a check by method A is relatively easy and will not cost very much.

Even when the printer is in use, a return check may be made by a method similar to B, by sending back the results of selection in selector mechanism or on the type bar or type wheel. It is feasible but has many difficulties to overcome. After all, it follows that method C is better in practical application. Under method C, however, the check being made is not on the final condition. For this reason, if the rate of errors that develop inside the printer is considerably high, the return check in itself may be meaningless. For example, when the error rates for the transmission path and the printer are fifty-fifty, a check by this method may justly be said to be nonsense.

In cases where the reperforator is in use, method B is not advisable as in the case of the printer. Method C may be better where the simplicity of send-back is a requirement. If a slight increase in cost is allowable, however, application of method A is the best in the case of the reperforator.

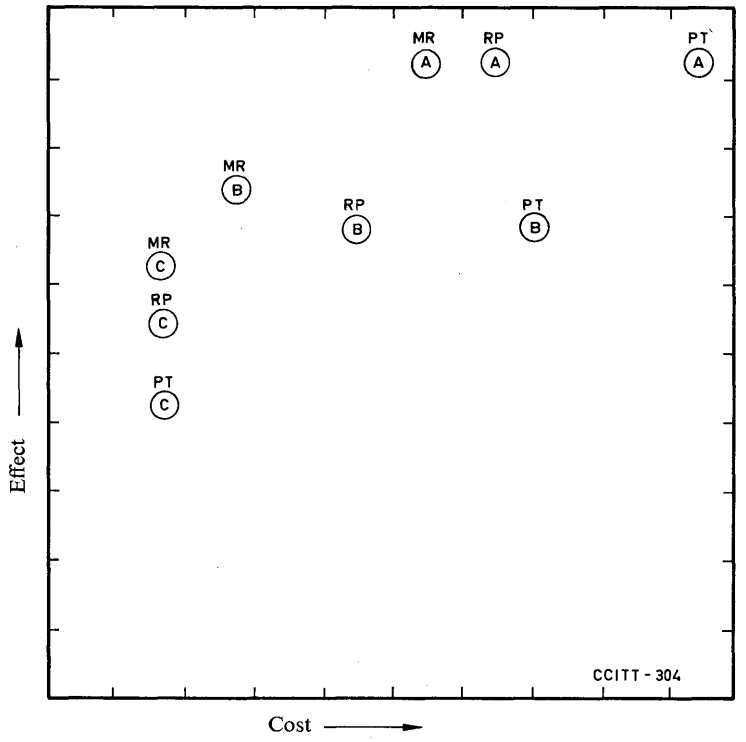
Similarly, when the magnetic recorder is in use, it is necessary to use method A. However, when the circuits inside the recorder and the recording system itself are maintained high in reliability by the parity check and dual recording method, a return check may be made adequately by method B or C to avoid the complexity of equipment necessary in method A.

Much the same thing can be said of the case where data are directly fed to the computer from a remote source.

Figure 3 is an attempt to illustrate the relationships between effect and cost mentioned briefly above.

In the aforementioned K.D.D. system, method C is used with the subscriber equipment designed for reception by the printer, that is, the method whereby signals available from the selector magnet circuit of the printer are sent back. Theoretically, it is not a perfect method but it offers the advantages of simplicity and economy. In the message relay equipment by the

application of the magnetic tape recording method, on the other hand, method B is adopted —namely, a method whereby signals available from the recording head are sent back. Under this method, however, signals once converted into bit-parallel must be reconverted into bit-serial signals. For this reason, if the circuit maintains a high reliability and extremely low error rate, we believe that switch to method C may be made to simplify the equipment.



Notes. — Numerical values have been omitted, because the figure is intended to show a general tendency.

PT = Printer
RP = Reperforator
MR = Magnetic recorder

FIGURE 3. — **One of the examples to illustrate the cost-effect relationships under each send-back system**

3. *Check system*

It is not difficult to determine which method to adopt for checking the sent-back signals.

There are two check methods, manual check and auto-check.

The manual check would serve the purpose fully when a small number of telegrams are handled in attendance of operator and there is no need for a thoroughgoing check.

However, when there is a large number of telegrams or when there is need for a thoroughgoing check or completely automatic operation, the auto-check is a must.

The auto-check may be divided into the electro-mechanical method which employs paper tape and the all-electronic method. Generally speaking, the former is relatively lower in cost but much inferior in reliability. The latter, on the other hand, costs more but ensures higher reliability. Generally, in the exchange system, the total propagation time of signal varies from point to point depending on the distance of the other party of the connection, so that each time it is necessary to measure the propagation time to make the auto-check correctly. It may justly be said that the electronic method is more advantageous for this purpose.

In the aforementioned K.D.D. system, the transmitting and receiving positions are provided with a parametron logic circuit designed for fully automatic operation. Therefore, the addition of auto-check capability to this circuit makes possible a check by the electronic method at comparatively low cost. Similarly, the electronic circuit makes easy the determination of the total propagation time by measuring the time required for the first letter transmitted to return home after completion of exchange connection (after answer-back reception). Incidentally, a check for its comparison with the original message is done on a character basis.

Supplement No. 16

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — July 1963)*

LOOP ERROR DETECTION — DISTORTION CORRECTION (NEED FOR REGENERATION)

1. *General*

One method of checking the errors encountered when transmitting telegraph signals over a circuit is to loop the circuit at the receiving end and compare the sent and received signals at the transmitting station (Fig. 1).

This system can be applied to two-way simplex circuits and is applicable to telex connections where duplex operation is possible.

Neglecting transmitter, receiver and comparator failures, this method would result in complete detection of all received errors provided the signals appearing at Y in Figure 1 are exactly the same as those appearing at X.

2. *Deficiencies of basic system*

The efficiency of the circuit shown in Figure 1 is modified by various factors and in the limit is reduced to unacceptability.

The efficiency in the detection of errors is reduced by the possibility that an element received in error may be corrected on the return path and assessed by the comparator as correct. This

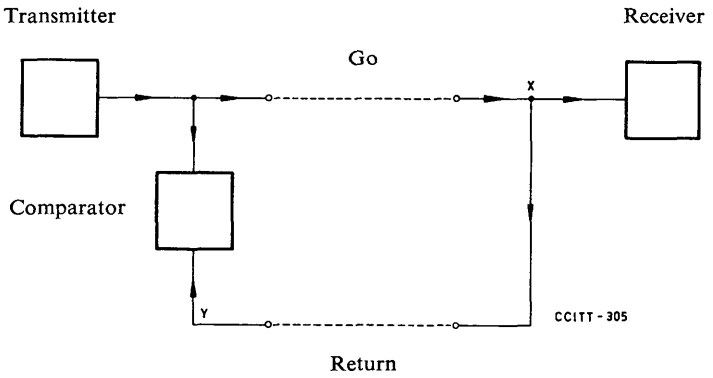


FIGURE 1

point is discussed in more detail subsequently. Two other factors may cause severe operational embarrassment. Defects which affect only the return path can cause a signal which was received correctly to be assessed as an error and involve an unnecessary retransmission. Where a connection is used which comprises a number of telegraph circuits in tandem as will be encountered on a switched network, the distortion of the signals on return to the sending end may be so high as to cause a signal to be assessed as faulty even though it was correctly transcribed at the receiving end. This will also cause an unnecessary retransmission. On a typical circuit formed of three voice-frequency telegraph circuits in tandem (i.e. when the signal received back at the sending end had traversed six channels in tandem) enhanced distortion caused an error indication to be given on average every five minutes although an actual error occurred on average each 100 minutes. These objections to the basic system can be avoided by the use of regeneration at the receiving end.

3. *Margin*

The receiving equipment at X will possess a certain receiving margin. If (as will be the case with telex) the terminal machine is a teleprinter the margin will be not less than ± 35%. Interruptions (and to a much less extent, telegraph crosstalk) will result in occasional elements being distorted beyond the receiver margin and a printed or punched error will be received. To enable this error to be detected the comparator must have a margin of not more than ± 35%, preferably somewhat less, say ± 30%, to allow some factor of safety. It cannot be assumed that the distortion of an isolated signal element will always be increased in its return to the sending end : it is equally probable that a signal element being sent back on the return path will appear at the comparator with either more or less distortion than it had at the receiving terminal.

4. *Distortion correction*

As has been stated, the original distortion of an isolated element returned from X to Y (Fig. 1) may be increased or decreased by the inherent distortion of the return channel. Consider a typical telegraph connection which has an inherent distortion of, say, 8% in both the send and receive channels. Consider also the situation where disturbances on the send path cause an element to be distorted to a value of 40%. This is liable to cause an error to be recorded on the receiving machine. There is a strong likelihood, however, that the distortion of such an element applied

to the return path would be reduced in the course of its retransmission to the sending end to a value that would cause it to be accepted as correct and the error would be undetected. The greater the distortion on the return path the greater is the probability of an element received in error being corrected on its return to the sending end and even on low distortion circuits the number of error corrections could still be significant.

5. *Tests*

Two telex subscriber stations were used for the tests. At the receiving station the send and receive wires were looped and the transmitting contact isolated. At the sending end the local record was disconnected and signals were transmitted from the automatic transmitter. Copy at both ends was both printed and reproduced on punched tape. Tests were carried out over two, three, four and five V.F. two-way simplex links in tandem. Random short interruptions were injected in the send line at the sending station. Except for those tests over artificial lines, connections were established on calls set up over the normal telex network.

The test results are listed in Table I. It is clear that under all conditions errors received at the looped terminal are frequently corrected on the return path and accepted as good signals by the comparator. Although sufficient tests have not been carried out to calculate an accurate relation between the errors corrected and those actually received an approximate relation is 1 : 10. Tests 12 and 13 show results much worse than this—the correction on artificial lines (non-relayed and relayed) being greater than those on V.F. circuits.

6. *Practical tests using an electronic comparator*

An electronic comparator has been used to check the efficiency of loop error detection. It was used in the circuit shown in Figure 1 and gave an alarm output and stopped transmission whenever an error was detected at the sending end in comparing the received signal, character by character, with the corresponding sent signals. The alarm output was coupled to a recording meter and an automatic resetting device. The latter equipment restarted the automatic transmitter and reset the comparator after the stoppage which occurred with each detected error.

Tests were made over a telex connection consisting of two V.F. telegraph circuits in tandem with and without a regenerative repeater in the loop path at the receive terminal. The results were :

a) Without the repeater at least 30 stoppages occurred in 3 hours, i.e. an average of one every 6 minutes.

b) With the repeater there were no stoppages in 8 hours.

Analysis of the stoppages under a) showed that most of them were caused by the total distortion of the signal after transmission over four V.F. telegraph channels in tandem exceeding the comparator margin and not by signal mutilation caused by interruptions.

The margin of the comparator had been adjusted to $\pm$ 30%. This means that some transitions in the range of 30-35% which had been correctly recorded at the receiving end might be shown as an error by the comparator. A comparator margin of $\pm$ 35% would reduce these false indications of error, but in the absence of regeneration at the receiving end it would be necessary to ensure that the receiver margin never fell below $\pm$ 35%.

TABLE I

Test No.	Circuit length	Number of interruptions (6 ½ ms interruptions)	Start-stop distortion		Number of errors		Remarks
			At receive end	On being returned	Receive	Return	
1	2 V.F. links	few	10 %	22 %	3	3	
2	3 V.F. links	,,	14 %	32 %	1	1	
3	5 V.F. links	,,	20 %	30 %	5	10	
4	4 V.F. links	,,	20 %	40 %	8	10	
5	Artificial line	,,	10 %	12 %	7	9	
6		,,	10 %	12 %	6	5	1 error corrected
7	5 V.F. links	500	30 %	40 %	19	78	
8	4 V.F. links	250	20 %	35 %	12	56	
9	5 V.F. links	300	30 %	40 %	11	74	
10	3 V.F. links	200	20 %	16 %	33	34	5 errors corrected
11	3 V.F.	100	20 %	20 %	16	20	
12	Artificial line	200	8 %	10 %	6	NIL	6 errors corrected
13	5 miles, 6½ lb per mile	400	8 %	10 %	7	NIL	7 errors corrected
		15 ms interruption every 6 seconds					
14	Artificial line 20 miles, 6½ lb per mile	92	10 %	16 %	92 error bursts		8 not received on return (i.e. corrected)
		10 ms or 6 ms interruption every 6 seconds					
15	2 V.F. links	74	18 %	20 %	74 error bursts		3 not received on return (i.e. corrected)

7. *Regeneration*

As stated in 6 b), when a regenerative repeater was included at the looped terminal no stoppage was recorded.

The many false indications of error that occur with a simple loop can be eliminated if the signal is regenerated immediately before the looping point at the receiving terminal. The regenerator ensures that all signals applied to the receiving instrument and retransmitted to line have very low distortion. The probability of error correction on the return path is now so low it can be considered negligible. The receiving equipment (regenerator) now has a wide margin (45-50%), consequently the comparator margin can be correspondingly wide. Hence negligible stoppages will occur due to normal distortion. All stoppages will be due to true errors either on the "go" or "return" or on both circuits. Stoppages due to other causes will be negligible.

The results of practical tests with and without a regenerator in the loop are listed in Table II.

TABLE II

Circuit	Signals	Errors received	Errors corrected on return	Loop
2 V.F. links	Mixed	150	NIL	With repeater
2 V.F. links	Repeated L/S	100	NIL	,,
2 V.F. links	,,	100	1	No repeater
2 V.F. links	Repeated R/X and L/S	62	NIL	,,
2 V.F. links	Test message	28	3	,,
2 V.F. links	Characters E - T - A	10	1	,,
2 V.F. links	Test message	19	3	,,
2 V.F. links	,,	16	NIL	With repeater
2 V.F. links	,,	125	NIL	No repeater
2 V.F. links	,,	19	3	,,
2 V.F. links	,,	16	NIL	With repeater

Interruptions were injected, as on the previous test, at the transmitting terminal, these being of 10 ms or 15 ms duration approximately every 6 seconds.

8. *Conclusions*

Loop error detection can be applied to telegraph circuits with or without regeneration at the looped terminal. Without regeneration it appears likely that of the order of one in every ten errors actually received will be corrected on the return path and thus remain undetected. In addition on long circuits the false indications of errors are so numerous as to be a severe operational embarrassment. Experience has shown that when a loop error-detection system without regeneration at the receiving end is applied to a telegraph circuit with an average basic error rate of 1 in 40 000 characters the undetected error rate would be of the order of 1 in 500 000 characters, but with regeneration it is difficult to postulate a realistic situation in which an error would remain undetected within the mechanical capabilities of the system. The application of regeneration at the receiving end has the additional advantage of allowing a greater range of distortion to be accepted and reduces to almost zero the operation embarrassment of false indications of error caused by the distortion on the return path.

Supplement No. 17

NETHERLANDS — *(Extracts from contributions COM Sp. A/No. 18 (September 1961) and COM Sp. A/No. 81 (August 1963)*

CONTRIBUTION TO DATA TRANSMISSION OVER ISOCHRONOUS SYSTEMS AND TO ERROR CONTROL

The Netherlands Administration has thought fit to give a short description of a test made with protected data transmission according to the TOR principle (Van Duuren system), as the experience gained with it may be useful.

Hereinafter a device is described that might be applied to 50-baud telegraph networks on which no use is made of regenerative repeaters or synchronous systems.

A test arrangement has been made in which the data recorded in a perforated tape according to the five-unit code are converted into a seven-unit code which is already being used on radio circuits allowing of automatic error correction by repetition (see Recommendation S.13).

The synchronously transmitted seven-unit signals consist of elements of 20 ms. The duration of one signal is then 140 ms, the modulation rate consequently being not smaller—in so far as no repetitions occur—than in the case of normal 50-baud start-stop transmission.

The equipment has been provided with a transmitter reading the five-unit perforated tape at a speed that is determined by the seven-unit converter.

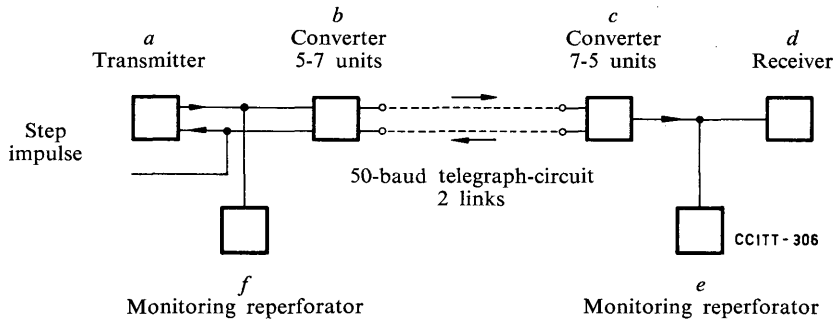
At the receiving end each signal, after having been converted, is sent to the receiver (in this case a reperforator) in the five-unit start-stop code. The transmitting and receiving converters have a memory in which three signals can be stored. These signals are repeated in case of faulty reception. The number of signals stored in the memory can in principle be extended if this is deemed necessary in connection with the expected delay time of the telegraph circuits.

The equipment is completely transistorized with the exception of some polarized relays, which have been provided in order to separate the circuits of the machines entirely from those of the electronic equipment.

Transmission took place via a telegraph circuit consisting of two V.F. links, beginning and terminating in one locality.

Right at the beginning of the test it appeared that the mechanically operating machines (transmitter and receiver) relatively caused so many errors that no clear insight could be obtained into the detecting properties of the seven-unit code.

Therefore some monitoring reperforators were connected to the equipment in order to locate the errors. The arrangement was as shown in the diagram below:



For the test in question four perforated tapes were used, viz:

1. The original perforated tape transmitted by means of transmitter *a*.
2. The perforated tape of the monitoring reperforator *f* containing an indication regarding the correct reading by transmitter *a*.
3. The perforated tape supplied by the ordinary receiver *d*.
4. The perforated tape supplied by the monitoring reperforater *e*.

The perforated tapes were automatically compared with one another. The figure on the opposite page shows the arrangement.

The comparison devices I and II compare the five units in the perforated tapes step by step. As soon as a difference is noticed, the comparison devices I and II are stopped and the alarm is given.

The figure shows that the signals received by receiver *d* are retransmitted by transmitter *a*.

If transmitter *a* reads a signal incorrectly, this is not indicated in the comparison devices I and II. However, the error remains in the perforated tape all through the transmission, so that it will be found afterwards.

If the TOR-system should not detect an error on the transmission path because the 3 to 4 ratio was maintained, this error is indicated by the comparison device I. If the monitoring reperforator of the transmitter *f* makes an error, this is indicated by comparison device I.

If the ordinary receiver *d* makes an error this is indicated by the comparison device II.

If the same error would simultaneously occur in receiver *d* and in monitoring reperforator *e* this would still be detected by the comparison device I. If monitoring reperforator *e* makes an error this is indicated by comparison device I.

Of course it is also possible that the comparison devices make reading errors. By comparing the read signals after a comparison device has stopped, it can be found out whether the devices stopped unnecessarily.

The provisional results are given below, covering fortnightly periods :

	Number of signals received	Number of errors in transmitter *a*	Number of errors in receiver *d*	Number of repetitions	Number of errors in TOR system
1st period	2 900 000	0	8	0	0
2nd period	2 650 000	0	18	23	0
3rd period	1 140 000	0	0	0	0
4th period	2 540 000	0	2	17	0
5th period	2 550 000	50	5	1727	0
6th period	2 460 000	52	0	0	0
7th period	1 180 000	0	0	20	0

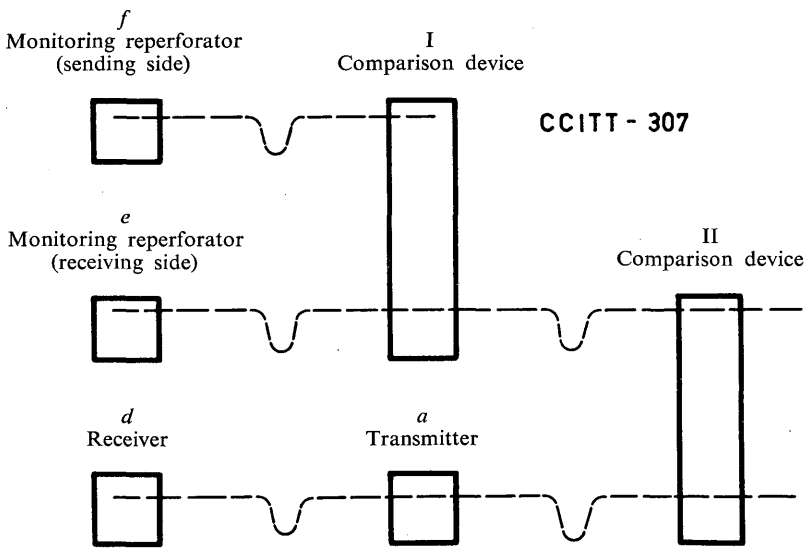
From the aforegoing provisional results it follows :

1. That it is necessary to keep a check also on the mechanical equipment.

2. That the converting and repetition equipment did not leave any errors undetected, the number of repetitions in the first four periods being so small, however, that they cannot be considered decisive for the reliability of the seven-unit code on telegraph circuits. The large number of repetitions in the 5th period was the result of a prolonged interruption of the telegraph connection as a result of incorrect manipulations of the staff in charge of the maintenance of the telegraph circuit.

3. That the test should be continued, in which case use should be made of a telegraph circuit that causes more errors.

Slightly more than 15×10^6 signals (characters) have been transmitted, no undetected errors were printed.



(Suppl. 17)

The number of repetitions (detected errors) was small, 1787. Of these 1787 repetitions 1727 were caused by a prolonged interruption of the telegraph connection as a result of incorrect manipulations of the staff in charge of the maintenance of the telegraph circuit.

If this accident is neglected, only 60 repetitions remain. Each repetition contains four characters in this equipment and therefore a loss of transmission speed of 1.6×10^{-5} is the average value.

Since the above-mentioned report, another series of tests has been made, this time on lines within Europe. About 18 million signals were transmitted. The number of repetitions is much greater in this case. On one of the circuits 8.2 million signals were transmitted and 43 500 repetitions occurred.

The other lines showed a more normal behaviour. With 10 million signals transmitted, 570 repetitions were noticed, but undetected errors were not printed during the transmission of all the 18 million signals.

The results seem to indicate that the protection with the system of automatic repetition and seven-unit constant ratio code is sufficient for transmission of data on 50-baud telegraph lines. It must be remarked that the protection of an ARQ system is better than the protection of the code alone. It can be shown that the maximum transposition rate occurs in a comparatively small region of the signal-to-noise ratio and that even this maximum is about one order lower than the maximum transposition rate of the code alone [1].

This may provide some explanation why the performance of the system is so much better than theoretical considerations at first sight seem to indicate. Another explanation can be that the characteristics of the mutilations on telegraph lines could be favourable for this particular code and this particular system.

Summing up the characteristics for an isochronous system suitable for data transmission on part of the world-wide telex network could be :

a) Protection per character with a seven-element constant ratio code with four Z and three A elements.

b) Repetition cycle of eight characters with seven characters in storage, allowing for a loop delay of 700 ms.

c) Duration of one signal element is 20 ms.

d) The simplest form of code conversion from five units to seven units is obtained if the maximum possible number of characters is formed by addition of two elements to the five of the five-unit code. This is possible for 25 characters ; the remaining seven characters are formed by not only adding two elements but also changing two or three elements of the five elements of the five-unit code. A possible solution is given in Figure 1.

The system can be adopted for half-duplex working. In that case the request for repetition is the change from — polarity to + polarity on the line at the traffic receiving end. With local repeaters, as given in Figure 2, the result is that at the traffic sending end the + polarity on the line is changed to — polarity irrespective of the position of the teleprinter sending contacts (in station A as well as station B).

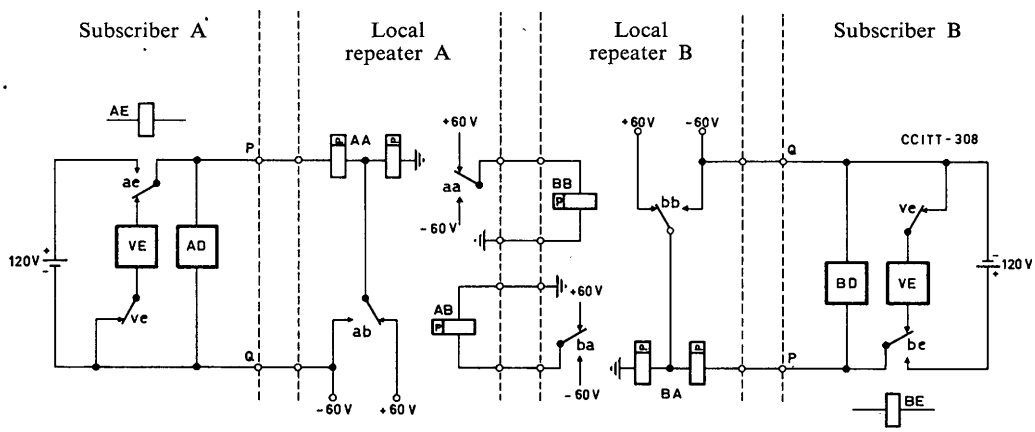
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[1] "Error probability and transmission speed on circuits using error detection and automatic repetition of signals", by H. C. A. van Duuren. *I.R.E. Transactions on Communications Systems*, March 1961.

With a different type of local repeater, it may not be possible to do it in the same way and it may be necessary therefore to provide in that case full-duplex facilities to the customer or change the local repeater according to Figure 2.

OOOOO	OXOXOXX	XOOXX	OXOOXXX
		XXOOX	XXXOOXO
XOOOO	OXXOXOX	XXOXO	XXXOXOO
OXOOO	OOXXOXX	OXXOX	XOXXOXO
OOXOO	OOXXXOX	XOXXO	XXOXXOO
OOOXO	OOOXXXX	OXOXX	XOXOXXO
OOOOX	OOXOXXX	XOXOX	XXOXOXO
XXOOO	XXXOOOX	XXXXO	OXXXXOO
OXXOO	XOXXOOX	OXXXX	OOXXXXO
OOXXO	XOOXXOX	XOXXX	OXOXXXO
OOOXX	XOOOXXX	XXOXX	OXXOXXO
XOOOX	XXOOOXX	XXXOX	OXXXOXO
XOXOO	XXOXOOX		
OXOXO	XOXOXOX	XXXXX	OXOXXOX
OOXOX	XOOXOXX		
XOOXO	XXOOXOX	α	XXXXOOO
OXOOX	XOXOOXX	β	XXOOXXO
		s_1	OXXOOXX
XXXOO	OXXXOOX		
OXXXO	XOXXXOO		
OOXXX	XOOXXXO		

FIGURE 1



VE = Teleprinter receiving coil
ve = Teleprinter sending contact
AE, BE } = RQ relay
AD, BD } = Line polarity guard device

FIGURE 2

Supplement No. 18

NETHERLANDS — *(Extract from contribution COM Sp. A/No. 18 — September 1961)*

ERRORS ATTRIBUTABLE TO ELECTRO-MECHANICAL EQUIPMENT — TEST RESULTS

The provisional results of the test, which took place during fortnightly periods, are given in the table below :

	Number of signals received	Number of errors made by transmitter *D*	Number of errors made by teleprinters		
			Type A	Type B	Type C
1st period	2 220 000	2	19	0	0
2nd period	1 573 000	2	71	2	41
3rd period	1 393 000	1	0	0	0
4th period	2 104 000	3	39	8	0

After each period the teleprinters were repaired when necessary. For the rest they were maintained normally.

However, only rather new teleprinters were available.

The testing period is still too short for us to draw reliable conclusions, although here too the results are an indication that it would be worth considering to protect also the transmitters and the receivers.

Supplement No. 19

AMERICAN CABLE AND RADIO CORPORATION — *(Contribution COM Sp. A/No. 73 — June 1963)*

COMBINED OPERATION OF A 1200-BAUD DATA TRANSMISSION CHANNEL AND 50-BAUD TELEGRAPH CHANNELS IN A SINGLE VOICE-BAND

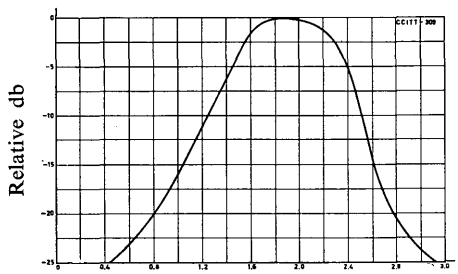
I. 1. For economical reasons it is desirable to combine on leased cable circuits a number of 50-baud telegraph channels along with a 1200-baud data channel.

2. a) Available modems for 1200 bauds are usually designed for operation in a full voice-band. A modem of the FSK type centred at 1900 c/s with plus and minus 400 cycles frequency

shift was tested alongside several 50-baud channels on the same voice-channel. Excessive interference was produced by the 1200-baud channel making all of the 50-baud channels unworkable. The modem used was ITT Type T1JB.

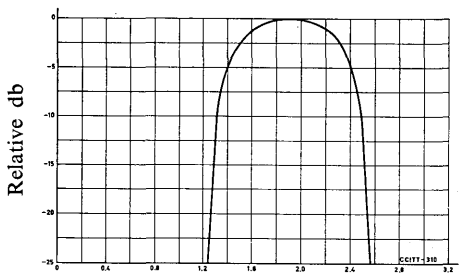
b) The reason for the interference may be seen from the original bandpass filter spectrum, Figure 1. A new filter with much sharper skirts was designed. As may be seen from Figure 2, the new filter retained at least the same bandwidth and rounding at the edges as the original filter. Therefore, it was expected and proven true that keying distortion was not increased as compared with the original filter. Both curves were made under dynamic conditions.

c) The tests were repeated with the Figure 2 filters substituted for the original filters. No interference was experienced on any of the 50-baud channels. Six 50-baud channels in the lower spectrum and five in the upper part could be worked satisfactorily. It is recognized that the number of five in the upper spectrum may reduce according to the condition of the voice-facility. As a minimum, six channels in the lower portion should be workable on even low-grade facilities.



Frequency : kc/s
T1JB data modem
1900 c/s ~ C.F., ± 400 c/s ~ shift
Random data spectrum
Original filtering
0 db = 1 mW in 600 Ω

FIGURE 1



Frequency : kc/s
T1JB data modem
1900 c/s ~ C.F., ± 400 c/s ~ shift
Random data spectrum after filtering with a 1300-2500 c/s bandpass filter
0 db = 1 mW in 600 Ω

FIGURE 2

II. 1. The initial investigation of the T1JB modems took place on the cable between New York and San Juan, Puerto Rico.

2. a) A new filter with a bandpass from 1300 to 2500 c/s and 180 db per decade attenuation was designed (Fig. 2). Preliminary tests showed that operation of the T1JB modem with this filter at 1200 bauds is feasible. The modem was set up to operate with a plus or minus 400-c/s frequency shift of a 1900-c/s carrier. Without a clock unit and regenerator, a small amount of jitter was noticeable. Insertion of the clock unit and the regenerator into the modem caused the amount of jitter to fall within the accuracy of the measuring equipment.

b) An investigation to determine the effects of carrier frequency centring on speed showed an apparent increase in allowable speed with a reduction in carrier frequency. At a carrier frequency of 1400 c/s and a shift of plus or minus 400 c/s, the ITT T1JB modem will operate at 1500 bauds. Previous attempts with the ITT T1JB modem with the original filter had resulted in a complete loss of output at speeds above 1280 bauds. The interesting observation of this experiment is that when operating in this mode the lowest sideband is completely cut off due to the action of the filter and the demodulator sees only the carrier and the upper sidebands and again only up to 2500 c/s.

c) In the initial investigation to determine the required transmission bandwidth for a data signal, it was first necessary to determine the spectrum of the signals with the ITT T1JB data modem. An experimental spectrum analysis of a random data signal was obtained by sending a long string of reversals into the data modem and noting the outstanding spectral characteristics for each case. When sufficient data were obtained so as to see the trends of sideband spreading and sideband amplitude behaviour with various keying speeds, a composite spectrum was plotted. This was done by dividing the V.F. band into arbitrary segments. All parts were checked to determine amplitude of sidebands that could be expected in any region of the spectrum.

d) The final result was, as expected, a Gaussian distribution since random events have this characteristic probability curve. From this plot the bandwidth for random data signals can be seen to be about 1300 to 2500 c/s at 6 db down when operating at 1200 bauds. With the insertion of a bandpass filter, 1300 to 2500 c/s, the frequency spectrum of the ITT T1JB data modem at 1200 bauds was plotted. The characteristics of this filter were specified to have a minimum cut-off of 160 db per decade but this particular filter had 180 db per decade cut-off. By using this filter on the modems at each end of the circuit between New York and San Juan, Puerto Rico, we have reduced the area of interference to and from 50-baud telegraph signals which are simultaneously impressed on a voice cable along with the 1200-baud data.

e) The present design of the Figure 2 filter allows for 4 milliseconds delay at the lower cut-off frequency, 1.7 milliseconds at the upper cut-off frequency and 2 milliseconds in the bandpass region.

In an effort to obtain a filter in which the envelope delay might be as low as plus or minus 1 microsecond, the cost of the filter increased by 150 times. Therefore, it has been determined that through proper equalization the present Figure 2 design should fulfil the needs for data transmission at 1200 bauds with simultaneous 50-baud transmissions in the same voice channel.

Supplement No. 20

SWEDEN — *(Extract from contribution COM Sp. A/No. 1 — June 1961)*

DATA TRANSMISSION TEST OVER LEASED CIRCUITS

Normally, a binary cycle of 256 characters, each having eight elements, has been used as a test message. This cycle is made up of all permutations of the eight elements ($2^8 = 256$). This test message is produced in the transmitter part of the "Error Rate Analyzer" as a pulse train which is transferred to the line via a modulator by frequency shift modulation of a carrier. The incoming signal is demodulated in the receiver of the data transmission system from where the distorted pulse signal goes to the receive part of the "Error Rate Analyzer" for comparison.

With consideration to the line delay each element is tested for its polarity in the middle of the interval, where it should come. Therefore, up to 100% isochronous distortion can be tolerated without errors being registered. The synchronizing signal, which is needed for the testing on the receiver side, was taken directly from the transmitter via a manually adjustable delay circuit. Of course, this is only possible with loop measurements.

With the measuring set-up the error rate as a function of signal-to-noise ratio, $p_e = F(R_{2850})$[1], was measured for different values of the frequency shift, modulation rate and channel mid-frequency, f_0.

At an early stage the optimum shift was found to be about 0.7-0.8 times the modulation rate. The modulation index used is 0.8. For practical reasons the number of modulation rates are reduced to the above-mentioned four, viz. 800, 1150, 1500 and 2400 bauds.

For each circuit and speed a number of curves, $p_e = F(R_{2850})\, f_0$, with f_0 as the variable parameter, were measured. Examples of these curve complexes are found in Figures 3, 6 and 7.

The measurements were performed with the speeds which have been usable in each particular case. The white noise level was held so high that p_e varied between about 10^{-4} and 10^{-2}, in order that the measurements should not be considerably influenced by the circuit's own noise. Out of each complex of curves the relationship $R_{2850} = F(f_0)\ p_e = 10^{-3}$ was found by reading the curves cross-section with $p_e = 10^{-3}$. This relationship is shown for each measured object. The minimum point for each such curve gives the optimum f_0. For these f_0s measurements were made on the data transmission system itself, loop-connected over an ideal line—i.e. a resistive attenuator. As a comparison these values have been marked on the diagrams by means of small circles. In some cases the circles can be above the minimum points of the curves. That will be because the line happens to equalize a phase non-linearity, which is found in the transmission system itself.

An investigation has been performed in order to get an idea of the measurement tolerances. If it is presupposed, by the error rate measurements with injected white noise, that the bropability of an element being conveyed incorrectly is constant and small, and if with each measurement the number of conveyed elements is large, then it can be shown that the resulting measurements lead to a Poisson distribution. With each measurement a minimum of 100 error registrations

[1] R_{2850} is the ratio between the signal power at input of the receiver and the power of an injected white noise spectrum with an equivalent bandwidth of 2850 c/s.

has been asked for, in order to secure 99% probability that measured error rates should not deviate more than 25% from the actual mean.

In order to check if the measured values were actually distributed according to the Poisson law, a series of measurements were made at one point. A comparison with a Poisson distribution showed a good fit. It was clear from this test that the dominating measurement tolerance was caused by the element error's statistical nature. A change of 30% in p_e corresponds to a deviation in R of about 0.3-0.5 db.

3. *Measurements without phase equalization*

Measurements without phase equalization have been performed on twelve different circuits, as can be seen in Figures 1-27. Nine of these circuits are loaded pairs and three consist of different carrier channels. The arrangement of each circuit, relative level and group delay, $\frac{d\theta}{d\Omega}$, are shown in the figures along with a diagram of the signal-to-noise ratio, R_{2850}, at $p_e = 10^{-3}$ versus channel mid-frequency, f_0. For this latter diagram some characteristics have been tabulated, i.e. optimum mid-frequency, suitable working range and function range.

1500 bauds could be transmitted over the loaded cable circuits 2, 3 and 4. However, this modulation rate could only be used for a very limited range of channel mid-frequencies with more than 10 db impairment in white noise performance. On circuits 1, 8 and 9 only 800 bauds could be used due to phase distortion caused by heavy loading or long distances (200, 256 and 500 km respectively). Both of the circuits 10 and 11 correspond to channels in a multiplex terminal for radio-link use. The case with four channels in tandem only allowed speeds up to 1150 bauds. However, in practice this case is certainly very uncommon. A measurement with 2400 bauds has been made on circuit 12, which is a high-quality toll circuit consisting of a standard channel in a coaxial-cable carrier system. As a return a programme circuit was used, which did not contribute to the total phase distortion.

The optimum channel mid-frequencies were found to be between 700 and 1000 c/s on loaded circuits and between 1800 and 2000 c/s on carrier circuits. This difference is due to the fact that the phase shift characteristic of a loaded cable circuit is of low-pass type while in a carrier channel most of the filtering is done by band-pass filters.

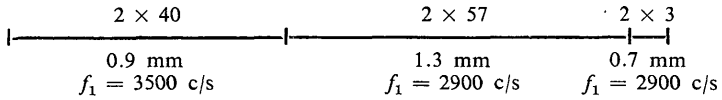
4. *Measurements with phase equalization*

The circuit characteristics can be improved by phase (group delay) equalization. This was done on five of the circuits mentioned above, i.e. circuits 1, 2, 6, 10 and 12. Circuit 13 is made up of circuits 2 and 10. By the measurements on circuits 1 and 2 the same type of equalizing network was used. This was an attempt to construct a compromise equalizer from a number of standardized networks. As can be seen in Figures 2 and 5, the suitable working ranges were considerably extended.

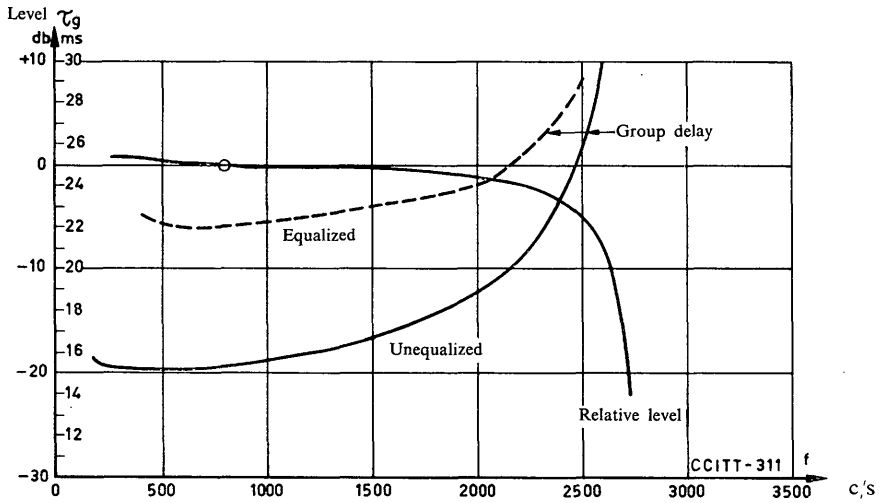
For circuit 6 an equalizer was used which was specially designed on the basis of a phase shift measurement of the circuit. With this method large improvements are always possible but the complexity of the procedure makes it less attractive for practical use. For the two carrier circuits, viz. 10 and 12, and the combined circuit 13, the result of the phase equalizing is seen in Figures 23, 27 and 29 respectively.

Measurements have been performed on circuit 8 both with and without level equalization. The difference in group delay was negligible. According to Figure 19, the equalization produced a slight improvement.

CIRCUIT ARRANGEMENT

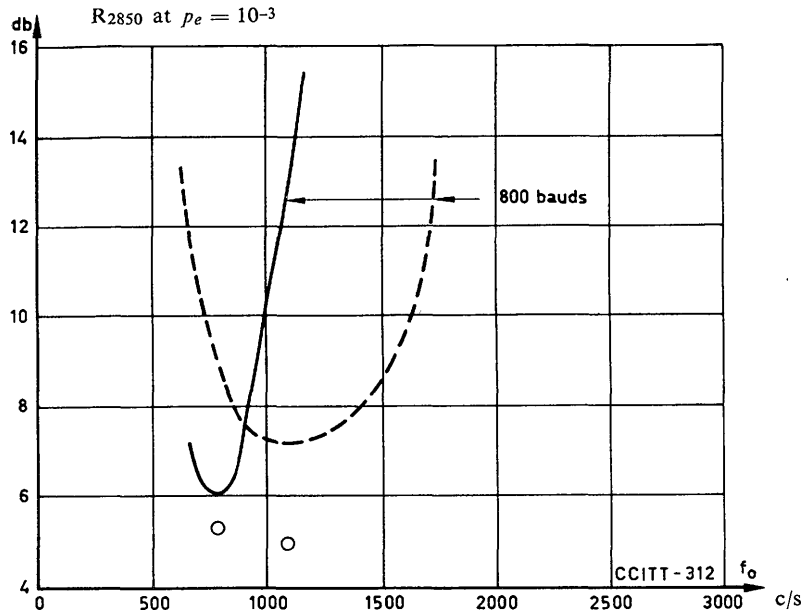


Total distance 200 km



Circuit No. 1

FIGURE 1. — **Relative level and group delay diagrams**

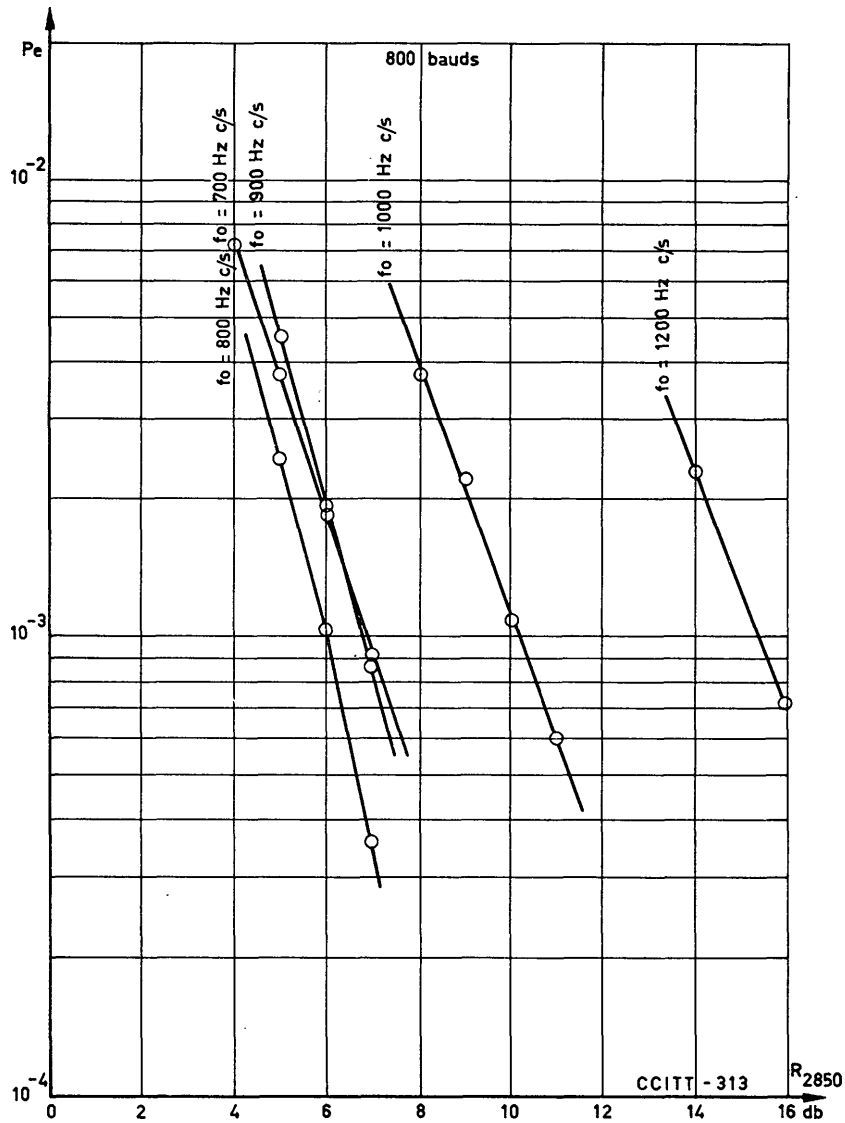


	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
			c/s
Optimum mid-frequency			800 (1100)
Suitable working range			-970 (750-1550)
Function range			-1280 (600-1700)

Note. — Dashed lines and figures within brackets refer to the case with group delay equalization.

Circuit No. 1

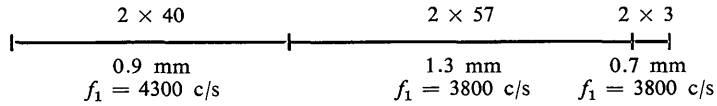
FIGURE 2. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**



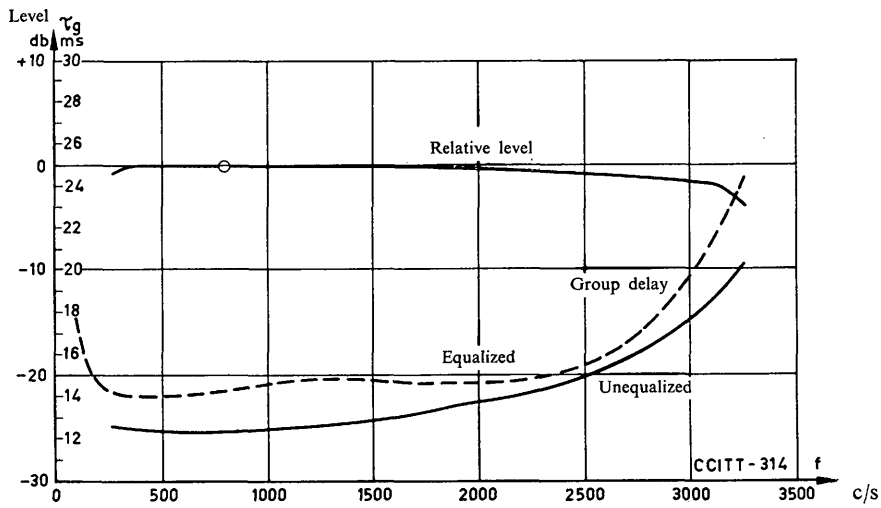
Circuit No. 1

FIGURE 3. — **Element error probability p_e versus signal-to-noise ratio R_{2850}**

CIRCUIT ARRANGEMENT

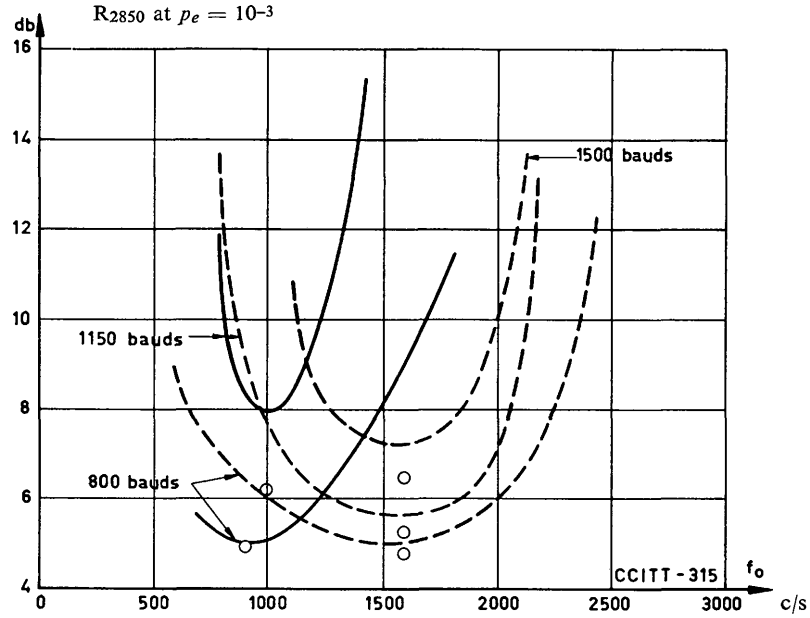


Total distance 200 km



Circuit No. 2

FIGURE 4. — **Relative level and group delay diagrams**

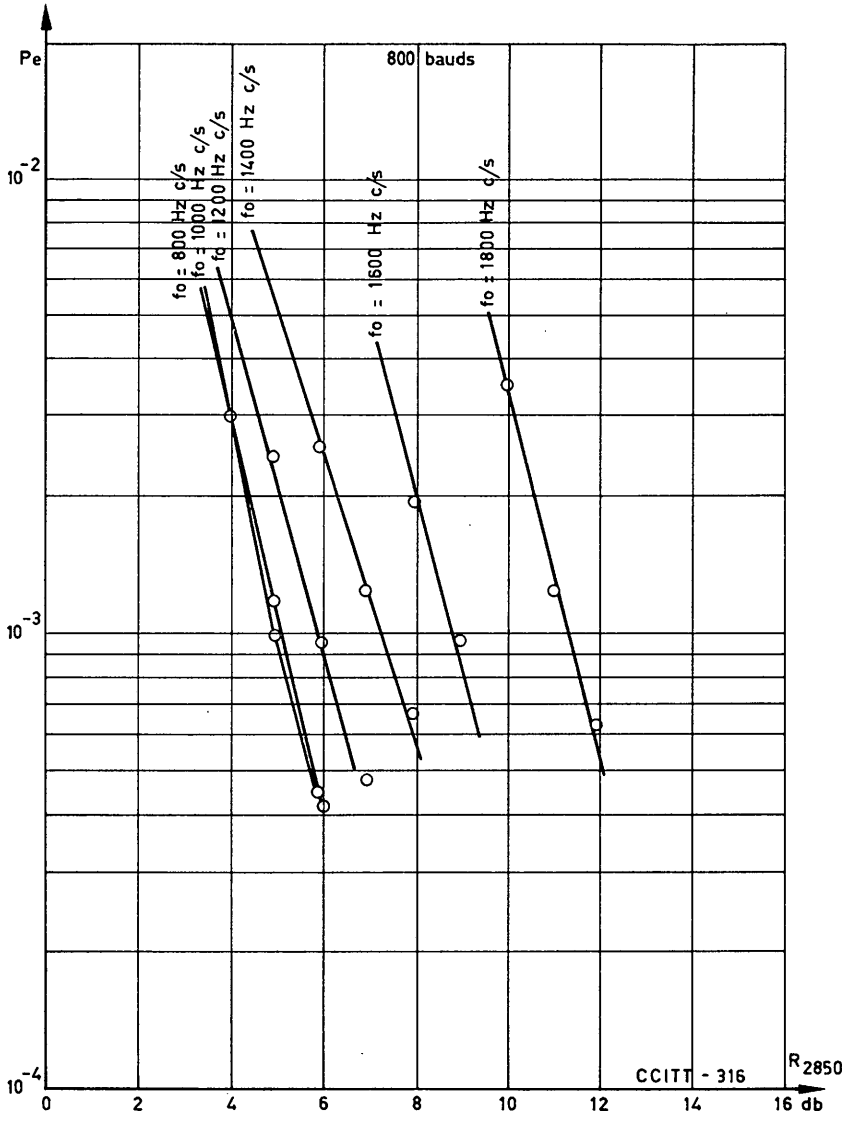


	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
	c/s	c/s	c/s
Optimum mid-frequency	(1600)	1000 (1600)	900 (1600)
Suitable working range	(1100-2000)	800-1300 (950-2100)	-1500 (650-2250)
Function range	(1050-2150)	700-1500 (750-2200)	(500-2450)

Note. — Dashed lines and figures within brackets refer to the case with group delay equalization.

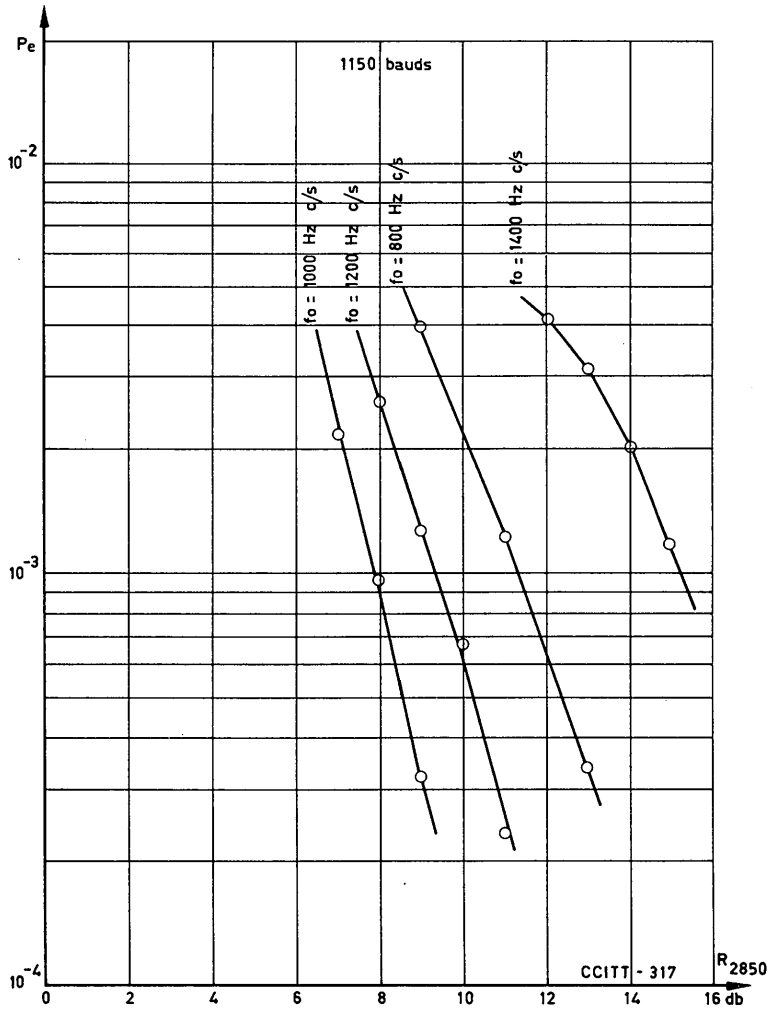
Circuit No. 2

FIGURE 5. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**



Circuit No. 2

FIGURE 6. — **Element error probability p_e versus signal-to-noise ratio R_{2850}**



Circuit No. 2

FIGURE 7. — **Element error probability p_e versus signal-to-noise ratio R_{2850}**

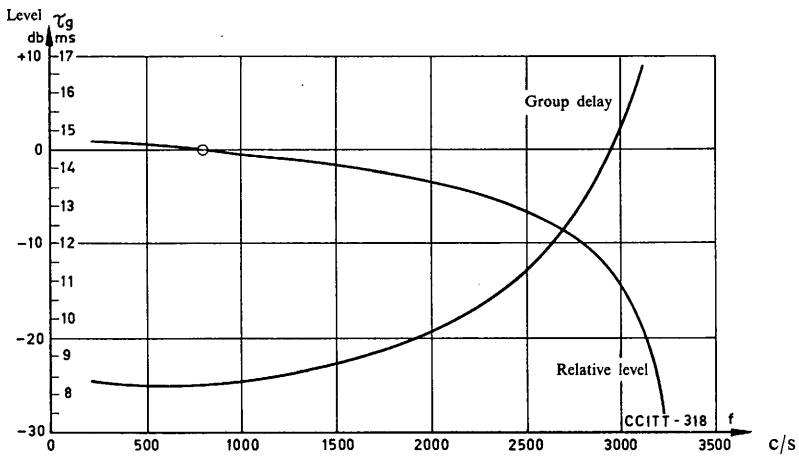
CIRCUIT ARRANGEMENT

2 × 47

0.9 mm

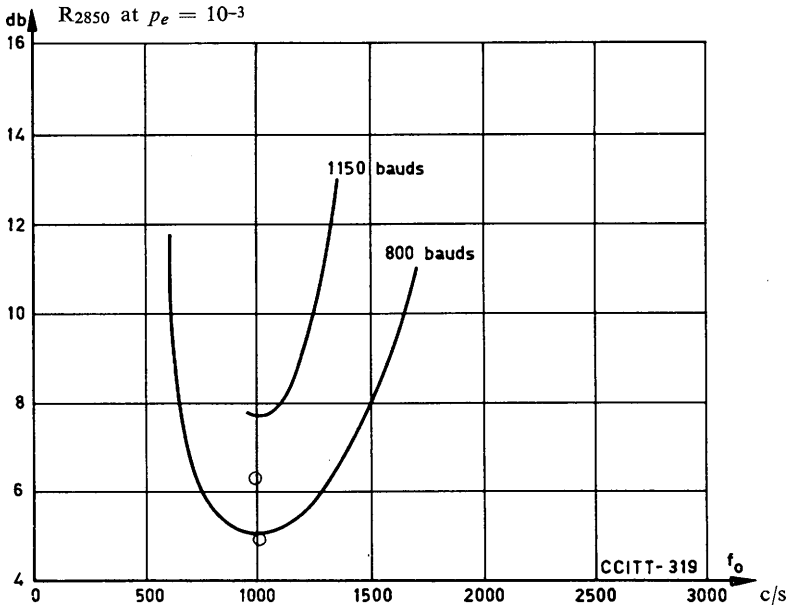
f_1 = 3500 c/s

Total distance 94 km



Circuit No. 3

FIGURE 8. — **Relative level and group delay diagrams**



	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
		c/s	c/s
Optimum mid-frequency		1000	1000
Suitable working range		-1275	650-1500
Function range		-1450	550-1900

Circuit No. 3

FIGURE 9. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

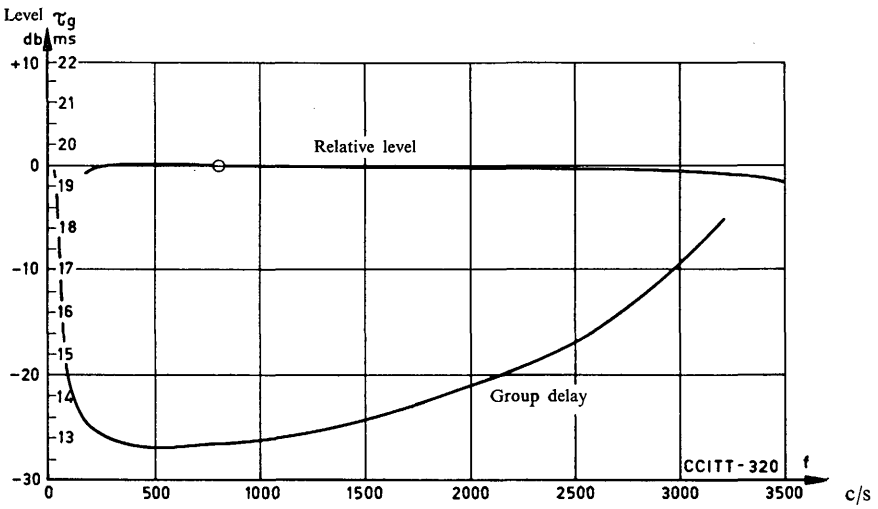
CIRCUIT ARRANGEMENT

4 × 47

0.9 mm

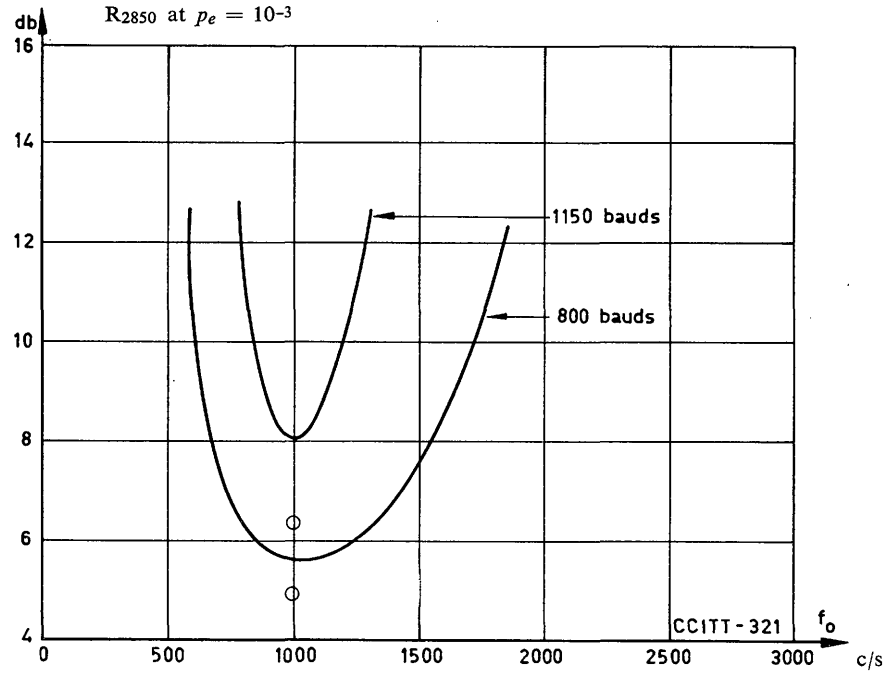
f_1 = 4300 c/s

Total distance 188 km



Circuit No. 4

FIGURE 10. — **Relative level and group delay diagrams**

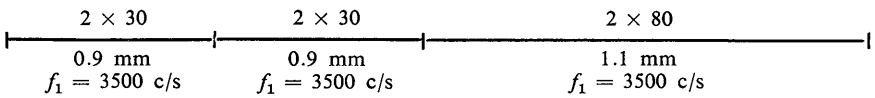


	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
		c/s	c/s
Optimum mid-frequency		1000	1000
Suitable working range		800-1250	650-1600
Function range		700-1350	550-1900

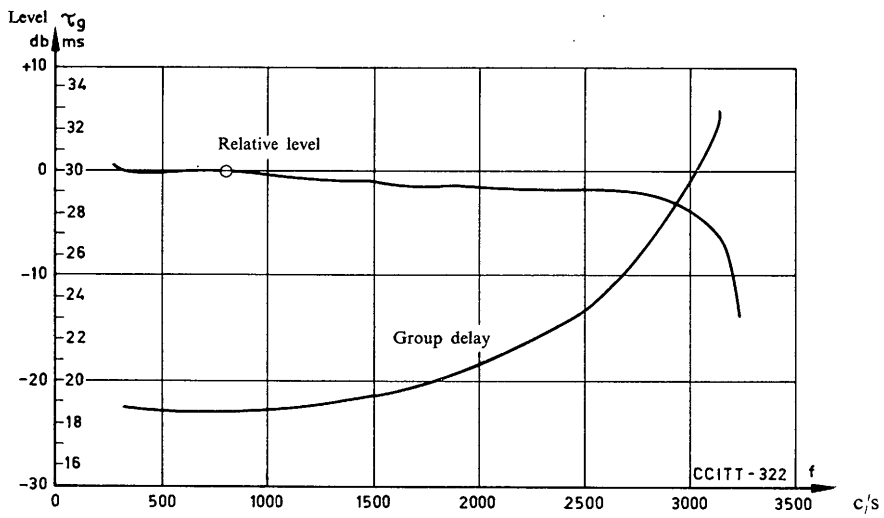
Circuit No. 4

FIGURE 11. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

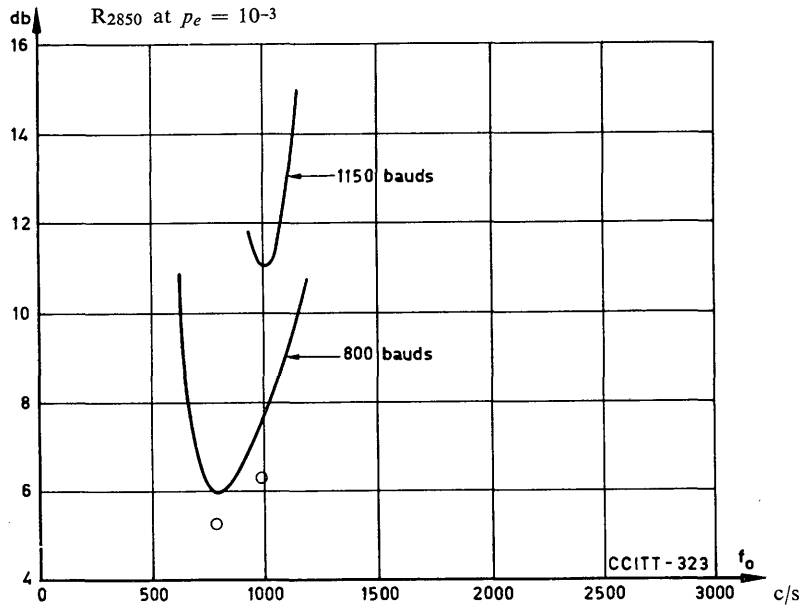


Total distance 280 km



Circuit No. 5

FIGURE 12. — **Relative level and group delay diagrams**



	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
		c/s	c/s
Optimum mid-frequency		1000	800
Suitable working range		-1100	650-1100
Function range		-1150	600-1300

Circuit No. 5

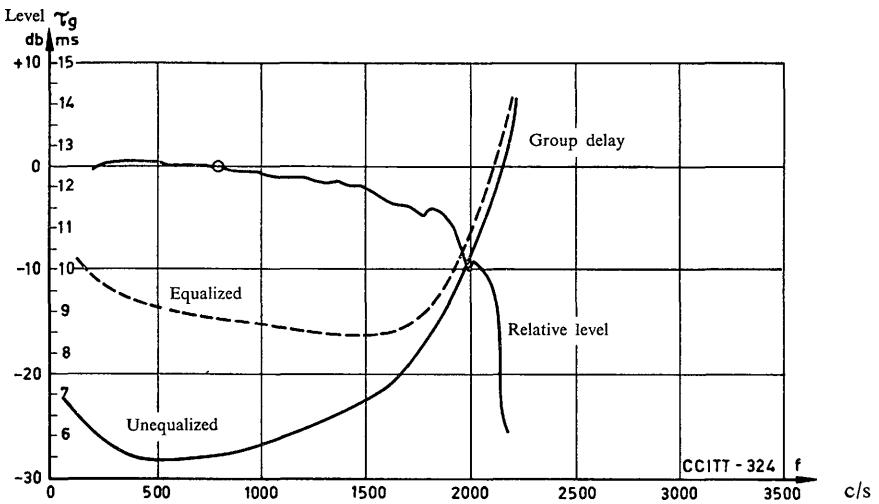
FIGURE 13. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

2 × 47

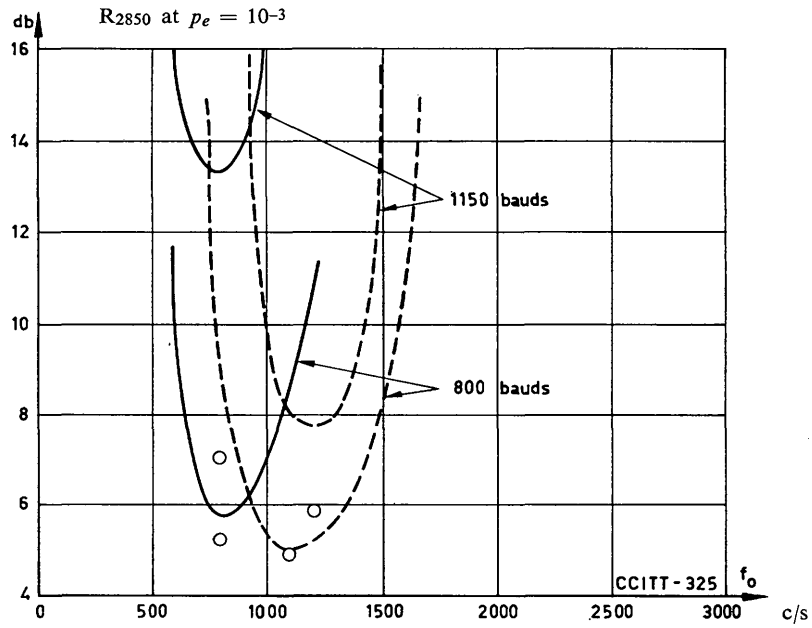
Wire diameter 0.9 mm
f_1 = 2400 c/s

Total distance 94 km



Circuit No. 6

FIGURE 14. — **Relative level and group delay diagrams**



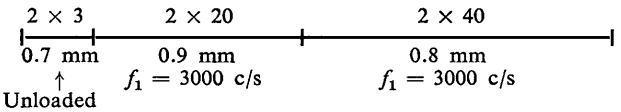
	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
		c/s	c/s
Optimum mid-frequency		800 (1200)	800 (1100)
Suitable working range		590-1000 (975-1450)	630-1120 (850-1475)
Function range		550-1100 (800-1500)	530-1300 (700-1700)

Note. — Dashed lines and figures within brackets refer to the case with group delay equalization.

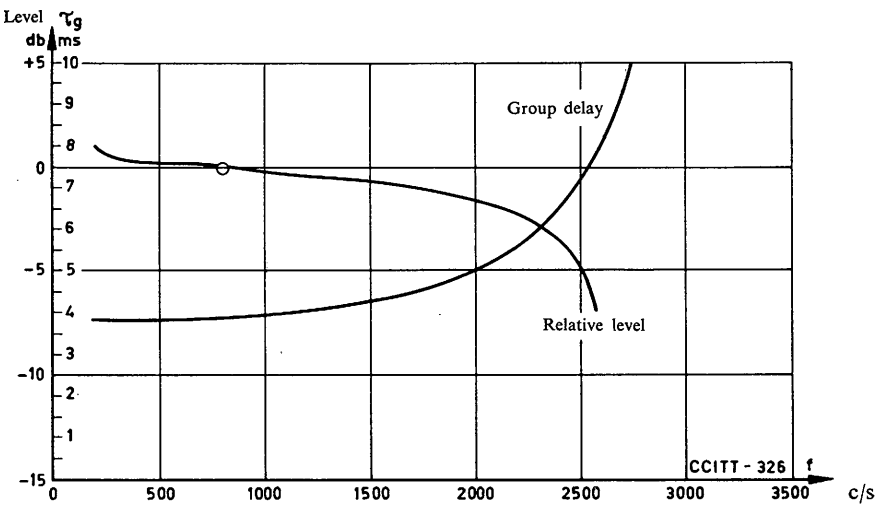
Circuit No. 6

FIGURE 15. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

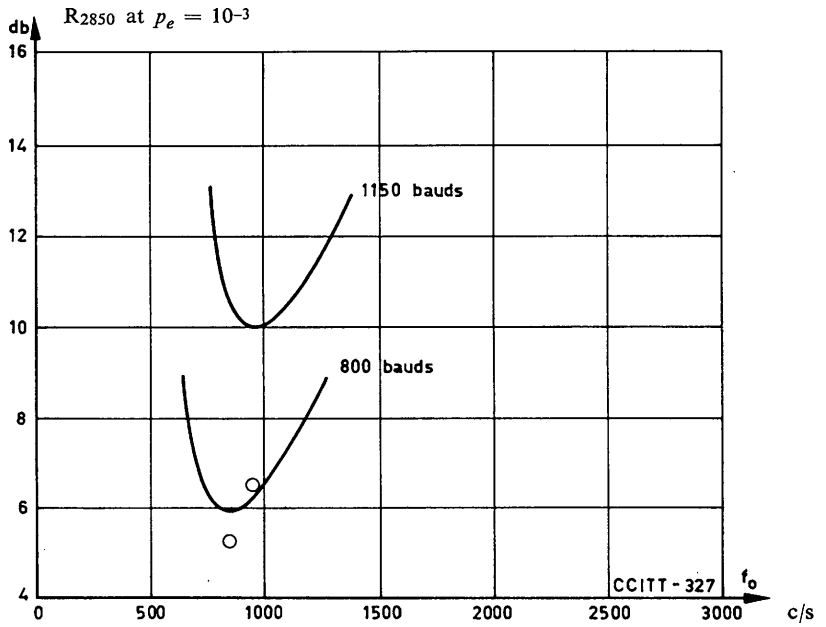


Total distance 126 km



Circuit No. 7

FIGURE 16. — **Relative level and group delay diagrams**

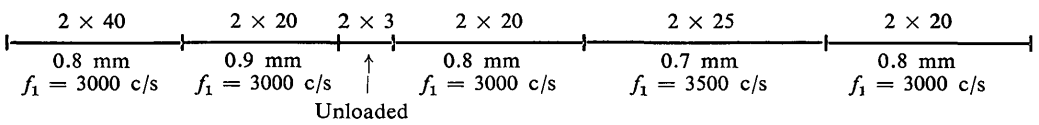


	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
		c/s	c/s
Optimum mid-frequency		950	850
Suitable working range		750-1400	650-1250
Function range		700-	600-

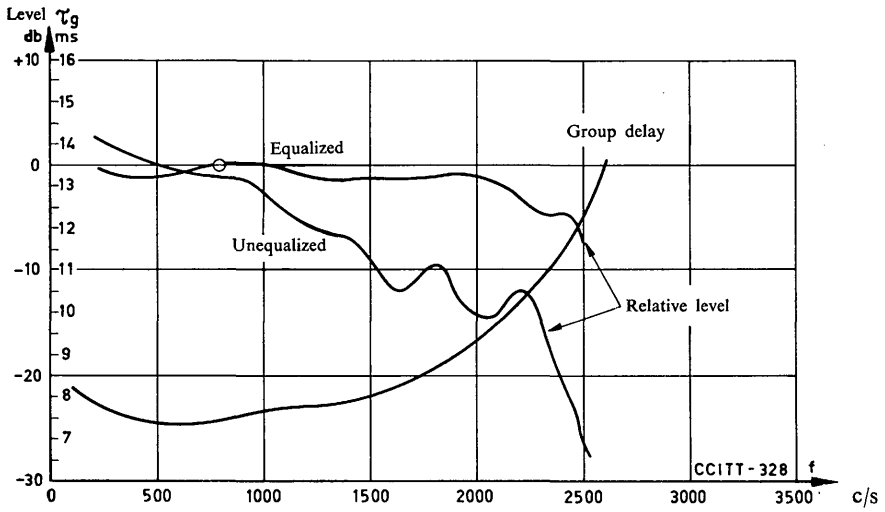
Circuit No. 7

FIGURE 17. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

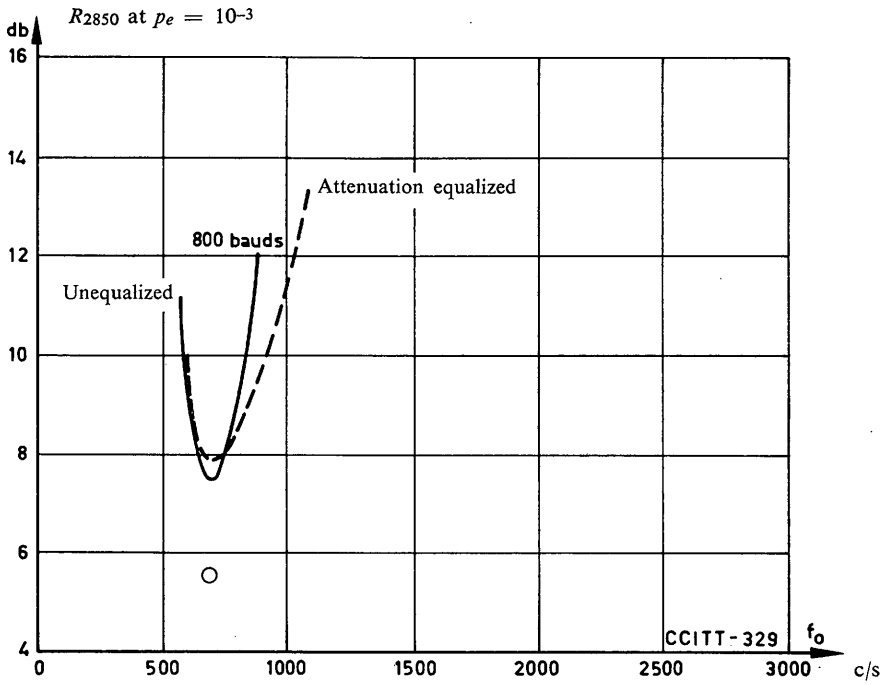


Total distance 256 km



Circuit No. 8

FIGURE 18. — **Relative level and group delay diagrams**



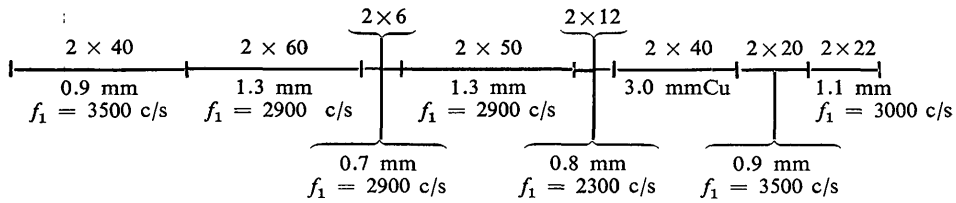
	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
			c/s
Optimum mid-frequency			700 (700)
Suitable working range			575-825 (575-925)
Function range			500-900 (500-1150)

Note. — Figures within brackets refer to the case with attenuation equalization.

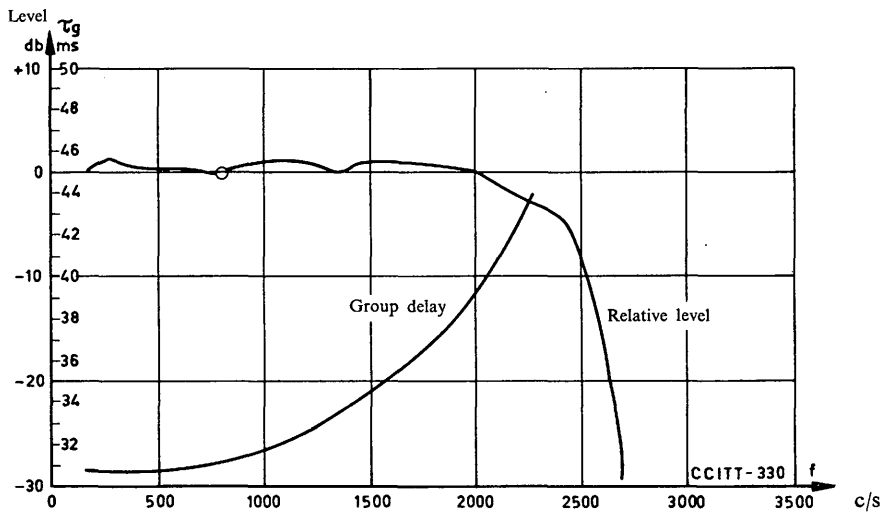
Circuit No. 8

FIGURE 19. — **Signal-to-noise ratio R_{2850} at $P_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

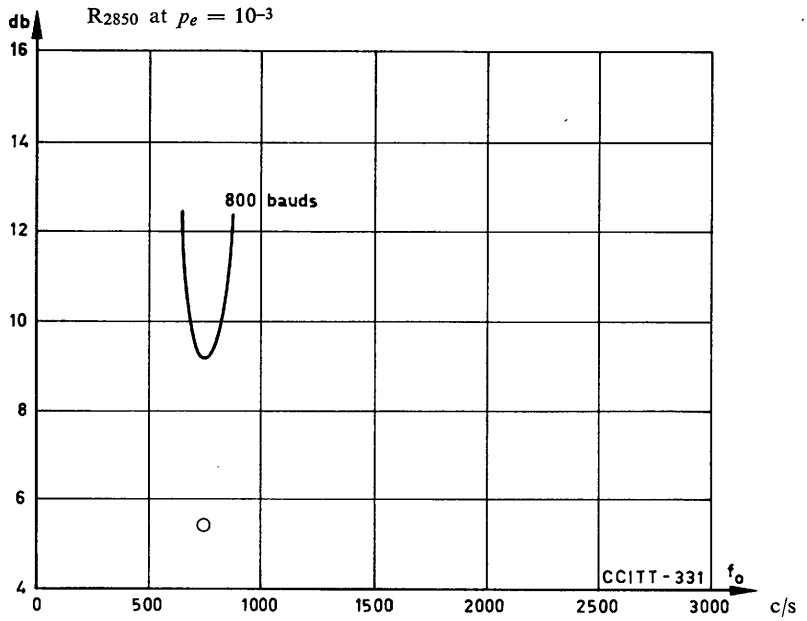


Total distance 516 km



Circuit No. 9

FIGURE 20. — **Relative level and group delay diagrams**

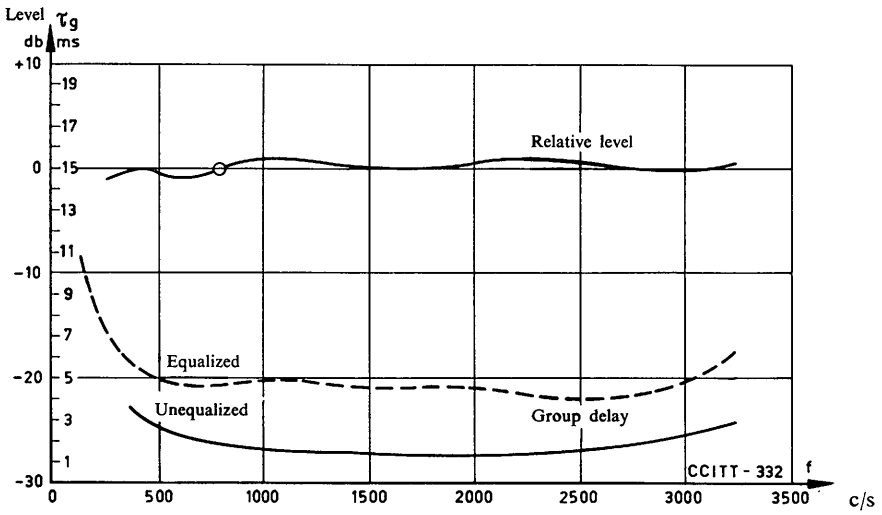


	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
			c/s
Optimum mid-frequency			750
Suitable working range			650-875
Function range			600-900

Circuit No. 9

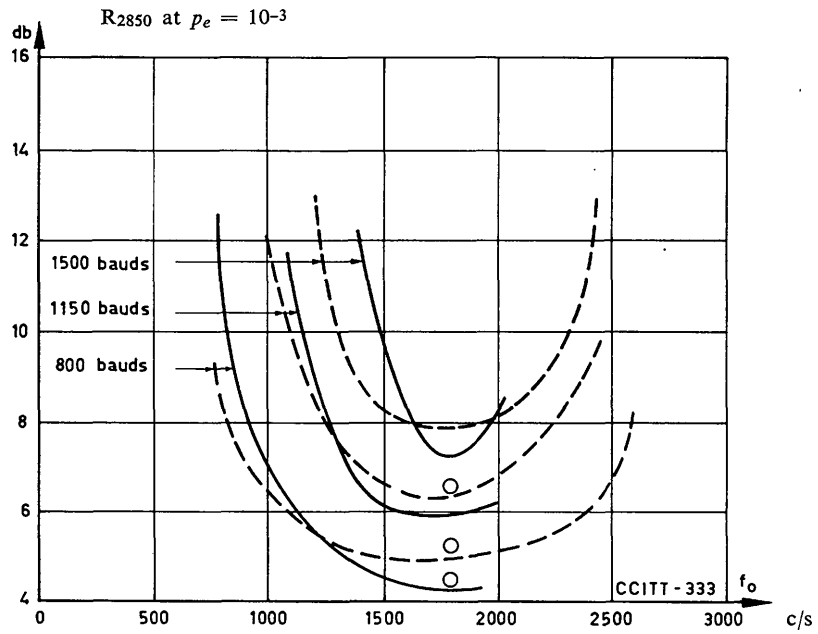
FIGURE 21. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

12-channel carrier system in local loop
Channels A1



Circuit No. 10

FIGURE 22. — **Relative level and group delay diagrams**



	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
	(c/s)	(c/s)	(c/s)
Optimum mid-frequency	1800 (1800)	1800 (1800)	1800 (1800)
Suitable working range	1475- (1275-2375)	1200- (1175-2350)	975- (850-2600)
Function range	1300- (1200-2450)	1000- (900-2500)	800- (700-2600)

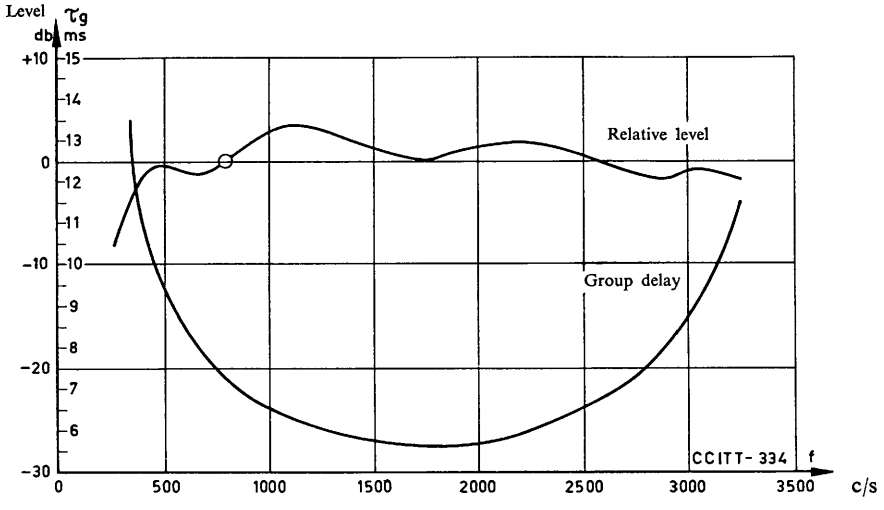
Note. — Dashed lines and figures within brackets refer to the case with group delay equalization.

Circuit No. 10

FIGURE 23. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

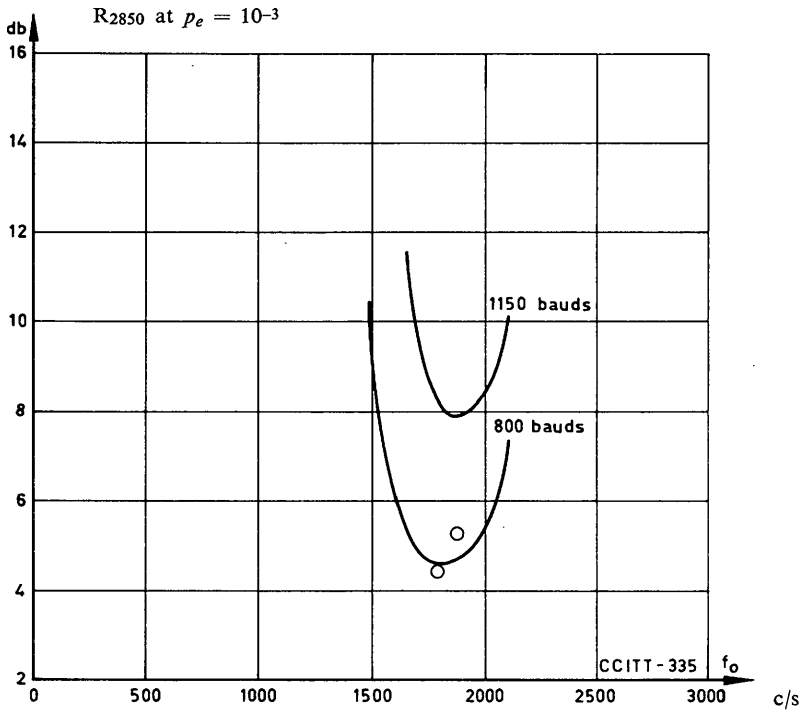
CIRCUIT ARRANGEMENT

12-channel carrier system in local loop
Channels A1 + B1 + A4 tandem coupled



Circuit No. 11

FIGURE 24. — **Relative level and group delay diagrams**

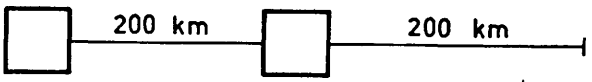


	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
		c/s	c/s
Optimum mid-frequency		1900	1800
Suitable working range		1650-	1550-
Function range		1600-	1400-

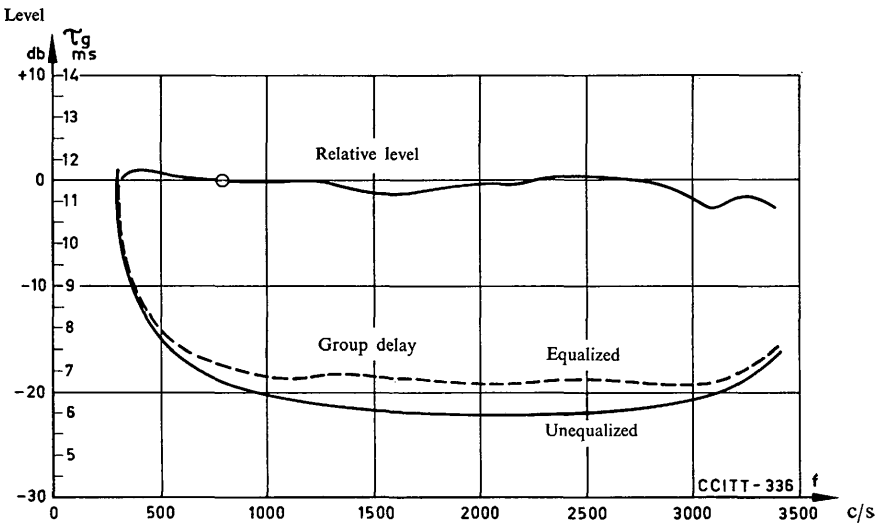
Circuit No. 11

FIGURE 25. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

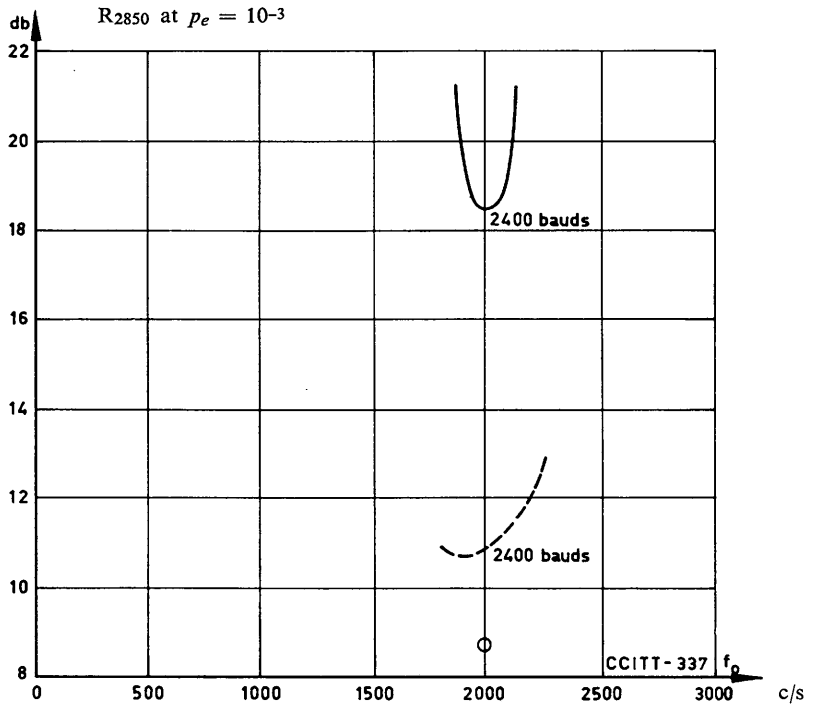


Coaxial cable and terminal equipment
Wire diameter 1.0-1.2 mm
$f_1 = 11\,500$ c/s



Circuit No. 12

FIGURE 26. — **Relative level and group delay diagrams**



	Modulation rates		
	2400 bauds		
Optimum mid-frequency	c/s 2000 (1920)		

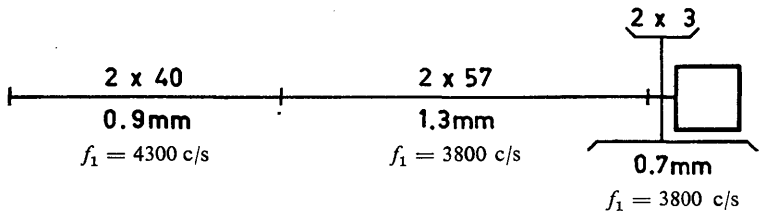
Note. — Dashed lines and figures within brackets refer to the case with group delay equalization.

Circuit No. 12

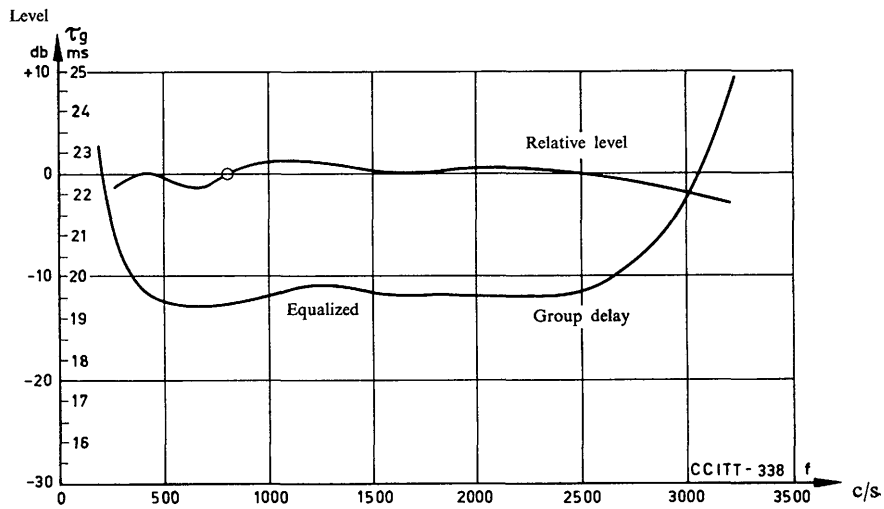
FIGURE 27. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

CIRCUIT ARRANGEMENT

12-channel carrier system in local loop
Channels A1

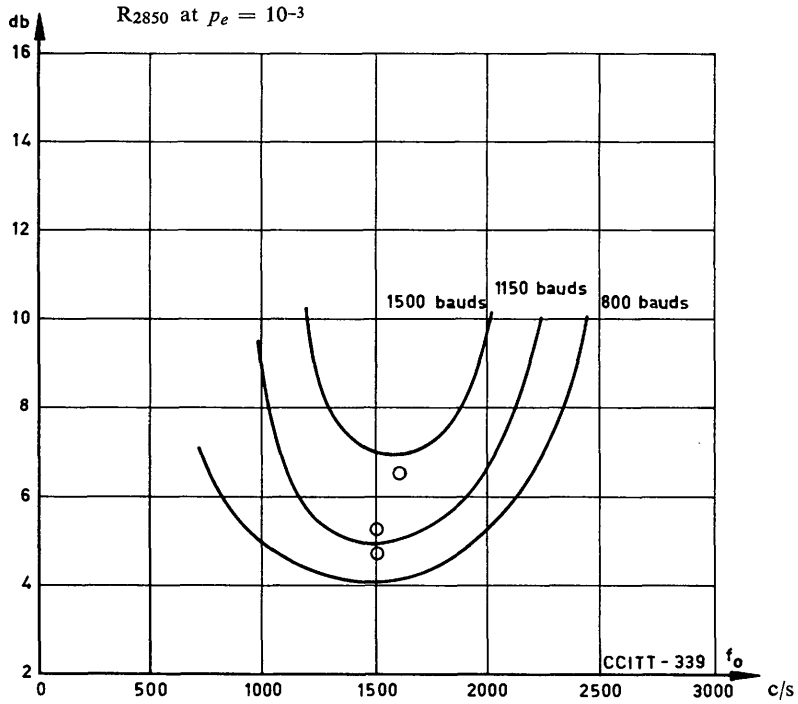


Total distance 200 km



Circuit No. 13

FIGURE 28. — **Relative level and group delay diagrams**



	Modulation rates		
	1500 bauds	1150 bauds	800 bauds
	c/s	c/s	c/s
Optimum mid-frequency	1600	1500	1500
Suitable working range	1200-2000	1050-2100	700-2250
Function range	1150-2100	950-2300	650-2500

Circuit No. 13

FIGURE 29. — **Signal-to-noise ratio R_{2850} at $p_e = 10^{-3}$ versus channel mid-frequency f_0**

SUMMARY OF FIGURES

Circuit No.	Type of circuit	Total distance	Usable modulation rate	Phase equalized	Figure No.
		(km)	(bauds)		
1	Loaded cable, physical	200	800	Yes	1 2 3
2	Loaded cable, phantom	200	800, 1150, 1500	Yes	4 5 6 7
3	Loaded cable, physical	94	800, 1150, 1500		8 9
4	Loaded cable, phantom	188	800, 1150, 1500		10 11
5	Loaded cable, physical	280	800, 1150		12 13
6	Loaded cable, physical	94	800, 1150	Yes	14 15
7	Loaded cable, physical	126	800, 1150		16 17
8	Loaded cable, physical	256	800		18 19
9	Loaded cable, physical	500	800		20 21
10	Carrier 12-channel system	—	800, 1150, 1500	Yes	22 23
11	Carrier 12-channel system	—	800, 1150		24 25
12	Carrier coax.-cable system	200	2400 [a]	Yes	26 27
13	Loaded cable, phantom + carrier, 12-channel system	(200)	800, 1150, 1500	Yes	28 29

[a] Circuit 12 has been measured only with 2400 bauds.

Supplement No. 21

SWEDEN — *(Extract from contribution COM Sp. A/No. 10 — August 1961)*

DATA TRANSMISSION TESTS OVER SWITCHED CIRCUITS

1. *Introduction*

The measurements have been carried out on dialled-up circuits consisting of rather long local circuits in pupinized pairs and on toll circuits of carrier type.

2. *Method of measurement*

The information consisted of a continuous sequence of 256 different signs in the eight-unit code, transmitted on a selected circuit loop. At the receiving end the information obtained was compared with a suitably retarded replica of the original information and the errors arising from transmission were registered. One-half of the test loop of the toll circuits consisted of a high-quality physical programme circuit, which did not significantly add to the number of errors recorded.

The measurements were carried out at a lower and at a higher speed. The lower speed was 750 bauds for the first part of the tests (local circuits), and 600 bauds for the latter part (toll circuits) after the final measuring equipment had been brought into use. The higher speed was consistently 1200 bauds.

The measurements were carried out at send levels of −6 and −12 dbm, respectively. The levels of reception can be obtained from the attenuation values for the different types of connections and the send levels. In these diagrams the phase distortion expressed as group delay was also traced in certain cases.

The number of errors caused through data transmission over the telephone circuits was registered, on the one hand, as a total and, on the other hand through indication of the number of wrongly transmitted data blocks of the order of 8, 16, 32, 64, 128, 256 and 512 elements.

As has been mentioned, the measurements were carried out on dialled-up circuits. On each circuit chosen, four measurements were carried out in succession, each with 12 minutes' effective measuring time, and at speeds of 750 (600) and 1200 bauds at the two levels −6 and −12 dbm.

3. *Modulating equipment*

In the measurements, a modem equipment of frequency shift type (FSM) was used, which had been designed on the basis of the investigation results reported in the Swedish document (June 1961) [1]. A relatively low mid-frequency (f_0) was accordingly chosen, as a compromise between the requirements of getting optimum performance from pupinized local circuits ($f_0 = 1000$ c/s) and from carrier circuits ($f_0 = 1800$ c/s). It was the intention to arrive at a suitable mid-frequency, in order to create a generally applicable modem system with sufficient reliability for use in commercial circuits, combining both carrier circuits and pupinized circuits.

Originally $f_0 = 1200$ c/s had been chosen as a compromise for measurements of local circuits. At measurements on carrier channels however, this f_0 value turned out to be too low at a speed of 1200 bauds, as disturbances from metering impulses from certain rate-fixing units had to be taken into account. For measurements on dialled-up *carrier* circuits, f_0 was therefore increased to

[1] Supplement No. 20.

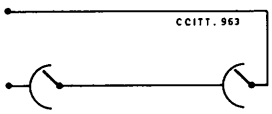
1320 c/s, a frequency which also seemed acceptable for a large part of existing loaded circuits. However, at this frequency, too, transmission is disturbed to some extent by rate impulses. It is only at a frequency of $f_0 = 1600\text{-}1800$ c/s that the influence of the rate impulses becomes negligible. However, such a high mid-frequency does not seem suitable for general use in a switched network. The proper choice of a mid-frequency is surely complicated.

Measuring results

	Modulation rates (bauds)	Send level dbm	Number of blocks in error for specified number of elements per block								Number of elements transmitted
			1	8	16	32	64	128	256	512	
SCHEDULE OF TESTS No. 1 **Circuit arrangements :** **Different 2×16.3 km local loops, including two "500-selectors" (Stockholm-Hässelby-Stockholm)** Send end → Receive end ← CCITT - 340 Mid-frequency : 1200 c/s Modem channel similar to type FS 480 but 25% broader	750	−6	83	63	63	63	62	56	52	44	12.4×10^6
SCHEDULE OF TESTS No. 2 **Different 2×16.3 km local loops, including two "500-selectors" (Stockholm-Hässelby-Stockholm)** Mid-frequency : 1200 c/s Modem channel similar to type FS 480 but 25% broader	750	−12	114	89	85	84	82	74	69	61	12.4×10^6
SCHEDULE OF TESTS No. 3 **Different 2×16.3 km local loops, including two "500-selectors" (Stockholm-Hässelby-Stockholm)** Mid-frequency : 1200 c/s Modem channel type FS 960	1200	−6	117	73	63	60	60	59	48	40	19.8×10^6
SCHEDULE OF TESTS No. 4 **Different 2×16.3 km local loops, including two "500-selectors" (Stockholm-Hässelby-Stockholm)** Mid-frequency : 1200 c/s Modem channel type FS 960	1200	−12	391	278	265	260	241	226	219	204	18.9×10^6
SCHEDULE OF TESTS No. 5 **Different 2×19.2 km local loops, including two "500-selectors" (Stockholm-Norrviken-Stockholm)** Mid-frequency : 1200 c/s Modem channel similar to type FS 480 but 25% broader	750	−6	24	15	15	13	11	11	11	11	14.0×10^6

	Modulation rates (bauds)	Send level dbm	Number of blocks in error for specified number of elements per block								Number of elements transmitted
			1	8	16	32	64	128	256	512	
SCHEDULE OF TESTS No. 6 **Different 2×19.2 km local loops, including two "500-selectors" (Stockholm-Norrviken-Stockholm)** Mid-frequency : 1200 c/s Modem channel similar to type FS 480 but 25% broader	750	−12	65	58	54	52	51	51	46	44	12.4×10^6
SCHEDULE OF TESTS No. 7 **Different 2×19.2 km local loops, including two "500-selectors" (Stockholm-Norrviken-Stockholm)** Mid-frequency : 1200 c/s Modem channel type FS 960	1200	−6	47	39	39	38	37	37	37	37	19.8×10^6
SCHEDULE OF TESTS No. 8 **Different 2×19.2 km local loops, including two "500-selectors" (Stockholm-Norrviken-Stockholm)** Mid-frequency : 1200 c/s Modem channel type FS 960	1200	−12	278	229	220	215	213	210	207	203	19.8×10^6
SCHEDULE OF TESTS No. 9 **Different 2×6.4 km local loops, including two "500-selectors" (Stockholm-Lidingö-Stockholm)** Mid-frequency : 1200 c/s Modem channel similar to type FS 480 but 25% broader	750	−6	28	20	17	16	16	15	15	15	20.5×10^6
SCHEDULE OF TESTS No. 10 **Different 2×6.4 km local loops, including two "500-selectors" (Stockholm-Lidingö-Stockholm)** Mid-frequency : 1200 c/s Modem channel similar to type FS 480 but 25% broader	750	−12	36	28	28	27	27	26	26	26	20.0×10^6

	Modulation rates (bauds)	Send level dbm	Number of blocks in error for specified number of elements per block								Number of elements transmitted
			1	8	16	32	64	128	256	512	
SCHEDULE OF TESTS No. 11 **Different 2×6.4 km local loops, including two "500-selectors" (Stockholm-Lidingö-Stockholm)** Mid-frequency : 1200 c/s Modem channel type FS 960	1200	−6	225	125	110	101	91	83	79	72	34.4×10^6
SCHEDULE OF TESTS No. 12 **Different 2×6.4 km local loops, including two "500-selectors" (Stockholm-Lidingö-Stockholm)** Mid-frequency : 1200 c/s Modem channel type FS 960	1200	−12	168	110	105	98	92	83	81	76	31.8×10^6
SCHEDULE OF TESTS No. 13 **Different 2×17 km local loops, including two crossbar selectors (Malmö-Klågerup-Malmö)** Mid-frequency : 1320 c/s Modem channel type FS 480	750	−12	9	3	3	2	2	2	2	2	10.3×10^6
SCHEDULE OF TESTS No. 14 **Different 2×17 km local loops, including two crossbar selectors (Malmö-Klågerup-Malmö)** Mid-frequency : 1320 c/s Modem channel type FS 960	1200	−6	55	30	22	20	18	17	14	13	17.2×10^6
SCHEDULE OF TESTS No. 15 **Different 2×17 km local loops, including two crossbar selectors (Malmö-Klågerup-Malmö)** Mid-frequency : 1320 c/s Modem channel type FS 960	1200	−12	65	52	49	48	47	46	45	42	17.2×10^6

	Modulation rates (bauds)	Send level dbm	Number of blocks in error for specified number of elements per block								Number of elements transmitted
			1	8	16	32	64	128	256	512	
SCHEDULE OF TESTS No. 16 **Circuit arrangements : Different 600-km carrier circuits, including one "500-selector" and one "crossbar selector" (Stockholm-Malmö-Stockholm)** Send end Music circuit  Receive end Carrier circuit Mid-frequency : 1320 c/s Modem channel type FS 480	600	−6	68	48	45	44	41	41	40	39	25.0×10^6
SCHEDULE OF TESTS No. 17 **Circuit arrangements : Different 600-km carrier circuits, including one "500-selector" and one "crossbar selector"** Mid-frequency : 1320 c/s Modem channel type FS 480	600	−12	140	113	102	99	97	97	96	95	21.9×10^6
SCHEDULE OF TESTS No. 18 **Circuit arrangements : Different 600-km carrier circuits, including one "500-selector"and one "crossbar selector"** Mid-frequency : 1320 c/s Modem channel type FS 960	1200	−6	242	182	176	172	169	163	159	153	45.4×10^6
SCHEDULE OF TESTS No. 19 Mid-frequency : 1320 c/s Modem channel type FS 960	1200	−12	956	704	675	655	642	636	631	620	18.9×10^6

Supplement No. 22

CHILE TELEPHONE CO. — *(Extract from contribution COM Sp. A/No. 7 — July 1961)*

MEASUREMENTS ON SWITCHED AND LEASED TELEPHONE LINES TRANSMITTING DATA AT SPEEDS OF 250, 800 AND 1000 BAUDS

1. *General comments on tests*

Since November 1959 an extensive series of tests has been carried out on a wide variety of switched and leased telephone circuits at data-transmission speeds of from 250 to 1000 bauds. In these tests special arrangements have been made to record the identity of every element received in error and, as a consequence, detailed information is available about the fine structure of the error patterns that occurred during any of these tests.

This detailed information has made it possible to analyse a number of parameters which are of interest for system design ; for example, specific information is available about the percentage of the data which would need to be retransmitted if a system employing error detection with correction by block retransmission were used.

It has also been possible to determine the percentage of errors which remain undetected with a large variety of different redundancy arrangements and with different block sizes.

Yet another interesting subject concerns the distance between the beginning and end of a burst of errors, particularly when examined from the point of view of a system using block retransmission.

The following comments relate to various clauses of the test programme for data transmission systems on telephone-type circuits contained in Section L of contribution GT 43, No. 20 (paragraphs as listed) :

(4) The signal levels used for test purposes were in accordance with the recommendations in respect to the maximum permissible power, it being assumed that these figures include both backward and forward signals.

(5) One conclusion of the analysis which has been carried out is that the sample size quoted is inadequate for many of the studies such as comparing different error-detection codes. For a great many of these studies the sample size which is important is the number of errors and not the amount of data transmitted. For code comparison purposes the useful portion of the sample may be restricted to that section in which more than three errors occur per block and in such circumstances an even larger sample is needed.
In the case of switched toll circuits, the majority of tests were made on connections of approximately 250 miles, but some tests used longer connections of nearly 750 miles. For the leased-circuit tests, the majority of tests used connections of 750 miles but a number of tests were also made over approximately 3000 miles.
The tests were restricted to the business hours of normal working days and it is considered that the attenuation conditions were representative.

(6) The test message contained 2000 bits and was made up of several sections each containing a different message characteristic.

(8) The error record can be analysed in blocks of any size. For convenience, blocks of 50, 100, 200 and 500 bits were found easiest to process and to maintain the identity of the bits in error.

(10) Characteristics of the connections were monitored and measured particularly as regards attenuation, impulsive noise, the signal level at the receiving apparatus, and the presence of switch noise and dial impulses. It was noted that for switched connections a large proportion of all errors were due to isolated and very sharp peaks of interference which influenced only one bit at 1000 bauds.

There was no evidence of unwanted operation of the toll signalling equipment.
Although the switched toll connections were normally switched at distinct centres on a two-wire basis, there was no evidence of trouble due to double echo effects on these connections.

No trouble was experienced by warning tones or metering signals.

(11) Interruptions exceeding 300 ms were excluded. On leased connections the elimination of these interferences had an important bearing on error rates. It is also to be noted that interruptions exceeding 300 ms were also experienced on switched connections without the connection being broken down. In practice it is likely that these longer interruptions would also increase the error rates.

Table I gives a brief summary of the tests to the end of June 1961.

The equipment used for the tests has been described in a number of papers :

1. Chile Telephone Company's reply to Working Party 43 (4 February 1960).

2. Cause, Prevention and Correction of Errors recorded during Data Transmission Tests, by E. P. G. Wright, M.I.E.E., *Proceedings of Seventh Annual International Congress on Electronics*, Rome. June 1960.

3. Error Rates and Error Distributions on Data transmitted over Switched Telephone Connections, by E. P. G. Wright, M.I.E.E., *I.R.E. Transactions*, Vol. CS-9 No. 1, March 1961, pp. 12-15.

2. *250-baud tests*

No tests were conducted at speeds lower than 250 bauds, but it is considered that the results obtained for 250 bauds operation on telephone lines would be directly comparable with 200-baud telegraph operation, as the bandwidth was suitably restricted to 300 c/s.

In the presentation of the results, emphasis has been directed at those aspects concerned with the design of error control, i.e. :

(a) block retransmission rates ;

(b) error distributions by blocks ;

(c) error separation characteristics.

TABLE I

Summary of data transmission tests

Item	Number of tests	Number of connections tested	Number of bits transmitted (in millions)	Baud speed	Connection	Number of errors
1	62	24	77	250	Local and toll switched	5 699
2	22	5	42	250	Local switched	3 136
3	11	4	11	simulated 500	Local switched	1 658
4	34	12	50	500	Local switched	4 350
5	10	6	13	simulated 1000	Local switched	3 918
6	4	1	14	simulated 1000	Local unswitched	9
7	58	22	150	1000	Local and toll switched	13 826
8	12	6	23	1000	Toll switched	1 611
9	24	12	55	1000	Toll switched (frequency attenuation measured)	2 228
10	10	4	20	1000	Toll switched (test message changed)	195
11	25	9	83	1000	Toll switched (receiver changed)	1 919
12	8	—	7.0	1000	P.O. noise tape	2 712
13	13	8	18.0	1000	Local and toll switched	172 078 [a]
14	2	2	5.0	1000	Toll switched	2 606 [a]
15	2	1	4.8	1000	Toll switched	834 [a]
16	11	4	33.2	1000	Toll switched	2 957 [a]
17	3	3	1.5	250	Loaded, unswitched, local and toll	0
18	2	2	2.6	500	Ditto	0
19	3	3	6.0	1000	Ditto	18 722
20	39	1	139.5	800	Leased, Series PA (microwave link)	1 083
21	26	1	636.6	800	Ditto (night tests)	634 410
22	39	1	134.2	800	Leased, series PA (submarine cable)	3 171
23	15	1	572.4	800	Ditto (night tests)	1 297
24	4	1	187.9	800	Local unswitched	2
25	80	1	300.0	800	Leased, series S	2 459
26	44	1	768.8	800	Ditto (night tests)	907 225
27	2	1	8.0	800	Leased, series PB	19
28	26	1	90.4	1000	Ditto	2 752
29	3	1	152.8	1000	Ditto (night tests)	682 663
Totals	594	138	3607.3			2 295 067
					plus	178 475 [a]

[a] Number of characters in error.

2.1 *Retransmission percentages*

Table II summarizes the results of element and block error rates and retransmission percentages for blocks of 50, 100, 200 and 500 bits. Included in the retransmission is the block following an erroneous block. Table III shows the effect on these three parameters of different receiving levels, the level quoted being that at the last switch in the connection.

The results shown in Table II appear in graphical form in Figure 1, which enables results to be read for intermediate block sizes.

TABLE II

250-baud tests on switched connections

Total number of elements transmitted	$E_t = 65.2 \times 10^6$				
Total number of faults	$F = 5367$				
Element error rate	$p_e = 8.25 \times 10^{-5}$				
Block size (bits)		50	100	200	500
Blocks transmitted		1.3×10^6	6.5×10^5	3.25×10^5	1.5×10^5
Blocks in error		1956	1717	—	—
Block error rate p_b		1.5×10^{-3}	2.64×10^{-3}	—	—
Blocks retransmitted [1]		3264	2904	2602	2252
Retransmission percentage		0.25	0.45	0.80	1.7

[1] Includes retransmission of the block following the erroneous block. It can be seen that the block following the erroneous block often contains errors.

TABLE III

250-baud tests — Effect of level variation

Level at last switch	p_e	Retransmission percentage for block size			
		50 bits	100 bits	200 bits	500 bits
L > − 15 dbm	2.76×10^{-6}	0.0103	0.0207	0.0413	0.0687
− 15 > L > − 25 dbm	5.7×10^{-5}	0.0825	0.135	0.226	0.486
− 25 > L > − 35 dbm	7.4×10^{-5}	0.099	0.167	0.284	0.618
L < − 35 dbm	1.26×10^{-4}	0.552	1.0	1.81	3.92

2.2 *Block error distributions*

Figure 2 shows the cumulative error distribution for 50- and 100-bit blocks.

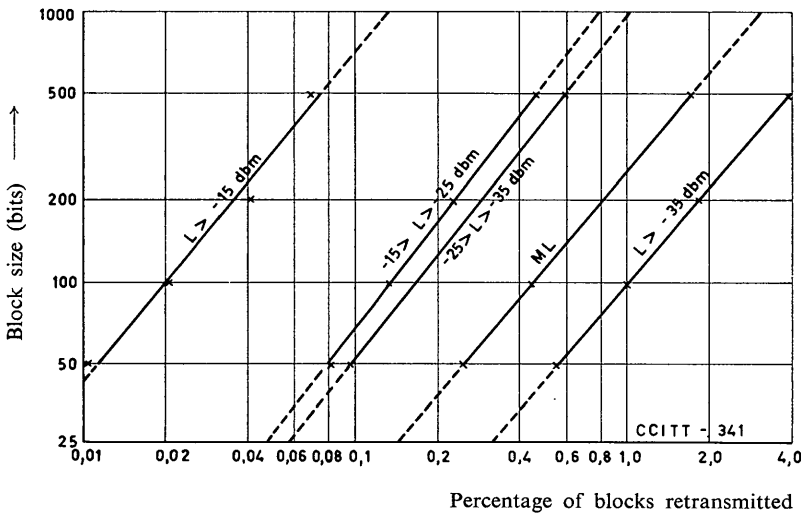
g_i is defined as the ratio of the total number of errors occurring "i" times in a block to the total number of errors, or :

$$g_i = \frac{i\ B_{fi}}{F}$$

where : I = Number of errors per block

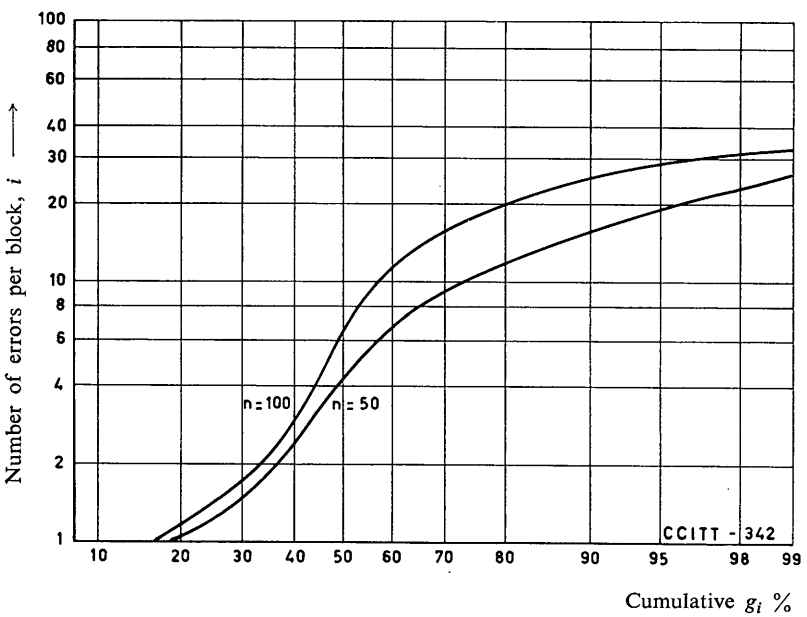
B_{fi} = Number of blocks containing exactly i errors

F = Total number of errors



L = Level at last switch

FIGURE 1. — **Retransmission percentages for tests at 250 bauds**



n = Block size (in bits)

FIGURE 2. — **Cumulative error distributions based on switched circuit tests in the United Kingdom at 250 bauds**

One of the factors influencing the g_l distribution is the message content. Many of the errors are caused by short breaks during which the receive relay remains on one side or the other. If the relay is in the "space" (or "0") condition, then only the transmitted "marks" (or "1"s) will be registered as errors and vice versa.

Since a succession of "0"s is quite likely in practical data transmission, this condition has been included in the standard message used throughout the tests. The remainder of the message consists of a period of reversals and a random pattern of "1"s and "0"s.

2.3 *Error separation characteristics*

The error tapes were analysed in a computer to give the separation in bits between errors. The result of this analysis for different tests is shown by the appropriate curve in Figure 3. It will be seen that in the case of 250-baud operation the probability of an error being succeeded immediately by another is some 36%.

3. *800-1000-baud tests*

Similar tests to those described above were conducted at higher speeds on local and toll switched circuits and on a number of leased international circuits. The results of these tests are summarized in Table IV and Figure 4.

3.1 *Block error distribution*

Figure 5 shows the cumulative g_l characteristics for tests on both switched and leased connections.

3.2 *Error separations*

The error separation characteristics for leased and switched lines are shown in Figure 3.

4. *Influence of time of day*

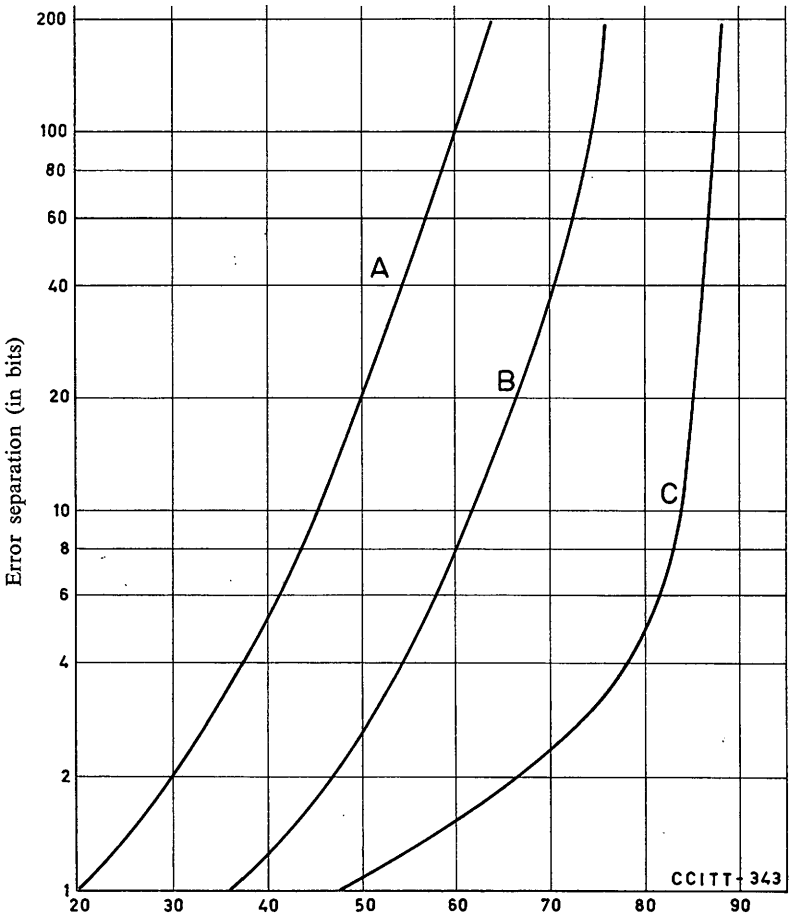
An analysis of the results reveals that there are occasional periods during the day during which the incidence of errors increases. On switched circuits, these are related to the busy-hour peaks in the morning and afternoon and, to a lesser extent, on leased lines. Probably the greatest effect, however, is due to maintenance operations and these may or may not coincide with the busy hour.

Figure 6 shows the variation in error rate for switched circuits, and Table V shows the variation for leased international circuits.

5. *Burst lengths*

Some of the leased-connection tests have been analysed to gain some information regarding length of error bursts. The results are given in Table VI.

For this analysis, a burst is defined as four or more errors occurring within a block, and the burst length is the number of bits between the first and last errors in the block.



A = Switched connections : 1000 bauds
F = 15 897
p_e = 5.9×10^{-5}

B = Switched connections : 250 bauds
F = 5367
p_e = 8.25×10^{-5}

C = Leased connections : 1000 bauds
F = 3512
p_e = 1.3×10^{-5}

F = Total errors in sample

p_e = Element error rate

FIGURE 3. — **Distribution of error separations**

TABLE IV

1. *1000-baud tests on switched connections*

1.1 *Local circuits*

Level at last switch	$= -28$ dbm
Number of tests	$= 30$
Total number of bits transmitted	$= 71.3 \times 10^6$
Number of errors	$= 5468$
Element error date	$= 7.67 \times 10^{-5}$

Block size (bits)	50	100	200	500
Number of blocks transmitted	1.43×10^6	7.15×10^5	3.58×10^5	1.43×10^5
Number of faulty blocks	3468			
Block error rate	2.43×10^{-3}			
Average errors per faulty block	1.58			
Number of blocks retransmitted	6164	5610	5086	4360
Retransmission percentage	0.43	0.78	1.42	3.05

1.2.1 *Toll circuits*

Level at last switch	$= -20$ dbm
Number of tests	$= 64$
Number of bits transmitted	$= 191.3 \times 10^6$
Number of errors	$= 6017$
Element error rate	$= 3.15 \times 10^{-5}$

Block size (bits)	50	100	200	500
Number of blocks transmitted	3.82×10^6	1.91×10^6	9.55×10^5	3.82×10^5
Number of faulty blocks	2833			
Block error rate	7.41×10^4			
Average errors per faulty block	2.12			
Number of blocks retransmitted	4756	4492	4180	3656
Retransmission percentage	0.12	0.235	0.44	0.96

1.2.2 *Toll circuits*

Level at last switch	$= -34$ dbm
Number of tests	$= 28$
Number of bits transmitted	$= 64.1 \times 10^6$
Number of errors	$= 9786$
Element error rate	$= 1.53 \times 10^{-4}$

Block size (bits)	50	100	200	500
Number of blocks transmitted	1.28×10^6	6.41×10^5	3.2×10^5	1.28×10^5
Number of faulty blocks	5080			
Block error rate	3.96×10^{-3}			
Average errors per faulty block	1.93			
Number of blocks retransmitted	8622	7564	6678	5642
Retransmission percentage	0.67	1.18	2.1	4.4

1.3 *Over-all retransmission percentage*

Block size	50	100	200	500
Retransmission percentage	0.3	0.54	0.98	2.1

2. *800- and 1000-baud tests on leased connections*

2.1 *Series PA* (800 bauds)

Sending level	$= -13$ dbm0
Number of tests	$= 54$
Total number of bits transmitted	$= 271.7 \times 10^6$
Number of errors	3528
Element error rate	$= 1.30 \times 10^{-5}$

Block size (bits)	50	100	200	500
Number of blocks transmitted	5.44×10^6	2.72×10^6	1.36×10^6	5.44×10^5
Number of faulty blocks	700	610	546	485
Block error rate	1.29×10^{-4}	2.25×10^{-4}	4.02×10^{-4}	8.93×10^{-4}
Average errors per faulty block	5.04	5.8	6.48	7.28
Number of blocks retransmitted	1134	1052	972	886
Retransmission percentage	0.021	0.039	0.071	0.163

2.2 *Series PB* (1000 bauds)

Sending level	= −13 dbm0
Number of tests	= 15
Total number of bits transmitted	= 85.5×10^6
Number of errors	= 2752
Element error rate	= 3.22×10^{-5}

Block size (bits)	50	100	200	500
Number of blocks transmitted	1.71×10^5	8.55×10^4	4.27×10^4	1.71×10^4
Number faulty blocks	726	519	379	270
Block error rate	4.25×10^{-4}	6.0×10^{-4}	8.88×10^{-4}	1.58×10^{-3}
Average errors per faulty block	3.79	5.24	7.25	10.19
Number of blocks retransmitted	976	714	550	428
Retransmission percentage	0.057	0.084	0.13	0.25

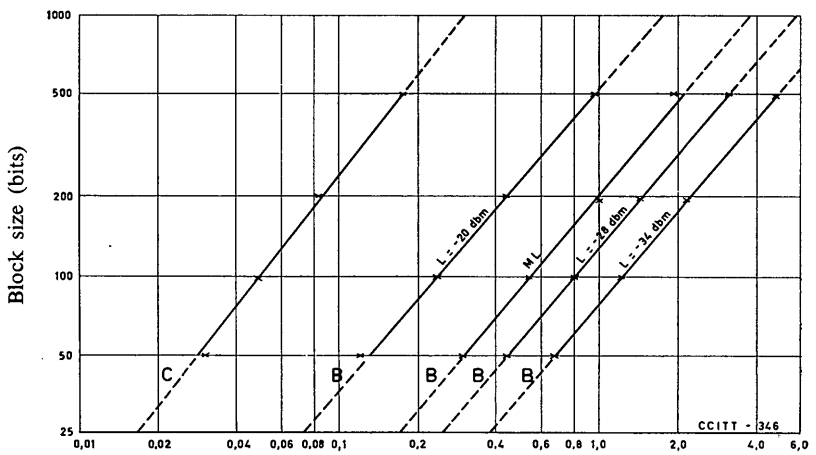
2.3 *Series S* (800 bauds)

Sending level	= −13 dbm0
Number of tests	= 37
Total number of bits transmitted	= 134×10^6
Number of errors	= 2262
Element error rate	= 1.69×10^{-5}

Block size (bits)	50	100	200	500
Number of blocks transmitted	2.68×10^6	1.34×10^6	6.7×10^5	2.68×10^5
Number of faulty blocks	588	442	331	237
Block error rate	2.19×10^{-4}	3.3×10^{-4}	4.94×10^{-4}	8.84×10^{-4}
Average errors per faulty block	3.84	5.11	6.81	9.54
Number of blocks retransmitted	808	626	476	362
Retransmission percentage	0.03	0.047	0.071	0.135

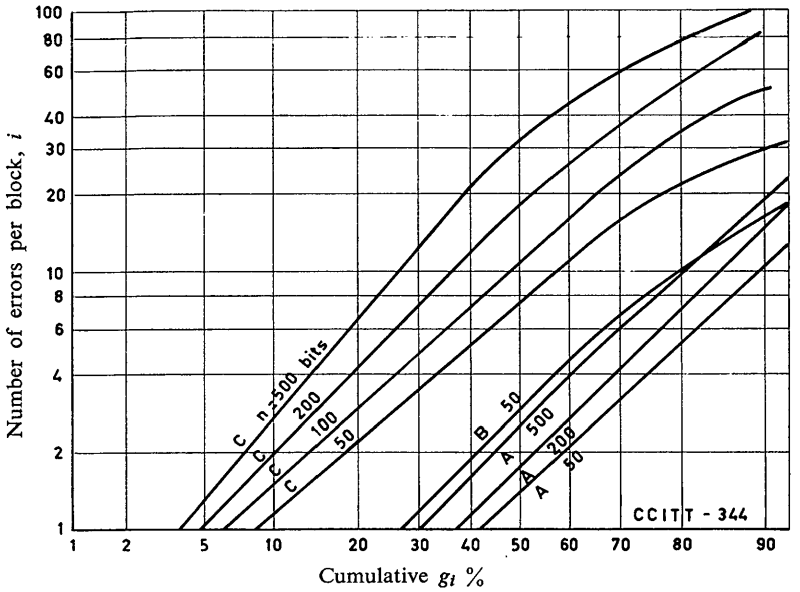
2.4 *Over-all retransmission percentage*

Block size	50	100	200	500
Retransmission percentage	0.03	0.049	0.082	0.17



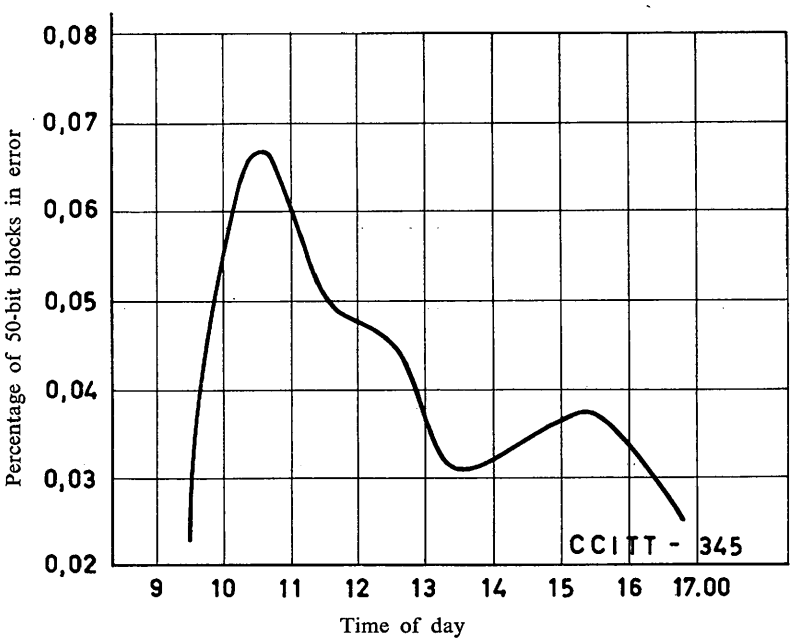
B = Switched connections
C = Leased connections
L = Level at last switch
ML = Mean of all levels

FIGURE 4. — **Retransmission percentages for tests at 1000 bauds**



A = 1000-baud tests on switched connections. Level at last switch = −28 dbm.
B = 1000-baud tests on switched connections. Level at last switch = −20 dbm.
C = 1000-baud tests on leased connections. Sending level = −13 dbm.
n = Block size.

FIGURE 5. — **Cumulative error distributions for 1000-baud tests**



Level at last switch ≃ −24 dbm

FIGURE 6. — **Variation in error rate during the day on switched connections at 1000 bauds**

TABLE V

800- and 1000-baud tests

Variation of error rate during the day on leased connections

Tests	Element error rate during morning (10-12)	Element error rate during afternoon (2-4)
Series PA	1.6×10^{-5} (112)	1.5×10^{-5} (86)
Series PB	4.9×10^{-5} (32)	2.5×10^{-5} (38)
Series S	0.8×10^{-5} (104)	1.02×10^{-5} (99)
Over-all	1.7×10^{-5} (248)	1.5×10^{-5} (224)

Numbers in brackets indicate millions of bits transmitted.

TABLE VI

(a) *Distribution of error bursts in 50-bit blocks*

Number of errors in burst	Number of bursts of length: 4	5-6	7-8	9-10	11-15	16-25	26-50 bits
4	5	8	6		3	3	10
5		6	1	1	2	2	14
6		1	2	2		2	6
7					2	3	5
8			1			2	3
9					2	2	7
10					1	3	6
11-15						5	16
16-25							17
26-50							9
Totals	5	15	10	3	10	22	93

(b) *Distribution of error bursts in 200-bit blocks*

Number of errors in burst	Number of bursts of length: 4	5-6	7-8	9-10	11-15	16-25	26-50	51-100	101-200 bits
4	2	3	1		2	1	2	6	9
5		5				2	2	4	8
6							1	2	6
7					1	2		3	4
8			1			2		5	6
9					1	1	1		5
10					1	2	2	3	1
11-15						2	1	1	2
16-25							8	3	4
26-50								2	11
51-100								2	5
Totals	2	8	2		5	12	17	31	61

Supplement No. 23

UNITED KINGDOM — *(Contribution COM Sp. A/No. 8 — July 1961)*

TEST FOR DATA TRANSMISSION SYSTEMS ON TELEPHONE-TYPE CIRCUITS

1. *Introductory note*

Several different data transmission systems have been tested in the U.K. over telephone type circuits; for the purpose of reporting on the results they have been given identification letters A, B, C, D, E. Previous tests on system A and tests on system B were described in Volume VII of the *Red Book*. This contribution describes further tests on system A and results of tests with systems C, D and E.

2. *Signalling levels*

It would perhaps be useful to explain the method of setting up the transmitted signals. In standardizing transmitting levels for devices using voice-frequency signals (including data transmission systems) over telephone-type circuits it appears necessary to meet two conditions:

i) To produce a simple form of specification which can be applied at the subscribers' line terminals.

ii) To meet the C.C.I.T.T. recommendations on data transmission signal levels which primarily relate to power levels at the zero transmission level point on main line plant.

The U.K. Administration therefore specifies the permissible level of voice-frequency signals on telephone-type circuits in the following way:

"The signal applied to a 600-ohm load in place of the line shall not exceed the following peak-to-peak values, determined in principle from observation of the signal on an oscilloscope:

a) "Leased" circuits, 1 V peak to peak, subject, for an international circuit, to meeting C.C.I.T.T. Recommendation V.2.

b) Exchange lines, 2 V peak to peak except that if the transmission loss between the subscriber's telephone and the group centre is less than 5 db the limit shall be 1 V peak to peak."

Note. — "Group Centre" may be taken as the point where the first trunk carrier system in the train of circuits is encountered and the measurement of loss would be made from the subscriber's telephone and the zero relative level point in this carrier system.

3. *System A*

The main features of this experimental system were as follows:

Method of modulation: on/off amplitude modulation, bit—synchronous, 1000 bauds.
Sub-carrier frequency: 1600 c/s.
Automatic gain control not provided.
Line pilot of 400 c/s at −20 dbm0.

Changes of circuit gain had the effects shown below :

Within the limits −2 to + 8 db *no* errors caused.
Outside the limits −3 to + 9 db errors caused.

Error detection. — The received signal was compared bit by bit with a delayed send signal over a period of 3 microseconds and an error recorded if a difference was detected. Errors were recorded separately as "failures" (F) when the line signal was lost, or as "extras" (E) when a spurious signal was recorded. Errors were recorded on counters and on a tape. The tape recording was subsequently analysed visually using a tape reading equipment. It should be noted that with the type of modulation employed a large increase in line attenuation would cause no error for any bit corresponding to "carrier-off".

Test message. — Random test signals at 1000 bits per second.

Test circuits. — Loop connection of three circuits routed on coaxial cable systems, plus two audio circuits. The total length of coaxial cable system was 501 miles, the total length of audio circuit was 8 miles. Circuit arrangement was four-wire throughout and corresponds in quality to a good private circuit.

Signal levels. — The standard level was 1 V peak to peak at the zero relative level point, corresponding approximately to a mean power of −10 dbm0 during transmission of the random test signal. Tests with the signal level on the line varied by −6 db and + 6 db were interleaved in time with the tests at standard level.

Sample size

3.6×10^7 bits at 0.5 V p-p
4.8×10^7 bits at 1 V p-p
3.6×10^7 bits at 2 V p-p
Time of day : 0900-1230 and 1400-1700 hours.

General notes on observations of circuit characteristics

a) *Error causes.* — Errors were mostly due to short breaks, either partial or complete, which generally sounded like single clicks when heard on a monitoring loudspeaker. In four cases (which were all single errors) and in one case (of 279 errors) the cause was not audible.

b) *Circuit gain.* — Varied over the range +2.0 to −2.0 db (measured once per day).

c) *Background noise.* — Varied over the range −49 to −51 dbm0 (measured once per day).

Summary of results with system A

Table 1

Average error rates

	− 6 db (0.5 V p-p)	Standard level (1 V p-p)	+ 6 db (2 V p-p)
Total number of bits transmitted	3.6×10^7	4.8×10^7	3.6×10^7
Total number of bits in error { F	287	229 [a]	21
Total number of bits in error { E	5	0	0
Average bit error rate	8.1×10^{-6}	4.8×10^{-6} [a]	5.8×10^{-7}
Total number of 250 bit blocks transmitted	1.44×10^5	1.92×10^5	1.44×10^5
Total number of 250 bit blocks in error [b]	12	17 [a]	4
Total number of 64 bit blocks transmitted	5.62×10^5	7.5×10^5	5.62×10^5
Total number of 64 bit blocks in error [c]	18	42 [a]	4
Average block error rate :			
250-bit blocks [b]	8.3×10^{-5}	8.9 [a] $\times10^{-5}$	2.8×10^{-5}
64-bit blocks [c]	3.2×10^{-5}	5.6 [a] $\times10^{-5}$	7.1×10^{-6}

[a] Two long bursts of "failure" errors excluded, 2642 errors in one 12-second period and 1673 errors in a 26-second period, probably caused by intermittent disconnection of the circuit.

[b] Assuming that (a) bursts separated by less than 125 bits affect 1 block only, and
(b) bursts of length in the range 125 $(2n-3)$ to 125 $(2n-1)$ bits cause n blocks to be in error where n is an integer.

[c] Assuming that (a) bursts separated by less than 32 bits affect 1 block only, and
(b) bursts of length in the range 32 $(2n-3)$ to 32 $(2n-1)$ bits cause n blocks to be in error where n is an integer.

Table 2

Error structure

Signal amplitude : 0.5 V p-p

Error-free runs	Duration	BURSTS
		Structure { F bits received incorrectly due to loss of line signal; E bits received incorrectly due to presence of spurious line signal; G bits received correctly
(bits)	(bits)	
3.36×10^6		
1.30×10^6	3	F 3
1.13×10^7	1	F 1
6.96×10^6	1	E 1
4.40×10^6	502	Burst of 279 F errors recorded on tape
1.88×10^6	2	E 2
6.0×10^4	1	E 1
4.2×10^5	2	F 2
	2	F 1
4.24×10^6		E 1
2.04×10^6	1	F 1

TABLE 3

Error structure

Signal amplitude : 1 V p-p

Error-free runs	Duration	BURSTS Structure { F bits received incorrectly due to loss of line signal; E bits received incorrectly due to presence of spurious line signal; G bits received correctly
(bits)	(bits)	
7.78×10^6	43	F 1 2 2 2 1 4 1 1 6 1 1 G 1 1 1 1 2 4 5 1 2 3
1.2×10^5	32	F 1 1 2 2 2 2 4 3 1 G 2 2 1 2 1 1 2 3
1.1×10^7	1.2×10^4	Burst of 2642 F errors recorded on tape
1.3×10^4	2.6×10^4	Burst of 1673 F errors recorded on tape
1.0×10^4	2	F 2
6.0×10^2	2	F 2
1.4×10^2	7	F 1 1 G 5
1.1×10^2	1	F 1
2.5×10^2	2×10^3	Burst of 177 F errors recorded on tape
1.5×10^4	1	F 1
1.22×10^7	1	F 1
6.42×10^6	7	F 1 2 G 4
1.04×10^7		

TABLE 4

Error structure

Signal amplitude : 2 V p-p

Error-free runs	Duration	BURSTS Structure { F bits received incorrectly due to loss of line signal; E bits received incorrectly due to presence of spurious line signal; G bits received correctly
(bits)	(bits)	
1.178×10^7	6	F 1 1 1 G 2 1
8×10^3	6	F 3 1 G 2
1.498×10^7	30	F 2 1 1 2 1 1 2 1 G 3 2 9 1 2 1 1
32	4	F 1 2 G 1
9.22×10^6		

4. *System C*

Main features of this system, of experimental type, were as follows :

Data system—variable speed 800-1000 bauds ;

Method of modulation—FM ; 1100 c/s = 1, 1900 c/s = 0.

Line pre-equalized at the sending end (nominally linear slope, 1 db per 200 c/s, over the range 900 to 2000 c/s). The sending level during transmission of the test message was −13 dbm at the zero level point on international circuits and −6 dbm at the subscribers' line terminals on switched connections.

Sending end units : modulator, sending filter (3 db bandwith of 1200 c/s), slope network.

Method of demodulation : frequency discriminator.

Receiving end units : receive channel filter, amplitude limiter discriminator, bi-stable output circuit.

Test message

Each test cycle contained 2000 bits, arranged as follows :

i) "m" consecutive 0 elements ;

ii) "n" reversals ;

iii) "p" elements of random signal.

Arrangement of test equipment

All tests were made on a looped basis using either go and return directions of four-wire circuits or by setting up a chain of connections via distant towns linking calling and called subscribers on the same exchange. The test equipment was housed in a laboratory some 30 miles away from the point where access was gained to the various kinds of circuit, these being extended to the laboratory via a high-quality audio circuit.

Conduct of tests — determination of errors

Two types of error measurement were used, one during daytime and the other during night-time tests. In the first type, a tape recording of every error was produced which detailed the precise position of each error. This tape was subsequently analysed by computer so that the distribution of errors could be found. The second type of measurement gave only a cumulative total of errors occurring during the testing period. In both types of test, errors arising during circuit interruptions of duration exceeding 300 ms were disregarded. In accordance with point L, paragraph 5, of document GT 43 No. 20, results of daytime tests only are given in this report.

i) *Tests on international telephone circuits*

Test circuits :

These were specially set up on spare plant, without switching, as four-wire circuits looped at the distant end.

Circuits used :

A. Through group ; coaxial cable system plus symmetrical pair submarine cable system, plus coaxial cable system.	Two circuits on the same route, approximately 490 miles long.
B. Through supergroup ; coaxial cable system, plus microwave radio system, plus coaxial cable system.	

C and D. As A and B, but extended at the distant end over a four-wire local circuit.

E. Through group ; coaxial cable systems plus 60-circuit symmetrical pair cable plus submarine cable system, plus coaxial cable system. 1110 miles long.

A summary of the test results is given in Table 5 and the error distributions are plotted in Figures 1, 2 and 3.

ii) *Tests on switched telephone connections*

Two series of tests were carried out on local exchange connections, set up by dialling from one subscriber to another, one series with a normal sending level and one at a lower sending level. A third series of tests was made over long-distance connections obtained by dialling from a subscriber through local and main switching centres.

The results are summarized in Table 6 and the error distributions are plotted in Figures 4, 5 and 6.

TABLE 5

Summary of test results with Data Transmission System C on international circuit

Circuit	A & B [1]	C & D [1]	E
Modulation rate (bauds)	800	1000	800
Sending level (at point of zero relative level on the international circuit)	− 13 dbm0	− 13 dbm0	− 13 dbm0
Number of tests	54	15	37
Total number of bits transmitted	2.717×10^8	8.55×10^7	1.34×10^8
Number of bits in error	3528	2752	2262
Average bit error rate	1.3×10^{-5}	3.22×10^{-5}	1.69×10^{-5}
Average block error rate :			
50-bit blocks	1.29×10^{-4}	4.25×10^{-4}	2.19×10^{-4}
100-bit blocks	2.25×10^{-4}	6.0×10^{-4}	3.3×10^{-4}
200-bit blocks	4.02×10^{-4}	8.88×10^{-4}	4.94×10^{-4}
500-bit blocks	8.93×10^{-4}	1.58×10^{-3}	8.84×10^{-4}
Average number of errors per faulty block :			
50-bit blocks	5.04	3.79	3.84
100-bit blocks	5.8	5.24	5.11
200-bit blocks	6.48	7.25	6.81
500-bit blocks	7.28	10.19	9.54
Number of blocks retransmitted [2] :			
50-bit blocks	1134	976	808
100-bit blocks	1052	714	626
200-bit blocks	972	550	476
500-bit blocks	886	428	362
Error distributions	Figure 1	Figure 2	Figure 3

[1] The results of tests on the two alternative routings A and B have been combined.
[2] Assuming that both the faulty block and the block immediately following are re-transmitted.

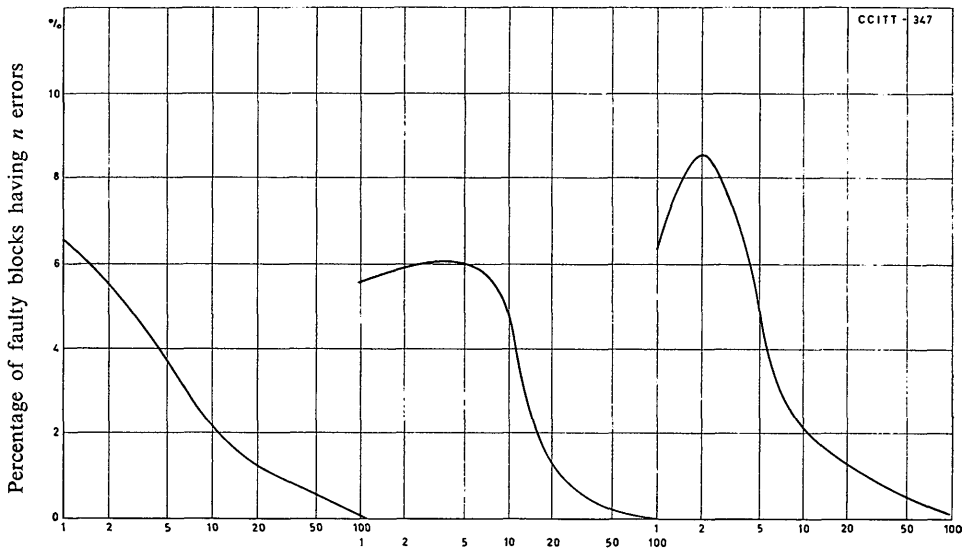


FIGURE 1. — *Circuits A and B : 800 bauds*

FIGURE 2. — *Circuits C and D : 1000 bauds*

FIGURE 3. — *Circuit E : 800 bauds*

Error distribution curves from tests with system C (FM) on international circuits — Block size, 100 bits

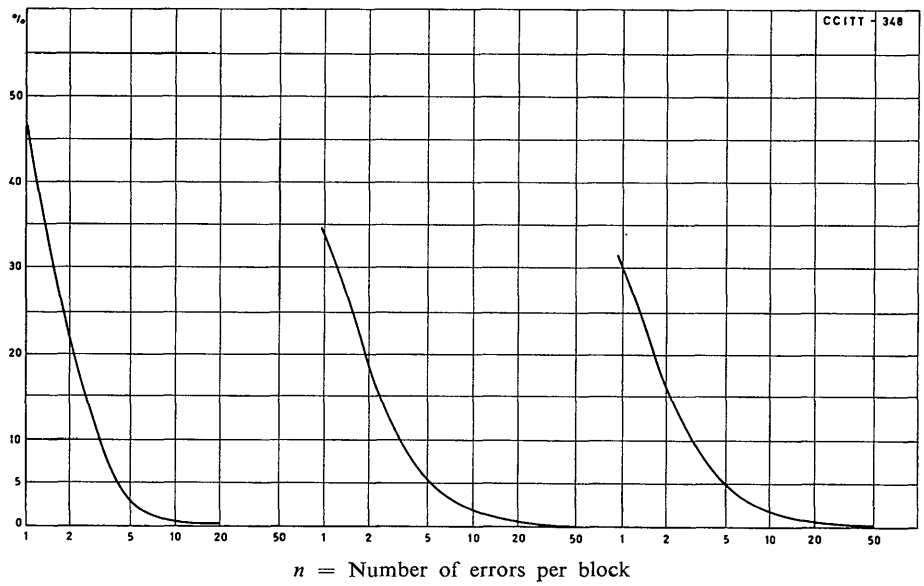


FIGURE 4. — *Local, switched connections Sending level —6 dbm*

FIGURE 5. — *Local, switched connections Sending level —12 dbm*

FIGURE 6. — *Long distance switched Sending level —6 dbm*

Error distribution curves from tests with system C (FM — 1000 bauds) on switched connections — Block size, 50 bits

TABLE 6

Summary of test results with Data Transmission System C on switched connections

Type of circuit	Local (first series)	Local (second series)	Long-distance
Modulation rate (bauds)	1000	1000	1000
Sending level at subscribers' line terminals (dbm)	−6	−12	−6
Receiving level (approx.) at subscribers' (dbm) line terminals	−31	−37	−30
Number of tests	30	8	64
Total number of bits transmitted	7.13×10^7	6.41×10^7	1.913×10^8
Number of bits in error	5468	9786	6017
Bit error rate	7.67×10^{-5}	1.53×10^{-4}	3.15×10^{-5}
Average block error rate : 50-bit blocks	2.43×10^{-3}	3.96×10^{-3}	7.41×10^{-4}
Average number of errors per faulty block : 50-bit blocks	1,58	1,93	2,12
Number of blocks re-transmitted [1] 50-bit blocks	6164	8622	4756
100-bit blocks	5610	7564	4492
200-bit blocks	5086	6678	4180
500-bit blocks	4360	5642	3656
Error distributions	Figure 4	Figure 5	Figure 6

[1] Assuming that both the faulty block and the block immediately following are re-transmitted.

5. *System D*

This was an early version of a production type of equipment and its main features were as follows :

Data system : variable speed, 600-1200 bauds.
Method of modulation : FM.
Frequencies used for 600-baud working ; 1400 c/s = 1 and 1900 c/s = 0.
For 1200-baud working ; 1200 c/s = 1 and 2200 c/s = 0.
Method of demodulation : amplifier limiter-differentiator-integrator-slicer.
Test message : pseudo-random pattern of 511 bits.

Arrangement of test equipment

All tests were made on an end-to-end basis, identical pattern generators being provided with the sending and receiving equipment, and arrangements made for the pattern generator at the receiving end to be synchronized with the received signal. The received signal after demodulation, etc., was compared bit by bit with the locally generated signal and any discrepancy, indicating an error, resulted in a record being made on a magnetic tape. A second track on the magnetic tape gave continuous timing information. The tape records will be subsequently analysed by computer and error statistics deduced ; preliminary deductions from observations made at the time of testing are given in the following tables and figures.

Conduct of tests

Tests were made between typical telephone subscribers' installations; the calls were set up in the usual way, mostly by dialling. Local and long-distance plant was involved. At each pair of locations, test calls were set up first in one direction and then in the other; on each call, data transmission tests were made in both directions.

Test transmissions were attempted first at 1200 bauds; if this was unsuccessful (usually owing to the presence of heavily-loaded cable in the connection), the modulation rate was changed to 600 bauds and tests made at the lower speed. The testing level used for all the tests was the appropriate standard level defined in section 2 of this report.

SYSTEM D — *Preliminary test results*

	At 1200 bauds	At 600 bauds
Total number of 15-minute tests	42	12
Total number of bits sent	4.53×10^7	6.2×10^6
Total number of errors	1340	494
Total number of bursts of errors	664	164
Average bit error rate	1 in. 4.53×10^4	1 in. 1.1×10^4

Distribution of error-free minutes:

Number of error-free minutes in a 15-minute test (N)	15	14	13	12	11	10	9	8
Number of test calls with "N" error-free minutes:								
at 1200 bauds	13	5	6	7	2	5	3	1
at 600 bauds	4	4	1	—	—	2	1	—

Note. — Nine calls failed to transmit at 1200 bauds, satisfactory transmission obtained at 600 bauds.

SYSTEM E — *Tests carried out on speech circuits using duplex data transmission alternately with telephony*

1. The equipment is referred to in the U.K. reply to Point J.

2. *Arrangement of test connection*

The signals were generated by an automatic telegraph transmitter at 50 bauds start-stop, the error rate was determined on a character basis, in accordance with Point G—first method.

3. *Signal level*

The tests were made on switched telephone circuits, with a sending level of −4 dbm.

4. *Description of equipment and tests*

The equipment provides speech or a 50- or 75-baud telegraph channel over a telephone circuit. The telegraph channel is duplex, using frequencies of 1680 and 1860 c/s, frequency-modulated ± 30 c/s shift. To study the feasibility of using this type of low-speed modulator on telephone circuits a programme of testing was carried out over the London telephone network. Calls were

set up by dialling from central London to the outskirts and also from points on the outskirts to other remote points, with the intention of using fairly long poor-quality circuits, i.e. circuits approaching the limit of speech transmission. The results can therefore be considered to include many more poor circuits than an average London network call would normally use ; the minimum line attenuation at 1800 c/s was 15 db and the maximum over 40 db.

5. *Test results*

TABLE 7

Characters transmitted : 6×10^6, taken on 43 days.
Number of calls which released automatically before test was completed : 7.
Number of operator interruptions : 9.
Number of double-connection occurrences : 4.
Number of error bursts : 169.
Fault rate was 1 incident (or burst) for 35 000 characters.

Error distribution

TABLE 8

	Characters in error per incident																	
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	Over 20
Number of incidents	108	8	11	9	5	3	4	—	1	1	—	—	2	4	1	1	1	8

TABLE 9

	Interval between incidents — minutes												
	0 to 10	10 to 20	20 to 30	30 to 40	40 to 50	50 to 60	60 to 70	70 to 80	80 to 90	90 to 100	100 to 150	150 to 200	Over 200 minutes between errors
Frequency of interval occurring	44	15	18	11	9	6	5	2	4	6	16	10	21

Comment

The difficulty in this test is to decide what errors to exclude from the summary as being a fault condition on the telephone circuits. There was one test in which over 300 errors occurred in 12 incidents without clearing the call down ; the circuit was faulty because there were three manual interruptions and two cases of double connection.

Excluding those tests which had "false clears" interruptions (over 300 ms) or double connections, the character error rate was 1/25 000 characters.

As the sample was 6×10^6 characters, the results should be representative of the use of this type of equipment on poor local telephone calls, and since the filters used in this equipment are only 120 c/s wide for each tone, hence avoiding signalling interference, the performance on combined local and trunk telephone calls should, on average, be better since the "tail" losses in the London outskirts are probably worse than in other towns in the United Kingdom.

Supplement No. 24

AMERICAN TELEPHONE AND TELEGRAPH CO. — *(Extract from contribution COM Sp. A/No. 13 — August 1961)*

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Supplement No. 25

N.V. PHILIPS TELECOMMUNICATIE INDUSTRIE — *(Extract from contribution COM Sp. A/No. 16 — September 1961)*

DATA TRANSMISSION TESTS ON LEASED TELEPHONE CIRCUITS

1. *Introduction*

In collaboration with the Société internationale de Télécommunications aéronautiques (S.I.T.A.), a data-transmission measuring and test programme has been carried out by Philips Telecommunicatie Industrie.

. . . . . .

The set-up, results and the conclusions of such a test on leased telephone circuits are described in the following points.

2. *Set-up of data transmission test system*

Leased telephone circuits between Paris and Hilversum (Netherlands) were used on a duplex basis.

The duplex circuit was looped at one end, transmitting, receiving and measurement equipment were connected to both ends of the duplex circuit at the other end.

The total length of the loop circuit was 1452 km.

The transmission level was maintained at −10 dbm.

Attenuation and phase equalization were applied.

The modulation rate throughout this test was 1000 bauds.

From preceding tests it was known that this modulation rate could be achieved on the majority of leased telephone circuits with a good performance, i.e. a low message-error rate.

It was realized that higher modulation rates are possible on many circuits, especially when the characteristics are in accordance with C.C.I.T.T. recommendations.

At present, however, there still is an appreciable number of circuits showing less favourable characteristics.

The circuits used are a case in point.

It can be seen from our conclusions that the most urgent problem seems to be line interruptions and this situation applies to all medium speed (600-2400 bauds) data transmission systems. The main object of this test was to get data on the over-all performance and feasibility of practical data transmission. The total number of messages tested was 1 169 476, i.e. about 100 million bits.

3. *Equipment and mode of operation*

3.1 *General*

Equipment of an electronic airline space reservation system was used for generating, receiving and checking test messages. The individual messages transmitted were composed of 19 five-element characters, i.e. 85 bits. The fifth bit in each character was a vertical parity bit. The last character in each message was used for a longitudinal parity check.

The choice of five element characters has no particular influence on the test of data transmission performance, each message being one sequence of 85 elements.

. . . . . .

3.2 *Modulator and demodulator*

The modem was of the phase modulation and demodulation type for a transmission speed of 1000 bauds.

a) *Modulator*

The data received was modulated with a 1500-c/s carrier. This carrier was shifted by 180° at every data signal transition.

The 1500-c/s carrier was taken from a reference signal of 1000 pulses per second.

The reference signal was supplied by the transmitter so as to guarantee a fixed phase relation between the carrier and the data signal received.

A 500-c/s pilot was added to the modulated data signal. The pilot was also taken from the reference signal mentioned above.

The frequency band transmitted was limited by means of a bandpass filter. For this test the transmission level is set to −10 dbm.

b) *Demodulator*

After entering into the demodulator the modulated data signal and the pilot were first passed through an amplitude equalizer and a variable phase equalizer.

The signals then went to a regulated amplifier. For a constant output level the input level of this amplifier may vary between −10 dbm and −30 dbm.

Subsequently, the modulated data signal was separated from the pilot.

There were three demodulators.

Demodulator A produced the data signal, demodulator B provided the carrier, and demodulator C a 1000-c/s signal.

In demodulator A the modulated data signal was demodulated by the carrier and this resulted in the data signal. In demodulator B the modulated data signal was demodulated by the data signal, which supplied the carrier.

The data signal may be distorted as a result of the characteristic distortion of the telephone channel and any spurious signals.

This is why the demodulator A output signal was regenerated by an integrating and sampling circuit.

In demodulator C the pilot signal was demodulated by the carrier obtained with demodulator B. Demodulator C produced a 1000-c/s signal which served as a phase reference signal.

The sampling pulses for regenerating the data signal were taken from the 1000-c/s signal by means of a differential network.

4. *Results of tests*

4.1 *Message-error rate*

Out of 1 169 476 transmitted messages, a total of 173 messages were received incorrectly (or lost). This corresponds to an average error rate of $p_m = 14.7 \times 10^{-5}$. In a system using error detection and repetition of faulty messages, this figure very closely approaches an average repetition frequency of one message in 6800 messages. There was, however, a considerable variation of the message-error rate in the course of the test period.

More details about this distribution are given in point 5.

4.2 *Characteristics of the telephone circuit*

During the test period of six weeks, line characteristics were measured several times.

a) *General*

The white noise level never exceeded −35 dbm, but was most of the time −40 dbm or less. Impulsive noise was recognized infrequently: crosstalk attenuation (on the looped duplex circuit) was −29 db.

When listening on the line, signalling tones and speech could sometimes be heard.

b) *Frequency departure*

The frequency departure between Hilversum and Paris, and vice versa, was found to be about 0.3 c/s.

No frequency departure was found in loop measurements, for instance, Paris-Hilversum-Paris.

c) *Delay*

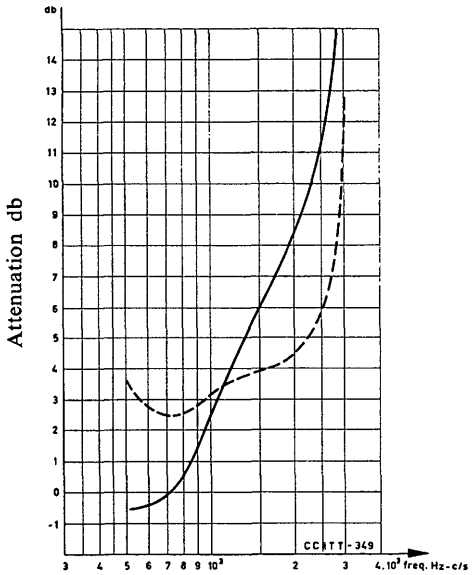
The total delay was 14.8 ms, i.e. 1.4 ms for the modem equipment, 2.2 ms for the variable phase equalizer, 11.2 ms for the line.

d) *Attenuation characteristics*

See Figure 2.

e) *Phase characteristics*

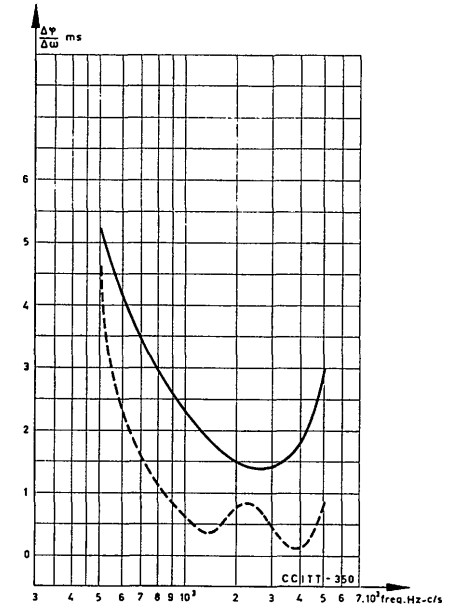
For the characteristics, see Figure 3.



----- Loop Paris – Hilversum – Paris with amplitude equalization

——— Loop Paris – Hilversum – Paris without amplitude equalization

FIGURE 2



----- Loop Paris – Hilversum – Paris with amplitude equalization and with phase equalization

——— Loop Paris – Hilversum – Paris without amplitude equalization

FIGURE 3

4.3 *Modem performance*

4.3.1 *Noise measurements*

In the course of special measurement periods white noise was added to the circuit noise. The total noise level, i.e. circuit noise plus the white noise from a noise generator, was measured with respect to the signal + circuit noise (the influence of the circuit noise is very small).

Messages were sent and checks were made for faulty messages under conditions of added white noise.

The added white noise level was increased from −15 db to −4 db. Up to this level all messages transmitted were received correctly.

Further increase of the added white noise showed the following results :

noise −4 db	20 messages	no error
noise −3 db	20 messages	2 errors
noise −2 db	20 messages	3 errors
noise −1 db	30 messages	11 errors
noise −0 db	30 messages	30 errors

4.3.2 *Measurements on the input sensitivity of the demodulator*

An attenuator was inserted between the modulator and the line in order to reduce the line level. Because the attenuator was located at the input of the circuit, the reduction of the level resulted in a deterioration of the signal-to-noise ratio. The measuring results are somewhat below the results for a normal circuit.

−20 dbm	10 messages	no error
−27 dbm	10 messages	no error
−30 dbm	10 messages	no error
−33 dbm	10 messages	no error
−35 dbm	10 messages	10 errors

5. *Distribution and causes of errors*

The total test period lasted from 31 March to 27 May 1961. During this period open-line conditions lasting *several hours* occurred seven times. These conditions were indicated by means of an open-line detector, which is a standard feature of the modem used and recorded on a pen recorder.

No tests were made during these prolonged open-line conditions. A distinction can be made between three observation periods.

Period 1 (31 March-10 April)

During this period tests were made only at night and on public holidays.

The total number of messages transmitted was 165 803. In 9 messages errors were found, i.e. one in 18 422 messages.

This corresponds to an average message error rate of $p_m = 5.4 \times 10^{-5}$. Compared with later tests which were performed during office hours or for 24 hours a day these results show a much lower message-error rate.

In this period the cause of the errors was not yet analysed.

Period 2 (20 April-9 May)

182 395 messages were transmitted in this period.

In 92 messages errors were found, i.e. one in 1982. This corresponds to an average error rate of $p_m = 50 \times 10^{-5}$.

One reason was found for this increase with respect to the first period.

The telephone circuit was changed twice without notice.

The attenuation and phase equalization was not matched therefore to the new telephone circuit for some time until this condition was known.

There still remained an increased error rate after proper matching of the equalization for which no reason could be found.

Period 3 (9 May-27 May)

In this period the most complete tests could be made, almost without interruption. Moreover, two recorders were used to correlate faulty messages to open-line conditions and sudden level variations. The number of messages transmitted was 821 278. In 72 messages errors were detected, i.e. one in 11 407 messages. This corresponds to an average message-error rate of $p_m = 8.8 \times 10^{-5}$. The distribution of the message-error rate over this period is shown in Figure 5.

In this graph further indications are given of the extent to which open-line conditions and sudden level variations are responsible for errors. For a certain number of faulty messages no recorded reasons could be found. They may have been caused by noise and very short line breaks which were not recorded.

Figure 4 represents a message-error distribution which was derived in the following way:

A number of observations (during test period 3, only working-days were chosen) was selected in such a way that the number of messages transmitted over 24 hours was constant per unit of time. The number of errors per chosen unit of time (half-an-hour) has been plotted in Figure 4.

It is shown quite clearly that most errors occurred between 07.30 and 12.10 hrs. and between 14.00 and 16.30 hrs. There is also a peak in the evening.

This distribution provides the same picture both for open-line faults and for level variations.

These faults may be due to peak hours in telephone traffic and maintenance work on the lines and the transmission equipment.

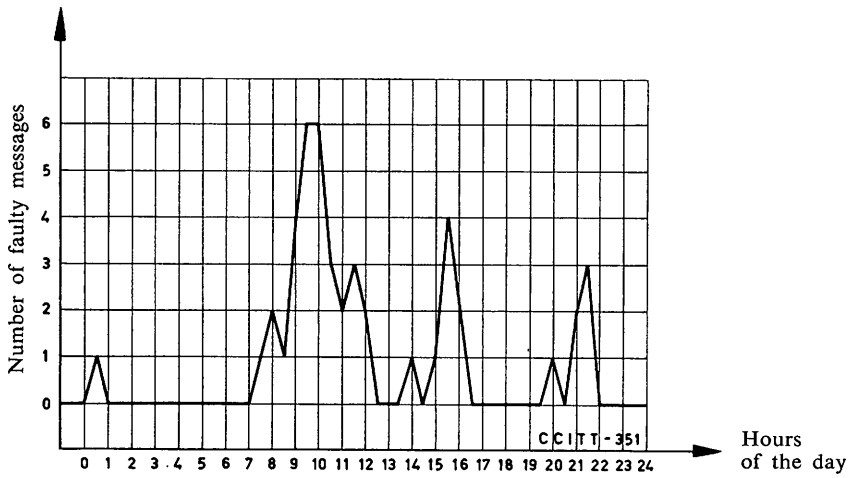


FIGURE 4. — **Typical distribution of erroneous messages during working-days**

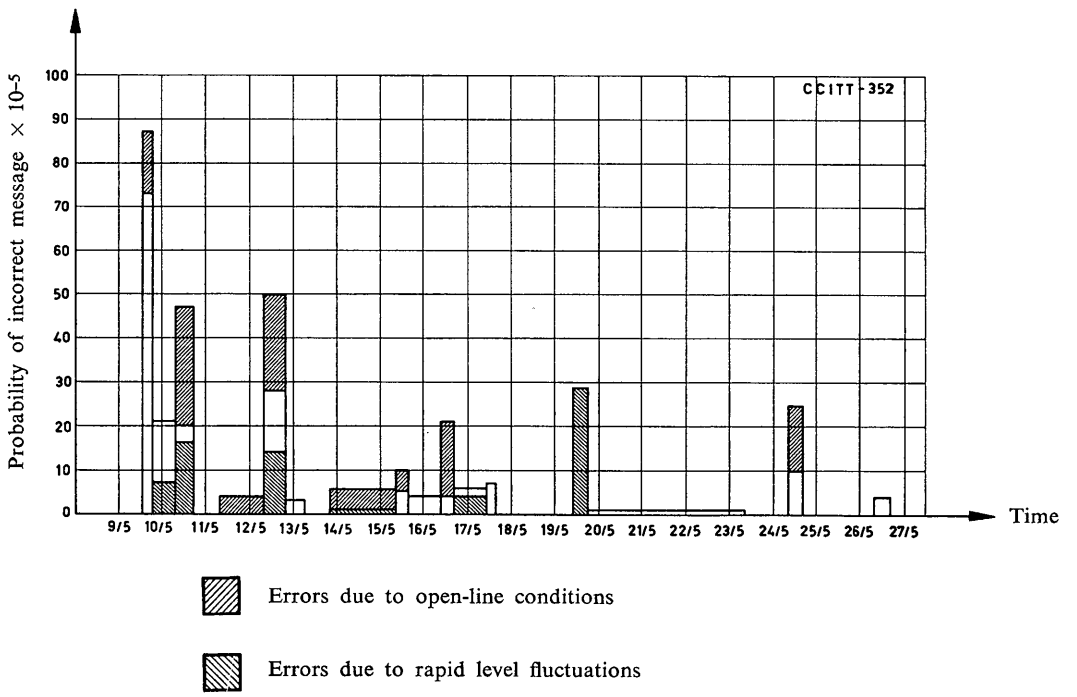


FIGURE 5

(Suppl. 25)

6. *Conclusions*

6.1 Open-line conditions are one of the most frequent causes of transmission errors.

6.2 During 48 days of transmission tests on international leased telephone circuits, seven prolonged open-line conditions occurred. These vary from about one hour to several hours and in one case a longer sequence of open-line conditions (of about one minute per case) was encountered.

It has to be stated, however, that no special arrangements were organized during the reported test period for quick replacement of circuits which showed prolonged open-line conditions.

6.3 It was noticed that the efficiency of data transmission is very high during periods without prolonged open-line conditions (on an average one message lost or containing errors in 6800 correct messages of 85 bits).

Taking account, however, of open-line conditions exceeding one minute, for example, the test has shown that these conditions would have caused serious trouble to data transmission facilities, especially if such transmission channels are used in automatically switched "on line" systems, unless means for automatic re-routing are incorporated.

We propose, therefore, that Special Study Group A should consider the following :

a) Methods and means should be studied in order to reduce prolonged and short open-line conditions to such an extent that a balance of errors caused by other sources of interference could be achieved.

b) Modem equipment should have means for detecting and signalling open-line conditions as a standard facility.

Supplement No. 26

FRENCH ADMINISTRATION — *(Extract from contribution COM Sp. A/No. 11 — August 1961)*

RESULTS OF DATA TRANSMISSION TESTS ON TELEPHONE CIRCUITS IN THE GENERAL SWITCHED NETWORK

I. *Choice and characteristics of the typical connections*

The rule was to use only fully automatic connections and not to make tests on connections established wholly or in part by manual switching.

It was decided to disconnect the meter for all trunk tests, thereby eliminating the regular sending of metering advices, which would momentarily interrupt the transmission of data.

Table I briefly outlines the set-up of the connections that were chosen.

It should be noted that more complex circuit layouts may occur in an automatic call between two provincial subscribers, but it was not possible to consider such cases.

TABLE I

Test	Connection	Switching network (input)	Trunk circuit (including any transit equipment)	Switched network (output)
1	Paris-Marseille	Rotary	800	Rotary
2	Paris-Lille	Rotary	800	Type R 6
3	Lille-Paris	Type R 6	800	Rotary
4	Paris-Lille	Rotary	800 (radio-relay link)	Type R 6
5	Paris-Beauvais	Rotary	800	Type crossbar
6	Beauvais-Paris	Type crossbar	800	Rotary
7	Paris-Saint-Quentin	Rotary	200	Type R 6
8	Saint-Quentin-Paris	Type R 6	200	Rotary
9	Paris-Beauvais	Rotary	200	Type crossbar
10	Beauvais-Paris	Type crossbar	200	Rotary
11	Paris-Albi	Rotary	800 + CT (R 6) + 800	Type crossbar
12	Paris-Montauban	Rotary	800 + CT (R 6) + 200	Type R 6
13	Paris-Mazamet	Rotary	800 + CT (R 6) + 200	Type crossbar
14	Paris direct exchange to exchange circuit			
15	Paris circuit with one transit exchange			
16	Paris circuit with two transit exchanges			

Type R 6 : R 6, R 6N, R 6N1, R 6N2, L 43, SRCT
Crossbar type : CP 400 Pentaconta
800 : HF circuit on symmetrical or coaxial cable (except test 4) belonging to a group circuit
200 : two-wire LF circuit amplified
CT : transit exchange

The data receiver was always situated in Paris, which corresponded to the strictest transmission set-up, since the Paris automatic network, more extensive and complex than those of the provincial towns, is crossed at low level.

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All the tests were made during weekdays and normal working hours.

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IV.3 *Statistical study of errors*

Apart from the customary countings (transmitting blocks, false blocks, wrong digits, etc.) the exact position of the errors was recorded on magnetic tape with the apparatus. This information was then transferred to perforated cards and machine-processed.

Each test was the subject of a separate study. No mention is made here of individual results in such or such a test, but a number of characteristic curves, valid for the tests as a whole, are given. Strictly speaking, these curves are not average curves but "median" curves dividing the

plan into two regions, each region containing half the characteristic curves of each test. In certain cases, extreme curves are shown.

As a help in understanding what is to follow, some definitions are given and the meaning of certain terms is explained.

Interval between errors. — Number of correct binary digits included between two consecutive wrong digits.

Interval between false blocks. — Number of blocks without errors included between two consecutive wrong blocks.

Groups of errors. — Set of errors, not necessarily consecutive, preceded and followed by a set of correct digits at least equal to n (during the statistical treatment n was taken as 10).

Length of a group of errors. — Total number of correct or wrong digits making up the group of errors ; this starts with the first and ends with the last wrong digit.

Figures 8 and 9 show the distribution of the intervals between errors. Figure 8 relates to tests actually carried out with equipment C which, as stated, comprised a coding system by change of condition. Figure 9 relates to a hypothetical transmission system, similar in every respect to the one tested but using a coding system by condition (phases 0 and π correspond, in this case, to digits 0 and 1).

In a system coding the information by changes of condition, a simple error on the sample signals is shown by two consecutive wrong binary digits. It is this well-known property of this type of coding which explains the very different shape of the curves in Figures 8 and 9 for low values of x.

If the errors appeared according to the Poisson process the curves would be straight lines starting from the point (0.100%), the slope being a function of the average rate of errors. The steep fall of the curve at low values of x shows the existence of groups of errors. The relationship between this portion of the curve and the straight line which prolongs it, situated around $x = 6$, gives an approximate idea of the average length of the groups of errors.

Figure 10 shows how the intervals between wrong blocks are distributed.

The non-linearity of the curve for low values of x proves the existence of grouped batches of errors. The chance that a block transmitted following a wrong block will itself contain errors is higher than the average rate of errors on the block.

Figure 11 gives some statistics on the length of the groups of errors. It can be seen that, for the series of tests made with Equipment C, 12% of the groups of errors were of a length above two ; 88% related to isolated errors on the signal sample.

Figure 12 completes the probabilist model of the group of errors. It shows that within a group of errors, the probability of a binary digit being marked by error is slightly less than 0.5, regardless of the position of the digit within the group.

V. *Conclusions*

The rates of error on the blocks are of the order of a few per thousand, and occasionally a few per cent or a few per ten thousand.

In actual service, these values, on a long-distance circuit, should be increased from 5 to 6 $\times$ 10^{-3} to allow for disturbance caused by the pre-metering tone (1 s every 180 s).

The French Administration considers that these results are encouraging, opening up the possibility of using the switched telephone network to serve the inter-connection network between subscribers to a general data transmission service.

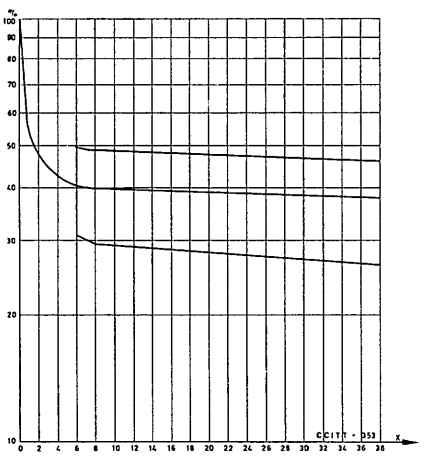


FIGURE 8

Percentage of intervals between errors comprising a number of binary digits equal to or higher than that shown on the abscissae

Median curve and extreme curves — Coding by condition change

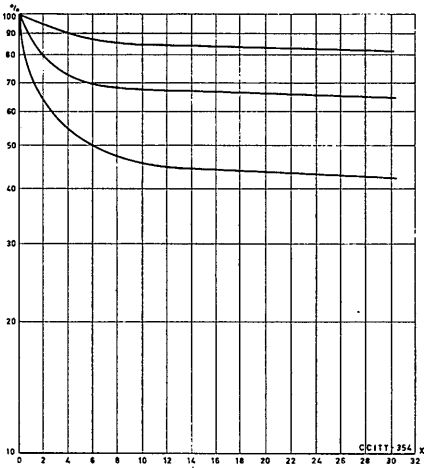


FIGURE 9

Percentage of intervals between errors comprising a number of binary digits equal to or higher than that shown on the abscissae

Median curve and extreme curves — Coding by condition

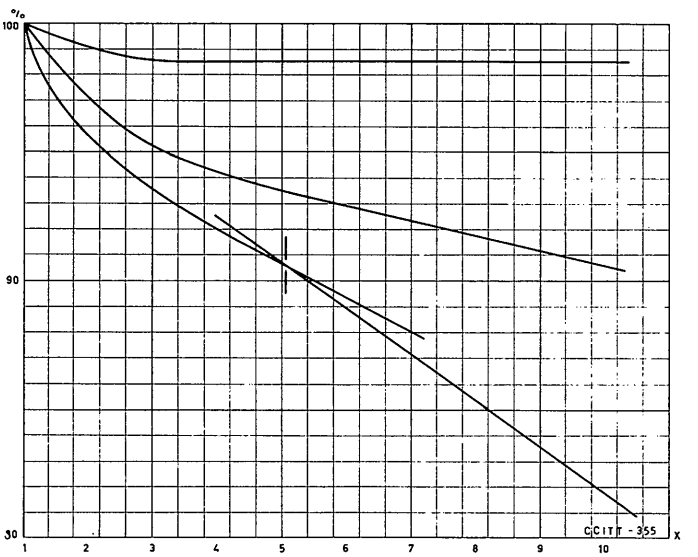


FIGURE 10. — **Percentage of intervals between wrong blocks comprising a number of blocks equal to or higher than that shown on the abscissae**

Median curve and extreme curves

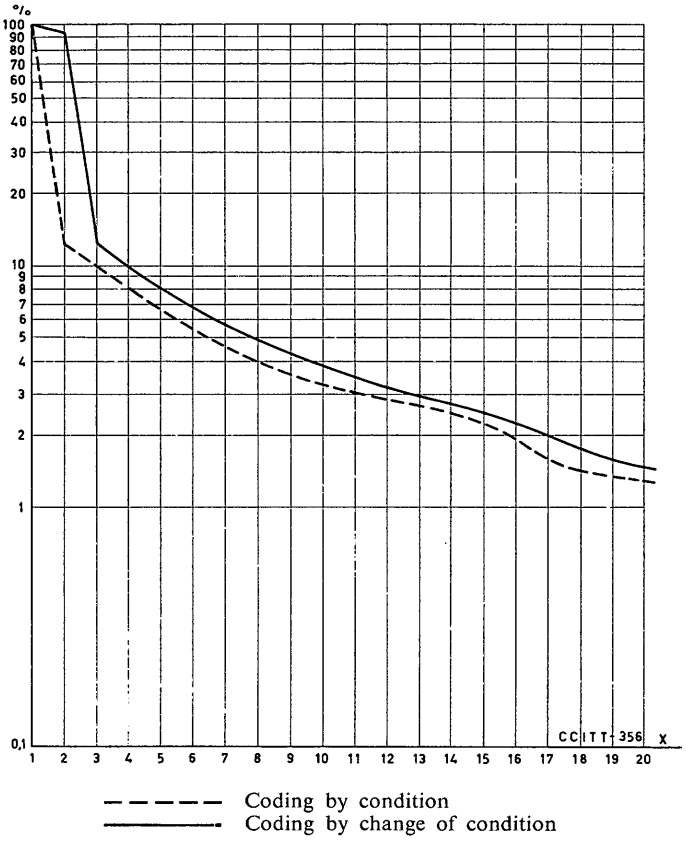


FIGURE 11. — **Percentage of groups of errors whose length is at least equal to the number shown on the abscissae**

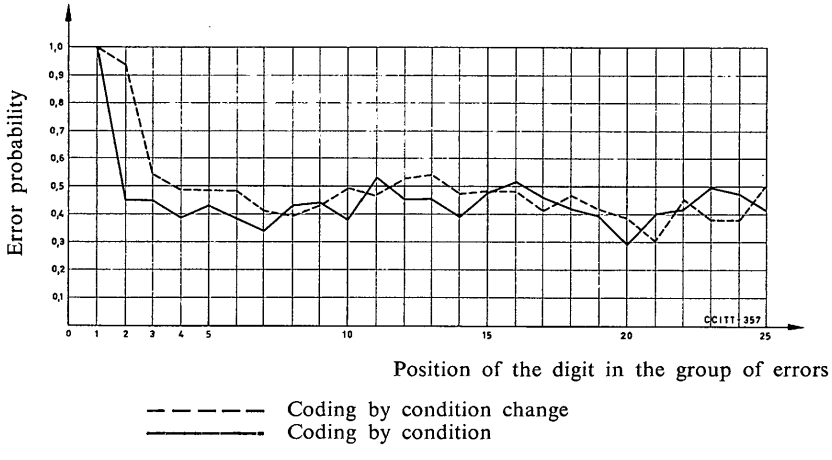


FIGURE 12. — **Probability of encountering a wrong binary digit as a function of the position of the digit in the group of errors**

Supplement No. 27

FRANCE — *(Extract from contribution COM Sp. A/No. 79 — August 1963)*

RESULTS OF DATA TRANSMISSION TESTS ON THE SWITCHED TELEPHONE NETWORK AT THE DATA-SIGNALLING RATES HIGHER THAN 1200 BITS/S

1. *Introduction*

The over-all aim of the series was to examine the advantages of the switched telephone network in high-speed data transmission (information flow : 1800, 2000, 2400 and 3000 bits/s).

. . . . . .

TABLE I

No.	Type of connection
1 E or R	Trunk : 800
2	Trunk : 800 + TC + 800
3	Trunk : 800 + TC + 800 + TC + 800
4	Trunk : 200 + TC + 800 + TC + 800
5 E or R	Local : direct exchange-to-exchange connection
6 E or R	Local : connection including 1 LTC

TC : Transit centre
LTC : Local transit centre
800 : HF circuit belonging to a group connection
200 : Two-wire amplified AF circuit
E : ROTARY line finder at the sending end
R : ROTARY line finder at the receiving end

Note. — The various 800 circuits used for the tests were all set up on 2.6-9.4-mm coaxial pairs ; the 200 circuits were coil-loaded (H 88) circuits set up on symmetrical pairs 0.9 mm in diameter.

The telephone circuits used were obtained by fully automatic means. In trunk calls, the charge meters were disconnected to eliminate the sending of the preavis and charging pulses.

Table I shows briefly the composition of the six types of circuit used.

Circuits 1 and 5 are examples of a relatively simple type of routing, which nevertheless is frequent in trunk calls and in local networks through several automatic switching equipments. Circuits 2, 3, 4 and 6, on the other hand, are more complex. Circuits 3 and 4, which were not included in the 1960/1961 programme, may be found in the French network when two subscribers in relatively distant provincial towns have to be connected.

The table does not show what kind of switching system obtains at the outgoing and incoming ends. However, the tests did show—and this is a point which will be explicitly dealt with in each table of results—that the point of contact of the Rotary line finder had a bad influence. The rate of error in the blocks varied considerably (multiplying factor between 4 and 40), depending on whether the Rotary line finder was at the receiving or at the sending end.

For the most part, the modems studied used a double-sideband two-condition modulation process (phase or frequency). It was thus possible to attain high modulation rates only at the expense of systematic correction of the phase and amplitude characteristics of the telephone circuits. For the equalizers, the manufacturers used typical curves showing the group delay distortion of a carrier channel and an H88 coil-loaded section. The policy followed in each case for the construction and adjustment of the equalizers will be described fully in connection with each test series.

The report on the tests also gives the results obtained with two modems of different design (vestigial side-band two-condition phase modulation, four-phase modulation), which included a non-adjustable compromise equalizer in the receiving equipment.

2. *Tests made with Ph 1 equipments*

2.1 *Main characteristics of signal-conversion equipments*

Three different modems were used successively :

1. a classical two-condition phase-modulation modem (double-sideband) ;
2. a two-condition phase-modulation modem (vestigial sideband) ;
3. a four-phase modem.

2.1.1 *Ph 1 — two phases — two sidebands*

Modulation rates — 1200, 1500, 1800, 2000 and 2400 bauds.

Modulation and coding — Two-condition phase-modulation (asynchronous), transition or condition coding.

Transmitter — The data signal, after it has been shaped, modulates a carrier frequency at 1800 c/s. The modulation products outside the telephone band are eliminated by a low-pass filter.

There is no coherence between the modulation rate and the carrier frequency.

Transmission level — 0 Nm at subscriber line output.

Receiver — At the input of the receiver there are a band-pass filter and an amplifier with automatic gain control which can reduce the reception area from 4.6 N to 3.5 dN.

The signals received are then demodulated synchronously in the usual way by means of a restored carrier (with a phase uncertainty equal to π).

When the data are condition-coded, the phase indetermination of the restored carrier is removed by sending a series of binary digits 1, lasting 25 ms, at the beginning of the transmission.

2.1.2 *Ph 1 — two phases — vestigial sideband (experimental modem)*

Modulation rates — 2400, 3000, 3600, 4000 and 4200 bauds.

Modulation and coding — Two-condition phase-modulation (asynchronous) ; vestigial side-band ; transition-coding.

Transmitter — The data signal, after it has been shaped, modulates a carrier frequency at 2500 or 2800 c/s (this figure depends on the modulation rate ; the value used in each test is given in Table III). A band-pass filter eliminates the undesirable modulation products and most of the upper sideband (vestigial sideband).

Transmission level — 0 Nm.

Receiver — The receiver is the same as the one described in 2.1.1. The carrier is restored by means of a narrow frequency band around 2500 or 2800 c/s, the amplitude of which is suitably equalized (this equalizing network must have a characteristic which is inverse to that used in transmission to "cut off" the vestigial sideband).

2.1.3 *Ph 1 — four phases (experimental modem)*

Information flow — 2000, 2400, 3000, 4000 and 4200 bits/s.

Modulation — Quaternary phase-modulation (asynchronous).

Principle of operation — It may be considered that transmission is carried out by associating two-condition channels of the kind described in 2.1.1, using two carriers in quadrature at 1800 c/s. On each channel, the data is transition-coded.

To switch the binary digits to another channel in transmission, one must have the terminal equipment time-signal, which is transmitted over the appropriate wire of the interface.

At reception, a clock should be incorporated in the signal-conversion equipment, in order to recompose the message from both series of binary digits. In this experimental demodulator, the clock in the receiving terminal equipment, synchronized by the transitions of the two two-condition channels, is used.

2.2 *Main characteristics of terminal equipments*

Transmission. A generator supplies joined blocks of 511 binary digits. The blocks are formed by a shift register looped to nine positions.

Reception. The blocks received are compared with blocks formed by a generator identical to the transmission generator and suitably synchronized.

The binary errors are detected in any position within the block, and the block is considered erroneous if it contains one or more binary errors.

When there is a *line interruption*, the counting of errors stops automatically at the end of the second block affected by the interruption and resumes only when the connection is re-established. During these tests, however, errors were recorded on magnetic tape to cut out most of the errors in blocks and digits caused by interruptions. Only the first erroneous block and the first two erroneous digits were kept.

Both the transmitting and receiving terminal equipments have a crystal-controlled clock.

2.3 *Attenuation and phase equalization*

Equalization is made at the receiving end, either in the audio-frequency band or in the basic-band.

One of the following three phase-equalizers may be used :

— In the basic-band, equalizer A with seven positions,

— in the audio-frequency band, equalizer B with seven positions (see Figure 1), or equalizer C with four continuous and independent settings.

During the tests, adjustments were made to improve the data signal waveform in the basic-band (eye pattern).

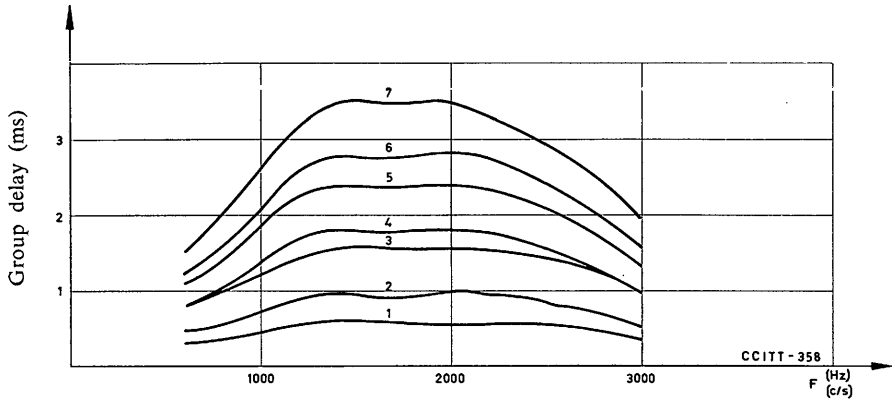


FIGURE 1. — **Frequency-delay characteristics of equalizer B in the equipment Ph1**

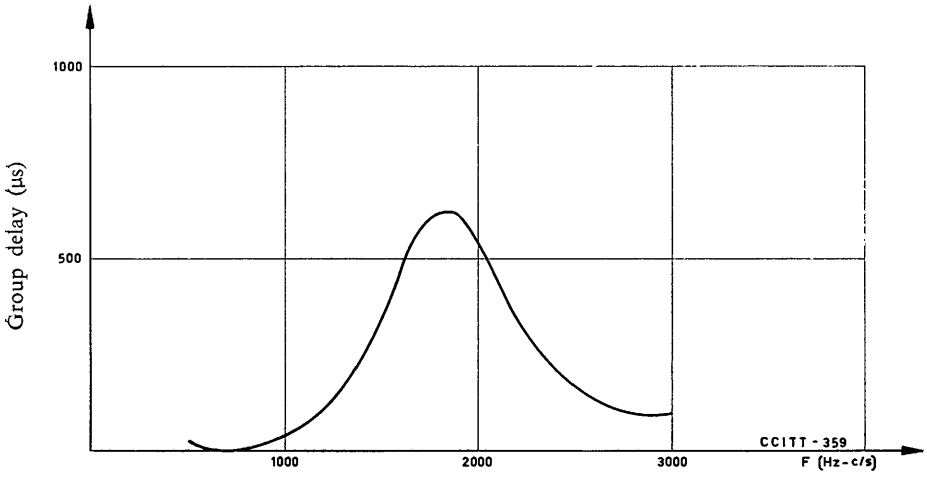


FIGURE 2. — **Frequency-delay characteristic of the compromise equalizer in the equipment Ph1**

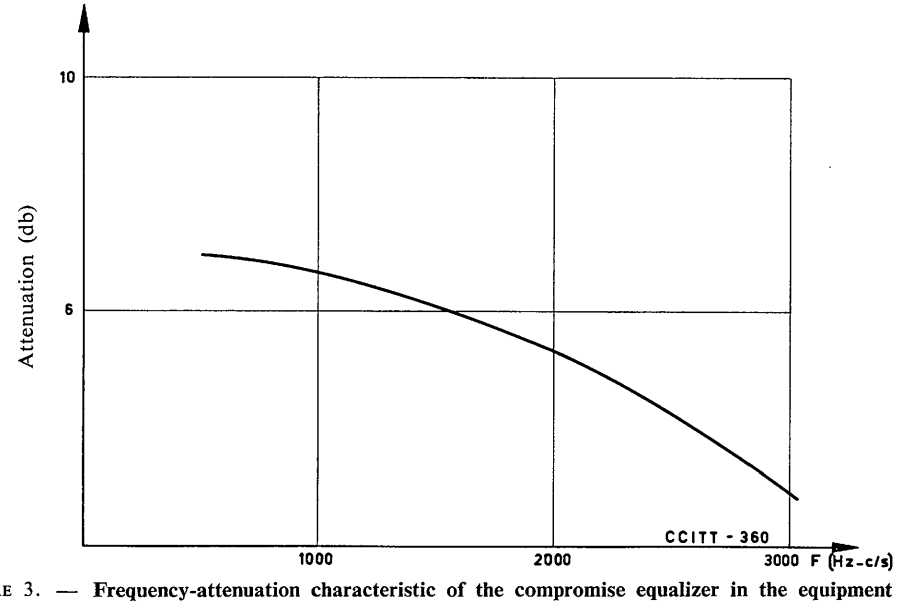


FIGURE 3. — **Frequency-attenuation characteristic of the compromise equalizer in the equipment Ph1**

For reception, vestigial sideband two-phase modems and four-phase modems have a compromise equalizer which is inserted permanently in the audio-frequency band and is not adjustable. This equalizer partially compensates for amplitude and phase-distortion in the circuit (see Figures 2 and 3).

2.4 *Test results*

The results of the three test series are summed up in Tables II, III and IV.

The continuous adjustment equalizer C gave good results on connections of types 1, 5 and 6. On the other hand, it seems difficult to make an appreciable improvement in the quality of trunk connections that are of a relatively complex structure. Figure 4 gives, by way of example, the shape of a frequency-attenuation curve of a type 3 connection. The many oscillations in this curve are due to the multiple echoes from the two (two-wire) transit centres and from the terminals. The frequency-group delay curve was not plotted but it must present the same characteristics. It is certain that the gradual introduction of four-wire switching will enable a great improvement to be made in these connections.

Type 4 connections were not used during the tests of the two experimental modems, since their characteristics are better than those of type 3 connections.

Table IV confirms that an information flow of 2000 bits/s can be attained with a four-phase modem on telephone connections of the switched network, if use is made of a non-adjustable compromise equalizer.

TABLE II

Equipment Ph 1 — two phases — two sidebands

Type of connection	Modulation rate (bauds)	Number of binary digits transmitted (in millions)	Error rate on blocks of 511 digits ($\times 10^3$)	Error rate on binary digits ($\times 10^5$)	Mean number of erroneous digits per erroneous block	Phase equalizer	Coding by transition (T) or condition (E)
1 E	1800	99.3	7.93	8.52	5.5	—	T
	2000	61.8	83.3	43	2.6	A (4 ou 5)	T
	2400	44.4	22.3	14.7	3.4	C	T
2	2000	118.8	85.5	26.7	1.6	A (3 ou 4)	E
3	1200	89.5	7.12	3.84	2.8	—	T
	1500	81.3	17.7	10.6	3	A (4 ou 6)	T
	1800	140.8	62.2	31.6	2.6	A (4 ou 5)	T
4	1500	67.9	6.72	4.83	3.7	—	73% E, 27% T
	2000	68.7	18.2	10.7	3	A (3 ou 4)	T
5 E	1800	26.5	0.28	0.16	2.8	—	T
	2000	78.3	0.75	0.35	2.1	—	T
	2400	47.5	0.23	0.13	3	C	T
6 R	1800	67.7	36.8	16.8	2.3	—	T
	1800	68.8	7.03	3.9	2.8	B (3 ou 4)	T
	2400	30.7	7.1	4.02	2.9	C	T

TABLE III

Equipment Ph 1 — two phases — vestigial sideband

Type of connection	Modulation rate (bauds)	Carrier frequency (c/s)	Number of binary digits transmitted (in millions)	Error rate on blocks of 511 digits ($\times 10^3$)	Error rate on binary digits ($\times 10^5$)	Mean number of erroneous digits per erroneous block	Additional phase equalizer
1 E	3000	2500	51.05	14.5	13.5	4.7	—
	3600	2800	33.9	48.7	23.2	2.4	—
	4000	2800	87.3	22.1	24.9	5.75	C
2	2400	2500	40.5	4.06	2.70	3.4	—
5 E	3000	2800	63.9	0.8	0.34	2.2	—
	4200	2800	76.6	1.27	0.7	2.75	C
6 R	3000	2800	25.7	16.3	23.4	7.3	—
	4200	2800	44.4	21.4	88.1	20.9	C

Note. — This equipment includes at the receiving end a non-adjustable (attenuation and phase) compromise equalizer.

TABLE IV

Equipment Ph 1 — four phases

Type of connection	Information flow (in bits/s)	Number of binary digits transmitted (in millions)	Error rate on blocks of 511 digits ($\times 10^3$)	Error rate on binary digits ($\times 10^5$)	Mean number of erroneous digits per erroneous block	Additional phase equalizer
1 E	3000	77.9	7.52	12	8.2	B (3)
	4000	57.7	25.6	16.6	3.3	C
2	2000	7.15	2.22	5.6	12.9	—
	2400	71.3	33.9	23.9	3.6	—
3	2000	84	20	12.7	3.25	—
	2400	24.5	35.2	103	15	B (7 ou 8)
5 E	3000	64.4	0.43	0.31	3.7	—
	4200	61.3	1.06	1.27	6.1	C
6 R	3000	51.1	25.6	87.8	17.5	—
	4200	87.4	28.5	117	21	C

Note. — This equipment includes at the receiving end a non-adjustable (attenuation and phase) compromise equalizer.

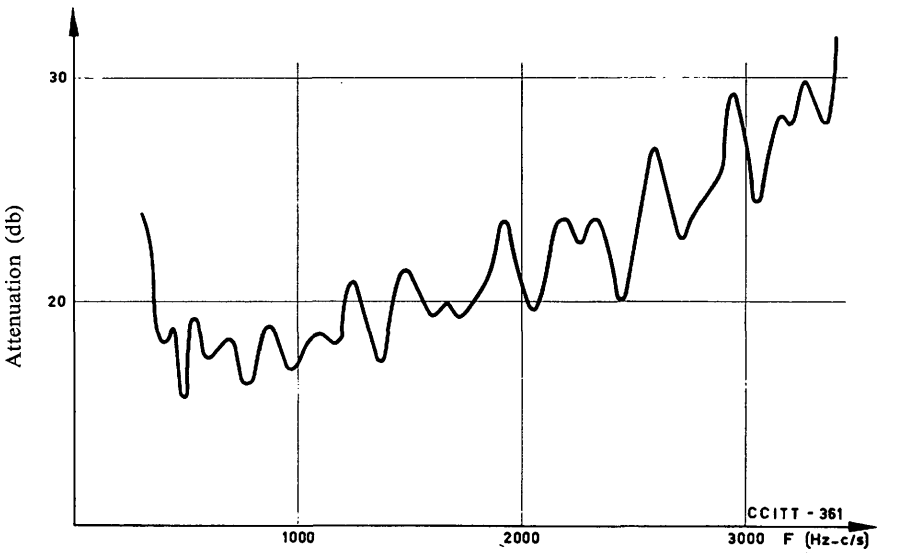


FIGURE 4. — **Frequency-attenuation characteristic of a type 3 connection**

3. *Tests carried out with Ph 2 equipment*

3.1 *Main characteristics of signal conversion equipment* (experimental modem)

Modulation rates — 1200, 1800, 2400 bauds.

Modulation and coding — two-condition phase modulation (asynchronous), coding by transition.

Transmitter — The data signal, after being shaped, modulates a carrier of 1800 c/s. The modulation products situated outside the telephone band are eliminated by a low-pass filter. There is no coherence between the modulation rate and the frequency of the carrier-wave.

Transmission level — 0 Nm at the subscriber line output.

Receiver — The data signal is filtered, amplified, and peak-clipped. The receiver adds modulus 2 of this peak-clipped signal to the carrier wave which has been recomposed in the form of a square signal (synchronous demodulation effected with logical circuits). At demodulator output, the signal is again filtered and amplified.

Transcoding — The binary digits are condition-coded on the interface wires. The necessary transcodings are effected in the signal converters (sending and receiving).

Clock — For the purpose of transcoding at the receiving end, a clock with a simple fly-wheel effect is used in the signal converter. A switching device enables the rhythm recuperation filter to be chosen as a function of the modulation rate.

3.2 *Main characteristics of terminal equipment*

Transmission. A generator supplies joined blocks of 127 binary digits. The blocks are formed by a shift register looped to seven positions.

They comprise :

— a start-of-block sequence of seven binary digits, and

— a message of 120 binary digits.

Reception. The received blocks are compared with blocks formed by a generator of similar characteristics that has been appropriately synchronized.

The logical rules for the detection of errors are summed up in the following table :

Correct start-of-block n_2 binary errors in the message	1 erroneous block	n_2 erroneous binary digits
n_1 binary errors in the start-of-block correct message	1 erroneous block	0 erroneous binary digit
n_1 binary errors in the start-of-block n_2 binary errors in the message	1 erroneous block	0 erroneous binary digit

These rules lead to the adoption of the following definitions for the block error rate and the digit error rate :

(1) Block error rate $= \dfrac{\text{Number of erroneous blocks}}{\text{Number of blocks received}}$

(2) Digit error rate $= \dfrac{\text{Number of erroneous digits}}{120 \times \text{number of blocks received with correct start-of-block}}$

Moreover, it can be considered that the effective length of a block is *127 binary digits* (considering the rate of errors as being proportional to the length of the block).

If there is a *line interruption*, the logical rules for error detection apply for the first four blocks affected by the interruption. The counting of errors is then stopped until the connection is re-established.

It should be noted that terminal reception equipment includes a crystal-controlled clock which is quite distinct from that which is fitted into the signal conversion equipment, which is frequency and phase-synchronized on the transcoded data signal.

3.3 *Phase and attenuation equalization*

This equalization is made at the receiving end in the audio-frequency band.

1200 bauds : no correction.

1800 and 2400 bauds : We have :

— a fixed-characteristic amplitude equalizer giving an attenuation variation of 2 N in the frequency band 600-3000 c/s ;

— three fixed-characteristic dephasing networks (Figure 5) : network A which equalizes the group delay distortion of a carrier channel in the frequency band 600-3200 c/s, network B which effects the same equalization in the 800-3000 c/s band, and network C which corrects group delay distortion of two carrier channels in the 800-3000 c/s band ;

— two five-position dephasing networks (networks D and E), one for the lower part and the other for the upper part of the telephone band (Figure 6).

The amplitude equalizer was used permanently during the tests. Networks A, B or C were inserted and networks D and E adjusted as a function of the composition of the link and of the degree of isochronous distortion in the received signals. Measurement of the degree of distortion played an overriding role in the choice of phase equalization for all the tests at 2400 bauds.

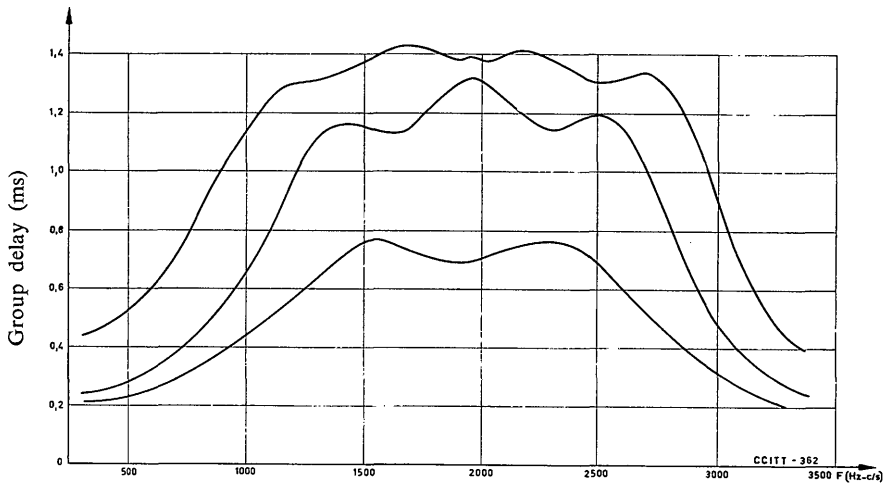


FIGURE 5. — **Frequency-delay characteristics of equalizers A, B and C in the equipment Ph2**

TABLE V

Equipment Ph 2

Type of connection	Modulation rate (bauds)	Number of binary digits transmitted (in millions)	Error rate on blocks of 127 digits ($\times 10^3$)	Error rate on binary digits ($\times 10^5$)	Mean number of erroneous digits per erroneous block	Degree of isochronous distortion (%)	Phase equalizer
1 R	1200	10.8	0.14	0.30	2.5	40	—
	1800	91.7	0.92	1.63	2.3	30 à 35	A
	2400	24.6	1.24	2.56	2.6	40	B + D (1) + E (4)
2	1200	18.7	2.14	5.85	3.3	30	—
	1800	121	2.31	6.75	3.8	30	A + B
	2400	24.2	0.56	1.42	3.1	40	B + C + D (5) + E (5)
3	1200	12.5	1.3	2.87	3	45	—
	1800	75.7	1.96	5.98	4	35 à 45	A + C
	2400	24.6	4.38	10.7	3	55	A + B + C + D (5) + E (5)
4	1200	14.7	2.7	8.53	4.1	40	—
	1800	85	9.41	32.9	3.3	50 à 60	A + B + D (1) + E (5)
5 E	1200	12.4	0.16	1	8.8	30	—
	1800	85	0.16	0.28	2.2	35	—
	2400	23.6	0.57	1.17	2.6	55	—
6 E	1200	12.7	0.41	0.87	2.9	25	—
	1800	97	0.64	1.09	2.2	35 à 40	—
	2400	23.3	0.60	1.24	2.7	55	—

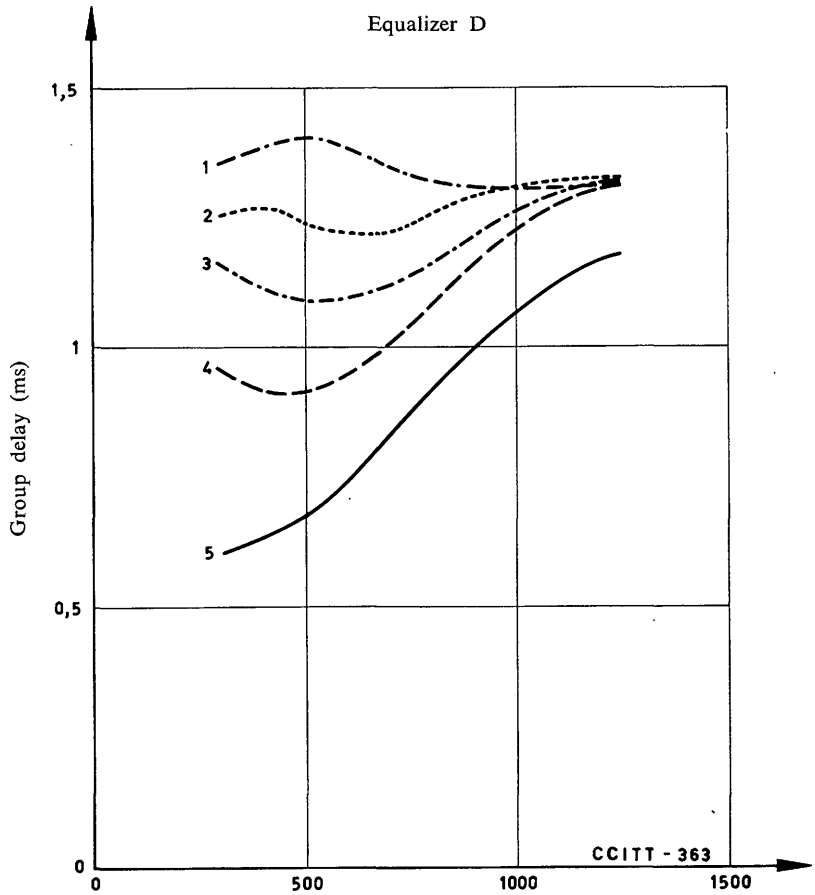


FIGURE 6a. — **Frequency-delay characteristics of equalizer D in the equipment Ph2**

3.4 *Test results*

The results of these tests are summed up in Table V.

It should be underlined that type 2, 3 and 4 connections were affected by sudden level variations of between 1 and 2 N. This fault was traced to a line finder in the transit centre but it could not be eliminated. Moreover, type 4 connections had a repeater with a cut-off frequency of 2600 c/s in the audio-frequency circuit.

These remarks also apply to the tests described under 4, 5 and 6.

4. *Tests with F 1 equipment*

4.1 *Chief characteristics of signal converters*

Modulation rates — 1200 and 1800 bauds.

Modulation and coding — Two-condition frequency modulation (asynchronous type), coding by transition [1].

[1] According to the equivalence 0 = transition, 1 = absence of transition. From this point of view, the new draft of Recommendation V.1 should be amended so as to define transition coding independently of the modulation procedure used.

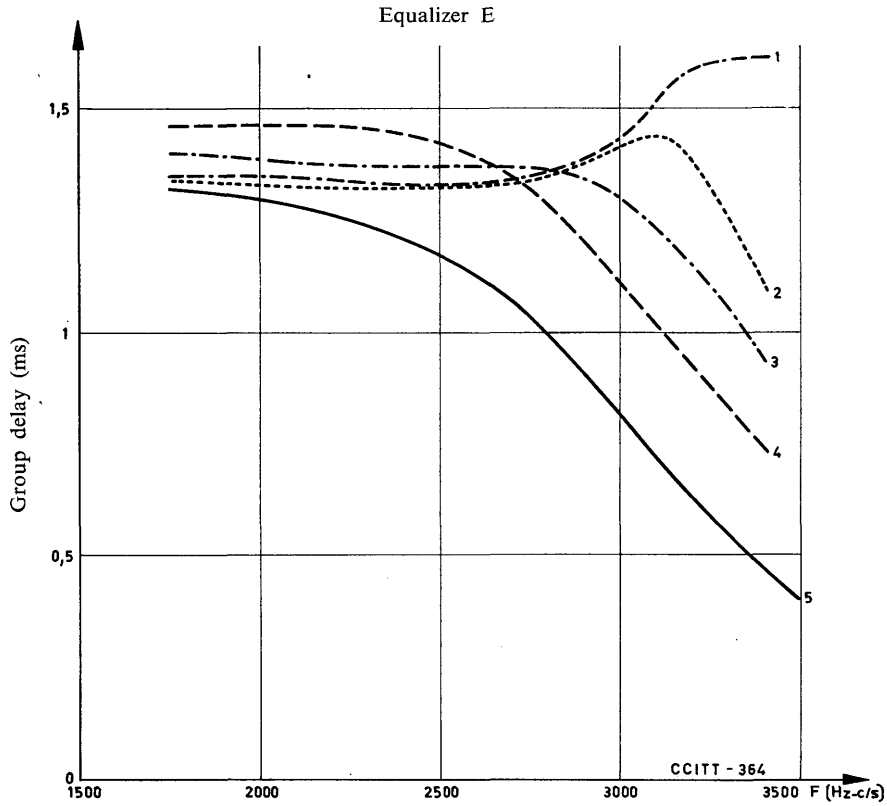


FIGURE 6 b. — **Frequency-delay characteristics of equalizer E in the equipment Ph2**

Characteristic frequencies :

1700 ± 400 c/s at 1200 bauds,
1900 ± 600 c/s at 1800 bauds.

Transmitter — The data-signal frequency modulates an oscillator of mean frequency 9.2 kc/s. The modulated signal is then translated into the audio-frequency band.

Transmission level — 0 Nm at the subscriber line output.

Receiver — The nominal receiving level is −3.5 Nm and the receiver operating margin ± 1.5 N on each side of the nominal level. An adjustable attenuator enables the high reception levels to be adjusted to the nominal value of −3.5 Nm.

The modulated signals are translated, after filtering, and discriminated in the classical way around 9.2 kc/s.

Transcodings — The binary digits are condition-coded on the interface wires. The necessary transcodings are made in the signal converters (sending and receiving).

Clock — For the purposes of transcoding at reception, a crystal-controlled clock is incorporated in the signal converter. It is frequency and phase-synchronized on the data signal supplied by the modulator.

4.2 *Chief characteristics of terminal equipment*

Transmission. A character generator supplies joined blocks of 307 binary digits. The blocks consist of 32 characters of eight binary digits separated by service digits. No two consecutive blocks are identical and the structure of a block is not a matter of change : in particular, the characters are emitted following the progression order of a binary meter :

0000 0000
1000 0000
0100 0000
1100 0000 etc.

Schematically, it can be said that the block is composed of :

— a start-of-block sequence of 19 binary digits, and
— a message of 288 binary digits.

Reception. The received blocks are compared with blocks formed by a character generator that is similar to the sending generator and appropriately synchronized.

The logical rules for the detection of errors are summed up in the following table :

Correct start-of-block n_2 binary errors in the message	1 erroneous block
n_1 binary errors in the start-of-block correct message	0 erroneous block
n_1 binary errors in the start-of-block n_2 binary errors in the message	0 erroneous block

The erroneous blocks are detected by means of the erroneous characters. This means that the number of erroneous digits in the message is not known.

The error rate on blocks is defined as follows :

$$(3)\quad \text{Block error rate} = \frac{\text{Number of erroneous blocks}}{\text{Number of blocks received with correct start-of-block}}$$

In these conditions, if it is wished to apply the rule whereby the error rate is proportional to the length of the block, the effective length of blocks must be considered to be *288 binary digits*.

When there is a *line interruption*, the first block is counted as an erroneous block (provided that the interruption comes after the start-of-block sequence). The following are automatically cut off.

4.3 *Attenuation and phase equalization*

Attenuation is pre-equalized at the output and the other equalizations are effected in the audio-frequency band at the receiving end.

The attenuation equalizer and pre-equalizer can be adjusted by contact studs and each comprises five positions. Most of the tests mentioned in the present document gave rise to optimum equalization on the basis of the frequency/attenuation curve of the connection. A compromise equalization could be defined through subsequent tests (pre-equalizer and equalizer in position 3, or 1.2 N from 800 to 2800 c/s). A phase equalizer, adjustable by contact studs, enables group

delay distortion of 1, 2 or 3 carrier channels and of 10, 20 or 30 coil-loaded H 88 sections to be equalized in an approximate manner. Adjustment was made as a function of the make-up of the connection. However, during tests with type 2, 3 and 4 trunk connections, measurements were made using the phase equalizer relating to only one carrier channel, so as to assess the characteristics and performances of a compromise equalizer.

4.4 *Test results*

The test results are summed up in Table VI. By comparing the results obtained on the 6R and 6E connections, it is possible to note the overriding influence of the call direction on the error rate (multiplying factor of the order of 4 to 40). As mentioned above, it is of some significance if the contact point of the Rotary line finder is crossed by a high-level or low-level signal.

TABLE VI

Equipment F 1

Type of connection	Modulation rate (bauds)	Number of binary digits transmitted (in millions)	Error rate on blocks of 288 digits ($\times 10^8$)	Attenuation equalization	Phase equalization
1 R	1800	97.4	1.65	optimum	1 CP channel
2	1800	91.8	5.4	optimum	1 CP channel
	1800	13.5	1.59	optimum	2 CP channels
3	1800	61.7	4.56	optimum	1 CP channel
	1800	29.2	0.86	optimum	3 CP channels
4	1200	29.7	15.5	compromise	1 CP channel
	1200	17.1	40	compromise	1 CP channel + 10 H 88 sections[1]
5 R	1800	79.8	2.45	optimum	—
6 E	1200	7.2	1.77	compromise	—
	1200	7.2	0.11	optimum	—
	1800	51.5	0.76	optimum	—
6 R	1200	5.4	7.5	compromise	—
	1200	7.2	4.75	optimum	—
	1800	46.1	18.5	optimum	—

[1] The AF circuit comprised 30 sections and the repeater at the end of this circuit had a cut-off frequency of 2600 c/s.

Moreover, the comparative tests carried out on type 2, 3 and 4 connections show that it is possible to effect only partial equalization of the delay distortion introduced by several carrier channels connected in tandem.

It is interesting to mention that other tests were carried out with terminal equipment including an error correction system involving the retransmission of erroneous blocks. In this case, a frequency multiplex return channel was used to request repetition of blocks. The characteristics of this return channel were as follows :

— frequency modulation ;

— characteristic frequencies : 420 ± 30 c/s ;

— transmission level at subscriber line output : −1.6 Nm.

The tests covered connections of types 1, 2, 3, 4 and 6. The results were satisfactory, but it was noted that the despatch of a waiting signal tone (400 c/s, cut up 20 times a second) on neighbouring local lines could interfere with the operation of the return channel.

5. *Tests with F 2 equipment*

5.1 *Chief characteristics of signal conversion equipment* (experimental modem)

Modulation rate — 1800 bauds.

Modulation and coding — Two-condition frequency modulation (asynchronous), coding by condition.

Frequency characteristics — 1700 ± 450 c/s.

Transmitter — The data-signal frequency modulates an oscillator of mean frequency 10.3 kc/s. The modulated signal is then translated into the audio-frequency band.

Transmission level — 0 Nm at subscriber line output.

Receiver — After filtering and amplification, the modulated signals are translated and discriminated in the classical way around 10.3 kc/s.

5.2 *Chief characteristics of terminal equipment*

Transmission. A generator supplies joined blocks of 128 binary digits. They are of a random structure which is predetermined by the wiring.

They comprise :

— a start sequence of four binary digits, and

— a message of 124 binary digits.

Reception. The received blocks are compared with blocks formed by a generator similar to the transmitting generator and appropriately synchronized.

The logical rules for the detection of errors are summed up in the following table :

Correct start-of-block n_2 binary errors in the message	1 erroneous block	n_2 erroneous binary digits
n_1 binary errors in the start-of-block correct message	0 erroneous block	0 erroneous binary digit
n_1 binary errors in the start-of-block n_2 binary errors in the message	0 erroneous block	0 erroneous binary digit

The definitions of error rates on the blocks and digits are similar to those indicated above (definitions (3) and (2) where factor 120 is replaced by 124).

Furthermore, the effective length of the block must be considered to be *124 binary digits* (if the error rate is regarded as being proportional to the length of the block).

When there is a *line interruption*, the first block is counted as faulty (provided that the interruption is after the start-of-block sequence). The following blocks are cut off automatically.

The sending and receiving terminal equipment includes crystal-controlled clocks.

5.3 *Phase and attenuation equalization*

The equalizations are made at reception in the audio-frequency band.

Use is made of an attenuation equalizer having seven positions and a phase equalizer having two cells and four independent settings (abscissa and ordinate of the maximum frequency-delay curve of each cell).

During the tests, the amplitude equalizer was used in position 2 (by way of exception in position 7 on type 4 connections). The optimum adjustment of the phase equalizer for one, two or three carrier channels had been determined in the laboratory. These adjustments have been confirmed in operating conditions by observing either the digit-error rate or the waveform of the data signals, in the basic-band, after demodulation.

5.4 *Test results*

The test results are summed up in Table VII.

TABLE VII

Equipment F 2

Type of connection	Modulation rate (bauds)	Number of binary digits transmitted (in millions)	Error rate on blocks of 124 digits ($\times 10^8$)	Error rate on binary digits ($\times 10^5$)	Mean number of erroneous digits per erroneous block	Phase equalizer (abscissa and ordinate of the maxima)
1 R	1800	97.3	2.5	4	2	0.8 ms — 1750 c/s 0.6 ms — 2240 c/s
2	1800	111	3.9	25.7	6.5	0.8 ms — 1570 c/s 1.2 ms — 2240 c/s
3	1800	89.6	17.8	157	11	0.8 ms — 1750 c/s 1.5 ms — 2240 c/s
4	1800	121	5.3	25.8	6.6	1.2 ms — 1750 c/s 0.6 ms — 2240 c/s
5 E	1800	80	0.21	0.76	4.4	—
6 E	1800	86.5	0.94	1.7	2.3	—

6. *Tests with F 3 equipment*

6.1 *Chief characteristics of signal converters* (experimental modem)

Modulation rate — 1800 and 2400 bauds.

Modulation and coding — Two-condition frequency modulation (of the synchronous type), coding by condition.

Characteristic frequencies :

1350 ± 450 c/s at 1800 bauds ;
1800 ± 600 c/s at 2400 bauds.

Transmitter — The modulator consists of logical circuits which make each unit interval correspond to a period or half-period of square signal of characteristic frequency.

Transmission level — 0 Nm at subscriber line output.

Receiver — Two demodulators were tested successively :

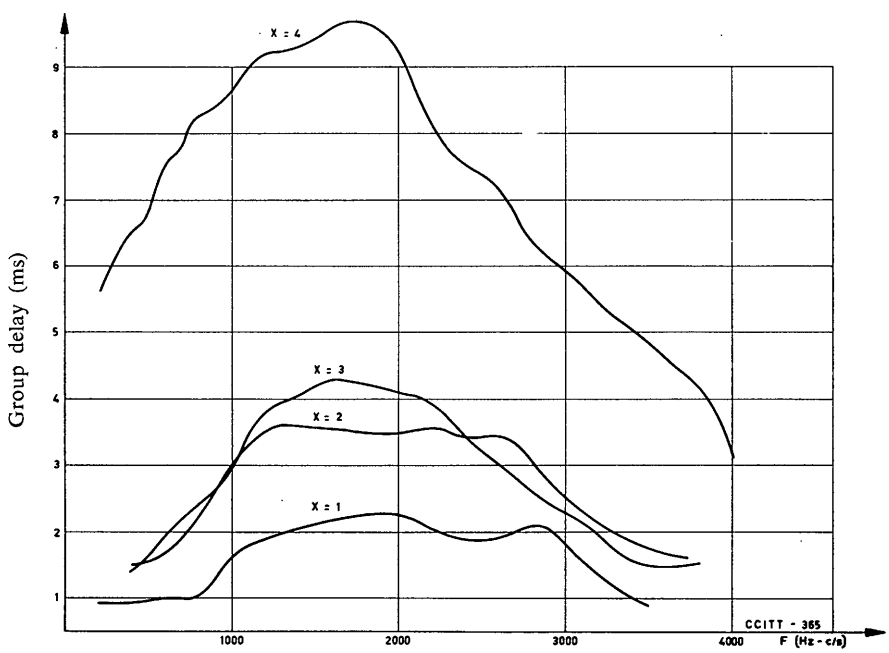
— an L demodulator having logical circuits ;
— a C demodulator of the classical type translating the audio-frequency band and discriminating around 8 kc/s (dynamic at input ± 1.7 N on each side of the nominal level).

6.2 *Chief characteristics of terminal equipment*

Transmission. A character generator forms unjoined blocks of 175 binary digits which comprise :

— 1 start-of-block character of seven binary digits ;
— 24 characters of seven binary digits constituting the message itself.

The structure of this message is not a random one (four characters repeated six times).



(*X* designates the type of connection used)

FIGURE 7. — **Frequency-delay characteristics of equalizers in the equipment F3**

Reception. The received blocks are compared with blocks formed by a generator similar to the transmitting generator and appropriately synchronized. The logical rules for error detection are the same as those defined under 5.2. In this case, the effective length of blocks may be considered to be *168 binary digits*.

6.3 *Phase and attenuation equalization*

The equalizations are made in the audio-frequency band at the receiving end.
The equipment comprises :

— a non-adjustable attenuation equalizer (1.2 N between 600 and 3000 c/s) ;

— an adjustable phase equalizer enabling 1, 2 or 3 carrier channels and 30 H 88 coil-loaded sections to be equalized. Figure 7 indicates the frequency/delay characteristics used during the tests on connections of types 1, 2, 3 and 4.

6.4 *Test results*

The test results are summed up in Table VIII.

TABLE VIII

Equipment F 3

Type of connection	Modulation rate (bauds)	Number of binary digits transmitted (in millions)	Error rate on blocks of 168 digits ($\times 10^3$)	Error rate on binary digits ($\times 10^5$)	Mean number of erroneous digits per erroneous block	Phase equalizer	Demodulator
1 R	1800 2400	82.8 81.7	12.5 9.83	16 17	2.2 2.9	— 1 CP channel	C
2	1800 2400	85.9 38.8	5.7 4.7	11.3 15	3.3 5.2	2 CP channels 2 CP channels	C
3	1800 2400	85.6 60.7	8.5 10.9	54 98	10.3 15	3 CP channels 3 CP channels	C
4	1800	61.4	13.2	61	7.7	2 CP channels + 30 H 88 sections	C
5 E	1800 2400	81.6 51.4	3.18 5.8	3.6 10.2	1.9 3	— —	L
6 E	1800 2400	83.4 89.6	1.36 1.26	2.6 2.1	3.2 2.8	— —	C

7. *Conclusion*

These tests show that data can be transmitted over the switched telephone network at an information flow of more than 1200 bits/s. To reach this objective, consideration may be given either to classical two-condition double sideband modulation procedure or more "sophisticated" procedures, such as two-condition vestigial sideband phase modulation or quaternary phase modulation.

In any case, these high speeds can be obtained only at the cost of systematic phase and attenuation equalization. And from this point of view, it would be desirable to lay down strict rules for the operation of the switched telephone network.

Two cases can be envisaged :

1) The signal conversion equipment at the receiving end is served by staff which operate the terminal equipment. It is to be expected that most of this staff will not have a very deep knowledge of electronics and data transmission. This being so, it does not seem possible to entrust this staff with the task of equalizing the connection, and the only practical solution would be to use a compromise (attenuation and phase) equalizer such as the one described under 2.3 at the receiving end.

2) If technicians are available at the receiving end (computation centre, for example) a different operating method could be adopted. Adjustable equalizers could be incorporated in the signal conversion equipment and this specialized personnel could be asked to make a simple and rapid choice.

One solution would lie in fitting contact-stud phase equalizers to compensate the delay distortion introduced by a variable number of carrier channels connected in tandem and by coil-loaded sections. On the other hand, it is very feasible to design a compromise amplitude equalizer since the composition of local lines — which are essentially responsible for attenuation distortion — is much less variable than that of trunk connections.

Since many trunk systems are homogeneous in structure, it might be considered practical to base the choice on the composition of connections. But apart from the fact that it would be difficult to keep such a directory of the network up to date, there would always be the difficulties inherent in the operation of heterogeneous systems.

It seems, in fact, wiser to base the choice on a rapid measurement of transmission quality, for example :

— measurement of the degree of isochronous distortion of peak-clipped signals ;

— observation with an oscilloscope of the waveform of data signals in the basic-band before the peaks have been clipped (eye pattern).

Subject to standardization of the modulation procedure or procedures, the French Administration considers that the first operating method should be authorized immediately. As regards the second procedure mentioned above, it feels that its adoption should be delayed pending the results of long-term operating tests bringing into play industrial equipment and a large variety of switched telephone connections.

Moreover, it should be mentioned that the tests carried out by the French Administration were not able to throw light on the possible interaction of these transmissions with audio-frequency signalling units, since the HF and AF circuits include signalling channels outside the telephone band. When the modulation rate is more than 1200 bauds, it might be desirable to choose modulation procedures which ensure the symmetry of the instantaneous spectrum with respect to the central frequency of the transmission channel. This symmetry would provide automatic protection against the possibility of audio-frequency signalling units being started by data signals.

Supplement No. 28

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 14 — August 1961)*

MEASURING THE ERROR STATISTICS OF SWITCHED TELEPHONE LINES (800 BAUDS)

The statistics reported herein were derived from measurements obtained with switched lines of the Stuttgart telephone network.

The modulating unit employed was an FM unit designed for a telegraph speed of 800 bauds and operating at a midband frequency 1800 c/s with the deviation ± 300 c/s.

The largest portion of the measurements were carried out in conjunction with an error protection unit that was utilized as the source of the test message. The design of this error protection unit is responsible for block lengths of 9 and 63 bits primarily occurring in the results.

Test set-up

The figures obtained are statistical means, that is, they contain no information as to the position of an individual faulty bit within a block. Faulty zeros and faulty "ones" can be separately counted across the gate circuit used for the comparison.

In most of the measurements, the test message was derived from the error protection unit and consisted of a sequence of identical blocks with 63 bits each. Each block was subdivided in six equal 9-bit words and a word of a different bit sequence, i.e.

010011010
010011010
010011010
010011010
010011010
010011010
101011110

The block contains the same number of bits and pairs of bits 0 and 1.

To detect the faulty blocks, a timing pulse marking the start and stop of a block has to be applied to the gate. This can be done either by marking the block length continuously, independent of error occurrence, or by counting a block starting with the occurrence of an error, thus correlating the block counting with the errors. The correlated method was favoured. It will be seen that it has almost no influence upon the result.

The attenuation of the over-all connection tested and of its sections is shown in *Appendix 2.*

The occurrence of errors is described by statistical mean values or probabilities.

The following quantities were measured :

$$p_E = \frac{\text{number of faulty bits}}{\text{total number of bits}} = \text{element-error rate}$$

$$p_B = \frac{\text{number of faulty blocks}}{\text{total number of blocks}} = \text{blocks-error rate}$$

$$p_{EB} = \frac{\text{mean number of faulty bits in a block}}{\text{total number of bits in a block}}$$

where $p_{EB} = p_E/p_B$

$$b = \frac{\text{number of blocks with } e \text{ faults}}{\text{total number of faulty blocks}}$$

$$g = \frac{\text{number of faulty bits in all blocks with } e \text{ faults}}{\text{total number of faulty bits}}$$

It is a condition that $\sum_{e=1}^{n} b = 1$, $\sum_{e=1}^{n} g = 1$ and $g = \frac{e}{np_{EB}} b$

where

e = number of faulty bits in a block

n = number of bits in a block = block length in bits

It must also be true that the mean number of faulty bits in a faulty block is:

$$\sum_{e=1}^{n} e \cdot b = n \cdot p_{EB} \sum_{e=1}^{n} g = n \cdot p_{EB}$$

which is the mean number of faulty bits per faulty block.

The value of the quantities above defined depends, among other things, on the signal/noise ratio. Hence, to make the measurements comparable with those from other sources, the signal level across the final selector is quoted as a measure of the signal/noise ratio that could not be tested. The level across the receiver input did not seem suitable as it may vary depending on the length of the subscriber line even when the noise conditions are the same. The final selector, contrariwise, is located closer to the noise source unless it even constitutes the noise source; hence the level at this point is a more reliable measure for the signal/noise ratio.

Of the quantities defined, the values of p_E, p_B, p_{EB} and the cumulative distribution of b were directly measured. The distribution of b and g plotted in the appendices were computed from the measurements.

The b and g distributions were presented, by exchanging the parameters, as functions of the block length and of the level. This rearrangement permits of conceiving a family of curves as a measuring unit in which one curve monitors and corrects the rest of the family. For this reason, the number of measurements is indicated not for a single curve, but for a family.

Another possibility to check the measurements is given by the factor np_{EB}, reliably known from direct measurements, that appears when the b curves are converted into g curves.

Test results

The measurements were taken with a transmit level of −2 N. This resulted in a signal level −3.6 N across the final selector (at 1800 c/s) and −5.4 N across the receiver input.

p_B varies in a very wide range between about 1 and 150 while p_{EB} varies at a ratio of less than 1 to 4. The fluctuating traffic is reflected by p_B. Obviously the traffic density has no effect upon p_{EB}; rather, the block-error rate p_B in night hours conspicuously decreases to 1/100 of the daytime value while the number of faulty bits in a block (np_{EB}) remains about constant on the average.

The p_B value at the beginning of a conversation period was extremely high and dropped to the average within one hour.

There were no complete line interruptions exceeding 80 ms (one block length). Shorter interruptions were not observed and are improbable as they would have become noticeable by a spontaneous increase of p_{EB}.

On the other hand, the connections were frequently interrupted by one of the offices in low-traffic periods. It is due to the fact that the maintenance staff, on their rounds during the night, test the selectors in operation as to whether there is a conversation on the line. If not, the selector is put into rest condition and the connection is released. Therefore, it was impossible to carry out tests during night hours without supervision. The two exchange offices concerned were not informed about the experiments in order to deal with real operating conditions.

Appendix 5 contains the distribution curves of the p_{EB} values. Although, owing to the low number of measurements, no smooth distribution curves are obtained, it will be seen that the spread is much lower in a single connection than in many connections. From this it may be concluded that p_{EB} is a suitable quantity to characterize the effect of noise on a line, with respect to the coding arrangement, independent of the daytime. The block-error rate p_B and the element-error rate p_E are unsuitable, therefore, because of their strong variation with the time of the day.

Tests with tape recorder

The purpose of this test series was to find the error distribution of a connection, typical for our tests. From the spread of the measurements it was concluded that a measuring time of about 60 minutes was required to determine, with some reliability, the mean value. To compute a single g distribution curve, at least 10 test points must be available from busy or other daytime hours. This means a measuring time of 28 days. Even if it were possible to maintain a connection for more than a month, the contact fritting appearing in due course shows that this connection would undoubtedly have different values at the beginning and at the end of the test period. Regular new dialling would have the effect that any distribution for a certain block length measured would apply to a discrete connection. Even if the parameters were statistically changed, the result would be a mean value of wide spread or standard deviation that might mar the picture of the disturbance introduced in data transmission by a particular connection.

For these reasons, a wideband recording was made of the typical noise of a line, using a tape recorder. About 88% of the recorded tape, free of any noise, was cut out ; the time of 42 minutes was thus condensed to 5 minutes. This condensed noise tape was joined to form an endless tape that was used as a noise source connected to the transmission system, as shown in Appendix 1 (bottom). Even after several hours of observation no periodicity of the errors detected could be established. A special difficulty inherent in this modified test set-up was to find an unambiguous measure for the signal/noise ratio. This problem was solved by associating the mean p_{EB} found in line measurements with the signal level -3.6 N measured across the final selector. Thus the determination of p_{EB} in tape measurements that yielded great accuracy permitted the association to the signal level across the final selector.

The values for p_E, p_B and p_{EB}, measured by the method described, are plotted vs. signal level across final selector in Appendix 6 for block lengths of $n = 9$ and $n = 63$ bits.

Two scales apply to p_B and p_E. The right-hand scale directly gives the results of tape measurements. The tape was recorded in the morning and cut at the ratio 1 : 8.8. This corresponds to a condensation of the noise at a ratio 1 : 17, referring to the whole day. Hence, the left-hand

scale, expanded by the factor 17, indicates the associated mean values for the full day of a certain connection. The value $p_B = 2.1 \times 10^{-2}$ ($n = 63$) for -3.6 N is well in accordance with the value directly measured on the line.

Only the left-hand scale applies to p_{EB}: since this quantity indicates the element-error rate in faulty blocks, it is not affected by cutting the noise-free section out of the tape. The lowest value possible for p_{EB} is $1/n$, which corresponds to the case where each of the faulty blocks contains not more than one error. This case is approximated at a high signal level in Appendix 6, when only a small number of a sequence of noise pulses exceed the response threshold in the receiver. The dependence on the level becomes still clearer when the curve b ($e = 1, n = 63$) is contemplated that indicates the probability of the occurrence of blocks containing one error only. With the increasing level, this curve steeply rises towards 1.

Appendix 7 shows the dependence of p_B and p_{EB} on the block length n for a constant element-error rate $p_E = 1.45 \times 10^{-3}$. These curves primarily serve the purpose of comparison of the test method using a correlated time scale ($^k p_{EB}$ and $^k p_B$) where the beginning of a block is made to coincide with the first disturbance with the second test method (p_{EB}, p_B) where the beginning of a block is independent of the disturbance. It will be seen that the difference is trifling. Hence, the correlated-scale measurements given below are in no way invalid.

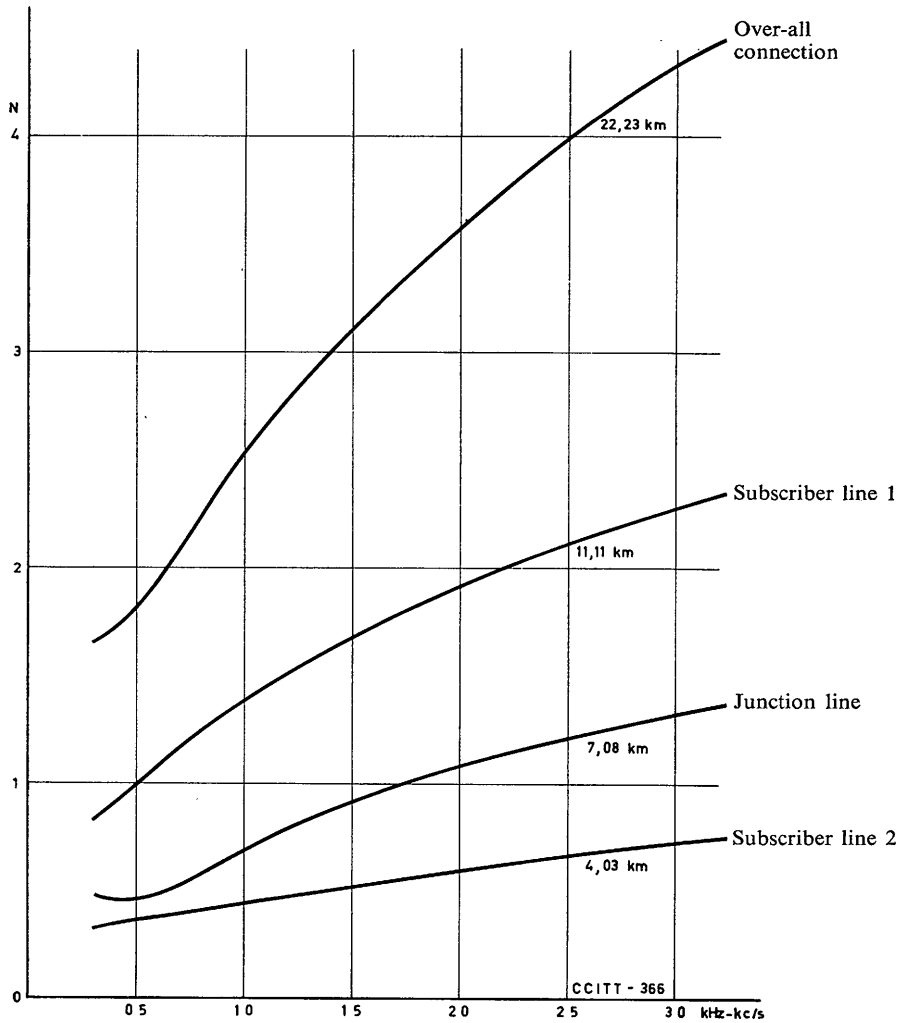
Appendices 8 to 11 show, in various presentations, the distribution curves of errors in blocks $n = 9$ and $n = 63$. The distribution curves b (e) giving the share of faulty blocks with e errors in the total number of faulty blocks, greatly depend on the level. The same level dependence will be found in the distribution curve g (e) in Appendix 12 representing the share of faulty bits in blocks with e errors in the total number of faulty bits.

Finally, Appendix 13 shows the distribution curves g (e) for various block lengths at a constant level.

The error distributions measured are representative for the conditions in the Stuttgart local telephone network only. Nevertheless, they suggest useful conclusions with respect to a generally applicable error-detection code. Since the distribution curves will be different in each connection (depending on the signal level or on the length of the subscriber line), the coding method should factor in all possible curves. In other words, it should not be based on one discrete curve, but on the whole family of curves to be expected.

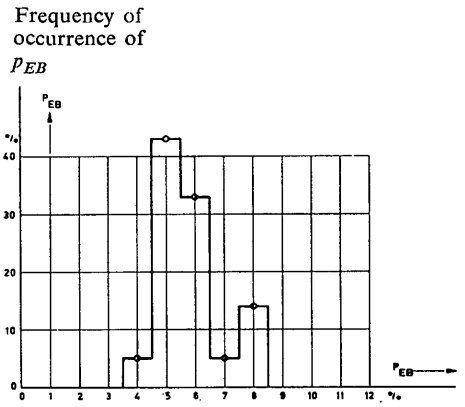
. . . . . .

APPENDIX 2



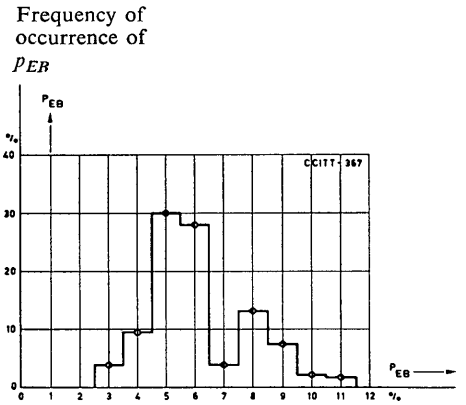
Attenuation characteristic of the lines investigated

APPENDIX 5



Duration of measurements : 11.4 hours
Number of blocks transmitted : 0.52×10^6

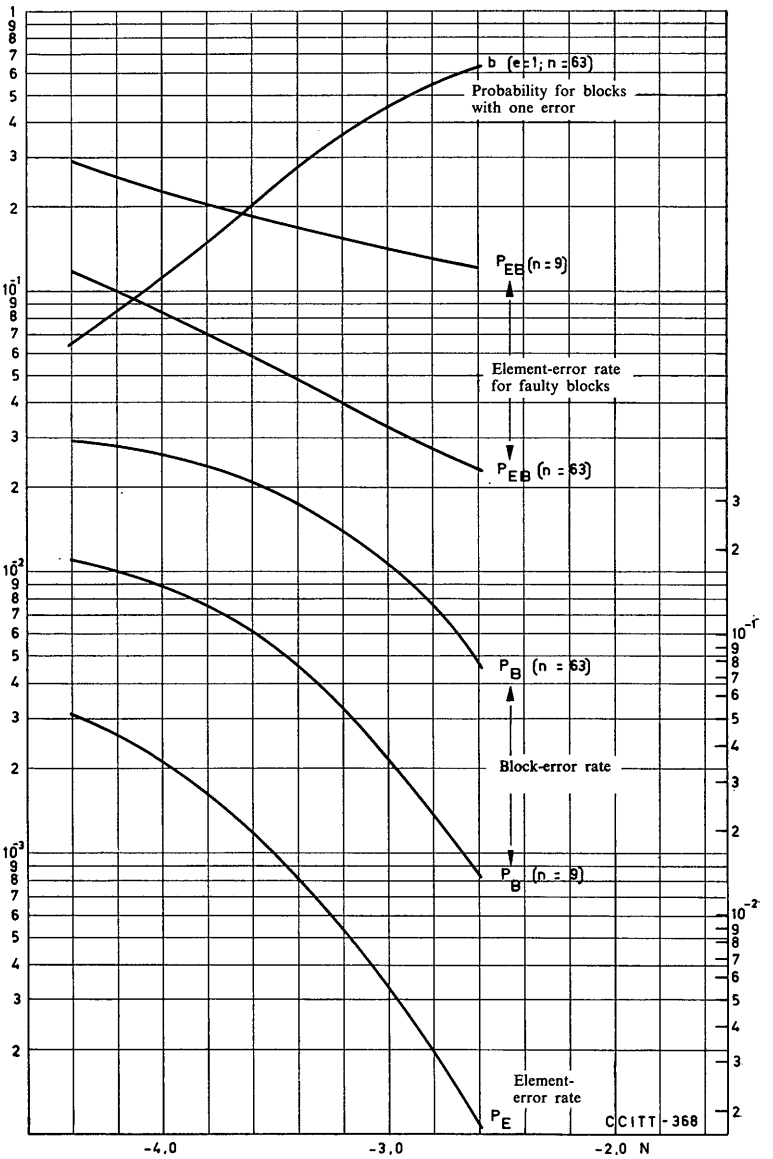
Distribution of the values for the element-error rate for faulty blocks, p_{EB}, measured on one connection



Duration of measurements : 52.5 hours
Number of blocks transmitted : 2.4×10^6

Distribution of the variation for the element-error rate for faulty blocks, p_{EB}, measured on 23 connections

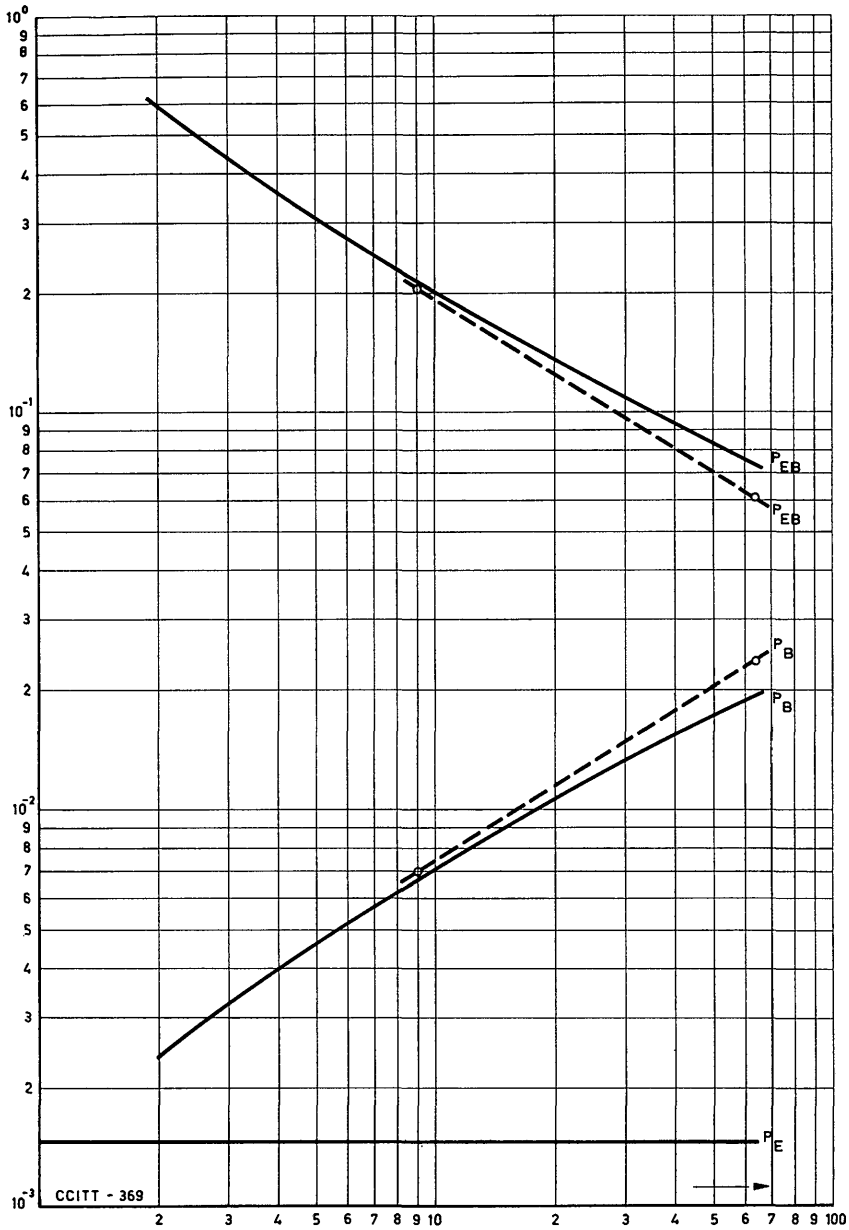
APPENDIX 6



Signal level at final selector

Tape record compressed 1 : 17 referred to the whole day
Number of elements transmitted : 18.7×10^6

APPENDIX 7

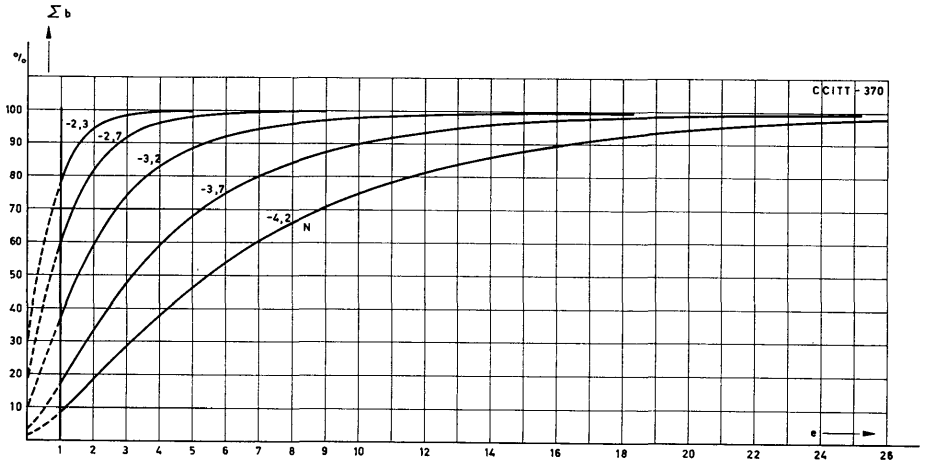


Measured with recorded noise, signal level corresponding to −3.6 N at final selector.

— — — — — Independent time scale
——————— Correlated time scale
Number of elements transmitted : 41.8×10^6

Variation of block-error rate p_B and of element-error rate for faulty blocks, p_{EB}, with block length n

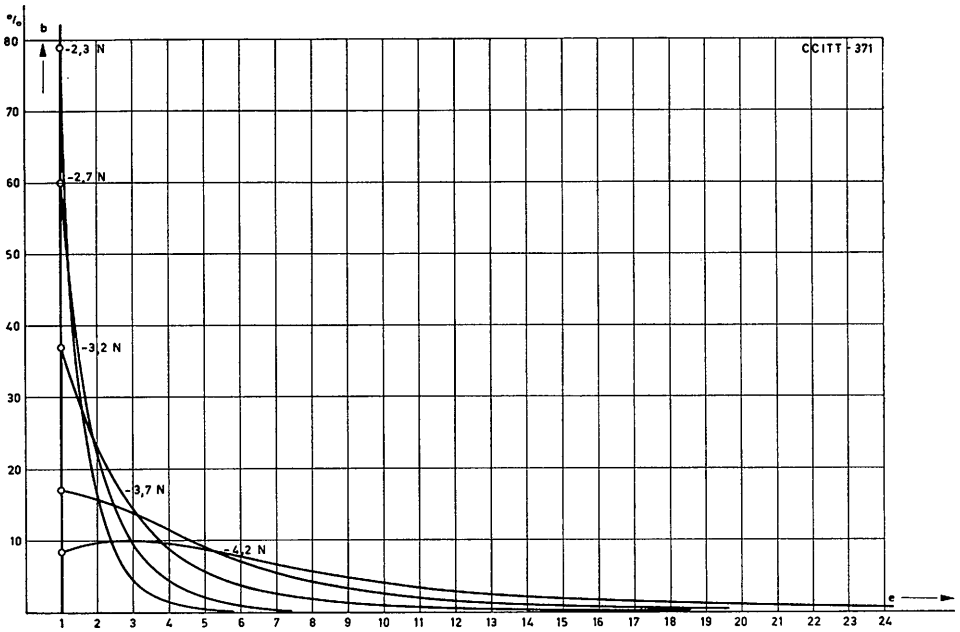
APPENDIX 8



Level at final selector: $N = -4.2$ N
Parameter: signal level at final selector
Block length: $n = 63$ bits
Number of bits transmitted: 9.4×10^6

Cumulative distribution Σb for the occurrence of blocks with e errors or less

APPENDIX 9



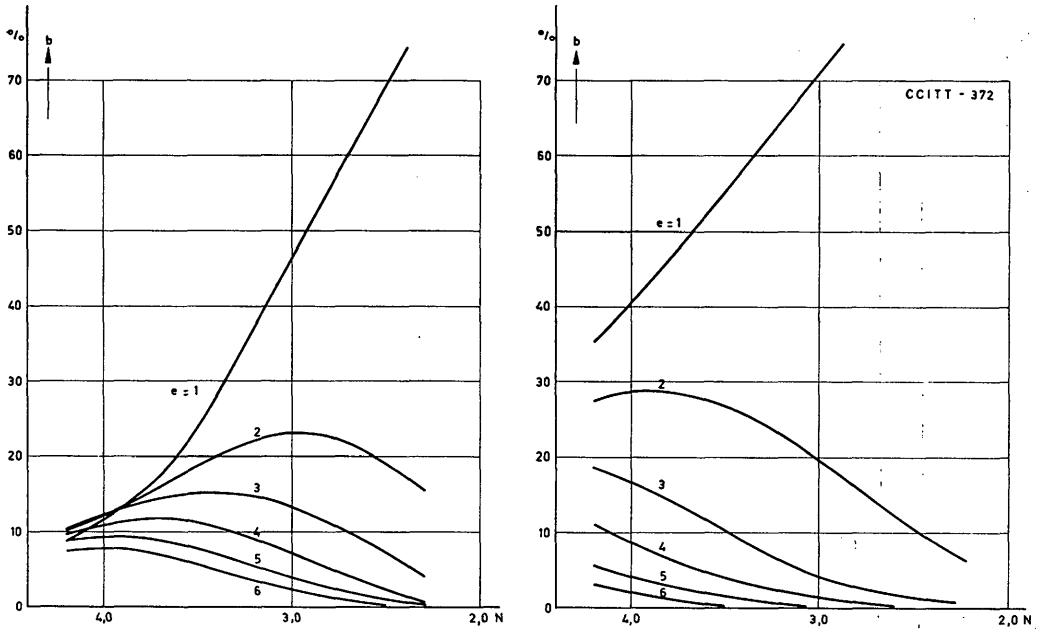
Parameter : signal level at final selector
Block length : $n = 63$ bits
Number of bits transmitted : 9.4×10^6

Level at final selector N	$^{k}p_{EB}$	p_{EB}	$n^{k}p_{EB}$	np_{EB}
−2.3	0.020	0.019	1.28	1.20
−2.7	0.029	0.025	1.73	1.57
−3.2	0.047	0.039	2.99	2.43
−3.7	0.078	0.063	4.88	3.96
−4.2	0.122	0.100	7.72	6.28

$n = 63$

Distribution b for the occurrence of blocks with e errors

APPENDIX 10



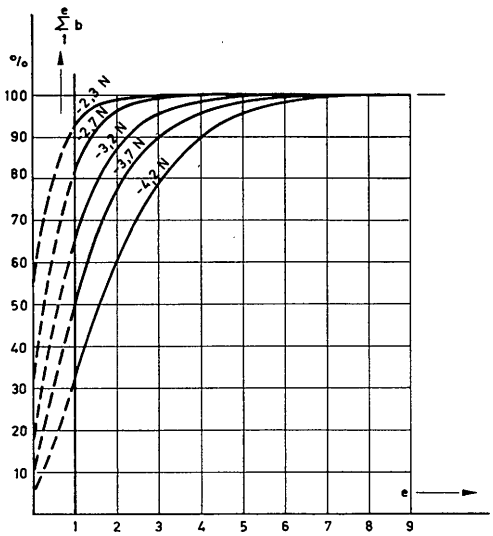
Number of bits transmitted : 9.4×10^6

Block length
$n = 63$

Block length
$n = 9$

Probability b for the occurrence of blocks with e errors as a function of the signal level at the final selector

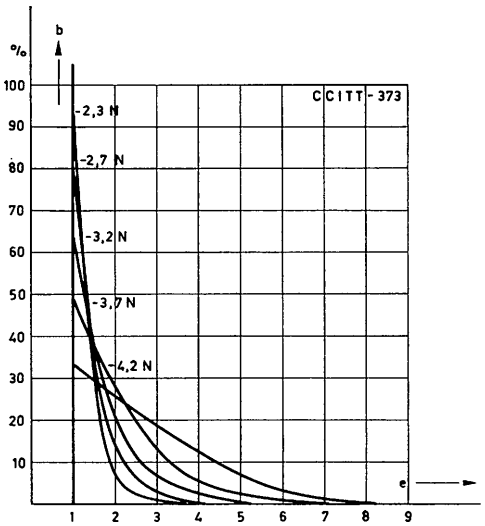
APPENDIX 11



Block length
$n = 9$

Parameter signal level at final selector
Number of bits transmitted : 9.4×10^6

Cumulative distribution Σb for the occurrence of blocks with e errors or less

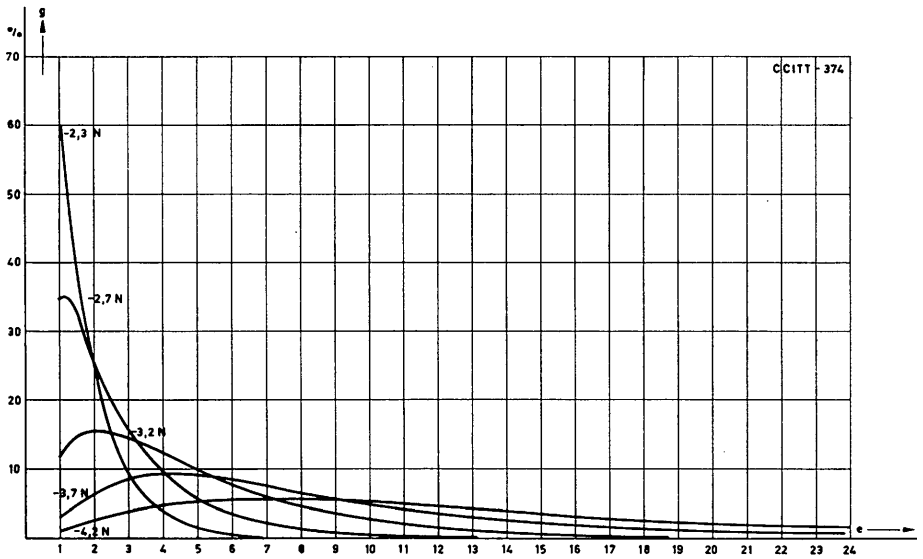


Block length
$n = 9$

	$^{k}p_{EB}$	^{p}EB
−2.3 N	0.122	0.117
−2.7 N	0.133	0.128
−3.2 N	0.175	0.159
−3.7 N	0.210	0.198
−4.2 N	0.267	0.258

Probability distribution b for the occurrence of blocks with e errors

APPENDIX 12



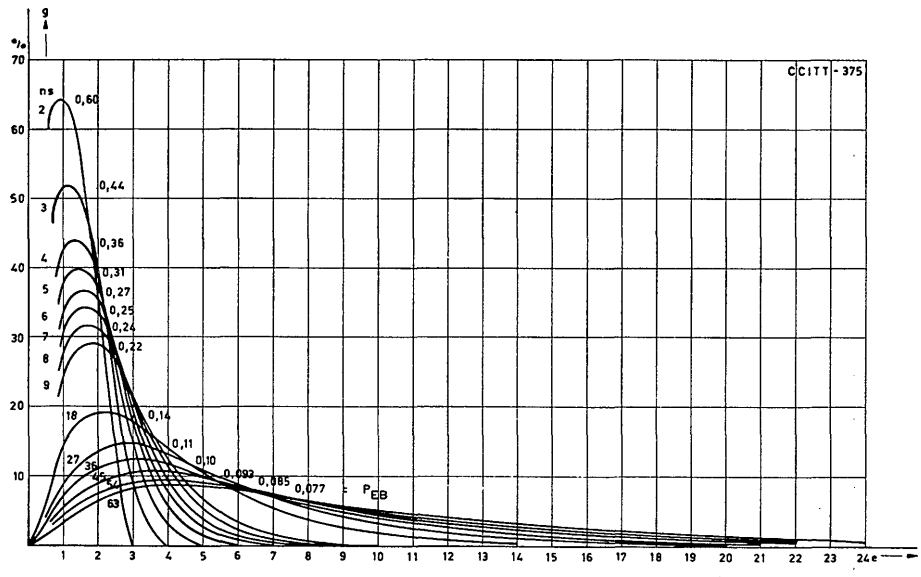
Parameter : signal level at final selector
Number of bits transmitted : 9.4×10^6

Level of final selector N	${}^{k}p_{EB}$	p_{EB}	${}^{n}{}^{k}p_{EB}$	${}^{np}{}_{EB}$
−2.3	0.020	0.019	1.28	1.20
−2.7	0.029	0.025	1.83	1.57
−3.2	0.047	0.039	2.99	2.43
−3.7	0.078	0.063	4.88	3.96
−4.2	0.122	0.100	7.72	6.28

Block length : $n = 63$

Probability distribution g for the occurrence of erroneous elements in blocks with e errors

APPENDIX 13



Parameter : block length n
Number of bits transmitted : 41.8×10^6
Signal level at final selector : -3.6 N

$$^{k}p_{EB} = \frac{1}{n} \times \frac{\text{errors}}{\text{block}}$$

Probability distribution for the occurrence of erroneous elements in blocks with e errors

Supplement No. 29

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 15 — September 1961)*

REPORT ON DATA TRANSMISSION TESTS ON THE GERMAN TELEPHONE NETWORK WHICH HAVE BEEN CARRIED OUT IN CO-OPERATION WITH I.B.M. GERMANY

1. *Introduction*

This Supplement is the summarized result of data transmission tests carried out with various connections over the German telephone network by the Federal Post Office in co-operation with I.B.M. Germany during the past six months.

2. *Test programme*

The test programme covered rented connections as well as dialled-up connections with a modulation rate of 1200 bauds.

In view of the recommendations outlined in document GT 43—No. 20 that the transmit level should be −10 dbm at the point of zero relative level (relatively −10 dbm0 ± 6 db) the tests were carried out with the following transmit levels (the values are given in −dbm0):

Rented connections	10,	16,	22,	28,	34
Dialled-up connections	10,	16,	22,	28	

For the dialled-up connection the zero relative level point was assumed to be at the first selector in the automatic exchange. The employment of a transmit level of −4 dbm0 was abandoned, since only a slight error rate resulted.

All tests were restricted to the main business hours, i.e. Monday to Friday, from 8 a.m. to 5 p.m.

3. *Test equipment*

3.1 *Complete measuring arrangement*

For the tests described in this contribution, I.B.M. Germany developed a measuring and evaluation method and supplied the respective equipments on which a short survey will be given here. The block diagram of the complete measuring arrangement can be seen in Figure 1.

3.1.1 *Test message*

A periodical binary message with 256 or 512 bits generated by a shift register with a feedback loop (pseudo character generator) was used for the tests (the principles of this generator are shown in Figure 2). The test message produced automatically covers the range of bit combinations that will also have to be expected in future operations. The modulation rate is determined by an AF signal generator. The periodically recurrent information block can be switched to the input of whatever modulation system is used.

3.1.2 *Recognition of transmission errors*

The reliability of the transmission depends on the detection of the received signal, which is composed of transmitted signal and noise, to obtain the appropriate binary digits. To check the received message for errors at the receiving end a clock signal in synchronism with the received data is needed to regenerate the original transmitted message to act as a reference.

The test message was coded by the NZR-method (no zero return). For this type of code the binary "1" is represented as a change of polarity, the binary "0" as no change of polarity in the data. Therefore, a defined transmission of the mark and space condition is not necessary.

For a comparison of the received data with the reference, the data are sampled in the middle of each bit, so as to tolerate a bigger bias distortion.

Two identical decoders are employed. They are assigned to the receiving and reference data and supply in the middle of the bit a sampled pulse for every binary "1" and no pulse for every binary "0".

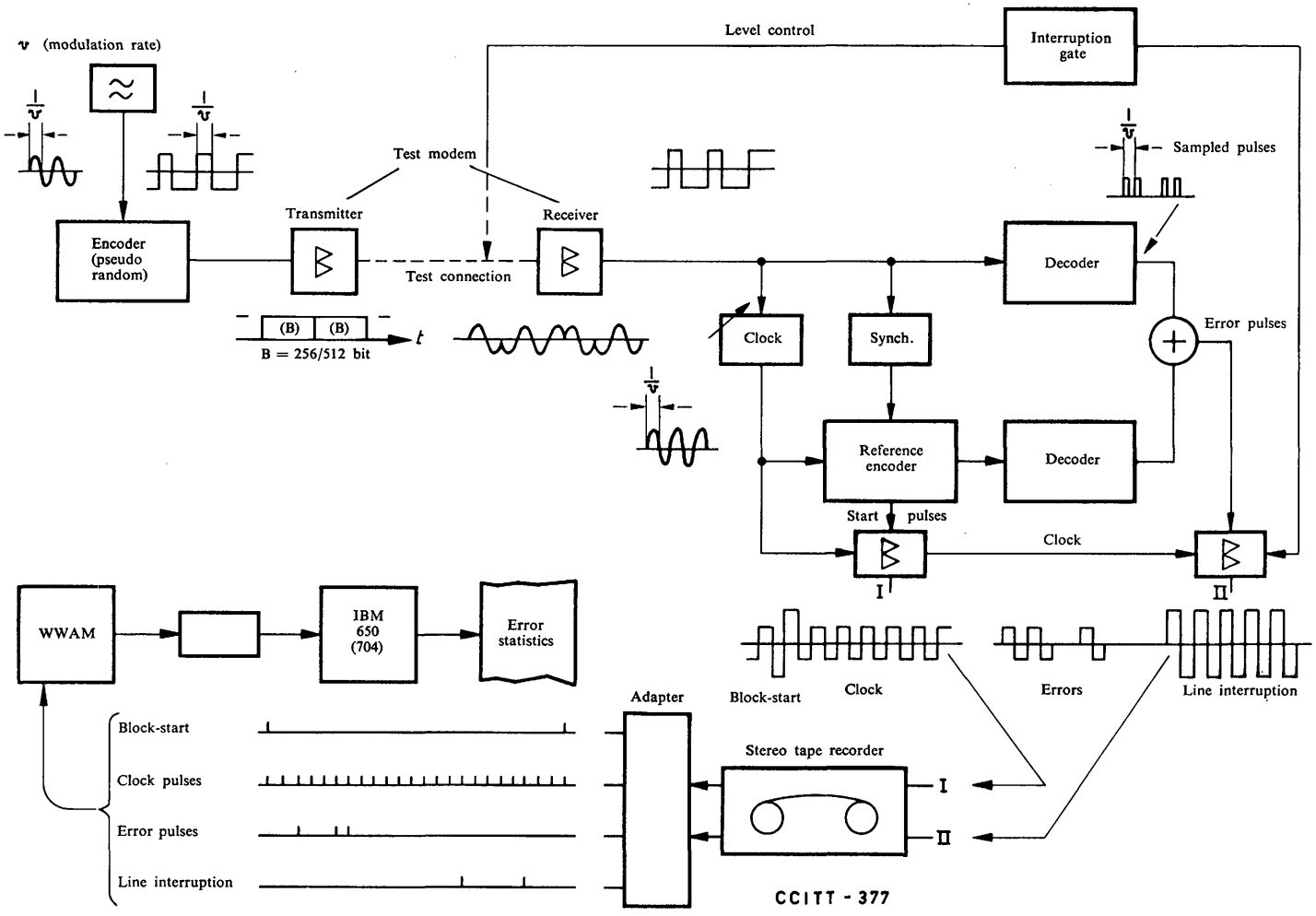
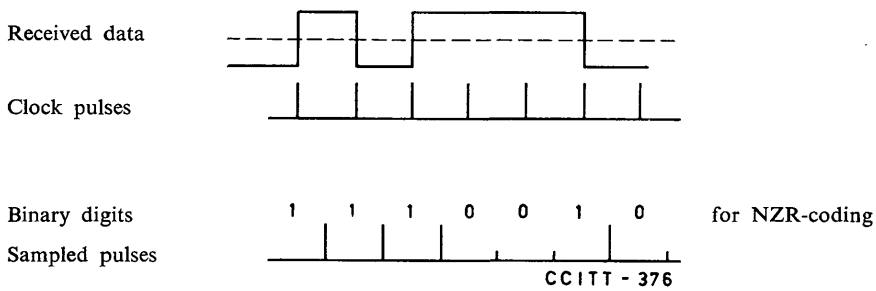


FIGURE 1. — **Serial data transmission on telephone connections. Complete measuring arrangement**

Example for the operation mode of the decoders :



Code conversion of received and reference data is carried out at the same time, i.e. in the middle of the bit. By comparing the sampled pulses gained, and when erroneous transmission takes place, error pulses are obtained which fix the location of these errors within the transmitted message.

Transmission errors which are due to line interruptions will be marked differently. This will be done by means of a level monitoring at the receiving end which produces a gate for the duration of the breakdown in the line.

3.1.3 *Recording of test results*

During the tests various measuring criteria can be obtained which are important for a further study of the transmission performance. By amplitude-modulating the clock, these criteria can be recorded on a stereo tape for future evaluation purposes.

Each of the tracks contains two criteria as follows :

Track 1 :

a) The clock signal ;

b) The block start of the reference message (first bit in the block).

Track 2 :

c) All error pulses (which are also counted and noted during the tests) ;

d) Duration of line interruptions.

Furthermore, precautions are taken to stop the recordings automatically whenever synchronizing faults or line interruptions $\geqq$ 300 ms occur.

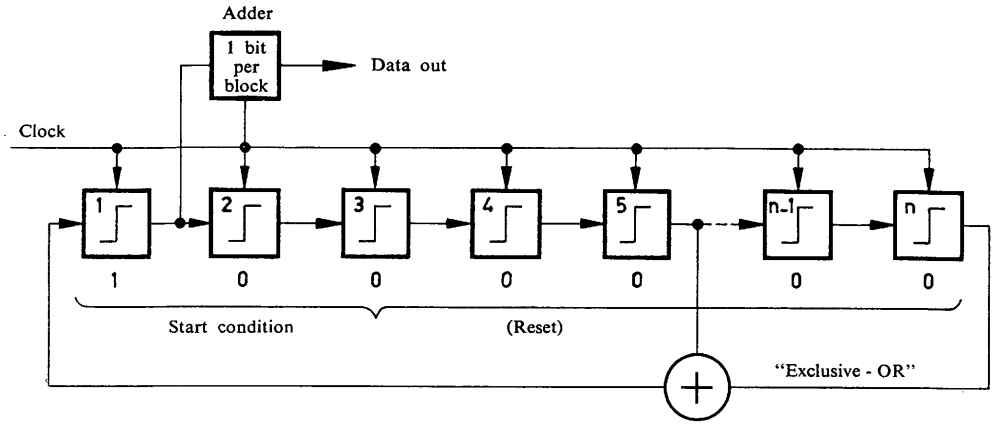
3.1.4 *Evaluation of test results*

With a relatively simple demodulation stage the entire tape information is regained. Given to a programmed computer I.B.M. Lab. type WWAM, the correct number of bits between successive errors or bursts of errors is counted, and the results are stored in punched cards which enable automatic evaluation of results with data-processing systems.

3.2 *Modulation system*

Description of the operation mode : the transmitter operates synchronously, i.e. modulated signal and input data are in fixed phase relationship.

Shift register arrangement for the production of periodic binary messages



Number of bistable multivibrators in tandem	n
Length of periodic binary messages	$L = 2^n - 1$ bit
Output block length (multiple of 8 bits)	$B = L + 1$ bit

for instance

n	B
8	256 bit
9	512 bit

Produced test message (data out)

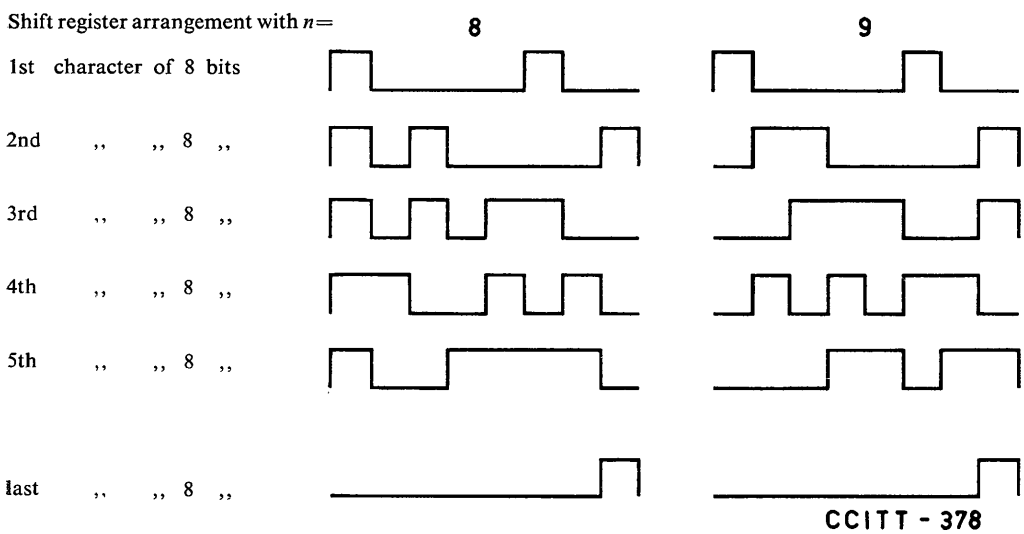


FIGURE 2. — **Test message-generator (Pseudo random characters)**

The clock controls a monostable multivibrator ; from its output signal the necessary carrier may be filtered out by means of a band-pass filter (carrier supply). Thus a carrier synchronous to the data is produced. During every change-over of their polarity, the binary data cause a 180° phase reversal of the carrier in a phase modulator. Because of the synchronous relationship this is always performed during the zero crossing of the carrier waveform. An amplifier and a matching symmetrical transformer which succeed the modulator adapt the output signal to the telephone connection.

Due to the 180° phase reversal of the carrier, two possible phase conditions exist, and hence the expression bi-phase modulation. Here the binary digit will be transmitted by frequency as well as by phase criterion. Therefore, a demodulation of the received signal is possible only when in the receiver the information on the carrier phase is known (synchronous detection). In order to realize this for non-synchronous transmission channels, both sidebands have to be transmitted to be able to find out the momentary phase condition of the demodulation carrier. The necessary transmission bandwidth for a bi-phase modulation system with double sideband is comparable with that of an amplitude modulation system.

However, this system is not able to transmit a well-defined Mark or Space condition. By no means would that be impossible, but it would be more expensive. Therefore, the tests could be carried out only for the NZR coding mentioned in the last chapter.

3.3

. . . . . .

4. *Test connections*

The test programme covered :

three rented connections ;

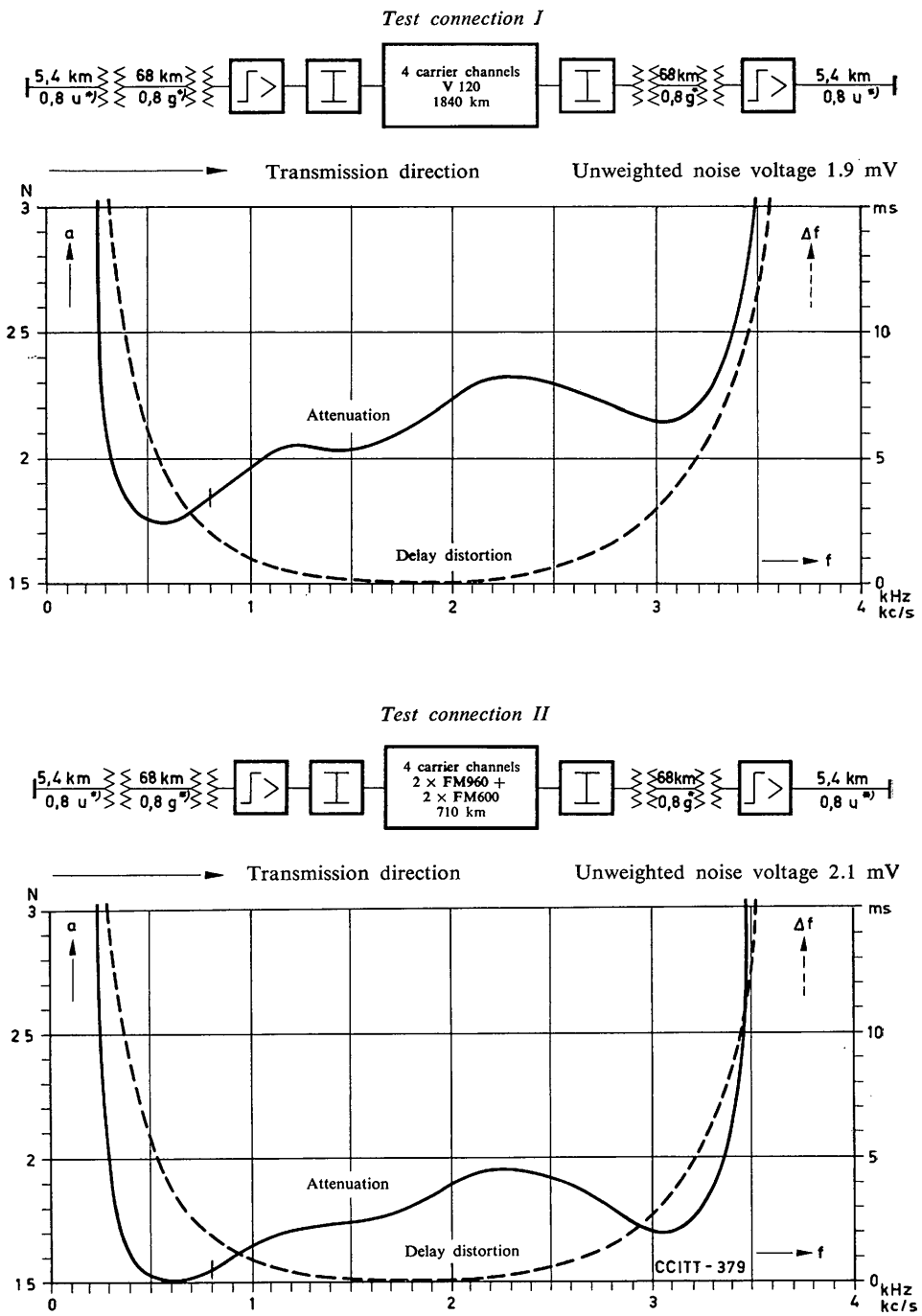
four dialled-up connections.

The rented connections were composed of sections with loaded and unloaded AF cables and carrier frequency channels. At two points these connections were equalized for amplitude distortion by special amplifiers. The individual sections were connected in loops. In order to take into account the frequency shifts which occur in single-sideband carrier transmission, and which could be compensated by loop conditions, each carrier frequency channel was used in one direction only.

All dialled-up connections—test connection VII excepted—are local connections between subscribers on different automatic exchanges in Frankfurt (Main). With test connection VII the attempt was made to simulate a dialled-up connection in the district traffic. (An equalization of the dialled-up connections was not attempted generally, since this would necessitate a compromise equalization which is not known.)

Further details about the set-up and the transmission characteristics on these test connections are contained in Figures 5 to 8. All data given there for the unweighted noise voltage are mean values from several individual measurements. Mostly the peak values of the impulse noise were about 20 db higher.

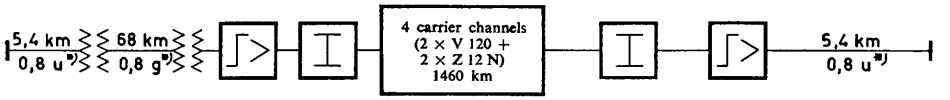
For the test on dialled-up connections two line-connection boxes (one at the transmitting and one at the receiving end) were used for switching over from the telephone set to the data equipment and vice versa. Each line-connection box contained a switch and a transformer and at the receiving end also a high-pass filter, the cut-off frequency of which was about 200 c/s. These two boxes were included in the measurements of the attenuation and delay distortion curves (Figures 7 and 8).



* u = unloaded connection
g = loaded connection with 80 mH, cut-off frequency 4.7 kc/s

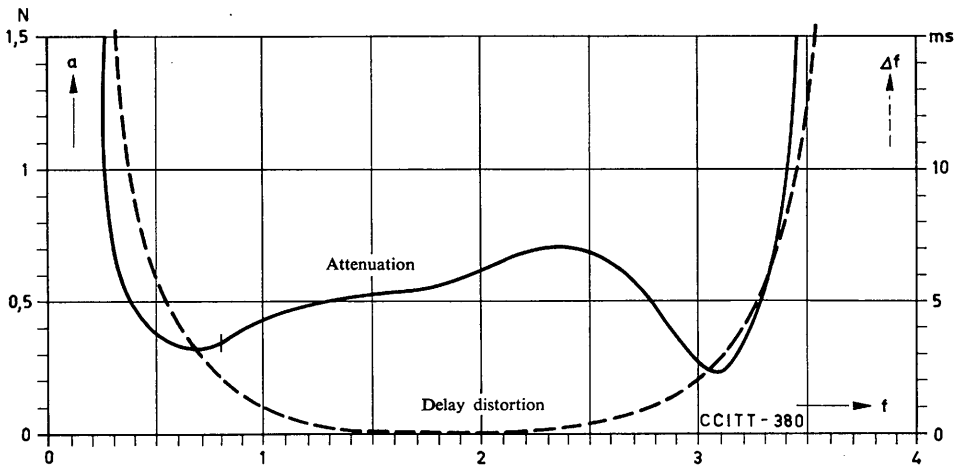
FIGURE 5. — **Test connections I and II (rented connections)—Circuits and characteristics**

Test connection III



Transmitting direction

Unweighted noise voltage 2 mV



* u = unloaded connection

g = loaded connection with 80 mH, cut-off frequency 4.7 kc/s

FIGURE 6. — **Test connection III (rented connection)—Circuit and characteristics**

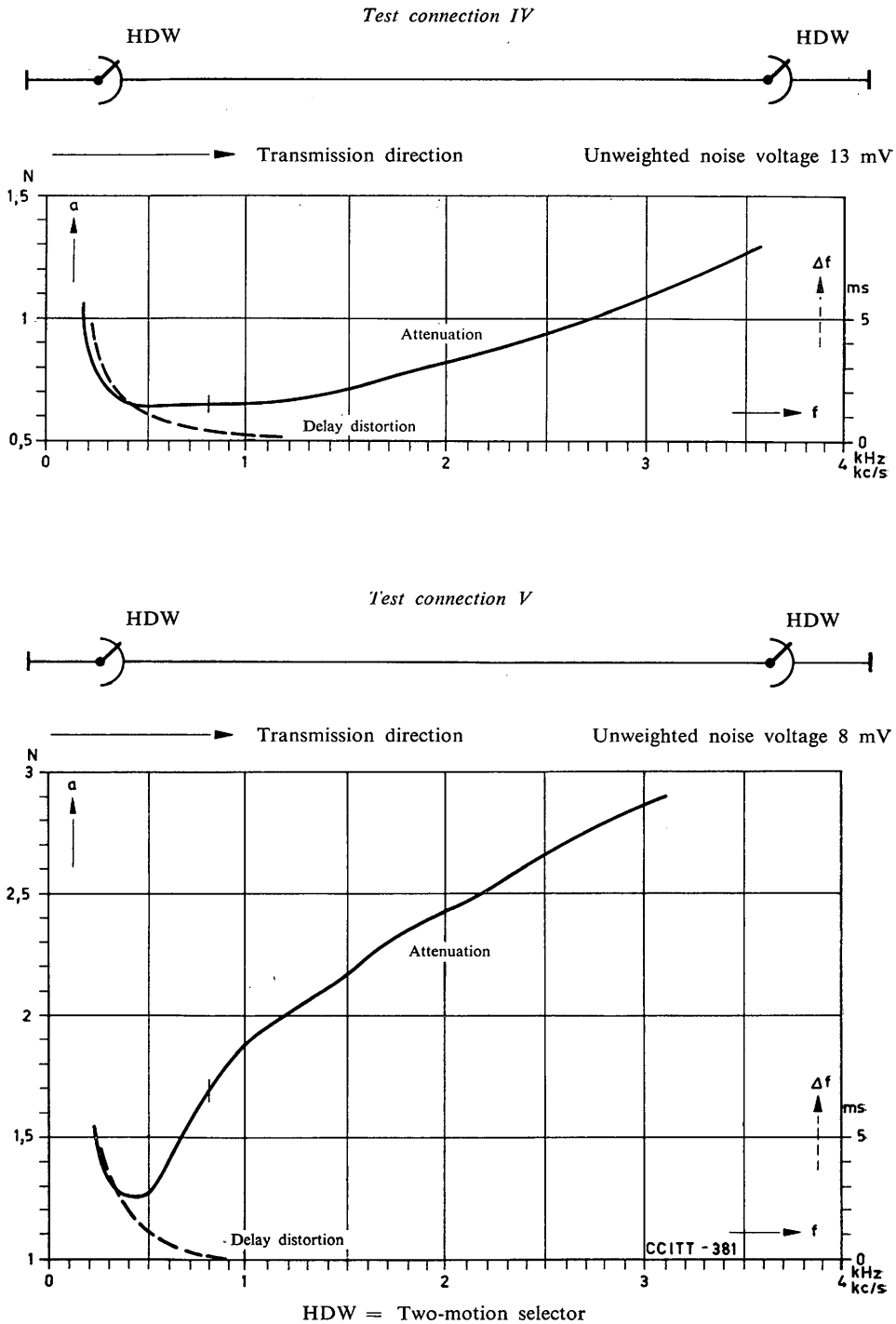
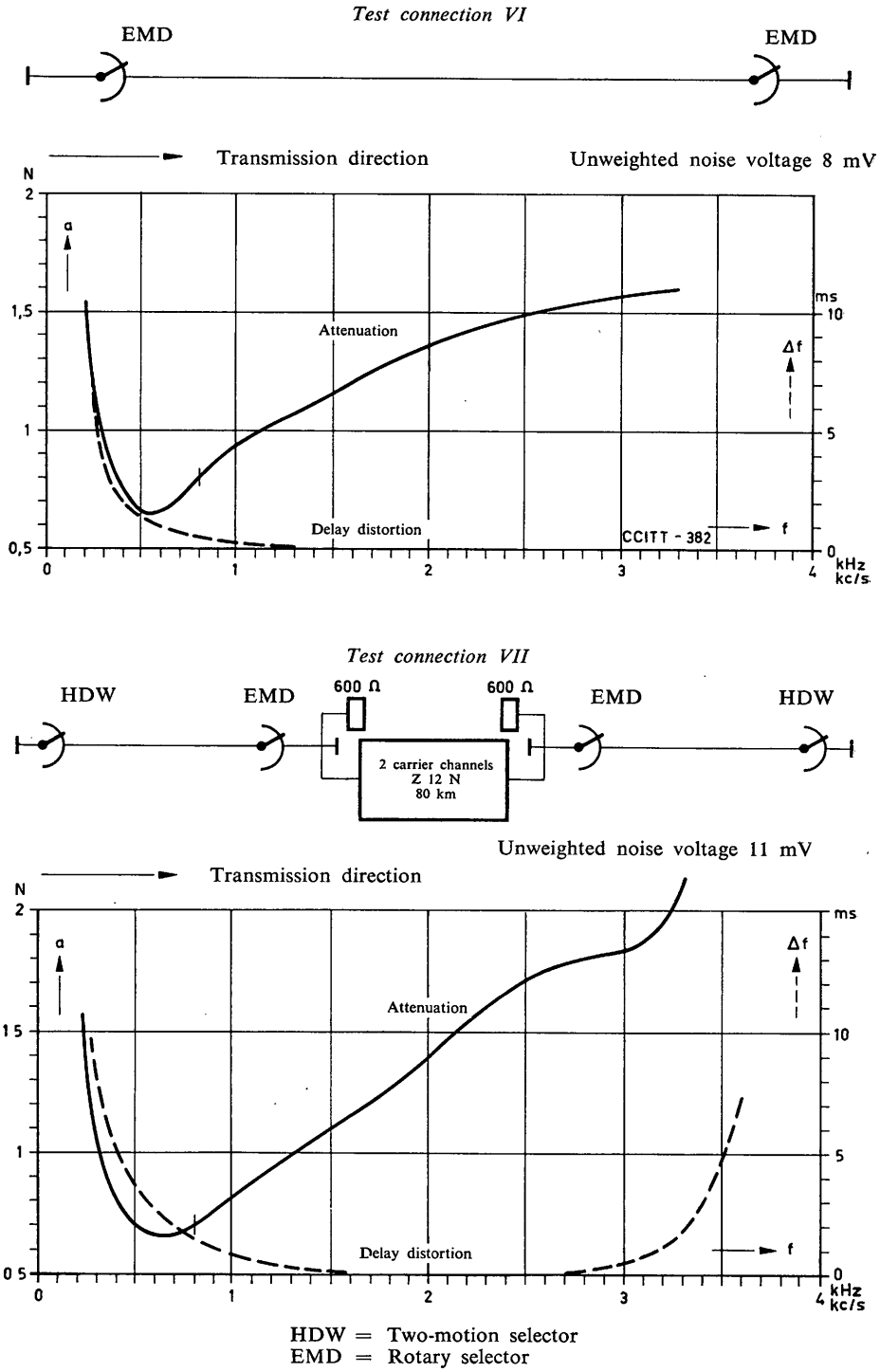


FIGURE 7. — **Test connections IV and V (dialled-up connections)—Circuits and characteristics**



HDW = Two-motion selector
EMD = Rotary selector

FIGURE 8. — **Test connections VI and VII (dialled-up connections)—Circuits and characteristics**

Test connection	Modulation rate in bauds	Total number of transmitted bits	Number of transmitted bits per transmit level					Number of bits in error per transmit level					Bit-error rate [a] per transmit level				
			− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 34 dbm0	− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 34 dbm0	− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 34 dbm0
I	1200	5.47×10^7		1.37×10^7	1.37×10^7	1.37×10^7	1.37×10^7		0	1	34	52		$< 7.3 \times 10^{-8}$	7.3×10^{-8}	2.48×10^{-6}	3.8×10^{-6}
II	1200	6.84×10^7	1.37×10^7	1.37×10^7	1.37×10^7	1.37×10^7	1.37×10^7	0	0	0	44	160	$< 7.3 \times 10^{-8}$	$< 7.3 \times 10^{-8}$	$< 7.3 \times 10^{-8}$	3.2×10^{-6}	1.17×10^{-5}
III	1200	5.47×10^7		7.2×10^6	7.2×10^6	2.02×10^7	2.02×10^7		0	1	5	44		$< 1.4 \times 10^{-7}$	1.4×10^{-7}	2.5×10^{-7}	2.2×10^{-6}

[a] $\frac{\text{Bits in error}}{\text{Transmitted bits}}$

FIGURE 9. — **Total survey of the test results — (bit-error rate [a] for rented connections)**

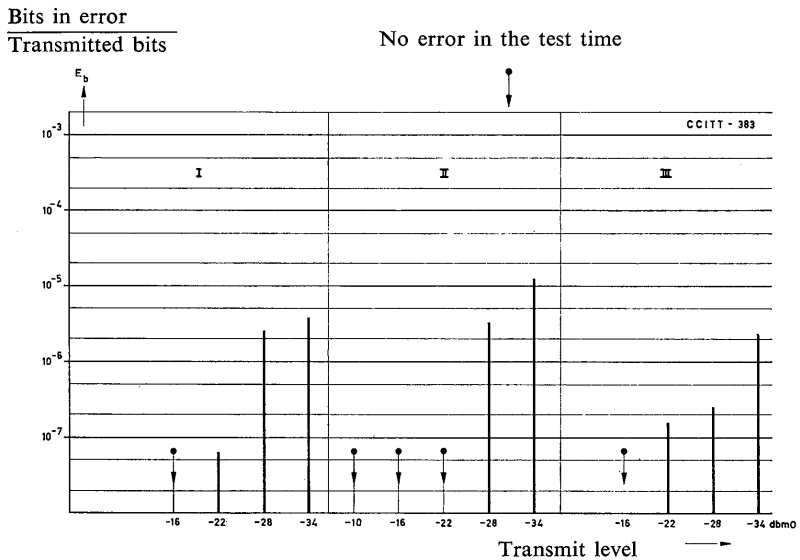


FIGURE 10. — **Total bit-error rate for rented connections**
Results of test connections I, II and III

Test connection	Modulation rate in bauds	Total number of transmitted bits	Number of transmitted bits per transmit level				Number of bits in error per transmit level				Bit-error rate [a] per transmit level			
			− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0
IV	1200	8.2×10^7	2.05×10^7	2.05×10^7	2.05×10^7	2.05×10^7	162	617	2 175	8 430	8.25×10^{-6}	3.0×10^{-5}	1.06×10^{-4}	4.12×10^{-4}
V	1200	8.2×10^7	2.05×10^7	2.05×10^7	2.05×10^7	2.05×10^7	513	2 272	6 137	15 880	2.5×10^{-5}	1.11×10^{-4}	2.99×10^{-4}	7.75×10^{-4}
VI	1200	8.2×10^7	2.05×10^7	2.05×10^7	2.05×10^7	2.05×10^7	21	29	28	62	1.03×10^{-6}	1.4×10^{-6}	1.36×10^{-6}	3.02×10^{-6}
VII	1200	8.2×10^7	2.05×10^7	2.05×10^7	2.05×10^7	2.05×10^7	79	417	3 293	14 156	3.85×10^{-6}	2.03×10^{-5}	1.6×10^{-4}	6.9×10^{-4}

[a] $\frac{\text{Bits in error}}{\text{Transmitted bits}}$

FIGURE 11. — **Total survey of the test results (bit-error rate [a] for dialled-up connections)**

Test connection	Modulation rate in bauds	Total number of transmitted blocks	Number of transmitted blocks per transmit level				Number of blocks in error per transmit level				Block-error rate [a] per transmit level			
			− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0	− 10 dbm0	− 16 dbm0	− 22 dbm0	− 28 dbm0
IV	1200	126 295	41 497	43 504	41 294		119	273	471		2.87×10^{-3}	6.27×10^{-3}	1.14×10^{-2}	
V	1200	122 718	40 991	40 615	41 112		229	597	1 359		5.6×10^{-3}	1.47×10^{-2}	3.3×10^{-2}	
VI	1200	*b*												
VII	1200	120 059	38 707	40 788	40 564		50	217	1 066		1.29×10^{-3}	5.32×10^{-3}	2.63×10^{-2}	

[a] $\frac{\text{Blocks in error}}{\text{Transmitted blocks}}$

[b] Block-error rates were not tested

FIGURE 12. — **Total survey of the test results (block-error rate [a] for dialled-up connections)**

5. *Survey of results*

. . . . . .

5.1 *Rented connections*

The total survey in Figure 9 shows the test results of this group of telephone-type circuits.

While using transmit levels of −10, −16, and −22 dbm0 no fixed relationship between total bit error rate and transmit level could be found. Therefore tests were made also for lower transmit levels (−28 and −34 dbm0) in order to establish the influence of the noise.

The bit-error rate for all tests performed for this group of connections was so small (for −10 dbm0 transmit level $<10^{-7}$) that no detailed error statistics and no study of the block-error rate were necessary.

5.2 *Dialled-up connections*

The total survey in Figures 11 and 12 gives a table for the test results of this group of telephone type-circuits. The bar graphs in Figure 13 here also enable an interesting comparison between the measured total bit-error rates obtained with test connection VI (pure rotary switching exchanges) and test connections IV, V and VII (two-motion switching exchanges).

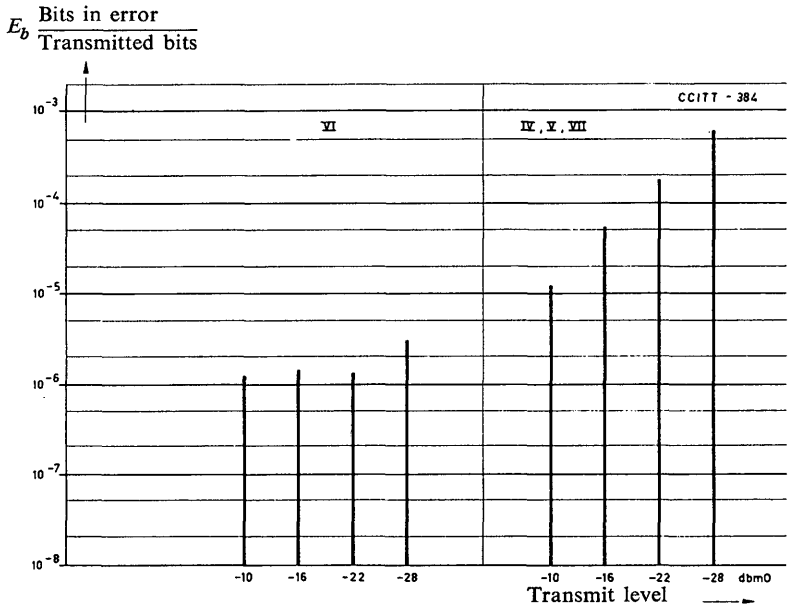


FIGURE 13. — **Total bit-error rate for dialled-up connections**
Results of test connections VI (EMD) and average values for IV, V, VII

The advantage of using modern rotary selector connections (EMD) may be recognized clearly. Most informative is the insignificant influence of the transmit levels. An exact evaluation of the error statistics for these EMD connections is not necessary. However, it will be necessary to have complete error statistics for connections with two-motion selectors (HDW), as had already been obtained for test connections IV, V and VII (see below). As is shown in Figure 13, the total bit-error rate for the HDW-connections will increase to unfavourable values as the transmit level decreases.

5.2.1 *Bit-error rate*

The peak value distribution curves of test connections IV, V, VII are shown in Figures 14 and 15. It is interesting to notice that the bit-error rate depends to a large extent on the transmit level. On the other hand, the mean values for the bit-error rate obtained for the various connections do not differ greatly.

For further evaluation of the results, the middle range of these peak value distribution curves is of foremost interest (say 20% to 80% values), since the outer ranges of these curves (0-20% and 80-100% values) are much more determined by accidental results. Therefore in Figure 16 the bit-error rate just reached is given for the 25, 50 and 75% values of the curves. The diagram at the bottom of Figure 15 shows a mean bit-error rate for the test connections IV, V, VII which is composed of the 25 and 75% values. From the characteristics of these curves it can be seen that the bit-error rate increases to unfavourable values by one order of magnitude as the transmit level is reduced by about 9 db. But also it can be seen that, by using a transmit level of, e.g., −4 dbm0, a considerable improvement may be achieved.

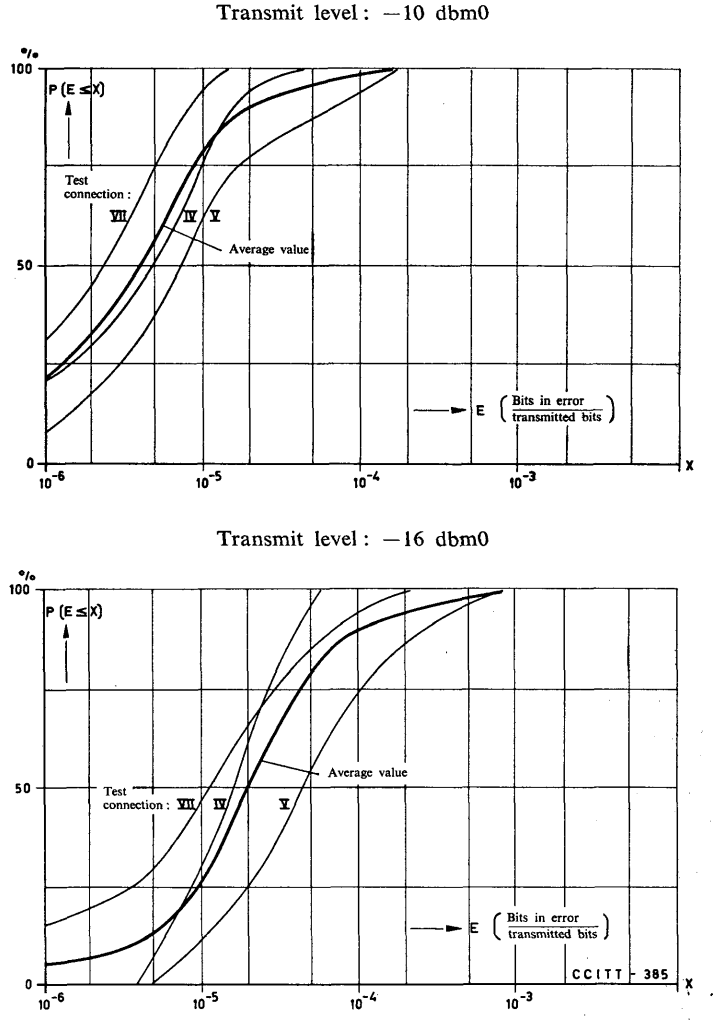


FIGURE 14. — **Bit-error rate (peak value distribution)**

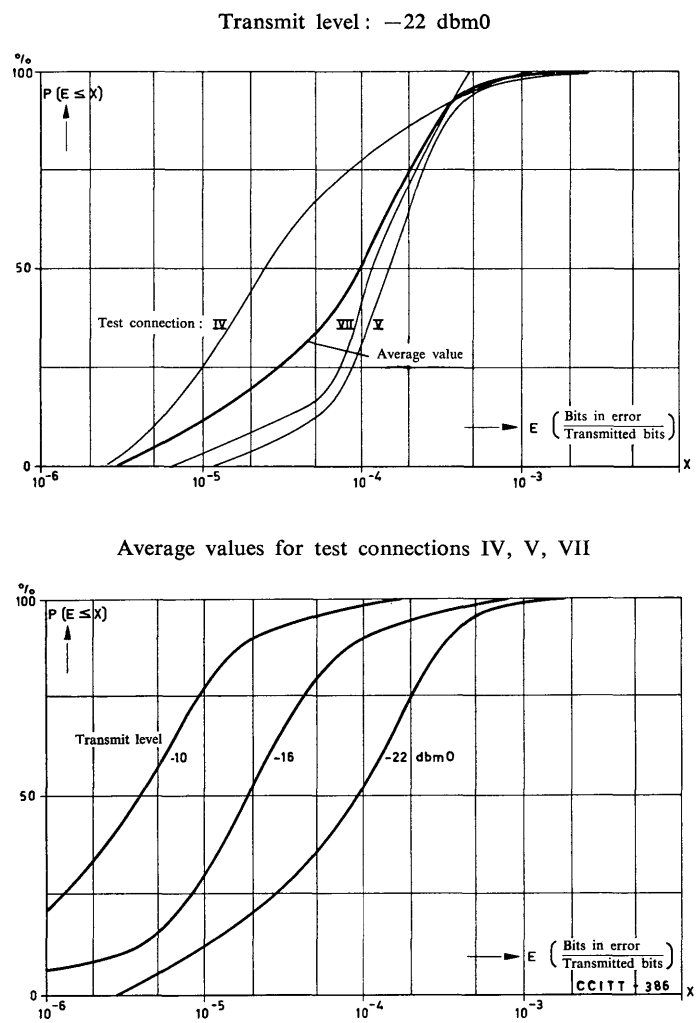


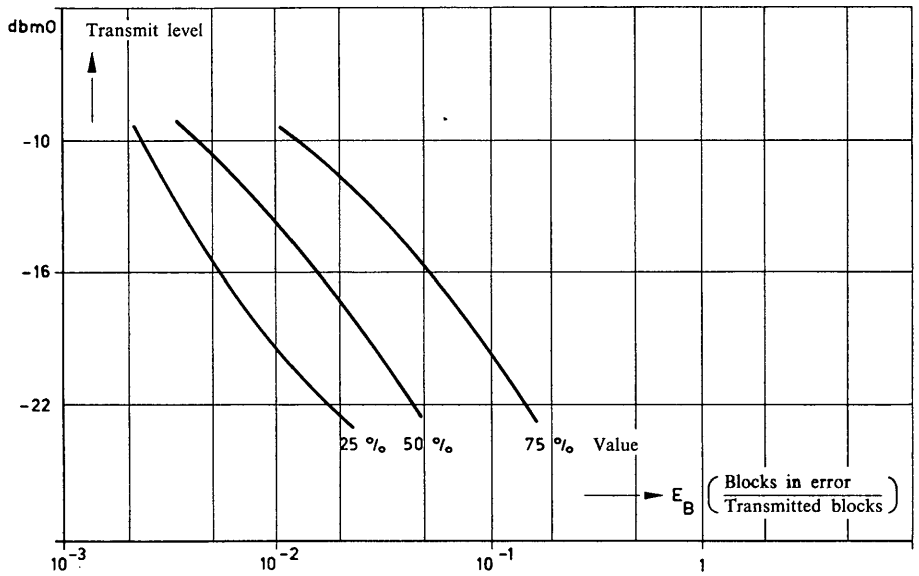
FIGURE 15. — **Bit-error rate (peak value distribution)**

5.2.2 *Block-error rate*

For reasons of reliability a data transmission system which will be employed in future will combine the information to be transmitted into individual blocks. In each case the length of these blocks will be determined by the transmission channel. For this reason the block-error rate (Figures 17 to 19) was found for the dialled-up connections whose bit-error rate is less favourable than that of the rented connections and which, moreover, showed that the errors more frequently occurred in bursts. For this purpose the test intervals were divided up into blocks of 512 bits each. For a better illustration of the conditions, the block-error rate was given in the same form as the bit-error rate specified previously.

For the transmit levels in question the block-error rate was found to be of the order of 10^{-2} approximately. As may be seen from Figure 19, the block-error rate tends to a value of 10^{-3} if a transmit of −4 dbm0 is used. Hardly any correlation between block- and bit-error rate can be noticed.

For the 25, 50 and 75 % values of the peak value distribution curves
(Fig. 15 bottom)



Mean bit-error rate
Composed of the 25 and 75 % values of the peak value distribution curves

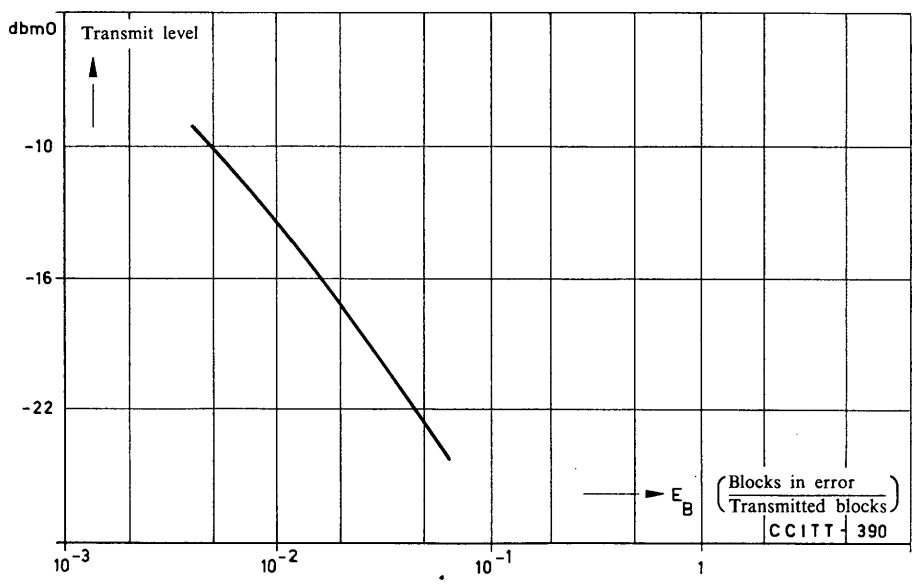
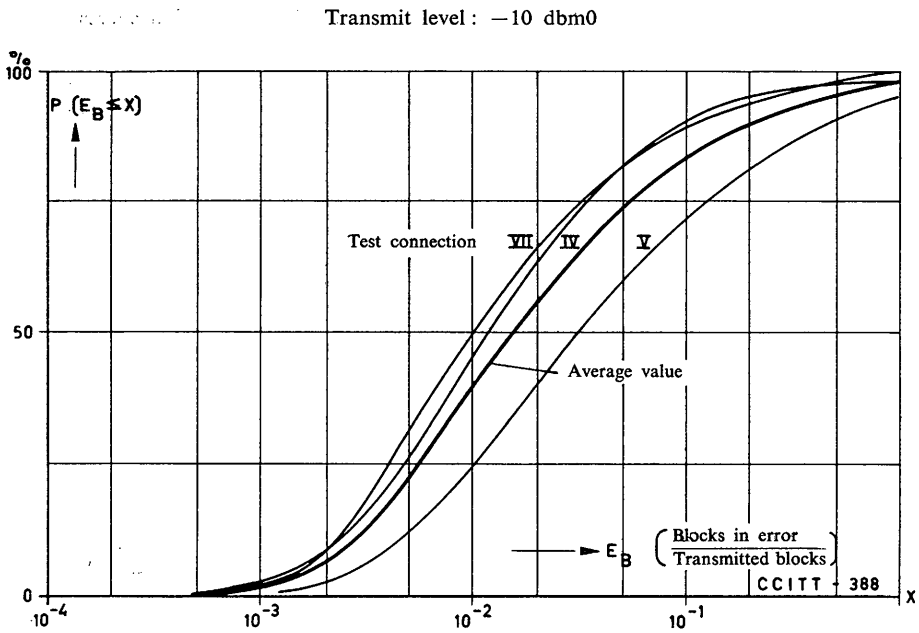


FIGURE 16. — **Bit-error rate (versus transmitting level)**
Average values of test connections IV, V and VII



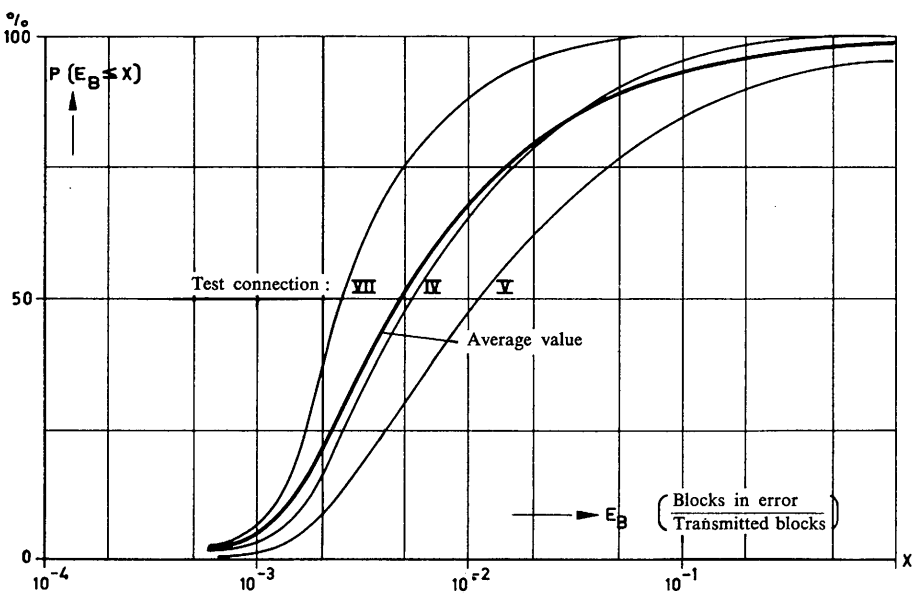
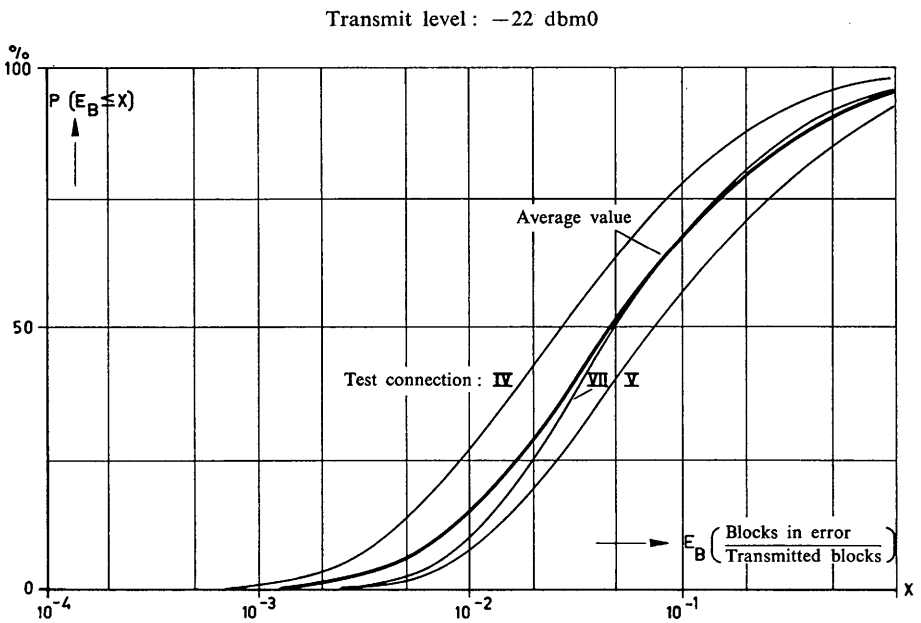


FIGURE 17. — **Block-error rate (peak value distribution)**



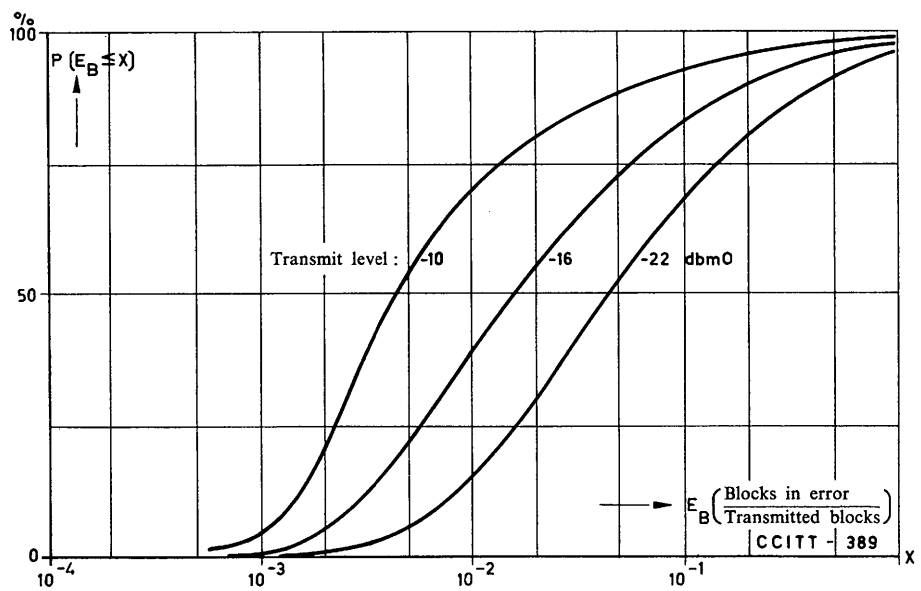
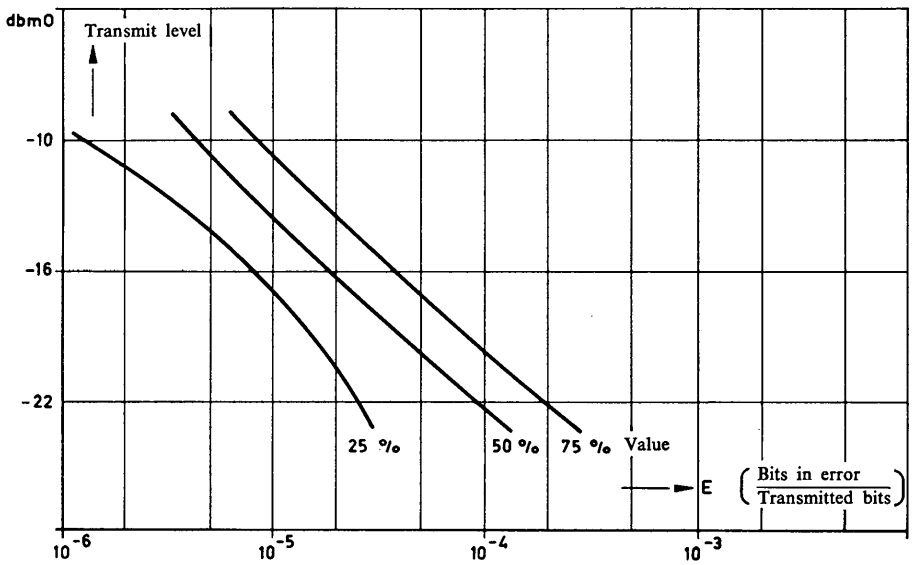


FIGURE 18. — **Block-error rate (peak value distribution)**

For the 25, 50 and 75 % values of the peak value distribution curves
(Fig. 18 bottom)



Mean block-error rate
Composed of the 25 and 75 % values of the peak value distribution curves

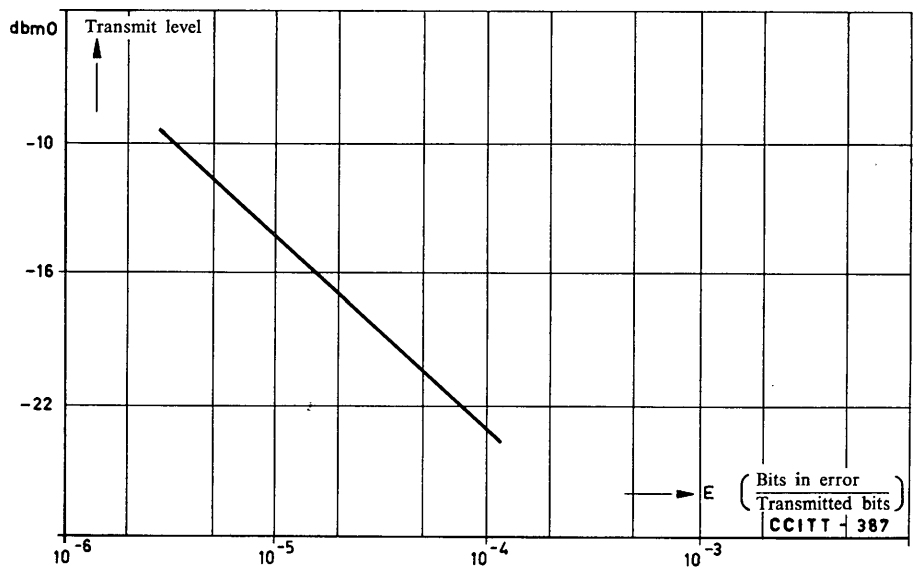


FIGURE 19. — **Block-error rate (versus transmit level)**
Average values of test connections IV, V and VII

5.2.3 *Total number of errors within erroneous blocks*

Since each erroneous block can contain several errors it will be as well to give a more detailed description as to their distribution (the total number of errors within erroneous blocks). In this case the single, double, triple, quadruple, and quintuple errors have been seized per block. Six and more errors per block are defined as bursts of errors and have been treated jointly. These tests were made for two different block sizes (512 bits per block and 3072 bits per block).

For a discussion of the error distribution the mean values of the results for the test connections IV, V and VII are plotted (bar graphs) in Figure 20 for 512 bits per block and for 3072 bits per block. In each case the percentage of the different classes of erroneous blocks relative to the total number of transmitted blocks is given.

As may be seen from the diagrams, the error distribution within the erroneous blocks is dependent on the transmit level. Interesting is the high percentage of single errors. The double, triple, etc. errors decrease, roughly following an exponential law. A more rapid decrease of the multiple errors can be noticed for operations with a higher transmit level. This, however, would be most desirable with regard to the reliable error recognition for a data transmission system. It is important to know that for a block size of 512 bits and a transmit level of −10 dbm0 the number of repetitions could be reduced by 80% if an error-correction code for single and double errors were used.

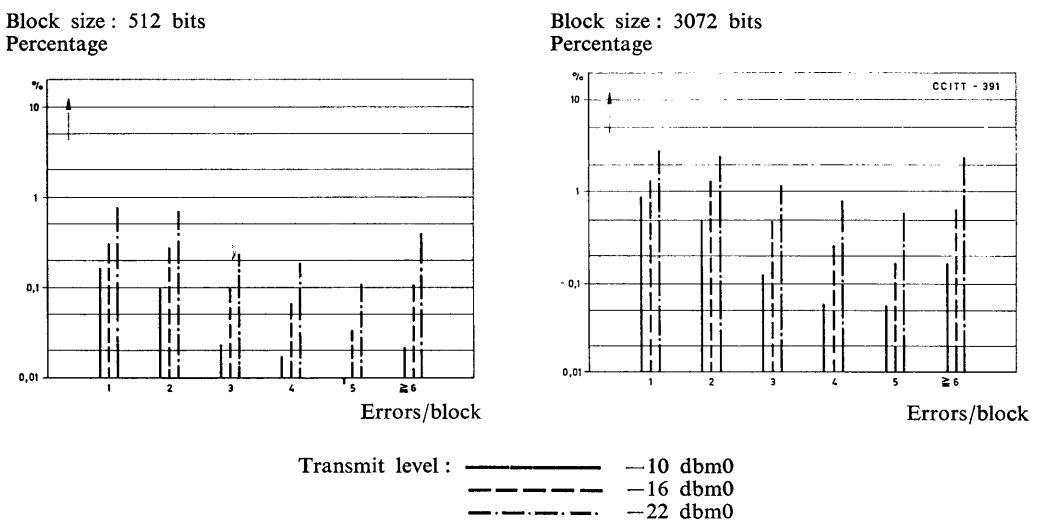


FIGURE 20. — **Total number of errors within erroneous blocks**
Results of the average value of test connections IV, V, VII

As may be seen in Figure 20, the mentioned error distribution within the erroneous blocks is valid for different block lengths also. Thus the result for a block size of 3072 bits shows that the above-mentioned application of the error correction code for single and double errors can reduce also in this case the number of repetitions by 80% approximately.

5.2.4 *Percentage of error-free blocks*

This diagram is plotted in Figure 21 for different block sizes. At increasing block size the percentage of error-free blocks will decrease, of course. For the transmit levels in question it may be assumed that in the cases under review about 95% of the transmitted blocks are error-free.

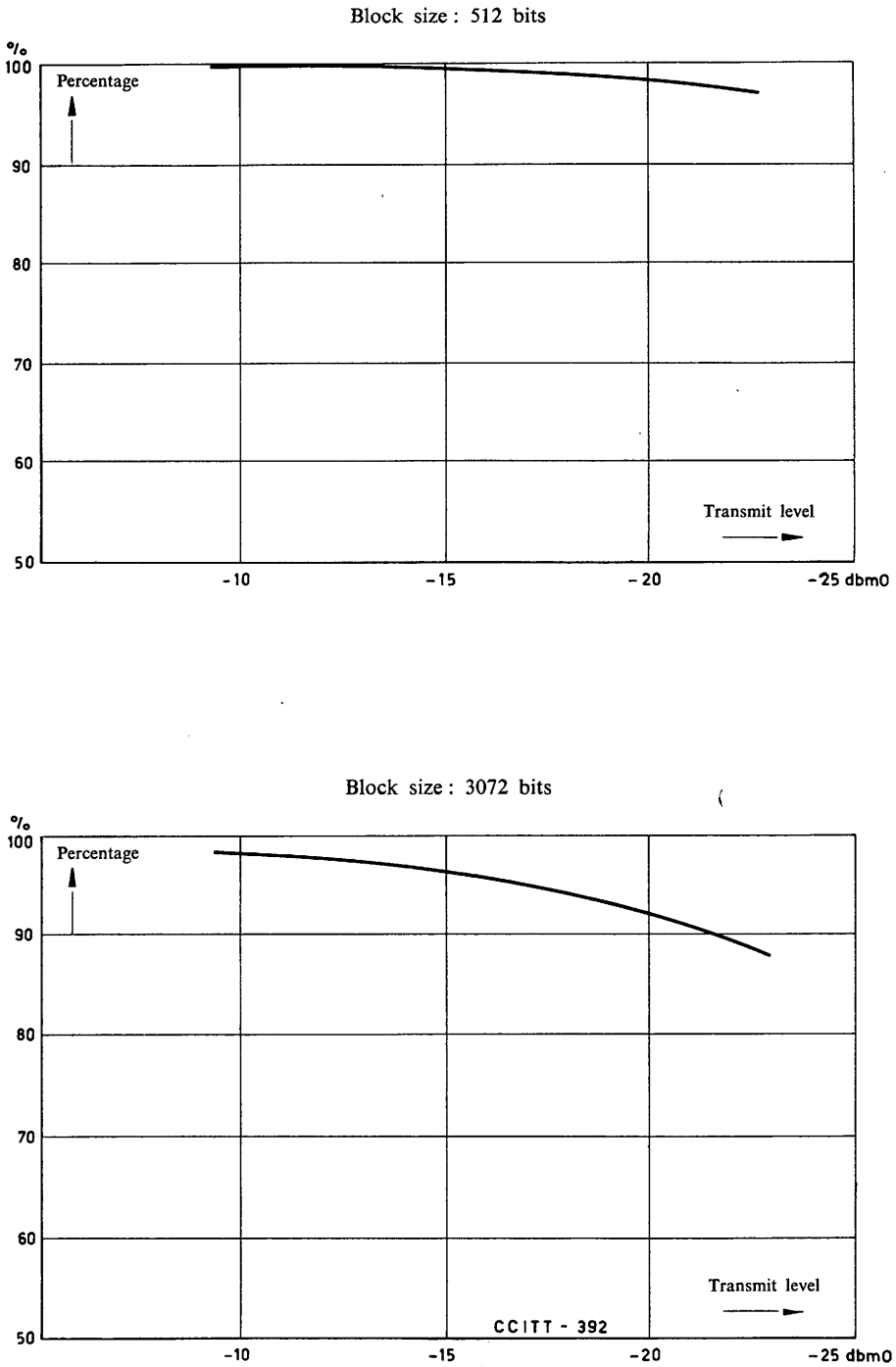


FIGURE 21. — **Percentage of error-free blocks**
Results of the average value of test connections IV, V, VII

5.3 *Line interruptions*

According to point L 11 of Question 1/A, the test apparatus contained an instrument which should disconnect the equipment automatically whenever line interruptions $\geqq 300$ ms occurred. Throughout the measuring period no such line interruptions were recorded.

6. *Analysis of results for prospective data transmission devices*

The results obtained from the line tests provide valuable knowledge for future operations of data transmission devices. In the following the attempt is made to adapt the obtained error statistics to the conditions of practical operation.

6.1 *Transmission in half-duplex operation*

Since many connections are accessible through two wires only, the half-duplex operation is an interesting test example (the possibility of using frequency multiplex will not be considered here). The transmission of data has to be controlled in such a way that occurring disturbances in the transmission channel may not reduce the transmission speed considerably. In order to achieve this, the data to be transmitted will have to be divided up into individual record lengths. Thus it is necessary to find an optimum record length that can be used, and which, last but not least, is determined by the transmission channel employed. A decisive factor in this respect is the time interval between successive information blocks, since it is dependent on the system (turn-around time). For the turn-around time, in the following not only the delay of the channel but also the time which is necessary for the transmission controls and for the passing on of the message will have to be considered.

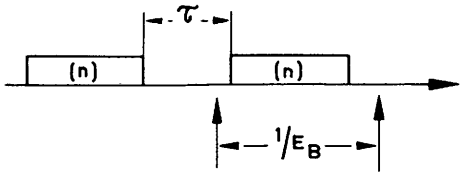
The example in Figure 22 deals with those conditions in concurrence with the applied definitions of the already discussed error statistics. Primarily an effective block rate is worked out and an efficiency is given which depends on the block-error rate and on the employment of integer multiples of the blocks (512 bits) as record length. In order to point out the influence of the turn-around time (τ) clearly, 100, 400 and 1000 ms have been chosen (similar to practical operations).

The given family of curves for $\tau = 100$ and 400 ms illustrates that for a small turn-around time a small record length should be used if an unfavourable block-error rate is to be expected (dialled-up connections). On the other hand, it is also obvious that for a better block-error rate, and while employing a greater record length, the efficiency can be increased only slightly (rented connections).

6.2 *Definition of an optimum record length*

As could be learned from the last section, the definition of an optimum record length makes sense only if employed on dialled-up connections. Since here the block-error rate is subjected to violent fluctuations (type of connection, time of day, etc.) the curves for the peak value distribution of the measured block-error rate (average value curve Figure 18) can provide information for a consideration of an optimum record length. For the given turn-around time $\tau = 100$, 400 and 1000 ms, it is shown in Figures 23 and 24 how the efficiency changes for these peak value distribution curves, for a transmitting level of -10 dbm0, if a record length is used which is the different multiple (n) of the blocks of 512 bits.

In specifying a favourable optimum record length high working efficiency should be guaranteed with the existing channel disturbances.



n = Number of blocks per record length
τ = Turn-around time
E_B = Block-error rate

$1/E_B$ = Distance between successive errors (error-free blocks)

The effective block-transmission speed (considering the repetition of erroneous records) can be written as:

$$V_{\text{eff}} = \frac{n}{\frac{n}{V_{\text{th}}} + \tau} e^{-nE_B}$$

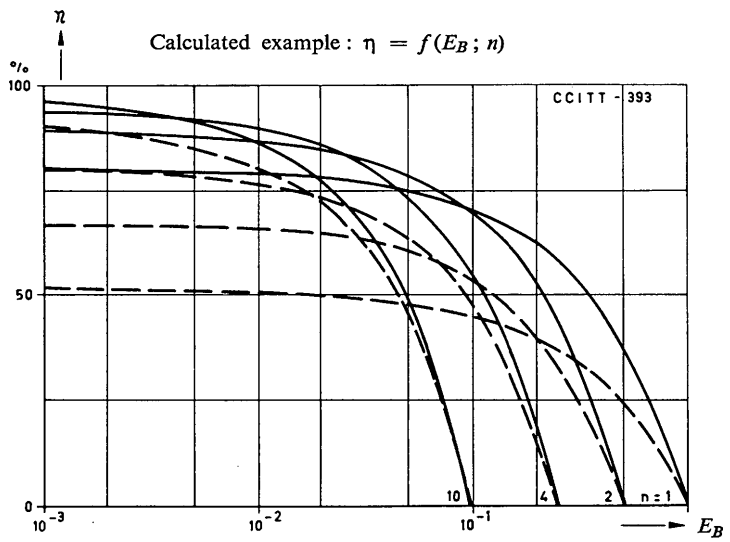
where $V_{\text{th}} = \frac{V_0}{m}$ theoretical block-transmission speed

V_0 = modulation rate

m = block size in bits

From the normal definition of efficiency: $\eta = \frac{V_{\text{eff}}}{V_{\text{th}}}$, then:

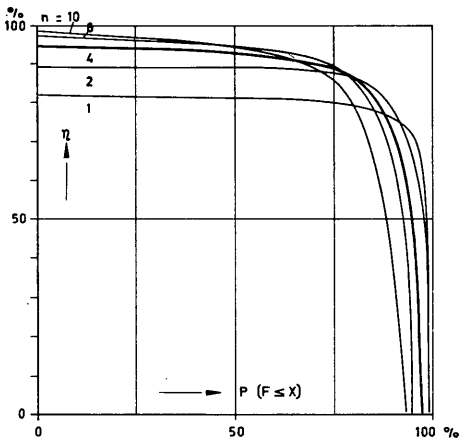
$$\eta = \frac{1 - nE_B}{1 + \frac{V_{\text{th}}}{n} \cdot \tau} \quad \text{if } nE_B < 1$$



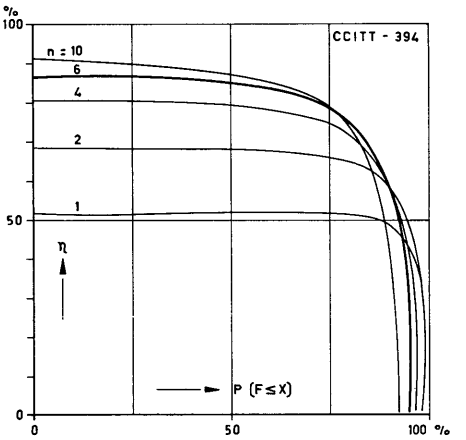
For: V_0 = 1200 bauds
m = 512 bits
τ = 100 ms ————
= 400 ms — — —

FIGURE 22. — **Data transmission in record lengths (effective transmission speed by half-duplex service)**

Turn-around time τ = 100 ms



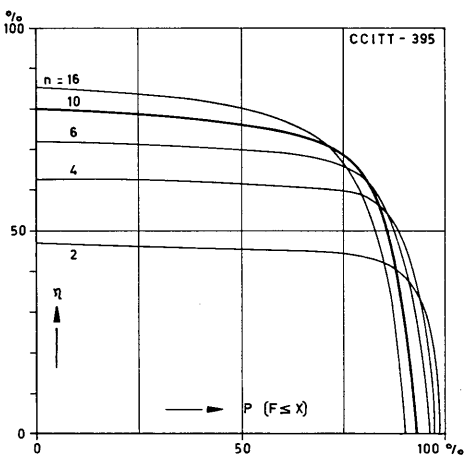
Turn-around time τ = 400 ms



n = Number of blocks of 512 bits (from Fig. 18 bottom)

FIGURE 23. — **Definition of an optimum record length**
Determined with the block-error rate of the peak value distribution curve for −10 dbm0

Turn-around time τ = 1000 ms



n = Number of blocks of 512 bits (from Fig. 18 bottom)

FIGURE 24. — **Definition of an optimum record length**
Determined with the block-error rate of the peak value distribution curve for −10 dbm0

From the results of these measurements it was found that for τ = 100, 400 and 1000 ms an optimum record length of n = 4, 6, and 10 blocks respectively may be considered as favourable (each block consists of 512 bits).

Finally it may be said that, if data transmission devices are employed on telephone-type connections, an optimum record length may be defined which depends on the turn-around time and also on the transmission rate. With regard to the various existing channel disturbances, a common optimum record length may be chosen, so that a favourable efficiency for a multitude of operations may be obtained.

The operator of the device thus is able to employ a recommended record length and has not to investigate it himself for each individual case.

6.3 *Efficiency of data transmission devices*

While working with a common optimum record length, the existing disturbances of the connections used determine the efficiency of the device. In Figures 25 and 26 this efficiency for the tested PM-modulation system is recorded. Finally, it will be shown to what extent it is influenced if the following parameters are changed: a) the transmitting level; b) the turn-around time.

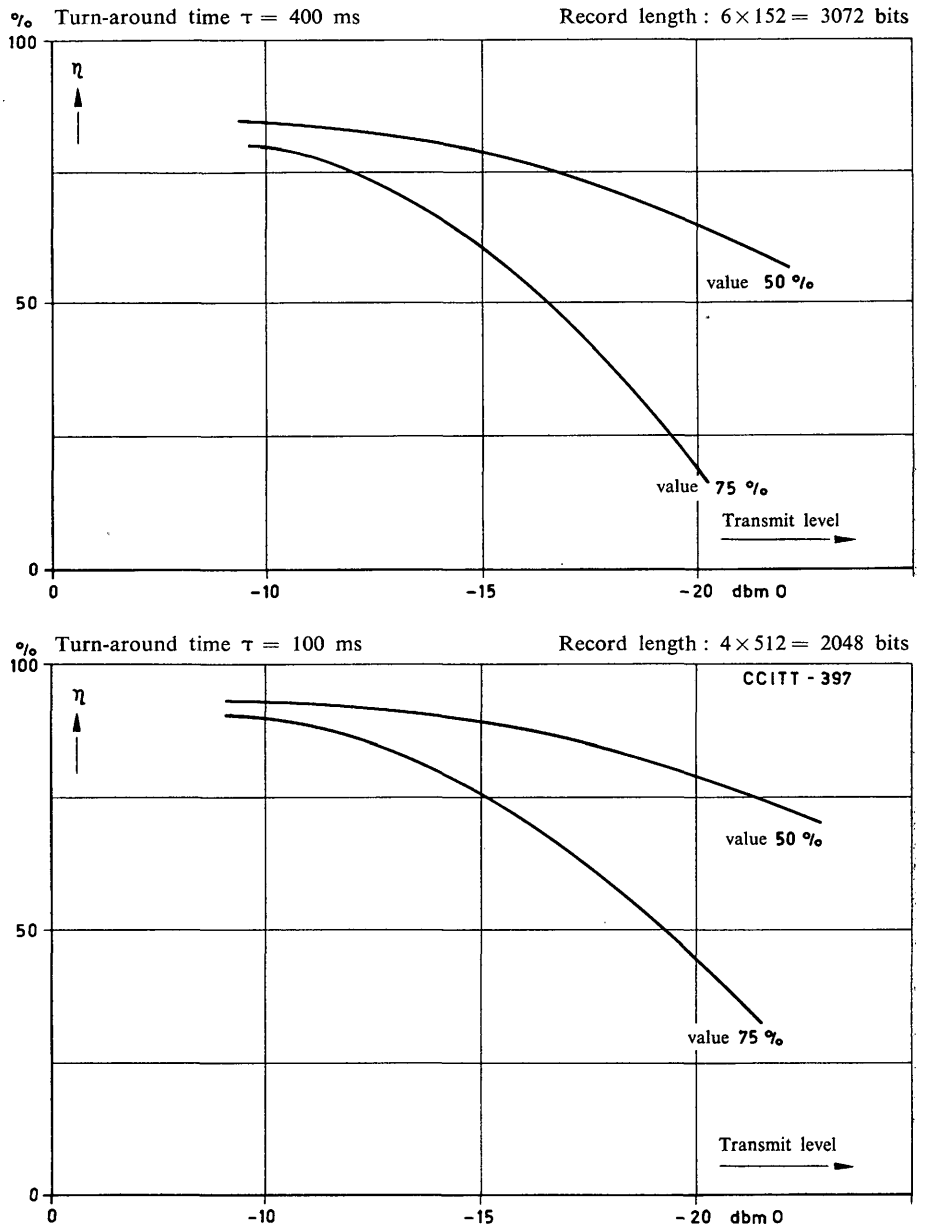
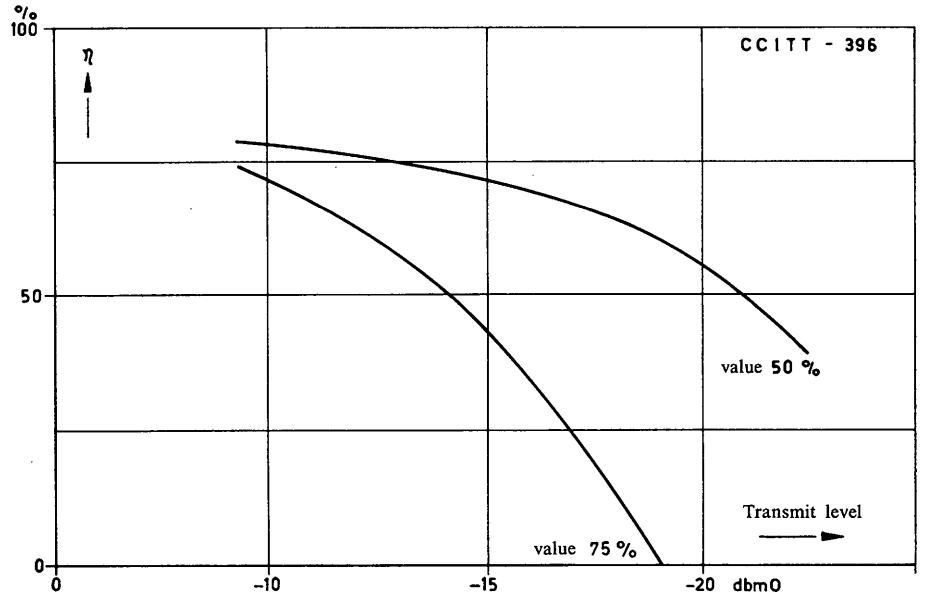


FIGURE 25. — **Efficiency of data transmission devices — Presentation of the total survey**
Determined with the block-error rate for the 50 % and 75 % values of the peak value distribution curve (Figure 19)

Turn-around time $\tau = 1000$ ms Record length $10 \times 512 = 5120$ bits



Efficiency and optimum record length versus turn-around time τ for -10 dbm0 transmit level

Blocks of 512 bits

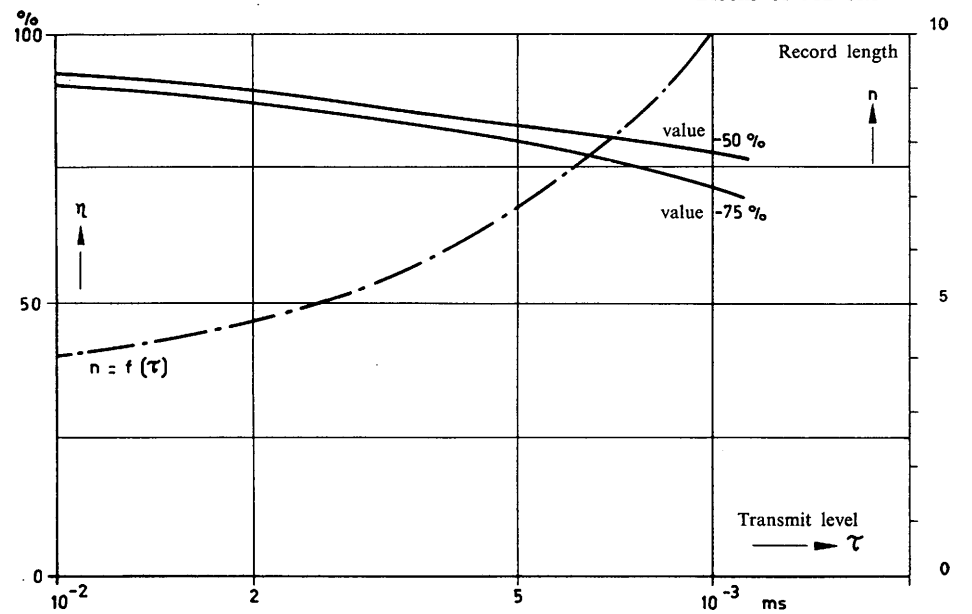


FIGURE 26. — **Efficiency of data transmission devices — Presentation of the total survey**

Determined with the block-error rate for the 50 % and 75 % values of the peak value distribution curve (Fig. 19)

As a basis for the set-up of this survey, the block-error rate just reached for the 50 and 75% values in the curves of the peak value distribution was used (Figure 19) as well as the values of the optimum record length obtained in the previous section and the calculated equations shown in Figure 22.

The total survey shows that the efficiency decreases if a transmission is effected with a transmit level of less than −10 dbm at the point with the relative level 0. This becomes especially clear if for a classification the less favourable measuring values of the peak value distribution curves are taken, e.g. the just reached block-error rate at the 75% value. From all plotted curves it may be seen that the transmission quality of the PM system is favourable.

In the diagram at the bottom of Figure 26 it may be seen that at increasing turn-around time a longer record length should be used and the efficiency becomes less favourable.

7. *Summary of results*

In all cases examined, a satisfactory transmission at a rate of 1200 bauds without any equalization of the test connections was possible. It is likely that the use of phase modulation would enable still higher transmission rates without equalization measures.

On the rented connections the bit-error rate obtained for the transmitting levels in question was $< 10^{-7}$. At 1200 bauds the reliability of a data transmission by records will practically not be reduced.

The bit-error rate of the dialled-up connections is, however, exposed to violent fluctuations (bit-error rate between 10^{-6} and 10^{-3}). This applies especially to connections which are switched over exchanges with two-motion selectors (which up to now are still predominant). The rotary switching connection which has been tested (modern automatic exchanges) on the other hand showed a much more favourable behaviour with regard to interferences, and is comparable with the rented connections since the bit-error rate has the order of magnitude of 10^{-6}.

For a consideration of the transmission quality of a dialled-up connection it is, however, of greater advantage to study the block-error rate because of the frequency of bursts of errors. Nevertheless, there could be recognized no scientific rule which relates the bit-error rate to the block-error rate. Such a rule would be determined to a larger extent by the error distribution which itself is dependent on the modulation system and the line characteristics. For the presently recommended range of transmit level −10 dbm0 ± 6 db, the block-error rate was found to be 10^{-3} to 10^{-2}.

According to the results obtained, the use of a common optimum record length for practical dialled-up connections appears suitable. This record length is determined to a large extent by the turn-around time and can be given for a fixed modulation rate.

The distribution of errors within erroneous blocks, plotted for various cases in Figure 20, shows that the percentage of single errors per block compared with the multiple errors does not prevail decisively. By comparing the percentages of single and double errors per erroneous block, it can be recognized clearly that the number of repetitions of erroneous records may be reduced considerably if a suitable error correction code is used. Thus, it can be seen from the results that 80% approximately of the record repetitions may be spared.

With regard to the efficiency of a prospective data transmission device, the total results obtained were exceptionally positive, all the more since the tests were carried out with the recommended transmitting level of −10 dbm0. These results complete the hitherto existing conceptions and demonstrate that also when dialled-up connections are employed with a modulation rate of 1200 bauds, in most cases an effective data transmission may be expected; always provided that additional delay and amplitude distortions due to band limitations (e.g. more heavily loaded lines) are of no essential importance.

Supplement No. 30

FEDERAL REPUBLIC OF GERMANY — *(Contribution COM Sp. A/No. 25 — October 1961)*

ABSTRACT FROM RESULTS OBTAINED FROM A DATA TRANSMISSION TEST WITH 1200 BAUDS OVER SWITCHED TELEPHONE CIRCUITS

1. Test loop, Munich-Nürnberg-Munich ;
   Munich-Nürnberg : private circuit, regarded as subscriber line ;
   Nürnberg-Munich : switched telephone circuits.

2. Transmitted data : blocks of 960 bits length. Each test run about 1000 blocks with a new dialled connection.

3. Modulation rate : 1200 bauds.

4. Transmission frequencies : 1500-2300 c/s, FM.

5. Sending level at the point of relative level zero in Nürnberg −10 dbm.
   Receiving level at last selector : −28 dbm.
   Receiving level at test receiver : −35 dbm.
   Note. — The last exchange is an old one with Strowger type selectors.

6. *Results :*

   Total number of transmitted blocks : 22.6×10^3.
   Total number of erroneous blocks : $2.47 \times 10^3 = 10.9\%$.
   Average number of erroneous bits per erroneous block : 10.
   All test runs between 8 h. and 17 h. on working-days only.

With an error detection method using a cyclic code and feedback shift registers with 12 cells, all blocks in error were detected.

The test was carried out in co-operation with Siemens & Halske.

Supplement No. 31

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 78 — August 1963)*

REPORT OF MESSRS. FELTEN UND GUILLEAUME FERNMELDEANLAGEN GmbH, NÜRNBERG, ON DATA TRANSMISSION TESTS OVER TELEPHONE CIRCUITS

1. *Introduction*

The test results which are laid down in this report were obtained from data transmission tests made by Felten und Guilleaume Fernmeldeanlagen GmbH Nürnberg on the P.T.T. telephone network in the Federal Republic of Germany. They were carried out by means of a frequency modem over several trunk dialling and point-to-point connections. The transmission speed was 1200 bauds. The transmit level generally was −10 dbm0.

The test results contain the following information :

a) bit-error probabilities, also separately for both possibilities of errors ;

b) block-error probabilities for blocks of 16, 32, 64, 128, 256, 512 and 1024 bits ;

c) error distribution ratio of blocks in error.

2. and 3.

. . . . . .

4. *The test loops*

The test results were obtained on three different dialled-up telephone connections and four rented connections.

The switched connection 1 was situated in the Frankfurt junction centre area. The selectors in both terminals were modern two-motion selectors and rotary selectors.

The switched connection 2 was situated in the Frankfurt main exchange area with modern rare-metal motor-driven rotary selectors (EMD-selectors). Two-motion selectors were also within the transmission path.

Figures 3a, 3b, 4a and 4b illustrate the arrangement of the connections and their attenuation curves.

Test loop 3 was an operational case of a data link in the presence of highly unfavourable conditions. This test loop consisted of test loop 1 connected in series with the Frankfurt-Hamburg-Frankfurt trunk dialling loop and test loop 2. It contained eleven selector stages (Fig. 3c) and had a high net attenuation (Fig. 4c). The establishment of this loop was made possible by means of a special arrangement in the trunk dialling centres of Frankfurt and Hamburg.

The four rented connections were composed—as shown in Figures 5a and 5b—of a number of carrier cable routes or of radio links. These figures also show the attenuation curves of these connections.

For the purpose of the tests, the tested loops were in no way equalized.

5. *The test programme*

The tests on the rented connections were performed over extended periods, i.e. over periods which sometimes exceeded the normal working-hours.

The binary test code applied to the modem was a signal of 80 bits emitted in a recurrent cycle as follows :

00000	10001	10010	11011	10101	00111	11000	00100
01100	10110	11101	01001	11110	00001	00011	00101

This sequence is the 2½ times repeated combination of 32 bits comprising all possible 5-bit combinations.

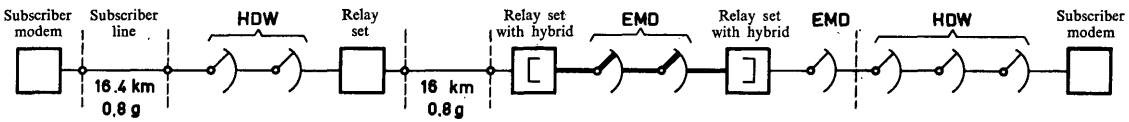
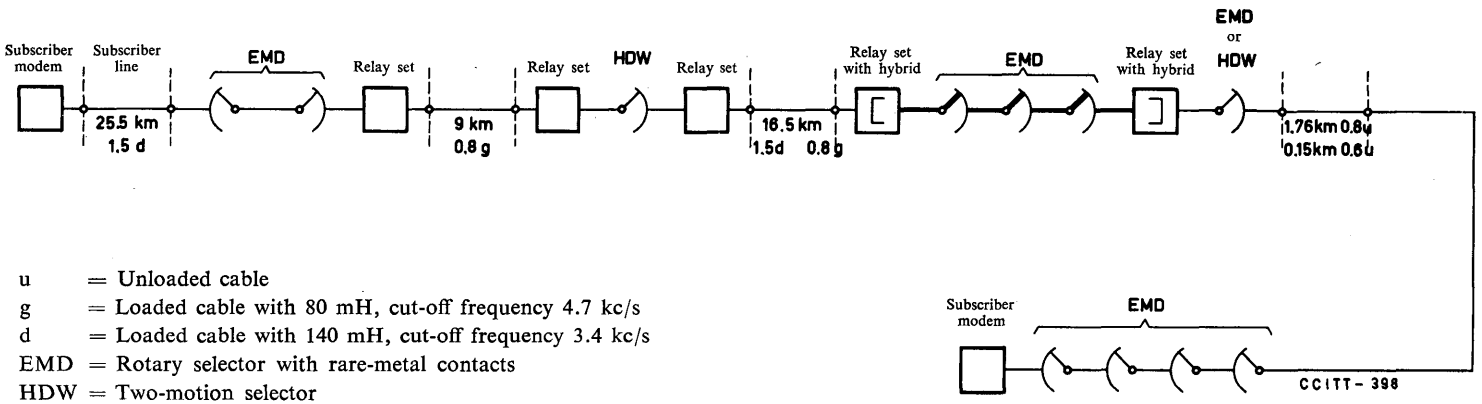


FIGURE 3 a — **Test circuit 1**



u = Unloaded cable
g = Loaded cable with 80 mH, cut-off frequency 4.7 kc/s
d = Loaded cable with 140 mH, cut-off frequency 3.4 kc/s
EMD = Rotary selector with rare-metal contacts
HDW = Two-motion selector

FIGURE 3 b. — **Test circuit 2**

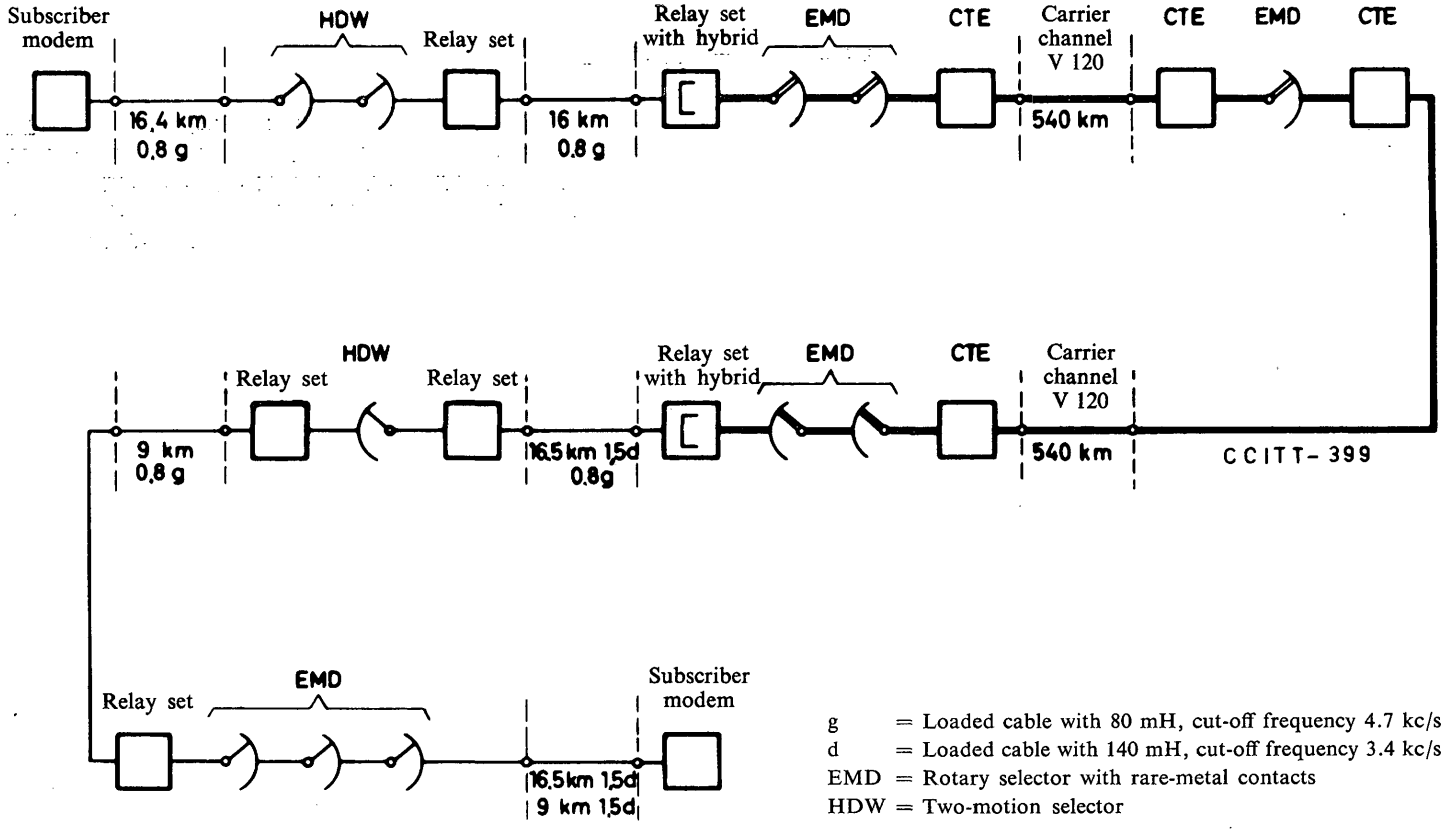


FIGURE 3 c. — **Test circuit 3**

6. *The test results*

Tables 1-6 and the diagrams shown in Figures 6a-6c are a summary of the test results drawn of the punched-tape records by means of an electronic computer.

The transmit level given in Table 1 is the level at the output of the modem.

The receive level was particularly in the dialled loops—subject to fluctuations. These were due to the various possibilities to establish the connections.

Table 1 contains a summary of the number of bits transmitted, the number of transmission errors, the bit-error rate, and the block-error rate.

The bit errors and their rate were recorded separately for both types of errors.

The block-error rate was ascertained by the partition in blocks of 16, 32, 64...1024 bits of the entire bit series transmitted for the test.

The tests of the dialled-up connections have shown—to a certain extent—a marked dissymmetry of the error ratio. No such dissymmetry is found on the point-to-point connections, with the exception of test loop I, but where the number of errors recorded is not high enough to draw up a statistical assessment.

For dialled-up connections (test loops 1-3) the block-error rate within the close compass of the tested range does not increase with the block length to such an extent as could be expected with an equiprobable bit-error distribution (Fig. 7a-7c). This is also the case for blocks of 1024 bits in test loop 3, which has shown a comparatively high number of bit errors.

Figures 6a to 6c are diagrams showing the extent of bit-error rate variations on the various connections of 15 minutes each. The diagrams show the cumulative distribution curve, the ordinate figure of which indicates the proportion of the connections having an error rate up to the value indicated on the abscissa.

For dialled connections, Tables 2 to 6 show the distribution of the faulty blocks (with 16 to 1024 bits) on the various numbers of errors.

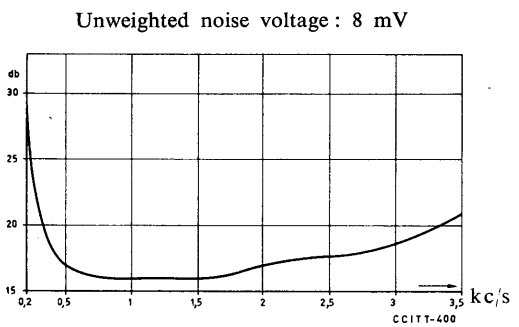


FIGURE 4a. — **Attenuation curve of test loop 1 (dialled-up connection)**

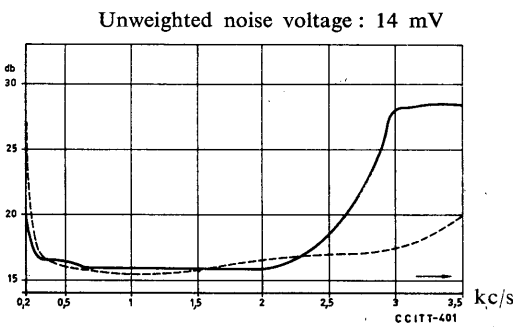


FIGURE 4b. — **Attenuation curve of test loop 2 (dialled-up connection)**

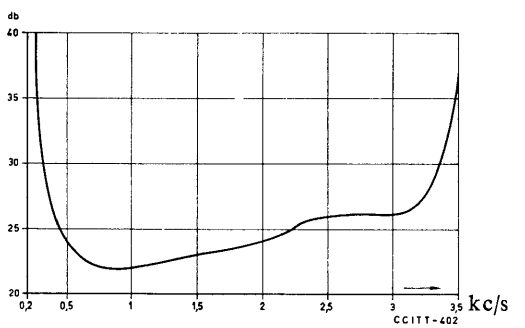


FIGURE 4c. — **Attenuation curve of test loop 3 (dialled-up connection)**

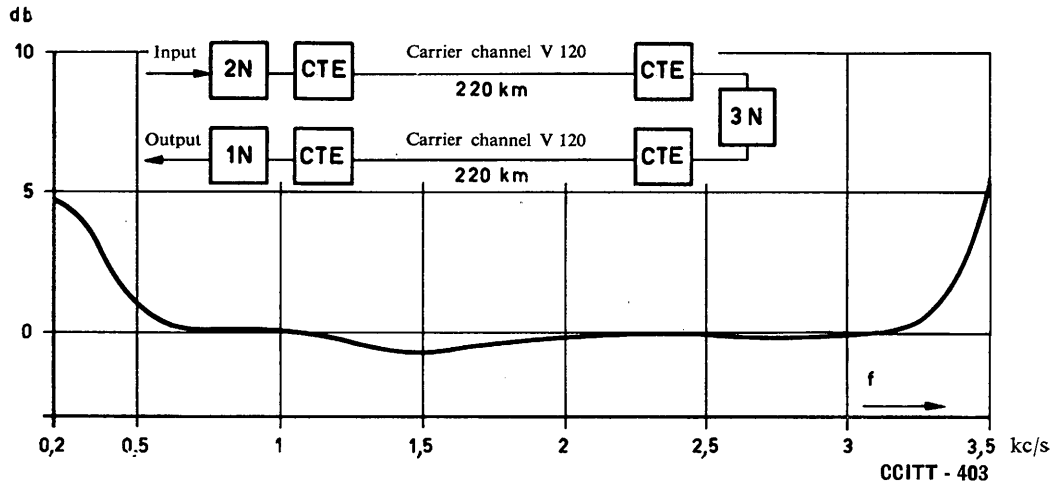


FIGURE 5a. — **Test loop I (rented connection)**

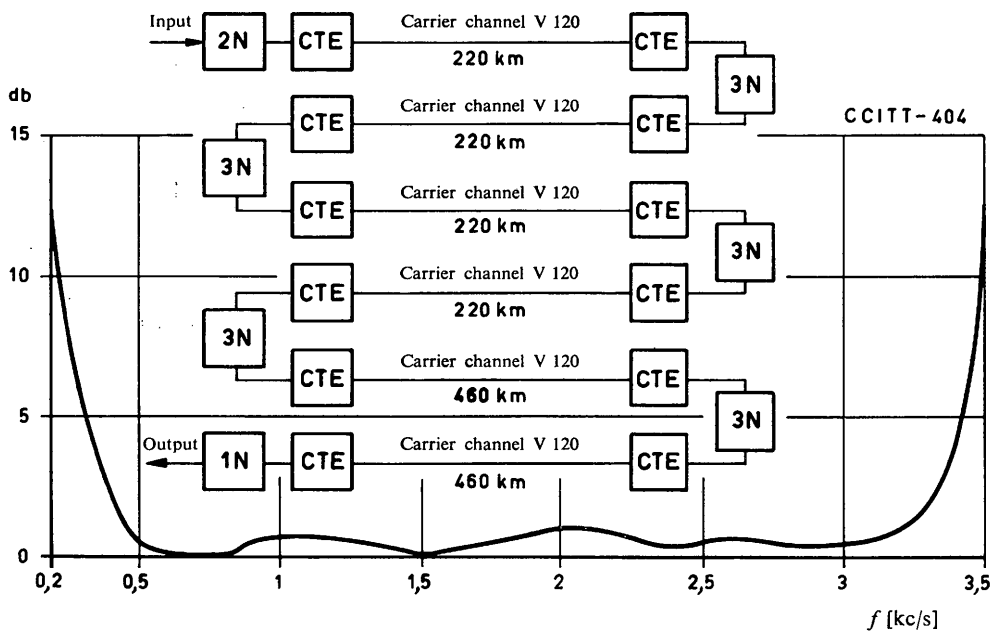


FIGURE 5b. — **Test loop II (rented connection)**

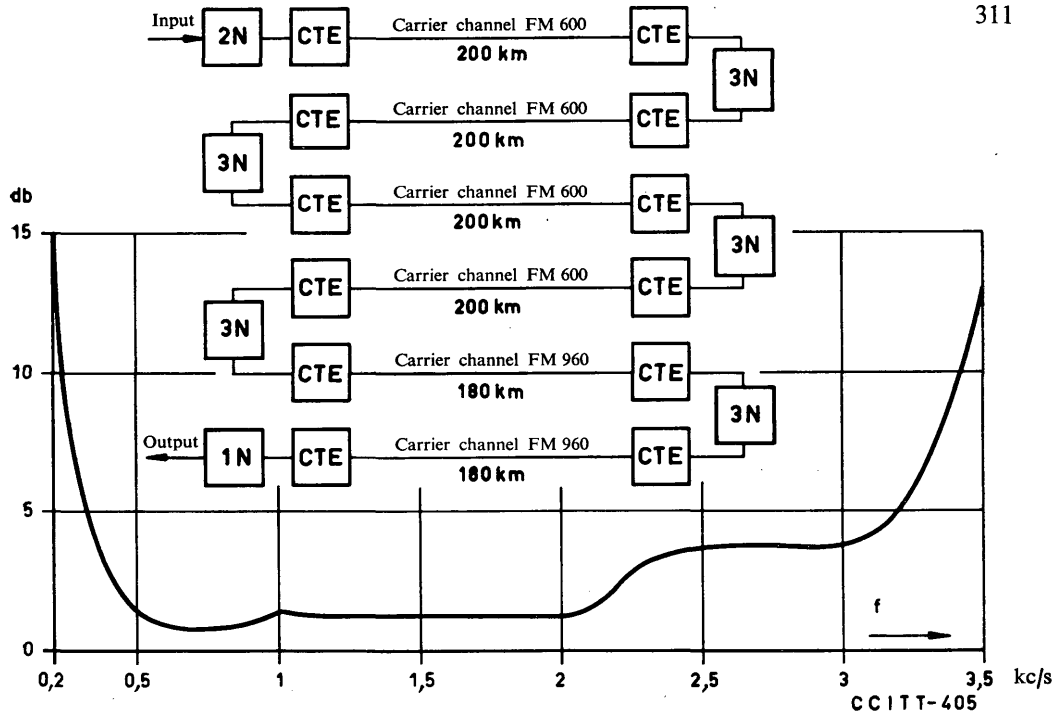


FIGURE 5c. — **Test loop III (rented connection)**

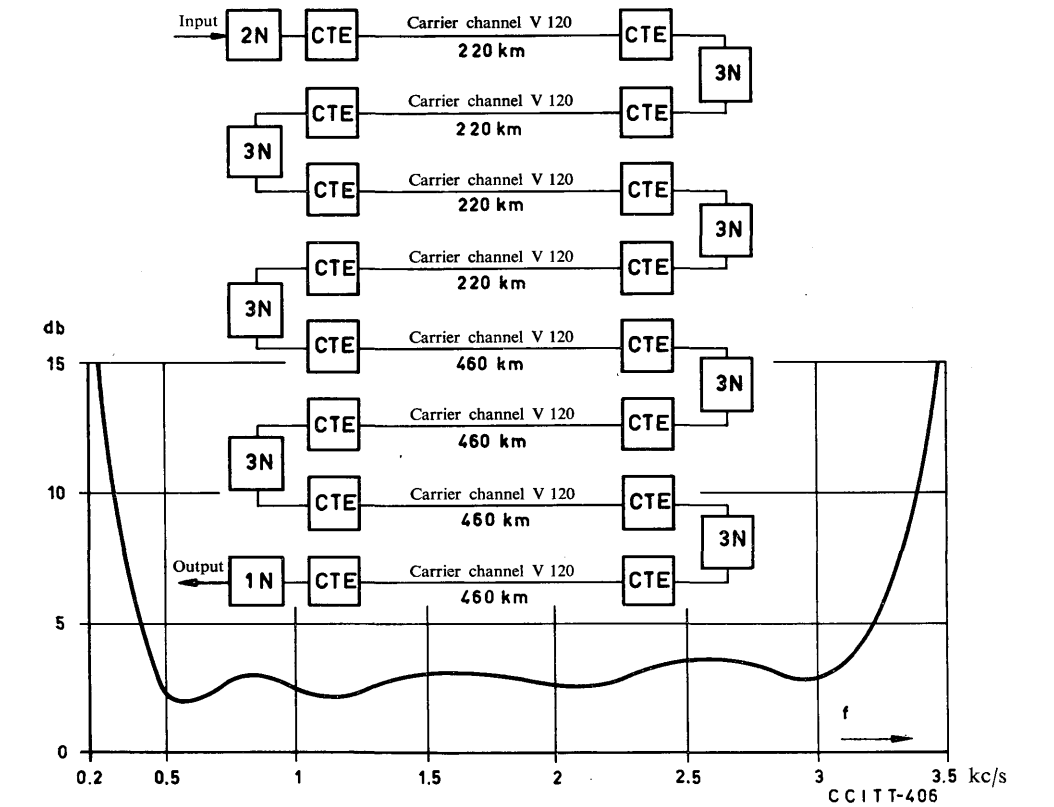
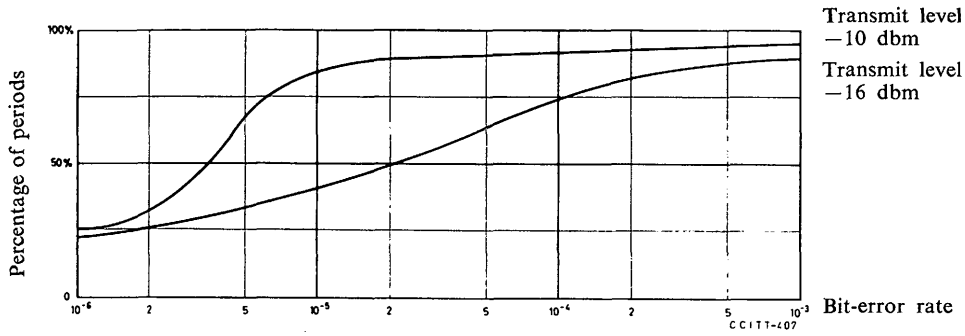
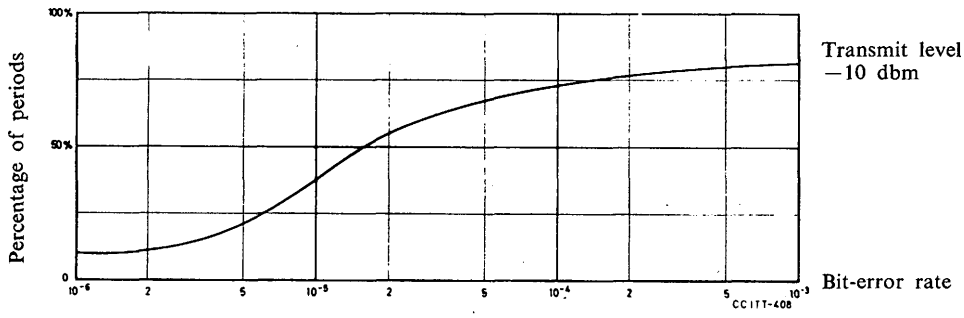


FIGURE 5d. — **Test loop IV (rented connection)**



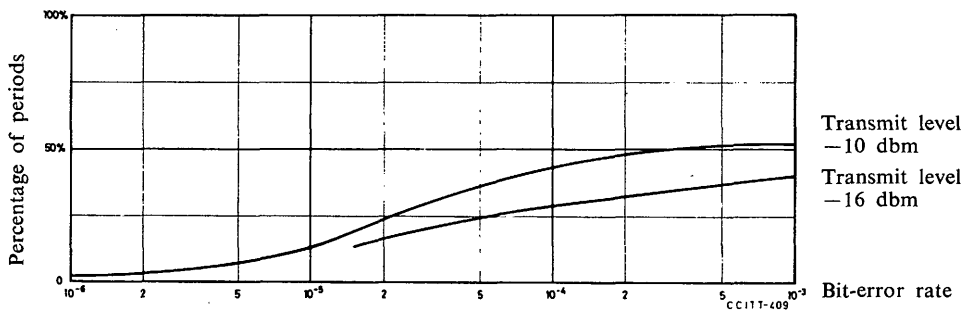
Test loop I

FIGURE 6a. — **Cumulative error-rate distribution of periods of 15 minutes (percentage of periods having an average error rate equal to or better than shown on the abscissa)**



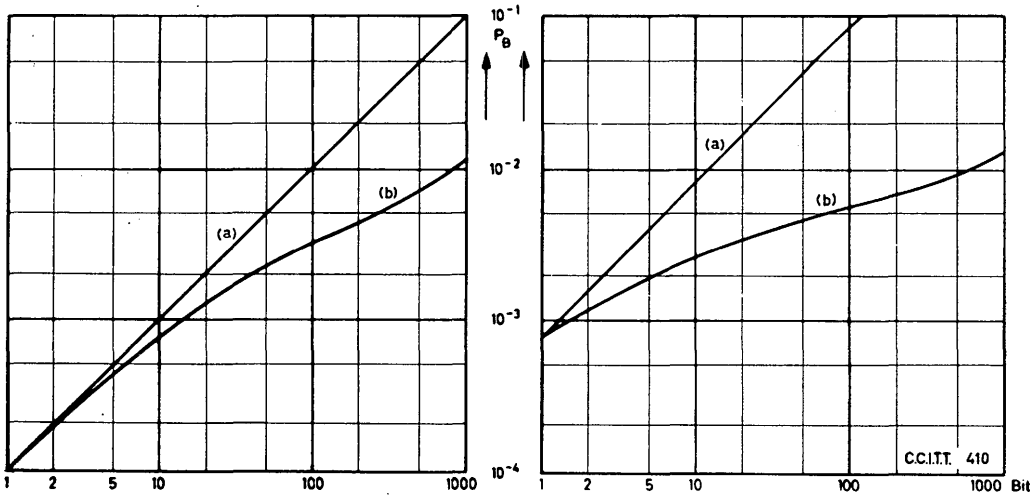
Test loop II

FIGURE 6b. — **Cumulative error-rate distribution of periods of 15 minutes (percentage of periods having an average error rate equal to or better than shown on the abscissa)**



Test loop III

FIGURE 6c. — **Cumulative error rate distribution of periods of 15 minutes (percentage of periods having an average error rate equal to or better than shown on the abscissa)**



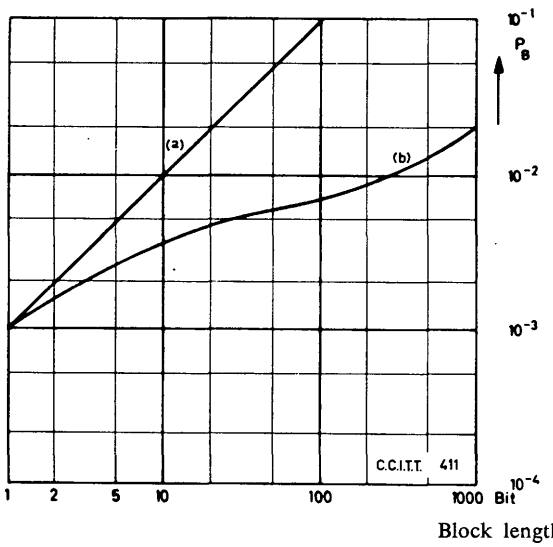
(a) = Equiprobable distribution
(b) = Measured distribution

Test loop I
Transmit level −10 dbm

(a) = Equiprobable distribution
(b) = Measured distribution

Test loop I
Transmit level −16 dbm

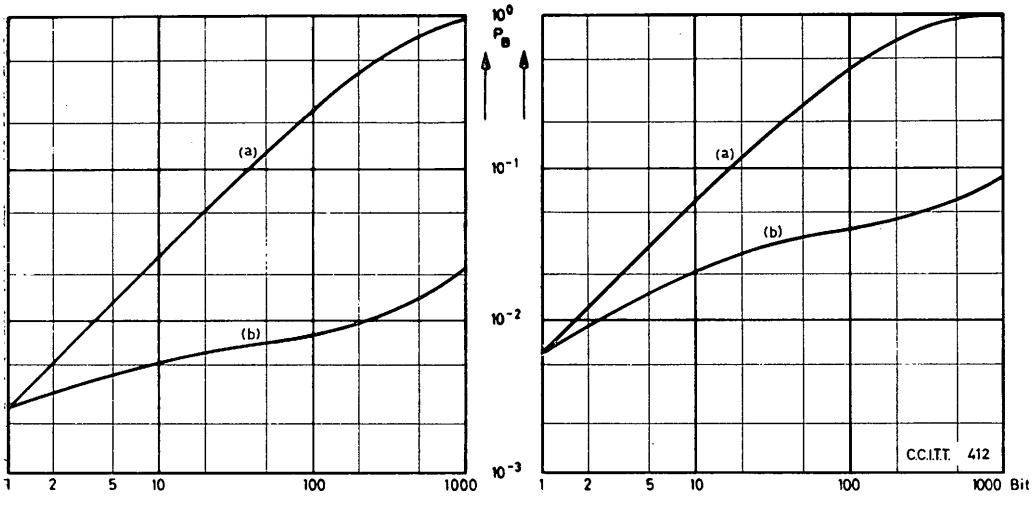
FIGURE 7a. — **Block-error rate P_B as a function of the block length**



(a) = Equiprobable distribution
(b) = Measured distribution

Test loop II
Transmit level −10 dbm

FIGURE 7b. — **Block-error rate P_B as a function of the block length**



(a) = Equiprobable distribution
(b) = Measured distribution

Test loop III
Transmit level −10 dbm

(a) = Equiprobable distribution
(b) = Measured distribution

Test loop III
Transmit level −16 dbm

FIGURE 7c. — **Block-error rate P_B as a function of the block length**

TABLE 1

Survey of test results

Test loop	Transmit level dbm	Number of millions transmitted			Number of elements in error			Element-error rate 10^{-6}			Block-error rate per block-length $\times 10^{-3}$ *d*						
		bits $\times 10^6$	O elements $\times 10^6$	L elements $\times 10^6$	Total	O	L	Total [a]	O [b]	L [c]	16	32	64	128	256	512	1024
1	— 10	47.62	25.6	22.0	4 977	3 723	1 254	104	145	57	1.1	1.7	2.5	3.5	5.0	7.3	12
1	— 16	29.9	16.1	13.8	23 804	19 179	4 625	818	1 190	335	3.2	3.9	4.3	5.9	7.6	8.7	13
2	— 10	32.2	17.3	14.9	32 178	18 969	13 209	995	1 095	888	4.4	5.3	6.2	7.4	9.3	12.9	20.1
3	— 10	104.5	56.2	48.3	277 841	181 834	96 007	2 850	3 230	1 990	6.0	6.7	7.4	8.4	10.6	14.5	22
3	— 16	7.06	3.8	3.26	42 941	29 617	13 324	6 082	7 800	4 090	25.9	31.0	36.0	41.2	48.7	61.8	86.5
I	— 10	104	55.9	48.1	37	9	28	0.36	0.16	0.58	0.005	0.009	0.02	0.04	0.07	0.15	0.30
I	— 16	104	55.9	48.1	13	5	8	0.13	0.09	0.17	0.002	0.003	0.007	0.01	0.03	0.06	0.11
II	— 10	292	157	135	67	36	31	0.23	0.23	0.23	0.002	0.003	0.006	0.01	0.02	0.04	0.07
III	— 10	192	103	89	246	114	132	1.3	1.1	1.5	0.015	0.03	0.06	0.11	0.22	0.43	0.81
IV	— 34	69	37	32	546	280	266	7.9	7.5	8.4	0.09	0.18	0.36	0.70	1.4	2.6	5.1

[a] Elements in error per transmitted bits.
[b] O-elements in error per transmitted O-elements.
[c] L-elements in error per transmitted L-elements.
[d] Blocks in error per transmitted blocks.

TABLE 2

Cumulative error pattern distribution of erroneous blocks

(Percentage of erroneous blocks having not more errors than indicated by the first line of the table)

Test loop 1	Transmit level −10 dbm															
Block length *n*	Number of errors within the block															
	1	2	3	4	5	6	7	8	9	10	11-12	13-15	16-20	21-25	26-30	> 30
16	62	83.5	92.5	97	99.5	99.8	100									
32	52.5	75	87	92.5	95	97	99	100								
64	42	64	75	83	88.9	92.8	95.5	96.4	98.2	98.9	99.8	100				
128	33.6	54.2	66	73.4	80	83.2	86	87.6	89.4	91.3	93.7	97.6	100			
256	27.6	46	56.6	66.8	73.8	77.9	80.6	83		87		91.1	93.3	96.3	99.8	
512	26	42.3	51.9	58.7	65.9	72.5		77.6		82.6	87	90.4	92.8	93.7	94.7	
1024	26.1	41.8	50.6	56.6	63.3	67.8		72.8		78.6	83.3	88	92.1	93.2	94.4	

TABLE 3

Cumulative error pattern distribution of erroneous blocks

(Percentage of erroneous blocks having not more errors than indicated by the first line of the table)

Test loop 1	Transmit level −16 dbm															
Block length *n*	Number of errors within the block															
	1	2	3	4	5	6	7	8	9	10	11-12	13-15	16-20	21-25	26-30	> 30
16	23.9	43.3	58.8	72	81.5	89	93	95	96.9	98.7	100					
32	16.1	27.4	37.9	48.3	56.3	65.1	70.1	75.8	80.5	85.9	92.8	97.0		100		
64	17.1	28.0	35.6	41.1	46.6	51.3	57.1	61.2	66.2	70.3	75.6	82.6	90.2	95.3	98.2	100
128	16.6	24.0	30.8	36.8	40.0	42.3	45.6	48.1	49.9	51.1	54.9	62.0	70.1	77.4	80.1	100
256	18.1	25.1	30.6	35.1	38.4	40.9	43.0	45.6	47.7	49.6	53.1	56.1	61.4	66.3	70.2	100
512	23.8	30.5	36.4	39.9	42.7	43.5	45.1	46.9	48.9	50.9	54.6	56.9	58.1	60.4	61.6	100
1024	25.0	33.8	39.5	44.4	47.5	49.3	51.1	53.2	54.2	55.7	58.0	60.8	64.7	67.5	69.6	100

TABLE 4

Cumulative error pattern distribution of erroneous blocks

(Percentage of erroneous blocks having not more errors than indicated by the first line of the table)

Test loop 2	Transmit level −10 dbm															
Block length *n*	Number of errors within the block															
	1	2	3	4	5	6	7	8	9	10	11-12	13-15	16-20	21-25	26-30	> 30
16	20.0	39.7	54.6	67.4	77.4	84.7	89.8	93.5	96.8	98.1	99.7	100				
32	13.3	24.1	34.3	43.6	51.4	60.9	68.8	75.4	80.9	86.1	91.6	95.1	100			
64	12.2	19.4	22.7	29.4	33.2	36.7	41.9	45.8	50.5	53.8	64.0	74.3	88.3	95.3	96.4	100
128	13.5	21.1	25.6	29.8	32.4	34.5	36.7	38.4	39.5	41.3	44.0	49.7	60.3	72.5	82.3	100
256	16.6	24.3	29.9	34.2	36.6	38.1	41.7	43.3	44.2	45.2		49.6	52.8	55.5	58.2	100
512	20.5	29.3	35.1	40.6	42.8	45.7	48.3	49.7	51.1	52.3	54.3	57.8	60.8	62.2	64.3	100
1024	24.2	34.2	40.4	45.8	49.1	51.5	54.3	55.5	57.0	57.8	59.8	63.8	67.7	70.4	72.5	100

TABLE 5

Cumulative error pattern distribution of erroneous blocks

(Percentage of erroneous blocks having not more errors than indicated by the first line of the table)

Test loop 3	Transmit level −10 dbm															
Block length *n*	Number of errors within the block															
	1	2	3	4	5	6	7	8	9	10	11-12	13-15	16-20	21-25	26-30	> 30
16	17.9	33.5	46.8	60.0	71.5	80.5	87.3	92.5	95.9	98.5	99.7	100				
32	11.9	20.2	27.8	35.0	43.4	52.1	57.3	64.5	71.0	76.8	86.4	94.5	100			
64	13.7	20.2	23.9	27.2	30.8	34.4	37.9	41.1	44.0	47.6	55.0	66.3	81.7	92.0	98.8	100
128	17.8	25.1	28.5	31.5	33.6	35.6	37.0	38.1	39.7	41.4	43.7	47.2	53.5	63.5	73.0	
256	22.5	31.4	35.8	38.8	40.9	42.8	44.2	45.1	45.9	46.7	48.3	50.3	52.7	55.4	57.8	
512	26.2	37.8	42.7	46.3	48.6	50.5	52.2	53.3	54.3	55.1	56.7	58.3	60.5	62.3	63.3	
1024	29.9	42.5	49.0	53.3	56.0	58.3	60.2	61.8	62.8	63.7	65.3	67.0	69.0	70.9	71.8	

TABLE 6

Cumulative error pattern distribution of erroneous blocks

(Percentage of erroneous blocks having not more errors than indicated by the first line of the table)

Test loop 3	Transmit level −16 dbm															
Block length *n*	Number of errors within the block															
	1	2	3	4	5	6	7	8	9	10	11-12	13-15	16-20	21-25	26-30	> 30
16	18.7	34.9	50.3	67.2	79.7	87.8	93.2	96.5	98.0	98.9	99.8	100				
32	13.5	21.9	29.6	36.5	46.0	55.3	64.4	73	79.8	85.2	94	98.5	100			
64	10.6	16.9	23.1	27.6	31.5	35.5	38.9	42.4	45.7	50.2	59.0	72.6	89.1	98.1	99.9	100
128	10.9	16.9	21.8	24.1	27.1	29.9	32.7	35.6	37.5	39.3	41.8	45	54.7	66.7	78.1	100
256	13.9	19.3	23.1	25.3	28	30	32.1	33.6	34.7	36.1	38.3	41.4	47.5	50.3	52.8	100
512	18	24.9	29.3	31.7	34.6	35.9	37.3	38.3	40.4	41.5	43.1	45.4	47.9	51.3	53.0	100
1024	22.6	30.8	36.1	38.8	41.5	44.2	45.3	46.3	47.8	49.2	50.5	52.4	54.7	57.5	58.7	100

Supplement No. 32

NETHERLANDS — *(Extract from contribution COM Sp. A/No. 18 — September 1961)*

TESTS OF SYSTEMS FOR DATA TRANSMISSION VIA TELEPHONE CIRCUITS

In co-operation with IBM, a data-transmission circuit was established via a number of dialled circuits in the Netherlands telephone system. The number of elements in error was counted. The results give a general impression of the possibilities in the Netherlands system.

Both local (Amsterdam) and trunk circuits were chosen.

Most of the trunk circuits included long local sections (up to 10 km).

The modem used was the A.T.T. FM subset, model IIIA (frequency modulation with zero-crossing). The transmission speed was 1200 bauds, the frequencies used were 1200 and 2200 c/s.

A compromise attenuation and delay equalizer was provided, which had been adapted to the Netherlands system. The transmission level amounted to −6 dbm. A group of 8 elements was continually repeated. The number of elements in error was counted. Errors were established by comparison of the corresponding elements of two subsequent groups.

The tests were exclusively effected during the daytime on ordinary working days and were spread over the whole day.

A total number of 43 circuits was selected of which 13 were local ones. The observation time varied from about 30 to 120 minutes per circuit. The element-error rate was as follows:

7 circuits between 0.1 and 1×10^{-5}

27 circuits between 1 and 10×10^{-5}

8 circuits between 10 and 100×10^{-5}

1 circuit 150×10^{-5}

There was no essential difference between the numbers of errors on the local and trunk circuits. In most cases in which there was a high percentage of errors many interruptions (bad contacts ?) were found to be the cause.

The division of errors according to time was also observed.

It appeared that the errors often occurred in bursts. An example that is characteristic for all observations is given below:

Number of elements transmitted: approx. 6×10^{6}

Number of elements received in error: 1127

Element error rate: 19×10^{-5}

Classification of the errors :

27 times	single error per burst
48 times	double error per burst
43 times	3-fold error per burst
27 times	4-fold error per burst
17 times	5-fold error per burst
13 times	6-fold error per burst
6 times	7-fold error per burst
7 times	8-10-fold error per burst
6 times	10-15-fold error per burst
5 times	16-20-fold error per burst
4 times	20-30-fold error per burst
3 times	30-50-fold error per burst
Once	114-fold error per burst
Total 207	

Supplement No. 33

DENMARK — *(Contribution COM Sp. A/No. 23 — October 1961)*

DATA TRANSMISSION TESTS ON SWITCHED TELEPHONE CIRCUITS

1. *Introduction*

The results obtained can be regarded as representative of the telephone network of the country as a whole. It should be noted, however, that, wherever it has been considered relevant, distinction has been made between exchanges with selector equipment comprising precious contact material (named type 1) and exchanges with selector equipment comprising base contact material (named type 2).

2. *Test equipment*

For the tests, data transmission equipment which is called "Data-Phone" (Digital Subset model 3A) was used. The equipment works with frequency modulation with the frequencies of 1200 c/s and 2200 c/s for the digits 0 and 1, respectively. Modulation rates up to 1200 bauds can be used.

3. *Conduct of tests*

All tests were made on direct dialled connections so that on all lines tests were carried out on the basis of calls from the transmitting station to the receiving station as well as from the receiving station to the transmitting station.

4. *Signal combination, lengths of blocks*

For the tests was used the C.C.I.T.T. standard text (see C.C.I.T.T. *Red Book*, Volume VII, Recommendation R.51, page 43).

The stopping signal being 2 bits, the length of the block will be 64 bits (a 64-bit block : 0111111101010011000010110010001101110111011011110001001100001111).

By combining four blocks of 64 bits each the block length of 256 bits is obtained (a 256-bit block).

5. *Modulation rates*

Tests were made at the rates 1200 and 600 bauds.

6. *Signal levels*

In all calls on both transmission speeds, tests were carried out at the transmission levels −4 dbm, −10 dbm, and −16 dbm.

7. *Calls and transmissions*

Thirty-four calls were made in connection with the tests. On each connection, six transmissions were carried out, one transmission for each transmission level and for each transmission speed having been made.

8. *Test period*

The tests were made during the period 4-31 July 1961, between 8.30 a.m. and 4 p.m. on weekdays.

9. *Amount of traffic*

The amount of traffic which forms the basis of the following tables is per transmission :

at 1200 bauds : 9.50×10^5 bits
at 600 bauds : 4.75×10^5 bits

Comment on the measurement results

Table 1

A comparison between the transmissions at the two transmitting speeds used shows no distinctive difference in the relative error frequencies, whereas the number of faultless transmissions is somewhat larger at 600 bauds than at 1200 bauds. At 600 bauds the errors are concentrated on the very poor calls to a larger extent than at 1200 bauds.

Table 2

The influence of the transmitting level on the error frequency has been examined in Table 2. It will be seen that there is no particular difference in the average error frequencies at −4 dbm and at −10 dbm, whereas the error frequency is much increased if the transmitting level is lowered to −16 dbm.

Table 3

Here a comparison has been made between calls established between type 1 exchanges and calls between exchanges of types 1 and 2. Both transmitting stations and six of the eight receiving stations belong to type 1 exchanges. The schedule shows that the error frequency is highest in circuits incorporating type 2 exchanges. A contributory reason for this is that base contact material has been used in type 2 exchanges, and that these exchanges are of older makes. It must, however, be noted that all calls incorporating a type 2 exchange are trunk calls, whereas one-sixth of the calls between exchanges of type 1 are local calls.

Table 4

Calls, which are made from a data transmitter to a data receiver, are different in principle from calls made in the opposite direction if either the exchanges of the transmitter and the receiver are of different type or if debiting pulses are transmitted on the calling subscriber's speaker circuit. Table 4 shows that the error frequency is highest if the receiver is the calling subscriber (the debiting pulses are added where the signal level is lowest). At the playing back it was established that the debiting pulses caused error bursts.

TABLE 1

Comparison of tests made at different modulation rates

Modulation rate		1200 bauds					600 bauds					Total				
With the relative bit error frequency lower than h, n transmissions are obtained, equal to m%	h	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}
	n	35	56	72	94	100	48	67	83	94	100	83	123	155	188	200
	m	35	56	72	94	100	48	67	83	94	100	42	62	78	94	100
Number of transmitted bits/10^6		33.3	53.2	68.4	89.3	95.0	22.8	31.8	39.4	44.7	47.5	56.1	85.0	107.8	134.0	142.5
Number of bits in error		0	72	633	8 013	18 105	0	37	338	2 779	8 352	0	109	971	10 792	26 457
The relative bit-error frequency/10^{-6}		0	1.4	9.3	90	190	0	1.2	8.6	62	180	0	1.3	9.0	81	190
Number of transmitted 64-bit blocks/10^4		52.0	83.1	106.9	139.6	148.5	35.6	49.7	61.6	69.8	74.2	87.6	132.8	168.5	209.4	222.7
Number of 64-bit blocks in error		0	62	393	2 608	4 091	0	32	279	1 184	2 127	0	94	672	3 792	6 218
The relative 64-bit block error frequency/10^{-4}		0	0.75	3.7	19	28	0	0.64	4.5	17	29	0	0.76	4.0	18	28
Number of transmitted 256-bit blocks/10^3		130	208	267	349	371	89.1	124.3	154.0	174.5	185.6	219	332	421	524	557
Number of 256-bit blocks in error		0	61	328	2 177	3 198	0	32	264	1 003	1 720	0	93	592	3 180	4 918
The relative 256-bit block-error frequency/10^{-3}		0	0.29	1.2	6.2	8.6	0	0.26	1.7	5.7	9.3	0	0.28	1.4	6.1	8.8

TABLE 2

Comparison of tests made at different transmitting levels

Transmitting level		− 4 dbm					− 10 dbm					− 16 dbm					Total
With the relative bit error frequency lower than h, n transmissions are obtained, equal to m%	h	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-2}
	n	39	48	56	63	67	30	42	54	64	67	14	33	45	61	66	200
	m	58	72	84	94	100	45	63	81	95	100	21	50	68	92	100	—
Number of transmitted bits/10^6		26.6	33.3	39.4	45.6	48.0	20.9	28.5	37.5	45.1	47.5	8.6	23.3	30.9	43.2	47.0	142.5
Number of bits in error		0	25	264	2 793	6 558	0	25	301	2 276	6 913	0	59	406	5 723	12 986	26 457
The relative bit-error frequency/10^{-6}		0	0.75	6.7	61	140	0	0.88	8.0	51	150	0	2.5	13	130	280	190
Number of transmitted 64-bit blocks/10^4		41.6	52.0	61.6	71.3	75.0	32.7	44.5	58.6	705	74.2	13.4	36.4	48.3	67.6	73.5	222.7
Number of 64-bit blocks in error		0	23	149	858	1 489	0	19	212	765	1 341	0	52	311	2 169	3 388	6 218
The relative 64-bit block error frequency/10^{-4}		0	0.44	2.4	12	20	0	0.43	3.6	11	18	0	1.4	6.4	32	46	28
Number of transmitted 256-bit blocks/10^3		104	130	154	178	187	82	111	147	176	186	33	91	121	169	184	557
Number of 256-bit blocks in error		0	23	121	726	1 169	0	19	182	667	1 112	0	51	289	1 787	2 637	4 918
The relative 256-bit block error frequency/10^{-3}		0	0.18	0.79	4.1	6.3	0	0.17	1.2	3.8	6.0	0	0.56	2.4	11	14	8.8

TABLE 3

Comparison of tests between exchanges of type 1 and tests between exchanges of types 1 and 2

Type 1 : precious-metal contacts
Type 2 : base-metal contacts

Types of exchanges used		Type 1					Type 1 and type 2					Total
With the relative bit error frequency lower than h, n transmissions are obtained, equal to m %	h	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-2}
	n	82	116	132	145	153	1	7	23	43	47	200
	m	54	75	86	95	100	2	16	49	92	100	—
Number of transmitted bits/10^6		55.6	81.2	93.1	103.1	108.8	0.5	3.8	14.7	30.9	33.7	142.5
Number of bits in error		0	92	479	4 931	16 933	0	17	492	5 861	9 524	26 457
The relative bit-error frequency/10^{-6}		0	1.1	5.1	48	160	0	4.5	33	190	280	190
Number of transmitted 64-bit blocks/10^4		86.9	127.0	145.5	161.1	170.0	0.7	5.9	23.0	48.3	52.7	222.7
Number of 64-bit blocks in error		0	77	296	1 833	3 603	0	17	376	1 959	2 615	6 218
The relative 64-bit block error frequency/10^{-4}		0	0.61	2.0	11	21	0	2.9	16	41	50	28
Number of transmitted 256-bit blocks/10^3		218	317	364	403	425	2	15	58	121	132	557
Number of 256-bit blocks in error		0	76	241	1 472	2 764	0	17	351	1 708	2 154	4 918
The relative 256-bit block-error frequency/10^{-3}		0	0.24	0.66	3.7	6.5	0	1.1	6.1	14	16	8.8

TABLE 4

Comparison of the tests divided according to call direction

Call direction		Receiver to transmitter					Transmitter to receiver					Total
With the relative bit-error frequency lower than h, n transmissions are obtained, equal to m %	h	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-6}	10^{-5}	10^{-4}	10^{-3}	10^{-2}	10^{-2}
	n	38	53	70	90	99	45	70	85	98	101	200
	m	38	54	71	91	100	45	69	84	97	100	—
Number of transmitted bits/10^6		26.1	36.6	48.0	64.1	70.3	29.9	48.5	59.9	69.8	72.2	142.5
Number of bits in error		0	44	550	6 457	18 926	0	65	421	4 335	7 531	26 457
The relative bit-error frequency/10^{-6}		0	1.2	11	100	270	0	1.3	7.0	62	100	190
Number of transmitted 64-bit blocks/10^4		40.8	57.2	75.0	100.2	109.9	46.8	75.7	93.5	109.1	112.8	222.7
Number of 64-bit blocks in error		0	38	383	2 413	4 209	0	56	289	1 379	2 009	6 218
The relative 64-bit block-error frequency/10^{-4}		0	0.66	5.1	24	38	0	0.74	3.1	13	18	28
Number of transmitted 256-bit blocks 10^3		102	143	187	251	275	117	189	234	273	282	557
Number of 256-bit blocks in error		0	38	364	2 031	3 337	0	55	228	1 149	1 581	4 918
The relative 256-bit block error frequency/10^{-3}		0	0.27	1.9	8.1	12	0	0.29	0.97	4.2	5.6	8.8

Supplement No. 34

I.B.M. (W.T.E.C.) — *(Contribution COM Sp. A/No. 6 — August 1961)*

EXTRACTS FROM THE I.B.M. (W.T.E.C.) CONTRIBUTION No. 1 ON DATA TRANSMISSION

1.1

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1.2 In the future, a theoretical frontier will be established between the telecommunication field and data-processing equipments; computer manufacturers have to know the statistical distribution of errors entering data-processing systems. The computer manufacturer is responsible to his customer for the over-all system performance; he has to make the customer well aware of the transmission efficiency and enable him to establish the computer programmes necessary to solve his problems. In other words, the programmer has to know how much redundancy in the transmitted data and what checking sub-routines must be applied in order to fit the safety conditions of his problems. Transmission efficiency has to be expressed under two main aspects: undetected error rate and actual speed (taking into account redundancy and checking).

1.3

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2. *Test apparatus and methods*

Tools of various power were used for the test programme. The apparatus belong to several groups:

Character or message generators with associated error checkers at the receiving end.
We have considered the clocking system and timing reconstruction as being included in this digital apparatus.

Modulators-demodulators (modems): various PM modems and one FM modem were used[1].

Recording apparatus: The demodulated signals were recorded on dual tracks audio recorders.

Data reduction: We call "data reduction" the conversion of raw recorded data into condensed form. The final set of data represents the successive distances between errors and their location in the message.

Statistical analysis: Final statistical analysis was performed on an I.B.M. 704 computer and the programmes are available for an I.B.M. 7090.

2.1 *Message generators*

Three kinds of message generators were used:

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[1] PM is phase modulation;
FM is frequency modulation.

2.1.1 *Eight-bit character generator*

This equipment sends continuously the same character without gaps. The bit sequence is adjusted manually by the operator. The repetitive conditions enable the study of waveforms with an oscilloscope in any point of the transmission link.

The checking of the messages at the receiver end is done by comparison of two incoming messages.

This simple apparatus enables the study of the worst bit sequences and affords good results for the error rate with a particular message.

2.1.2 *A pseudo-random character generator*

In order to get results with random messages, a pseudo-random character generator was built in several samples. This equipment delivers a 511-bit message, the sequence of which is given by a feedback-loop shift register, the ninth position being added (modulo two) to the fifth position (Figure 1).

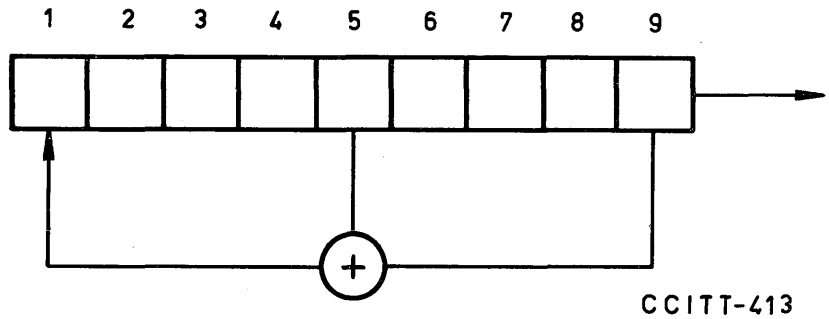


FIGURE 1

The output of the ninth position feeds the character generator. This system furnishes all possible sequences of 9 bits (except 9 zeros).

At the receiving end, a similar device compares the incoming sequence with a local pseudo-random generator and delivers error signals.

The receiver clock is reconstructed through an LC network synchronized by the data and the system has a small inertia.

2.1.3 *Data transmission test equipment*

The system is larger than the previous ones and works closely to an actual complete data transmission system on a half-duplex basis over two-wire telephone channels.

This data transmission test equipment or "*DATATEST*" *equipment* is able to send messages multiple of a basic 80-bit message, the content of which is set by the operator prior to the transmission. The message is sent by the transmitter terminal and compared bit by bit at the receiving end with the same message stored in this terminal. Once the message is sent, the transmitter waits for an answer and receives a short control message back from the receiving end. This message means either "correct message" or "erroneous message" according to the results of the receiver checking.

Then a new message is sent by the transmitter and the same loop of operation is started again.

Both terminals have clocking systems controlled through crystal oscillators which need an initial synchronization time ; this phase-locking operation is done at the beginning of the test period through an automatic sequence.

This test equipment enables not only the study of transmission problems in terms of error distribution, but also the study of problems related to synchronization for example :

— Clock synchronization

— Character synchronization

— Start and end of message recognition.

In addition, the variable message structure enables the study of various codes efficiency on an experimental basis. The 4-out-of-8 character code and the longitudinal parity checking were submitted to detailed investigations. A more general study was undertaken about cyclic codes and their ability for error detection and correction.

2.2 *Modulators-demodulators*

Various PM and FM modems were used, associated with the previous data generators.

2.2.1 *PM modems*[1]

Phase modulation principle is based upon the phase reversal of a carrier. The receiver reconstructs the carrier from the line signals and delivers the data signals. Several kinds of receivers were experimented according to the different way of carrier reconstruction.

Then a new message is sent by the transmitter and the same loop of operation is started again.

One of these systems is based upon a feedback loop as displayed in Figure 2.

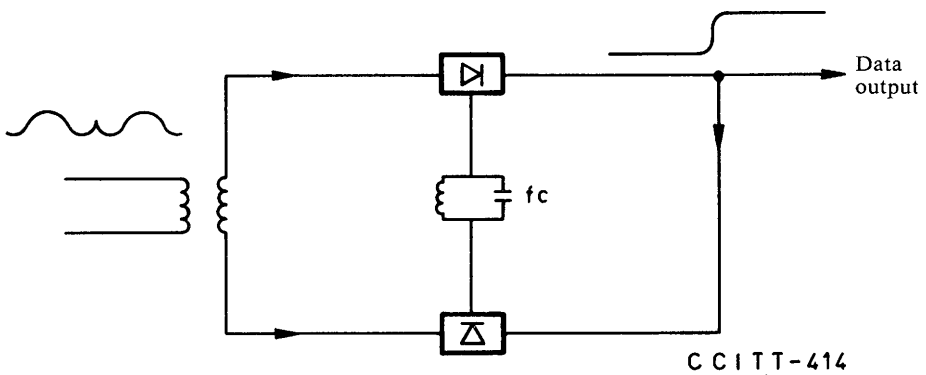


FIGURE 2

Information coding was done in two ways :

— With phase changes of the waveform or "Transition coding". This system is independent of the waveform phase entering the receiver.

— By comparison of the waveform phase with the reconstructed carrier phase.

In this latter system, it is necessary to synchronize, prior to any modulation, the reconstructed carrier phase with the waveform entering the receiver in order to have a non-ambiguous phase reference.

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[1] For reference, see Mr. E. Hopner's paper in "IBM Journal of Research and Development", April 1961.

This synchronization is done at the beginning of each message and needs a few cycles of the carrier.

Almost all our tests were performed with the first way of coding (phase transition) which necessitates a simpler equipment.

However, comparative tests have displayed similar results with equipment of each kind taking account of systematic doubling of the errors with transition coding (see below "Statistical Analysis", paragraph 2.5).

2.2.2 *FM modem*

Parts of the experiments were performed with an FM modem already available to the customers. This modem is based upon zero crossing detection at the receiving end.

2.2.3 *Hook-up of modems to the test equipment*

Hook-up of modems to test equipment was done under recommendations of last Working Party 43, that is to say under E.I.A. recommendation RS 232.

In the case of PM modems, an additional synchronization signal is sent from the data generator to the modems in order to allow a carrier generation synchronous with the bits generation. However, parts of the experiments were done asynchronously and the associated transmitter equipment modified to avoid this synchronization problem. The results are very similar in the two cases.

2.2.4 *Level of transmitted signals*

As the C.C.I.T.T. has recommended a maximum for output signal power but has not specified a definite level, government organizations and telephone companies have not requested the same level for their particular test programme. The signal levels were between 0 and −10 dbm. Special care of this parameter was taken in the statistical evaluation, because the results are obviously very much dependent on the signal levels. A good data transmission service will be ensured in the future if the transmitted power is close to the highest level compatible with the present telecommunication network.

2.3 *Recording apparatus*

Test equipment receivers are able to deliver four different digital signals which are recorded on a dual track audio-tape recorder :

— Start of message and clock signals are recorded on track one ;

— Errors and line breakdown are recorded on the second track.

The results are recorded at 3.75 inches per second in the various telecommunication centres where the tests are performed and are ready for data reduction.

2.4 *Data reduction*

All tapes recorded at the various testing locations are played back at 7.5 inches per second in the data reduction centre which delivers punched cards suitable for further statistical studies.

This has been done with a data-processing machine running under control of a special programme. This machine computes the intervals between errors in terms of the number of messages and bits and loads its memory with these successive numbers.

At the end of the process, the data-processing machine punches cards. Each card displays 15 error intervals plus a test-recording number. The consecutive numbers can represent two errors, the distance between which is from one bit (adjacent errors) to 100 000 bits. If the distance is larger than 100 000 bits, zeros are filled up in the error counting position and the next position is used for further counting.

Finally, the "data reduction" process enables the user of the punched cards to avoid the long length of time which exists between errors and prepares the data to be processed for statistical studies.

2.5 *Statistical analysis*

Once the cards have been punched, the exact location of errors within messages is known and 704 programmes were established for statistical analysis of various tests. Each test was processed separately since a general mixing of data dependent on so many different parameters of the lines would have mixed the results. However, similar phenomena were observed on different tests. The parameters vary according to the testing conditions, but certain basic statistical laws appear to be invariant.

The statistical analysis was conducted under a double scope :

— At first, to know what is the error distribution got through the transmission system.

— Secondly, to establish the efficiency of data-processing machines hooked up to transmission systems.

Programmes were established to get the following information from the raw data :

— Average error rate.

— Average number of errors per message.

— Erroneous message rate.

— Length of bursts study [1].

— Error-rate changes in respect to the time of the day.

— Error-rate in respect to the bit position in the message.

— Symmetry of errors : comparison of zero-to-one changes versus one-to-zero changes.

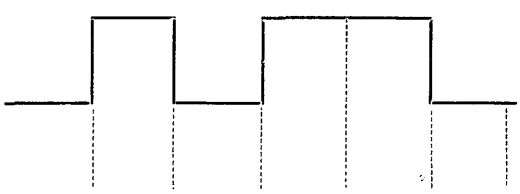
— Efficiency of the 4-out-of-8 character code.

— Efficiency of longitudinal parity checking.

— Undetected erroneous message rate with a count of erroneous messages which would have been accepted by the IBM 7701 tape to tape transmission machine.

In addition, a special sub-routine of the programme was established to convert the results gained with "No zero return" into "Zero return" bit coding. "No zero return" or NZR is a "transition coding" and "Zero return" or ZR is a "level coding".

As an example, consider the following waveform :

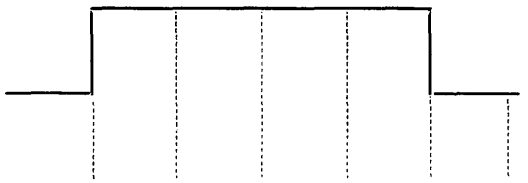


it is coded 1 1 1 0 1 0 in the first case (NZR)

or 1 0 1 1 0 0 in the second case (ZR)

[1] A single burst was classified in respect to guard spaces (or a given time interval free of errors). Two bursts are classified as different if the distance between the last bit in error of the first burst and the first bit in error of the second burst is longer than the guard space.

If the reconstructed waveform is disturbed as below :



it is coded 1 0 0 0 1 0 in NZR system

or 1 1 1 1 0 0 in ZR system

It appears that the error pattern depends on the coding selected : *the NZR system delivers systematic double errors.*

The special sub-routine permits error distribution studies of both coding systems with test runs performed with one of the two codes.

3. *Statistical results*

A very large amount of data was recorded and processed with the computers. There were performed more than :

— 750 hours of tests in the field

— 400 different tests.

The data reduction centre was running for six months with a few interruptions. Almost all possible kinds of lines were tested ; however, a great emphasis was requested by telecommunication organizations on lines which represent typical telephone networks.

Local test results will be published by telecommunication organizations if they have such a need and it is not our purpose to discuss here the relative quality of telephone lines. However, we can display the common laws encountered over various telephone networks, even with different modems ; this is of primary importance for the design of future data transmission systems, outside account of a particular telephone network requirement.

Some of the laws appear quite obvious but the statistical results permit the measurement of various parameters.

Most of the tests were performed at 1200 bauds, which seems a convenient speed for the modems used and the general switched telephone network. Over heavily loaded cables, the speed was reduced to 600 bauds. In some countries, 800 and 600 bauds were used. In all cases, the results are similar and we do not present separate curves in this summary of the general studies.

Comparative studies of PM and FM modems have confirmed the theoretical higher quality of phase modulation. However, error distribution in both cases is quite similar when using ZR coding.

3.1 *Error distribution studies*

3.1.1 The first study deals with *the distance between errors.* For each test, curves have been drawn which represent the percentage of intervals greater or equal to the length given in abscissa (in bits).

All curves have the following average shape :

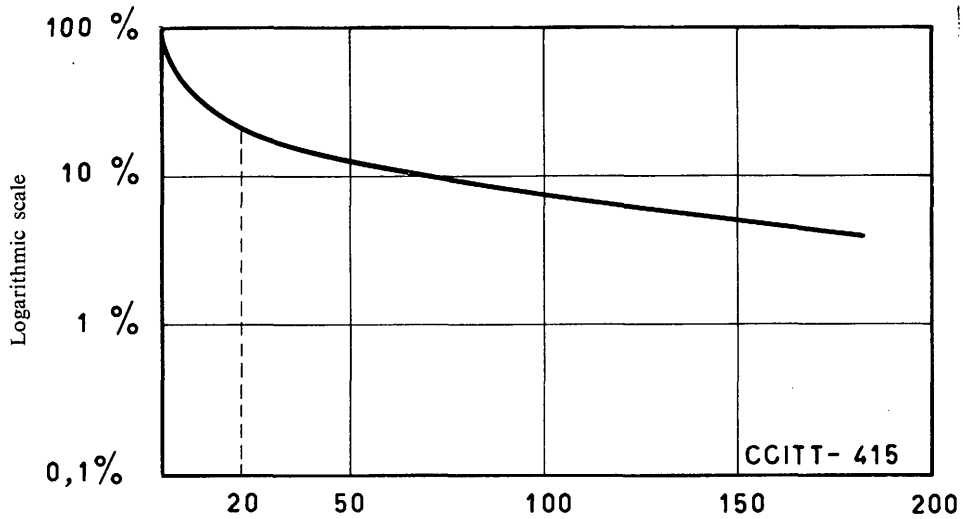


FIGURE 3

The curves display a short-range correlation up to a "guard space" of 15 to 30 bits and a Poisson distribution for larger distances.

We shall use this guard space to separate bursts of error.

3.1.2 The curve below displays the *distribution of the length of bursts* and indicates two varieties of bursts. A large amount of bursts are of two bits duration (NZR coding). This indicates disturbances of a duration less than 1/1200 second. The longer bursts have an exponential distribution in length which indicates an exponential law for the duration of disturbances.

The slope of part BC depends on the guard space used.

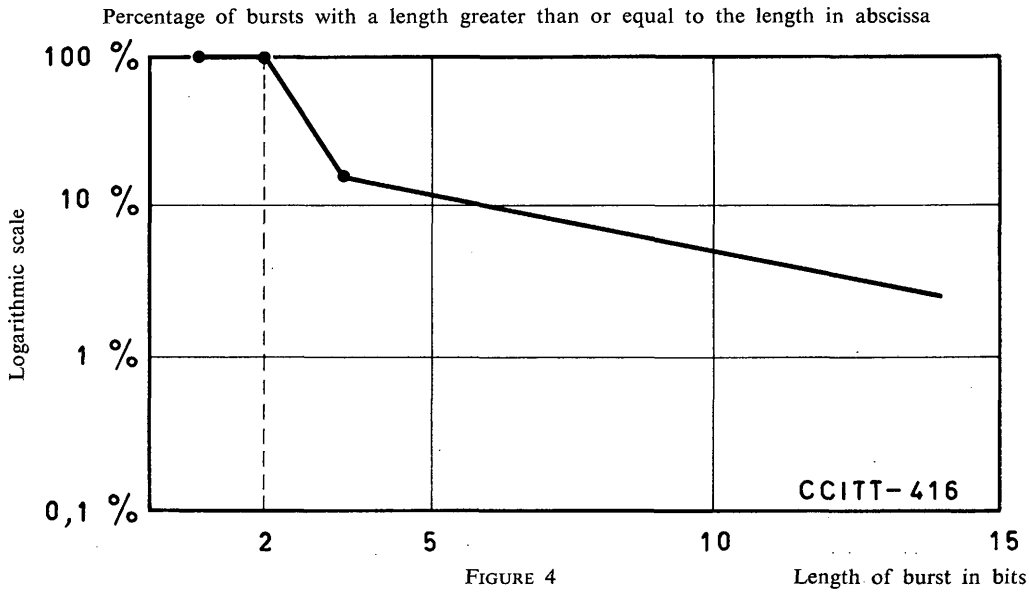


FIGURE 4

Within a burst, the frequency of *bit errors* is a little less than 0.5. This verifies the probabilistic burst models and at the same time indicates that the longer bursts correspond to strong line disturbances which are difficult to combat effectively.

Nevertheless, improvements to the modems can be used to reduce the number of single isolated errors.

3.1.3 *Bursts themselves* tend to form *bursts of bursts*

Long-range correlation curves (see below) point out a suitable error-free "guard space" between bursts of bursts of about 5 to 10 seconds.

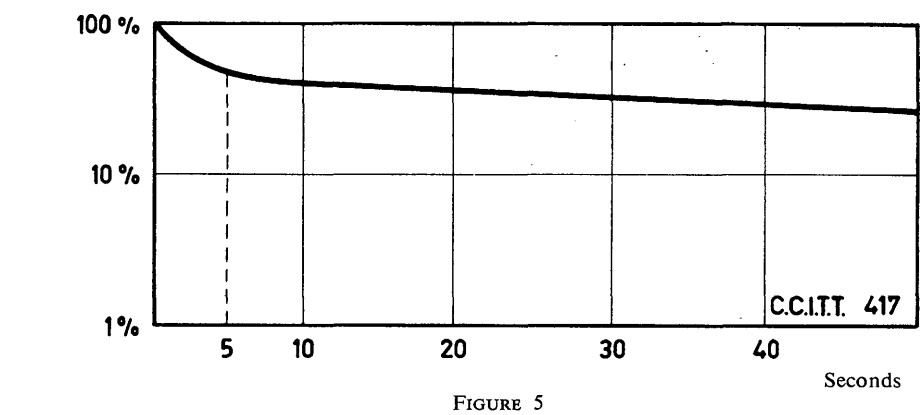


FIGURE 5

If we consider a "burst of bursts" as a disturbance and consider these disturbances as occurring randomly, then the theoretical erroneous message rates (for records longer than 800 bits) agree almost perfectly with the actual results.

3.1.4 *Error within message studies*

The content of binary messages transmitted has an important effect on the error distribution. For example in the PM systems we used, the last bit of a string of zeros has an error rate several times higher than the average rate. This can be explained by delay distortion phenomena.

In the FM system the probability of an erroneous bit is also dependent on the message structure but with different laws.

The results of this study can be used to point out the critical part of a transmission system. They can be used too for the completion of test messages, the contents of which have to involve the most critical bit patterns of possible messages. In some cases an excessively simple test pattern can be transmitted without trouble, and yet information messages cannot be properly transmitted over the same line.

3.1.5 *Time distribution of the errors*

A strong correlation of the error rate does exist with the busy hours of human activity. This phenomenon is the same for local switched lines and for long-distance calls. During daytime, the lowest error rate period is between 1 and 2 p.m. A user who can wait until the hours of least traffic can expect a better efficiency from his machine (on an average basis).

3.2 *Machine efficiency*

As an example, we shall discuss the effect of line performances for a complete system such as the I.B.M. 7701 unit.

The I.B.M. 7701 magnetic tape read-write unit can be used to send or receive digital information to or from a computer (or another 7701 unit) in a semi-duplex operation.

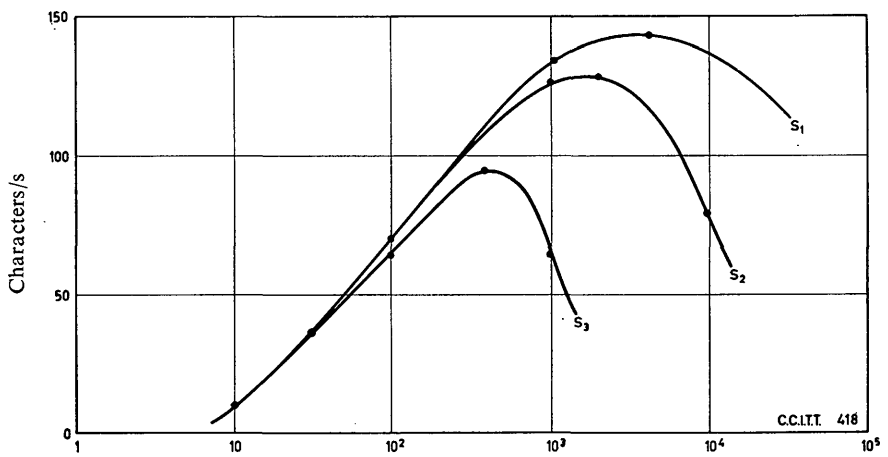
The message length is variable and can be pre-set on the magnetic tapes. At the end of a message, a longitudinal parity checking character should represent the sum modulo two of the characters' bits.

Each character is represented by the 4-out-of-8 code (four ones out of eight bit positions) and each character code is checked at the input of the receiver. The regular transmission speed of the machine is 1200 bauds or 150 characters per second.

If a received message is found erroneous, a retransmission is required of the transmitting unit.

The average bit-error rate does not mean anything to compute the machine efficiency. Even over lines with bit-error rates as high as 10^{-2}, the use of the 7701 machine appears feasible with an efficiency better than 50% of the maximum speed. The message-error rate based on independent errors is far above the actual rate, but the rate based on independent bursts of errors does represent a good approximation to the actual message transmission speed (and anyway gives a lower bound for this value).

The curve below (Figure 6) shows the effective speed of the machine versus the record length for various error bursts assumed to be independent.



S_1 Average of 1 burst every 1000 seconds
S_2 Average of 1 burst every 100 seconds
S_3 Average of 1 burst every 10 seconds

7701 — Speed of information transmission in characters per second versus message length for various error burst rates

FIGURE 6

If the average distance between bursts is D bits, the effective information transmission speed is :

$$S = \frac{n}{t + \frac{n}{B}} e - \frac{n}{D}$$

n being the message length

B the modulation speed (here 1200 bauds)

t the time gap between messages used for checking, answer back, etc.

As the error rate depends on the time of day, it is appropriate to choose a message length shorter than the average optimum because the speed variations will be smaller for short messages.

The study of machine efficiency from actual test results displays a use of good lines close to the maximum speed and a use of poor lines with a reasonable percentage of good messages transmitted. This is due to the grouping of errors within bursts. If the bit-error rate increases, the number of errors per bursts increases in parallel and the burst rate increases more slowly.

As an example, the curve below (Figure 7) shows the density of erroneous bits for 11 tests of 5 minutes each on three different lines with the same transmitted signal.

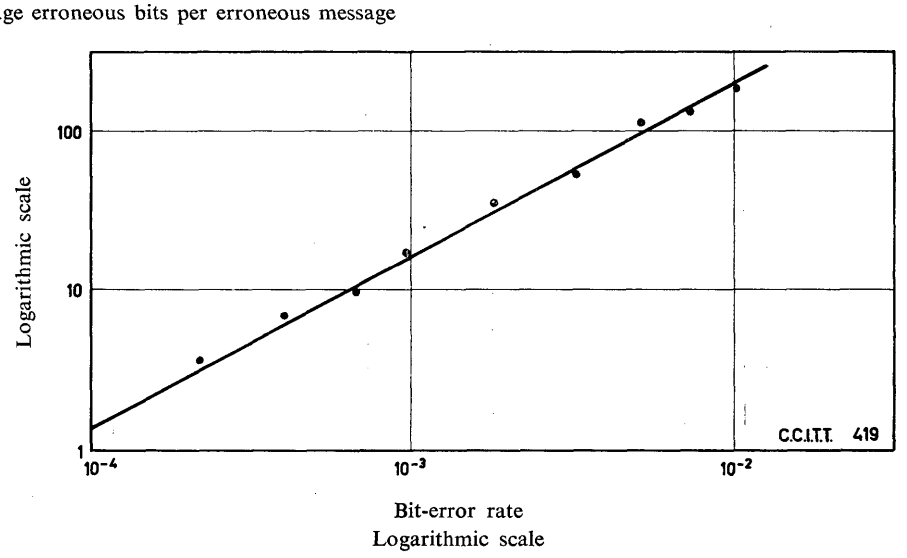


FIGURE 7

The test results are close to the straight line :

$$Y = 1.2\,(10^4\,X)^{0.87}$$

The concentration of erroneous bits into bursts leaves the greatest number of messages free of errors even if the line is poor in terms of its average bit-error rate. In the case of 511-bit

messages, 50% of switched lines are able to transmit 99% of these messages without error. Exceptionally, bad lines transmit less than 90% of messages without error. On Figure 8 is plotted the efficiency of lines in respect to various lengths of messages.

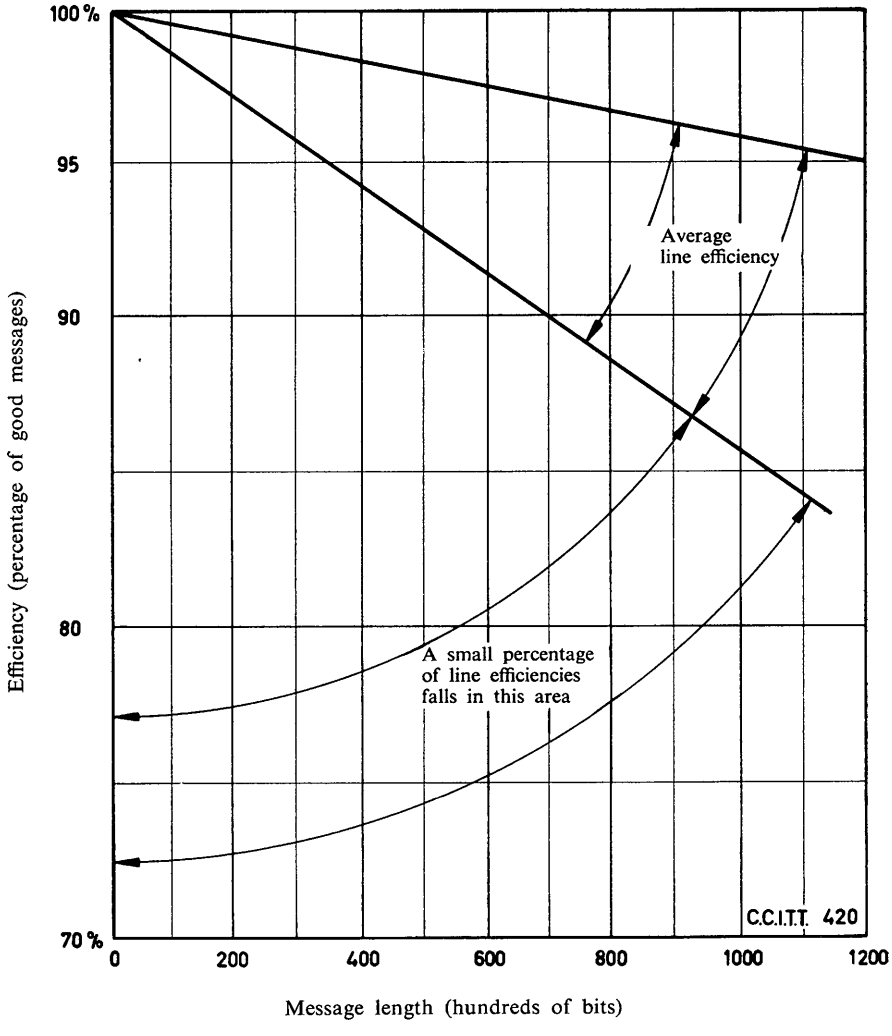


FIGURE 8

3.3 *Error detection in messages*

After the effective transmission speed, the undetected error rate is the most interesting parameter. This study involves coding structures in the message and for the 7701 machine two means are used to detect the errors :

— A 4-out-of-8 code per character,

— A longitudinal parity check character.

The longitudinal check system leaves by itself a theoretical percentage of 0.4% of erroneous messages, assuming that all error patterns are equally likely ; the statistical results give a percentage closer to 1% which is a good efficiency for such an error detection system.

The combined use of a 4-out-of-8 code (with a ZR coding) and a longitudinal character are sufficient to ensure quasi-error-free transmission by itself, that is to say undetected error rates as low as those encountered in data processing through computers. However, according to the particular problems he has to deal with, the programmer can add the necessary redundancy in the information itself to be transmitted. This is of common practice in the work done with computers in data-processing centres.

The use of new codes in the future, like cyclic codes, will enable to increase the reliability of transmission (i.e. the undetected error rate) up to the desired level. With 511-bit messages, a cyclic code using 12 check bits can detect almost all errors we have recorded in our test experiments; a simulation with an I.B.M. 704 computer has checked the theory with our actual data. For 3.5×10^5 messages, we found 5.2×10^3 erroneous messages and only *one* erroneous message undetected.

These considerations point out the question of the over-all system operation responsibility. In most of the cases, three organs are involved in the process :

— the customer who operates the system,

— the computer manufacturer who provides the data-processing equipment,

— the telecommunication agency which provides transmission facilities.

It appears that the customer will require the computer manufacturer to ensure a given operation quality for his system and that the computer manufacturer has to deal with the telecommunication facilities as they are : in certain areas, this operation quality could be increased in the future by some improvements of the telecommunication network.

The over-all quality of system operation depends on the line error distribution, on the character and message codes and on the checking sub-routines involved in the main programmes ; the computer manufacturer is able to take all these factors into consideration.

3.4 *Error correction in messages*

The 7701 machine uses error-detecting codes. What is the effective speed we can reach with error-correcting codes ?

At first, the single burst correcting analysis shows that 80 to 90% of erroneous 240-bit messages are corrected. The burst length was set to 7 bits ; increasing the length of burst beyond 7 bits has negligible effects. Again the reason for this phenomenon is the "burst of bursts". If a larger percentage of erroneous messages has to be corrected, multiple burst correcting devices have to be used in the data transmission system and then we can reach a rate of uncorrected messages less than 1%.

To reach this 1% percentage over three lines with a message in error rate of 0.25%, we found it necessary to correct up to 10 bursts of four bits wide.

Finally, if it is quite easy to correct a large amount of wrong messages, it needs a very powerful (and expensive) equipment to correct the last few per cent still in error.

However, error-correcting devices can be very useful in the future to solve various problems, some of them being yet unknown.

As an example, error-correcting devices can be applied to systems working very close to their maximum speed and where a few per cent of the bit patterns cannot be properly transmitted. In this case, message retransmission is not a sufficient tool because the line is not able to carry properly the signals, and error correction is necessary to avoid continuous retransmission of the same message.

Error correcting will obviously be of strong help for simple transmissions where no feedback controls are possible.

Of course, there exists the possibility of using the error-correcting codes merely to reduce the number of retransmissions. However, for good lines, we found an average rate of retransmission at 1200 bauds and with 3200-bit messages less than 5%. Single-burst error correction would afford a performance improvement of perhaps 3%, which is of no interest if the equipment has to increase too much its complexity.

A general statement which can be made is that, except for very special cases, efficient error detection and correction by retransmission is one of the most important means of error control. However, it would be illusory to expect that error correction means may provide by themselves sufficient efficiency for computer communication ; these means have to be used jointly with the best modulator-demodulator equipments.

Supplement No. 35

NIPPON TELEGRAPH AND TELEPHONE PUBLIC CORPORATION — *(Extract from contribution COM Sp. A/No. 72 — August 1963)*

DATA TRANSMISSION TESTS OVER TELEPHONE CIRCUITS

1. *Introduction*

It is very difficult to suggest any substantial difference between the FM system and the PhM system from the point of view of reliability, complexity and flexibility. However, if a comparison is restricted within the transmission performance, the difference between the FM and PhM systems can be made rather definitely clear and, furthermore, it is helpful to make clear the difference. To compare the transmission performances of various kinds of data modems, measurements on the transmission characteristics of the data modems should be done under strictly controlled conditions. In a field test, it is very difficult to keep the conditions constant, but in a laboratory the comparative test could be done rather easily. In this case, the main factors which cause transmission error of binary data should be simulated as closely as possible. All the modems which were compared in our test programme were designed to transmit binary data over telephone circuits compatible with C.C.I.T.T. recommendations. Although these modems were all prototype equipments, the differences of the transmission performance between the FM and PhM system are easily obtained.

2. *Outline of the experiment*

2.1 *General considerations*

The data information is represented in the binary form. The quality of the transmitted signal is reduced due to limitations of the frequency band, variations of over-all circuit loss with time, circuit noise, frequency difference at the two ends of a carrier circuit, etc.

For the purpose of the investigation of the reduction in quality, to take "the isochronous distortion" for the measure of reduction in quality is very useful. But the quality of the binary data transmission system cannot be expressed by distortion characteristics only, because sometimes it is probable that the margin for noise is near 0% in spite of little distortion. Therefore, in our laboratory test, the isochronous distortion, element error rate and signal-to-noise ratio threshold at which transmission errors begin to appear were used for the measurement of data transmission quality. That is, the isochronous distortion is used to measure the reduction in quality due to limitation of the frequency band, variation of loss with time and frequency difference at two ends of the carrier circuits, and element-error rate is used to measure the reduction in quality due to random noise and finally signal-to-noise ratio threshold is used to measure the reduction due to sine wave interferences and impulsive noises.

2.2 *Test circuit for data transmission systems*

The set-up for the test which realized the above described points is shown in Figure 4. In this figure the data transmission test equipment 3, to which the distortion meter 1 supplies the master-clock signal, generates the test patterns of 128 bits repeatedly and can count the number of errors in the received signals. The distortion meter can measure isochronous distortion of

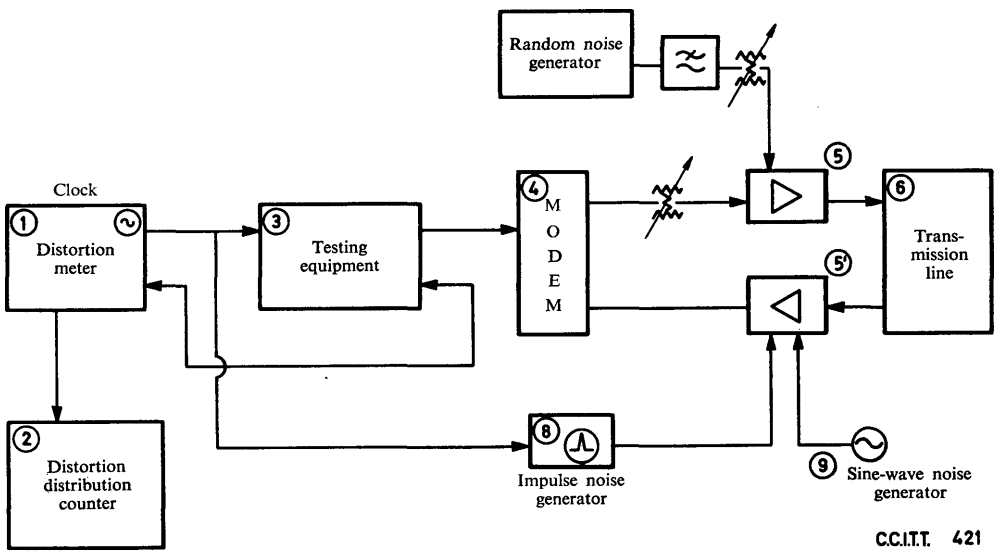


FIGURE 4. — **Test circuit for data transmission system**

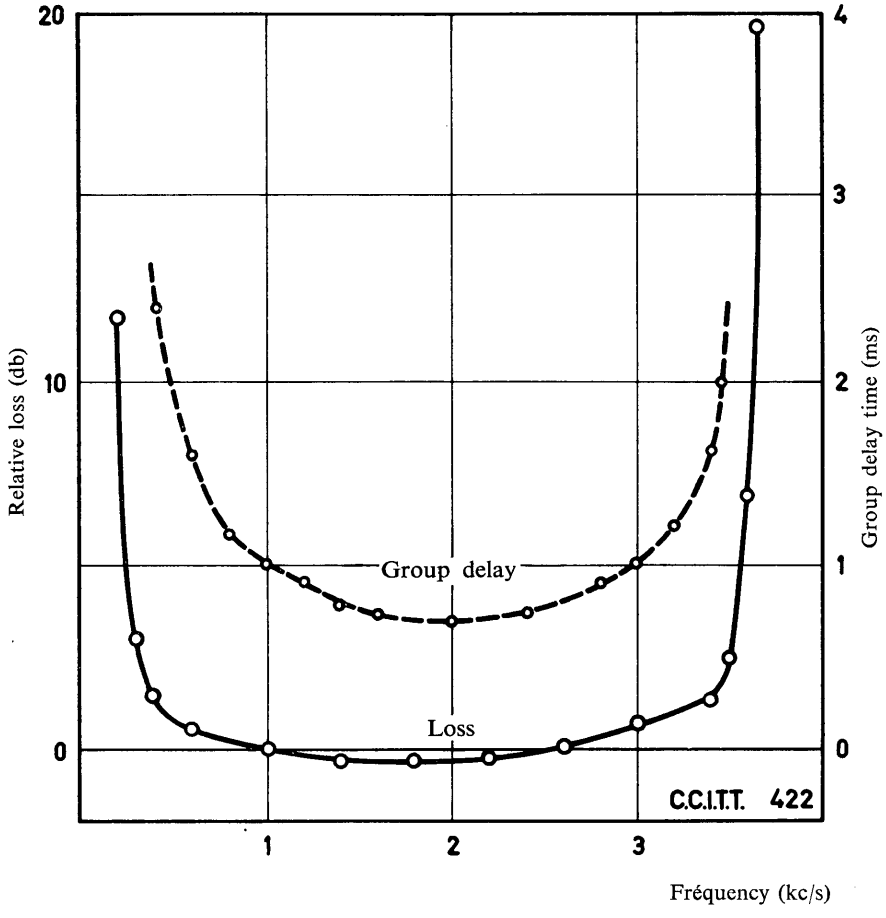


FIGURE 5. — **Transmission characteristics of one link of a carrier telephone**

the received signals and can classify the significant instants of the codes according to the magnitude of distortion. Two amplifiers 5 and 5′ have an input terminal to which noise can be added and mixed with the transmitting signals if necessary.

To simulate a wider band transmission line, one link and two links of carrier telephone channels were used which consisted of the pre-group modulators, group amplifier and pre-group demodulator of the 12-Mc/s coaxial cable telephone system. Frequency characteristics of one link of the telephone channel is shown in Figure 5. For narrower band transmission, as in the case where a channel is divided into a data channel and a supervisory channel, one link of carrier telephone channel and the band-pass filter, whose frequency characteristics are shown in Figure 6, were used.

In this test, each type of modem was examined concerning the margin to white random noises, single frequency interferences and impulsive noises, after the distortion performances were measured. It is unnecessary to illustrate the structure of random noises and the single frequency interferences but it is necessary to illustrate the generation of impulsive noises. That is, the shape of the impulsive noises is a square wave whose time-width can be varied from 30 μs to 100 μs. Its

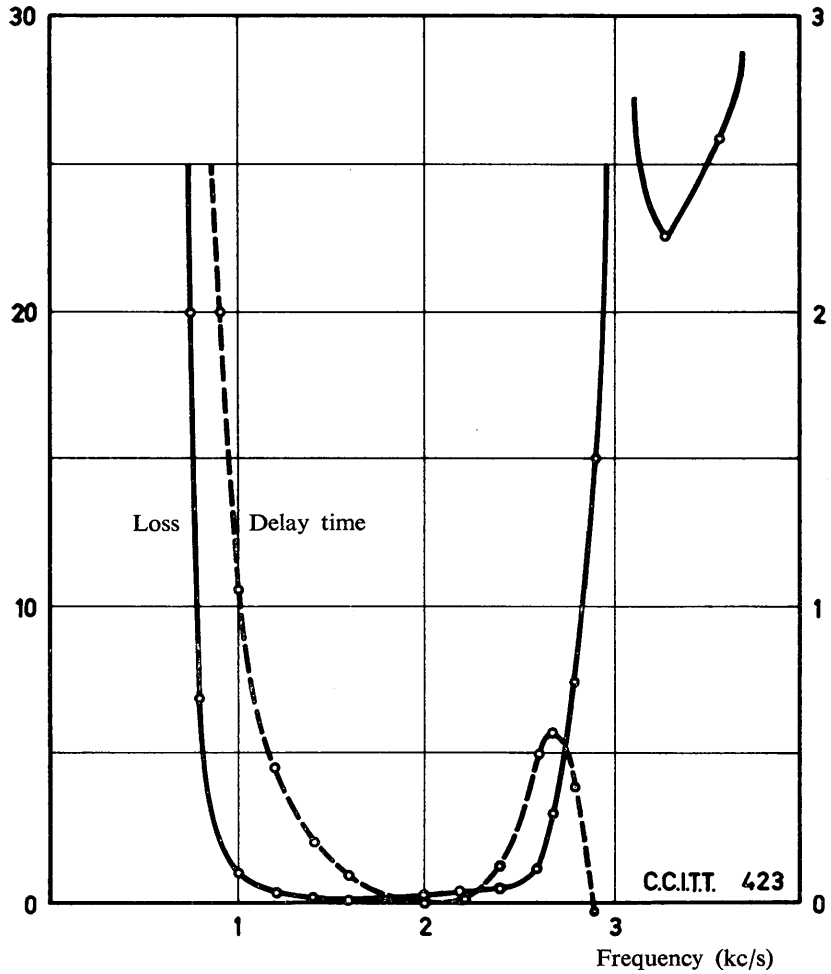


FIGURE 6. — **Frequency characteristics of narrow band-pass filter**

repetition cycle synchronizes to the elementary bit of the test patterns and the impulse is generated once for every 15 bits of the test patterns. The occurrence time position is adjustable about 1.5 ms forwards and backwards. Since 128, the number of bits of the test pattern, is incommensurable with 15, the impulsive noises can be superposed in every position of the test pattern elements.

2.3 *The items measured*

Each modem was compared with others concerning the following items :

i) Isochronous distortion of received signals. Transmission rate was changed from 1000 bauds to 2500 bauds at every 250 bauds.

ii) Distribution of characteristic distortion. The significant instants of received binary signals from the output of each modem were classified by their distortions and counted respectively. After drawing the curves of the distributions from the results, the variations of the distribution were represented by the distortion width which includes from 16% to 84% and denoted as σ.

iii) Performance reduction by random noise. Measurements were made of such S/N ratios that the error rates lie in the orders of 10^{-3} and 10^{-4}. These ratios were measured at the input of the modems.

iv) Error threshold by single frequency interferences. Seven frequency sine waves were used as interferences and minimum values of S/N when errors occur were measured by a level meter.

v) Error threshold by impulsive noises. Since the numbers of errors at the same S/N differ according to the relative time position between significant instants of the test pattern and impulsive noises, the error thresholds were obtained at the relative position where errors were most susceptible to occur.

In addition to the above items, the distortion performances of modems were examined for the variations of signal level and for the frequency translations of carrier telephone circuits, but these results are omitted as there are no significant differences between the systems.

2.4 *The modems tested*

The systems of modem tested were the following:

i) PhM modem with coherent detection (1) (PhM (1) in abbreviated form). The block diagram of this system is shown in Figure 7.

ii) PhM modem with coherent detection (2) (PhM (2) in abbreviated form). This modem is the same as PhM (1) except that the phase characteristics of one link of carrier telephone circuit is first equalized by a pulse-shaping filter (refer to Figure 5). It is noted that the equalization in the base-band system can be obtained by less components than in the band-pass system.

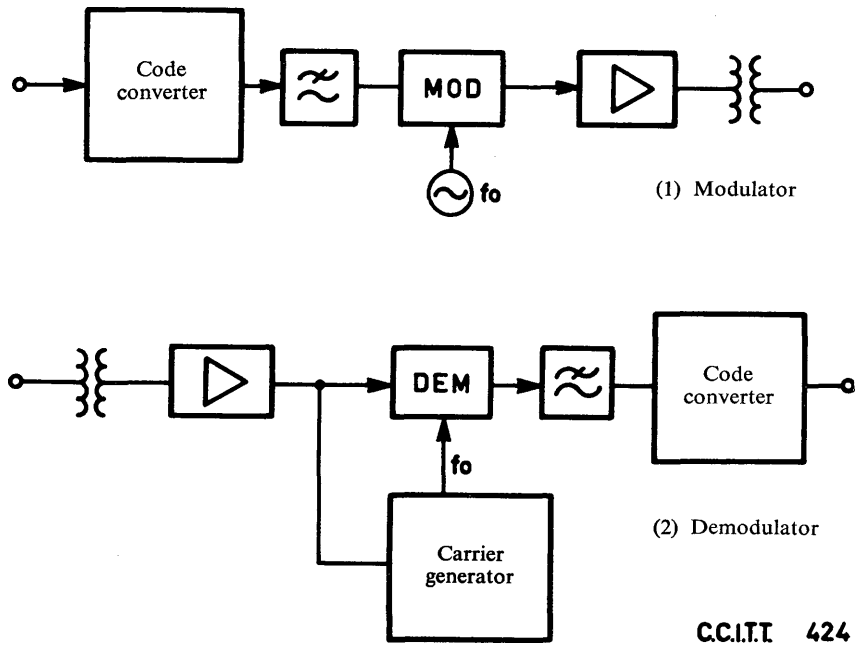


FIGURE 7. — **PhM modem using coherent detection**

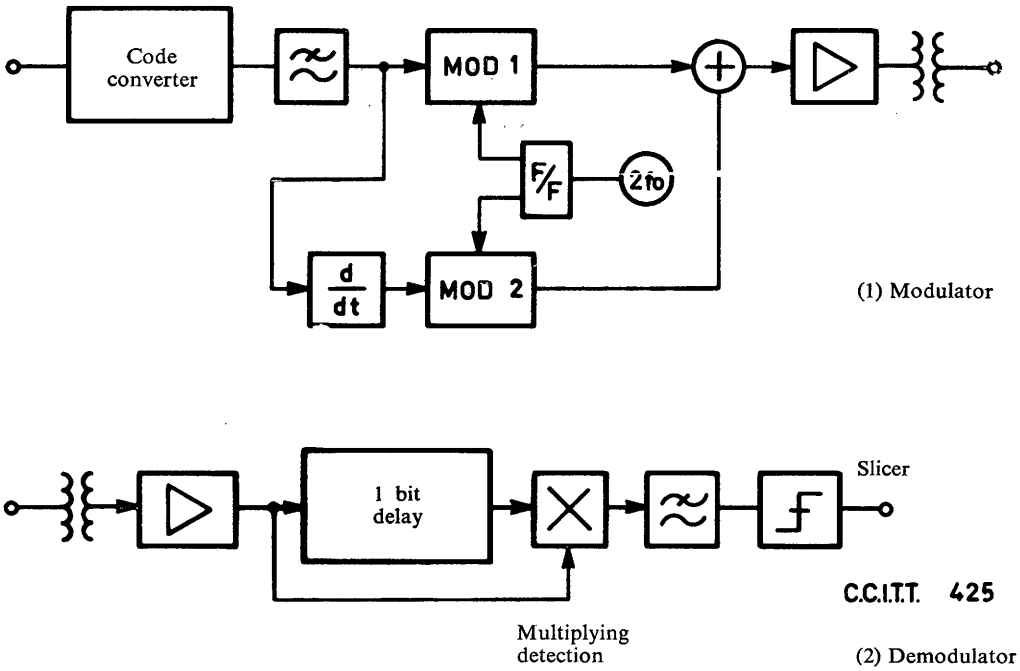


FIGURE 8. — **PhM modem using delay line detection**

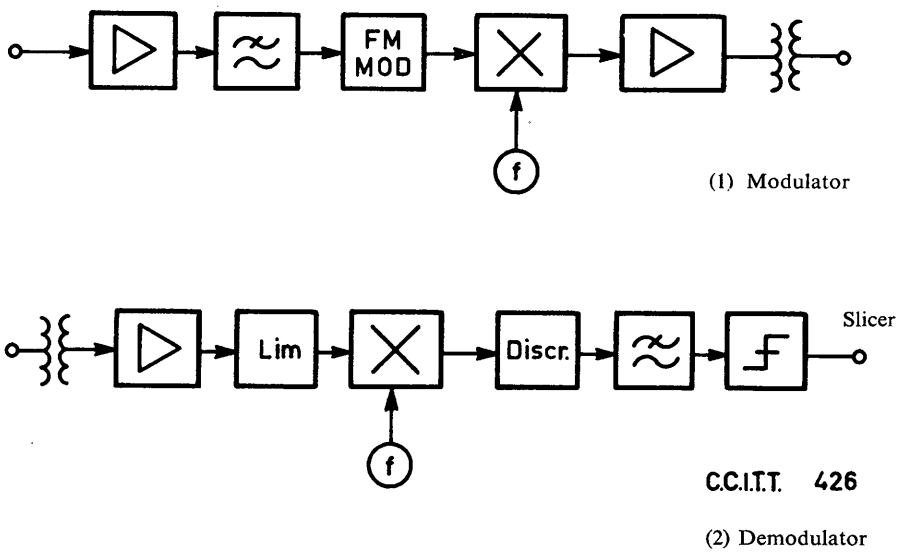


FIGURE 9. — **FM modems**

iii) PhM modem with delay line detection (PhM (3) in abbreviated form). As in Figure 8, the converted codes from the ordinary binary codes modulate the inphase carrier on the one hand and they modulate the quadrature carrier on the other hand, after being differentiated. Both modulated signals are added algebraically and sent to the line. The spectrum of the output signals is a symmetric. In this system, demodulation can be carried out with delay-line detection.

iv) FM modem with zero-cross detection (FM (0-cross) in abbreviated form).

v) FM modem with discriminator detection. In this system, two modems were tested; the shift-frequency of one was 1.8 ± 0.8 kc/s for wide-band transmission and that of the other 1.7 ± 0.4 kc/s, denoted FM (± 0.8) and FM (± 0.4) respectively in abbreviated form. The block diagram of these modems is shown in Figure 9.

3. *Results of the test*

Various performances of the modems are shown in Figures 10 to 21 and Table 4 for wide-band transmission.

For narrow-band transmission, the performances of the modems are shown in Figures 22 and 23 and Table 4.

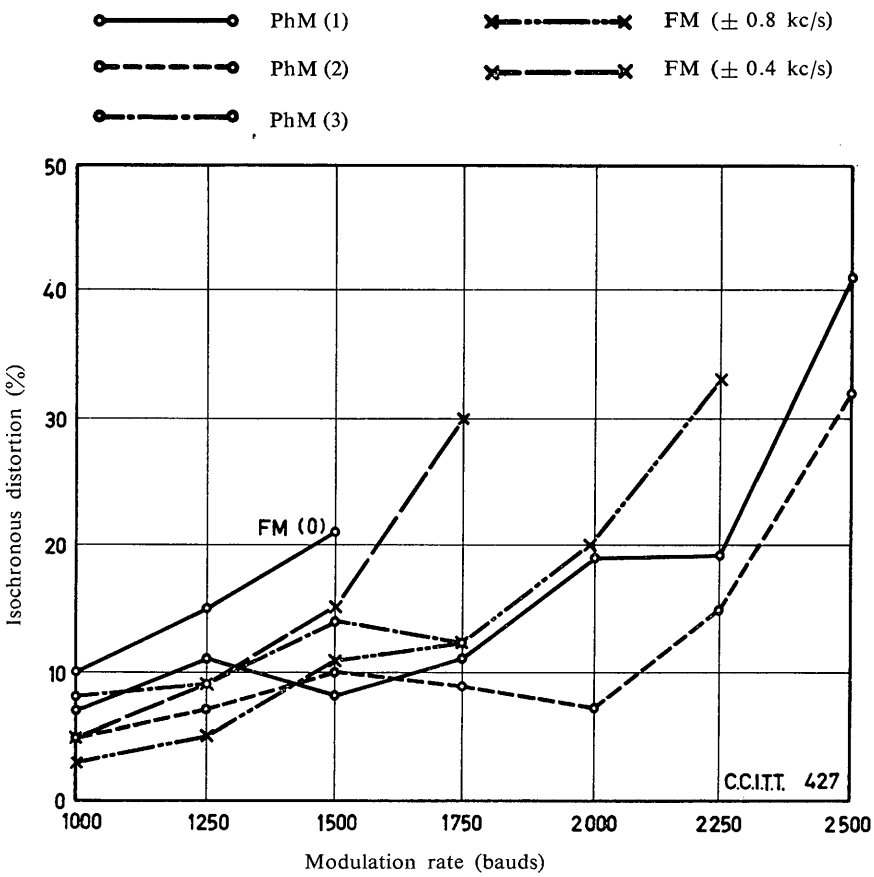


FIGURE 10. — **Characteristic distortion (carrier telephone circuit, one link)**

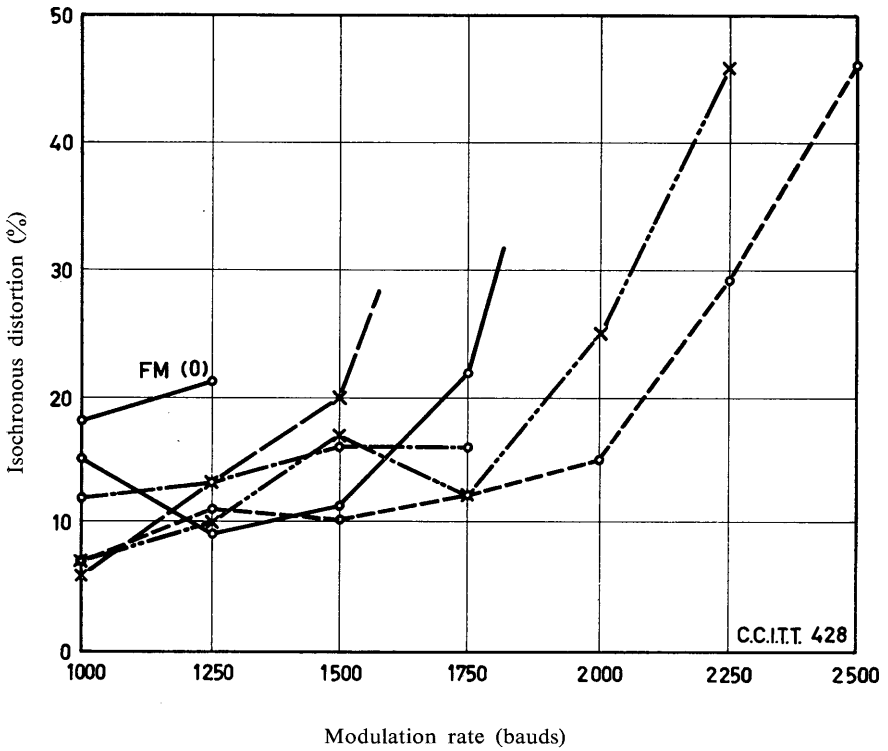


FIGURE 11. — **Characteristic distortion (two links)**

4. *The comparison of the modems from their performances (except for FM (0-cross))*

4.1 *On the wide-band channel (PhM (1), PhM (2), PhM (3) and FM (± 0.8) are compared with one another)*

There are no notable differences in distortion performances of each modem at the modulation rates under 1750 bauds, but the differences are obvious over 1750 bauds. The highest modulation rate, so that the isochronous distortion is within 30%, is 2250 bauds for PhM (2) and 2000 bauds for FM (± 0.8). (This rate will be reasonable as to how each modem can transmit.) With regard to the variations of distortion (σ), the PhM (2) is superior to the FM (± 0.8) at 1500 bauds, while the FM (± 0.8) is superior to the PhM (2) at 1250 bauds.

From the random noise performances of modems shown in Figure 14 and Figure 15, it can be seen that the PhM (1) and PhM (2) have a 3 to 4 db advantage in signal-to-noise ratio over the FM (± 0.8) at 1500 bauds.

Figure 16 indicates that the random noise performances of the PhM (2) and PhM (1) are invariable with changes of signalling rate, but that of the FM (± 0.8) modem is deteriorated with increases of signalling rate. The difference of the random noise performance between the PhM (2) and FM (± 0.8) is about 5 db at 2000 bauds.

It is noted that the immunity from the single-frequency interferences is almost the same for all systems. For error threshold to impulsive noises at 1250 bauds, the FM (± 0.8) would have a small advantage over the PhM (2), but the PhM (3) is the system most immune from the noises.

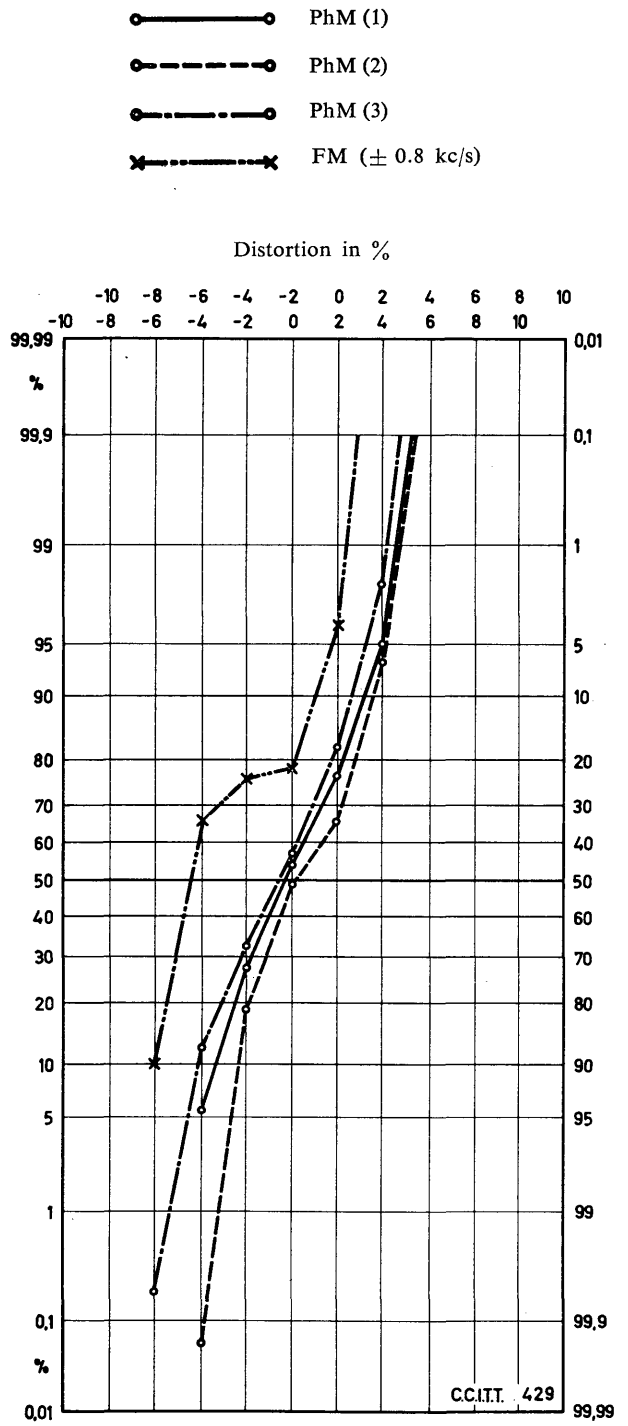


FIGURE 12. — **Distribution of distortions of significant instants of transmitted codes at 1500 bauds for each type of modem (one link)**

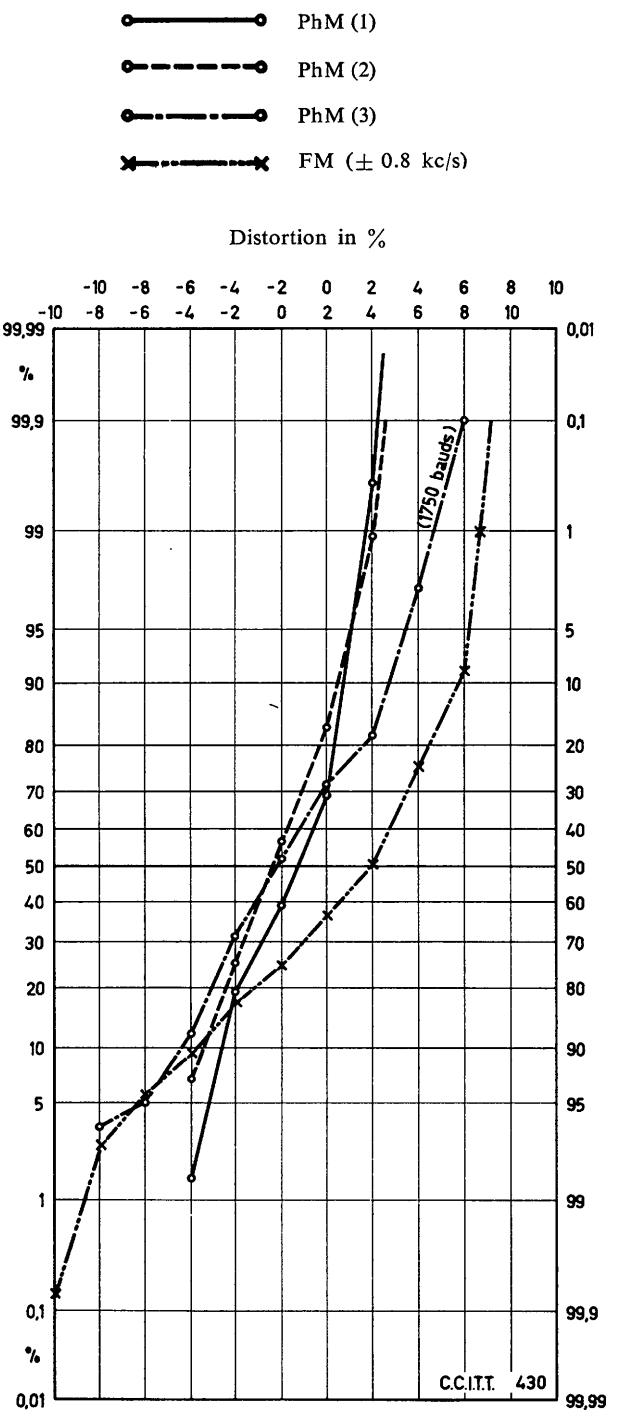


FIGURE 13. — **Distribution of distortions of significant instants of transmitted codes at 1500 bauds for each type of modem (two links)**

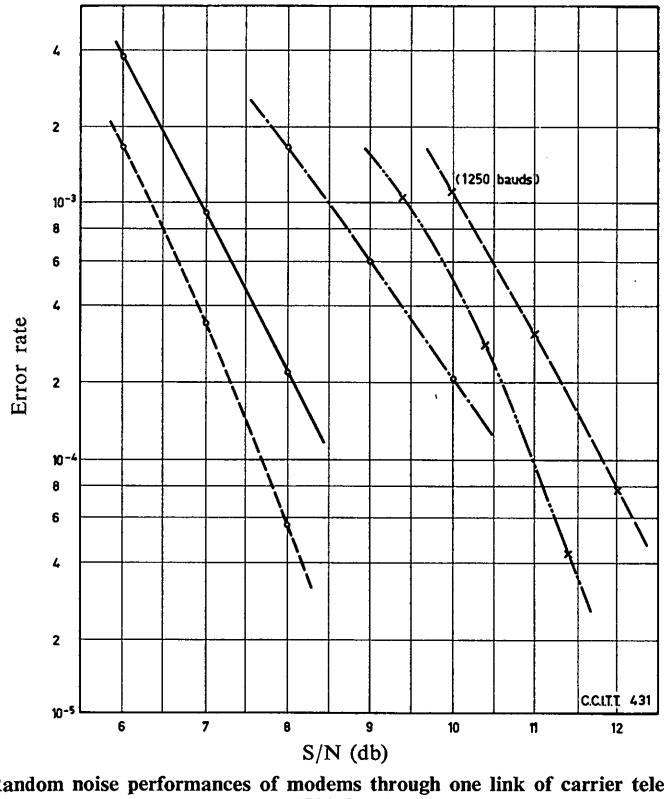


FIGURE 14. — **Random noise performances of modems through one link of carrier telephone circuit at 1500 bauds**

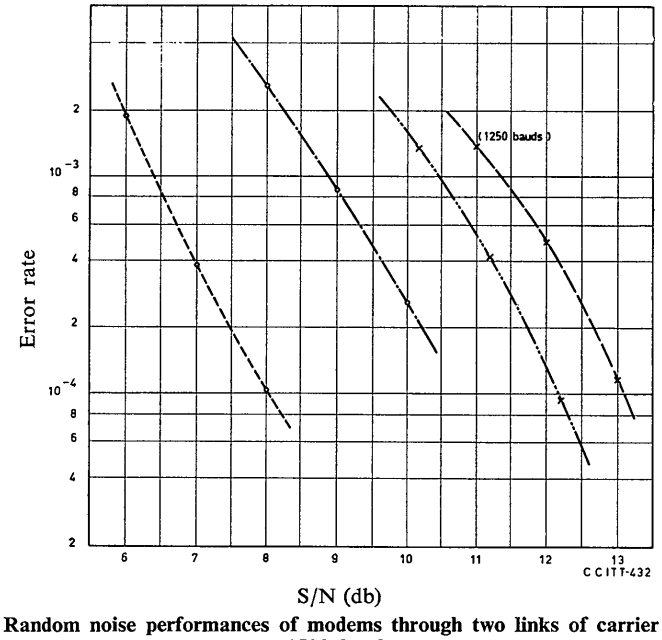


FIGURE 15. — **Random noise performances of modems through two links of carrier telephone circuit at 1500 bauds**

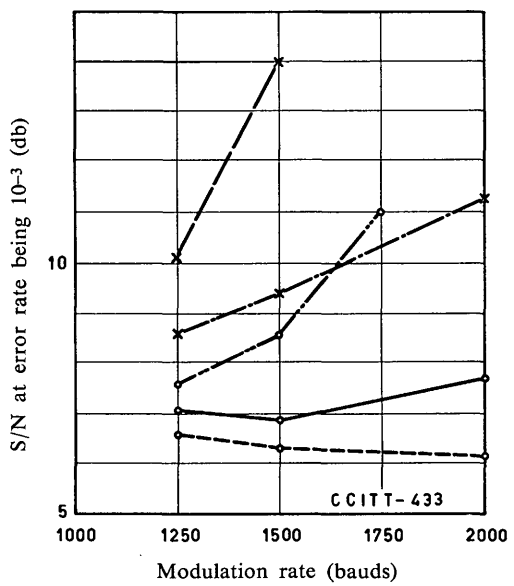


FIGURE 16. — **S/N (db) at error rate being 10^{-3} to modulation rates (one link)**

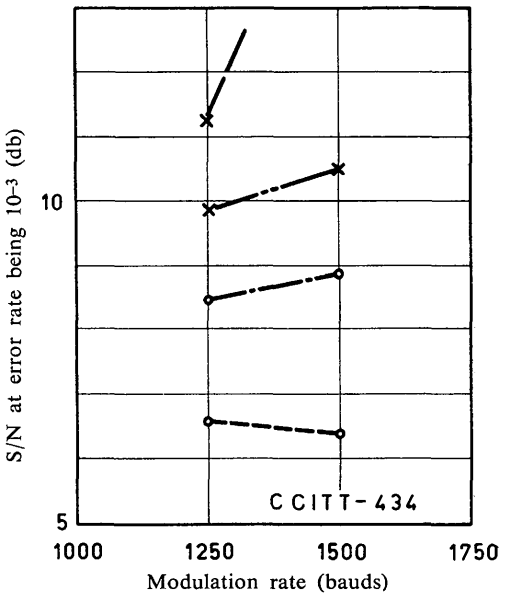


FIGURE 17. — **S/N (db) at error rate being 10^{-3} to modulation rates (two links)**

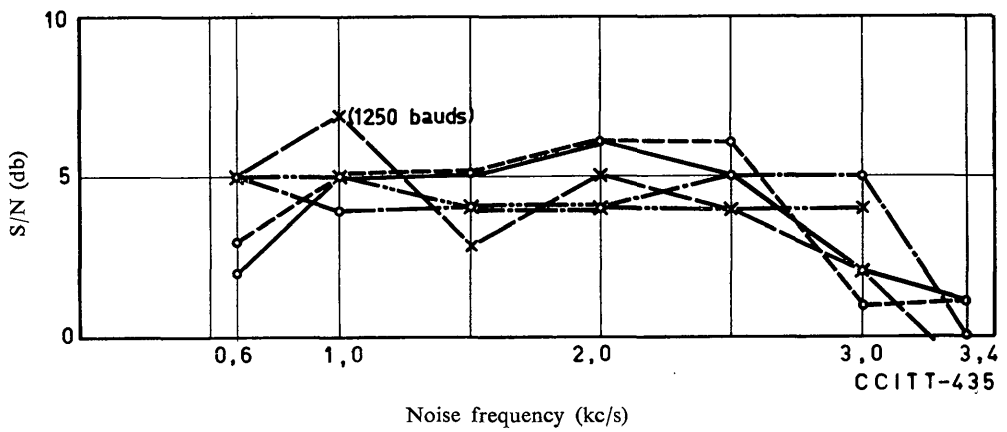


FIGURE 18. — **Error threshold to sine-wave noises for various modems (one link at 1500 bauds except for FM (± 0.4 kc/s))**

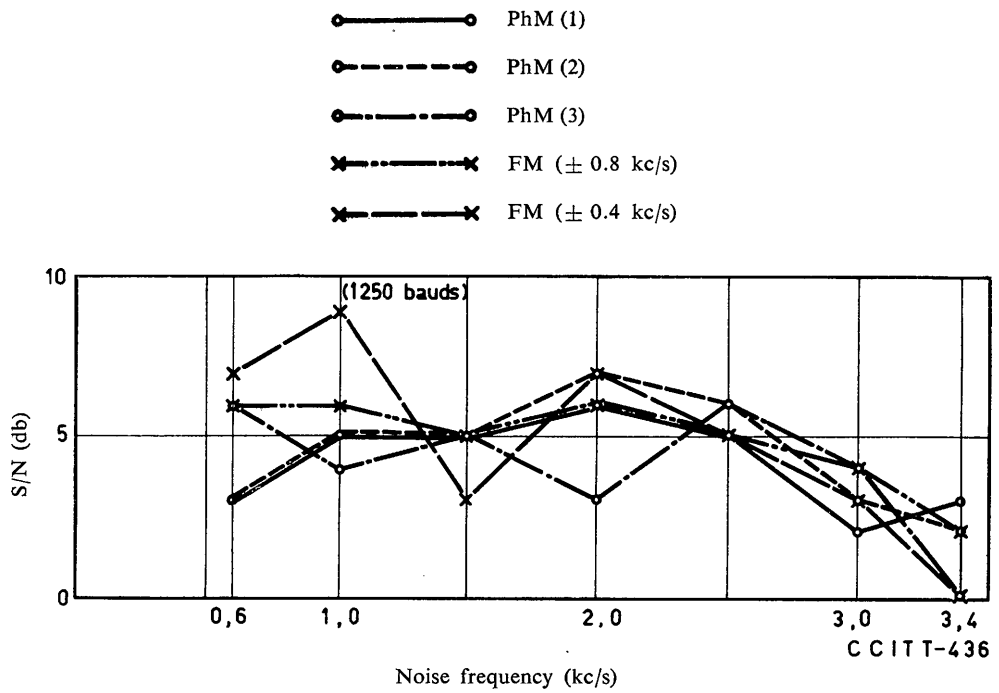


FIGURE 19. — **Error threshold to sine-wave noises for various modems (two links at 1500 bauds, except for FM (± 0.4 kc/s))**

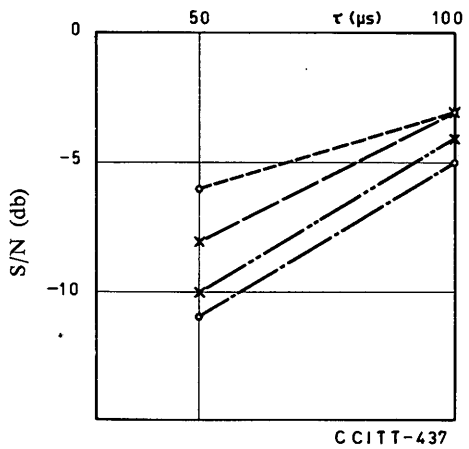


FIGURE 20. — **Error threshold to impulsive noise (1250 bauds, one link)**

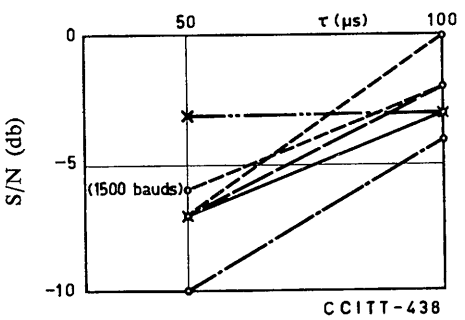


FIGURE 21. — **Error threshold to impulsive noise (1250 bauds, two links)**

4.2 *On the narrow band channel (PhM (1), PhM (2), PhM (3) and FM (± 0.4) are compared with one another)*

Except for the PhM (3), all systems have the same characteristic distortion performances at 1000 bauds and 1250 bauds (Figure 22).

It can be seen, however, that there is a remarkable difference between the random noise performances of modems at 1250 bauds. That is, the PhM (2) modem has a 5.5-db advantage over the FM (± 0.4) modem in S/N.

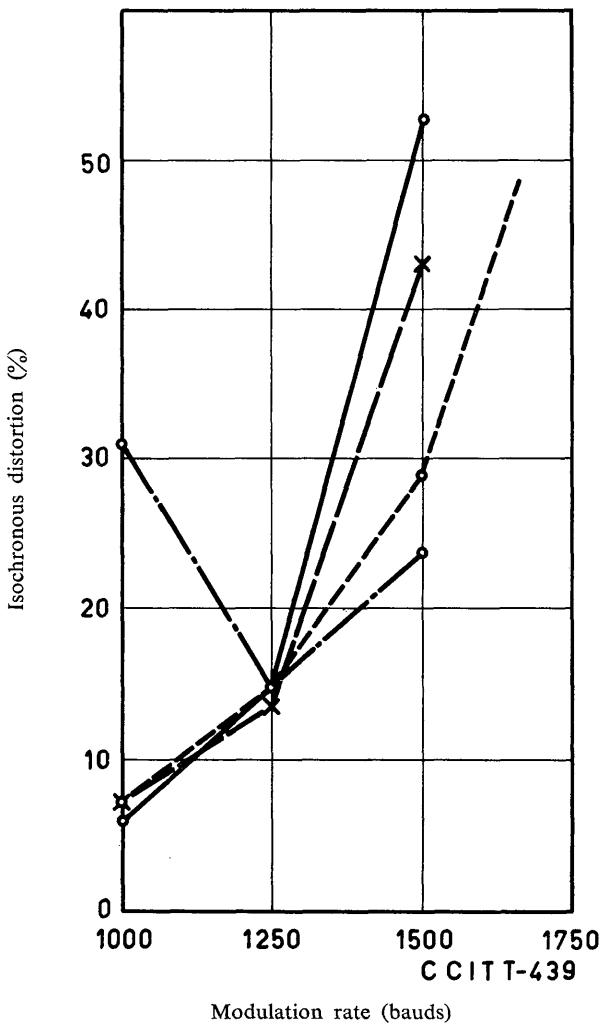


FIGURE 22. — **Characteristic distortions over narrow-band circuit (0.9-2.7 kc/s) at various rates**

TABLE 4

σ obtained from the distributions of significant instants to speeds through three transmission lines for various modems

(in percentages)

System of modem		PhM (1)			PhM (2)			PhM (3)			FM (0-cross)			FM (± 0.8)			FM (± 0.4)		
Transmission line		1 link	2 links	Narrow band	1 link	2 links	Narrow band	1 link	2 links	Narrow band	1 link	2 links	Narrow band	1 link	2 links	Narrow band	1 link	2 links	Narrow band
Transmission rate (bauds)	1000										4.8								
	1250			6.0	3.6	4.8	4.8	6.4	5.0	6.4	8.8			2.3	4.5		6.3	6.9	7.6
	1500	5.4	4.7		5.0	4.8	9.8	5.6		14.0	11.2			6.2	9.6		10.0	11.8	
	1750							6.2	7.6										
	2000	8.0			4.8	6.6													
	2250					10.0													

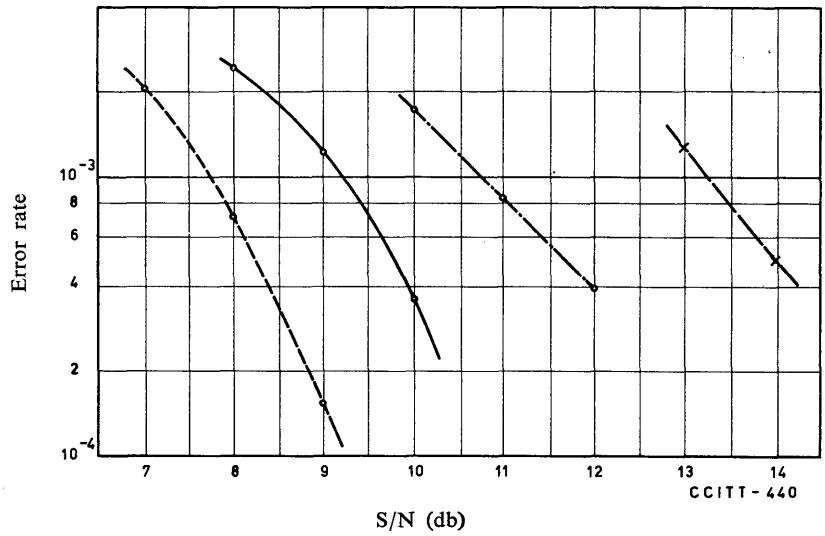


FIGURE 23. — **Random noise performances of modems through narrow-band circuit at 1250 bauds**

4.3 *Conclusion*

From the point of view that the reliable data transmission can be attained at as high a speed as possible for a given bandwidth, the PhM modem with coherent detection is superior to the FM modem by about 5 db in signal-to-noise ratio at the limiting modulation rate. In the case where the transmission rate is low compared with channel bandwidth (e.g. at 1200 bauds over a wide-band channel), however, the PhM and FM modems would have roughly the same performances, because the real noise would have an intermediate peak-factor between those of the random noise and impulsive noises used in this test.

On the narrow band transmission where a supervisory channel is required in the same channel, the rate of 1200 bauds approaches the limiting rate and the PhM modem has an advantage over the FM modem of more than 5 db in signal-to-noise ratio.

Supplement No. 36

NIPPON TELEGRAPH AND TELEPHONE PUBLIC CORPORATION — *(Extract from contribution COM Sp. A/No. 22 — October 1961)*

DATA TRANSMISSION TESTS OVER POINT-TO-POINT TELEPHONE CIRCUITS ON COAXIAL CABLES

1. *Introduction*

Data transmission tests over telephone circuits on coaxial cables between Osaka and Tokyo were made from 11 September 1961 to 22 September 1961, according to test specification "L" in the study programme annexed to Question 1/A.

2. *Test circuit*

Three telephone circuits were used as data transmission path, circuit interruption monitoring circuit and noise measuring circuits.

2.1

. . . . . .

2.2 *Characteristics of the test circuit*

1) Frequency response of the test circuit

The net loss characteristics and envelope delay response are shown in Figure 2. The envelope delay distortion was not equalized but the net loss distortion, due to non-loaded cables, was equalized.

2) Noise

The test was carried out almost all the time under S/N of the data transmission channel being 41-43 db. At the same time, the noise level of the noise monitoring circuit was being measured

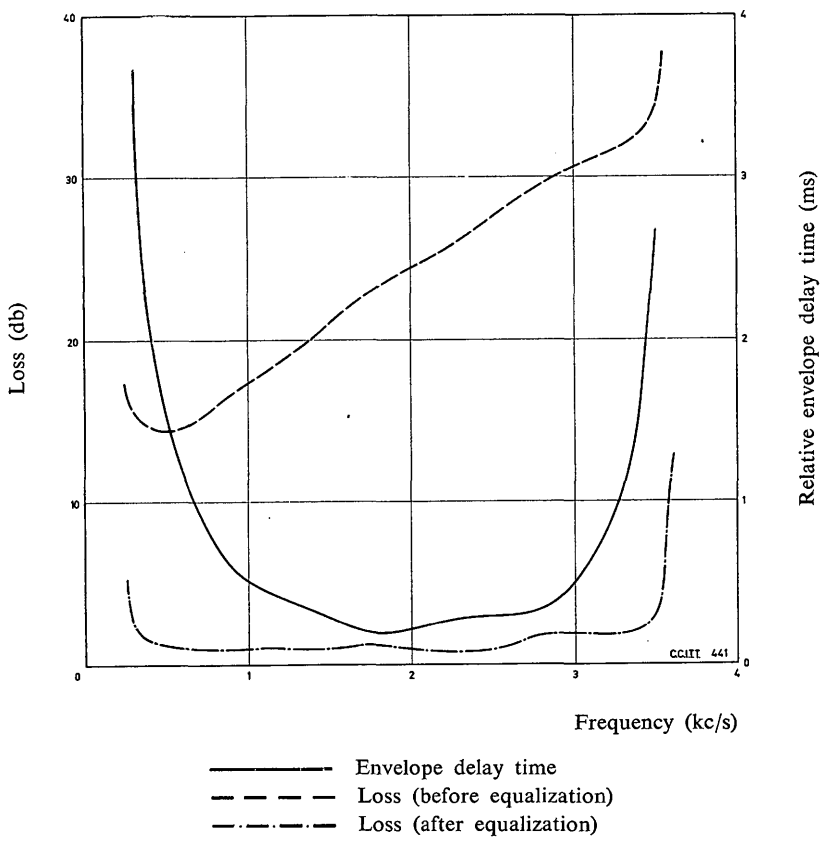


FIGURE 2. — **Frequency characteristics of test circuit**

and the fact was found that impulsive noise which occurred rarely reduced the S/N of the channel about 20 db. Moreover, it was observed that the occurrences of these impulse noises correlated to the occurrence of error bits.

3) Circuit interruption

The number of circuits interruption from 15 September to 22 September and their causes are tabulated in Table 1. The time lengths of circuit interruptions are shown in Table 2.

TABLE 1

The number of circuit interruptions and their causes

Causes of interruptions / Date	Change-over of the flat amplifier	Change-over of the line amplifier	Change-over of the carrier supplies	Noise	Miscellaneous and unknown	Total
September 15	5	5		1	2	13
„ 16						0
„ 17						0
„ 18		7		3	2	12
„ 19				2		2
„ 20						
„ 21			12			12
„ 22						
Total	5	12	12	6	4	39

TABLE 2

The time length of circuit interruptions

Time length of circuit interruptions	Percentage of the number of interruptions to the total number (%)
Longer than 300 ms	2.6
Shorter than 300 „	97.4
„ 200 „	94.9
„ 100 „	92.3
„ 70 „	92.3
„ 50 „	89.7
„ 40 „	89.7
„ 30 „	89,7
„ 20 „	89.7
„ 10 „	87.2
„ 5 „	79.5

TABLE 3

Time length of interruptions and error bits caused by change-over of the power supplies system

Kinds of test	Number of stations under the test	Time length of interruptions	Error bits
Giving-out of power supplies	10	0	8
Recovery of power failure	10	0	0
Change-over of MGG to induction voltage regulator	9	0	0
Change-over of induction voltage regulator to MGG	9	0	0
Change-over of MGG : normal equipment to stand-by equipment (AC)	1	0	0
Change-over of MGG : stand-by equipment to normal equipment (AC)	1	0	0
Change-over of MGG : normal equipment to stand-by equipment (DC)	1	40 ms	26
Change-over of MGG : stand-by equipment to normal equipment (DC)	1	12 ms	10

TABLE 4

Time length of interruptions and error bits caused by change-over of the line amplifiers and the flat amplifiers

Types of change-over	Numbers of stations under the test	Time length of interruptions	Error bit	Notes
Line amplifier : normal equipment to stand-by	10	1-2 ms	1-2	
Line amplifier : stand-by equipment to normal	9	1-3 ms	1-3	
Flat amplifier : normal equipment to stand-by	11	1-2 ms	1-3	Change-over at only one station causes interruption 13 ms
Flat-amplifier : stand-by equipment to normal	11	1-3 ms	1-3	Change-over at only one station causes interruption 10 ms

3. *Circuit interruptions caused by maintenance operation and their effects on data transmission*

In advance of data transmission test, the time length of interruptions caused by maintenance operations and their effects on transmission performance were measured. This test was carried out at 12 repeater stations.

3.1 *Circuit interruptions caused by change-over of the power supply system*

Results are shown in Table 3.

3.2 *Circuit interruptions caused by the change-over of line amplifier and flat amplifier*

Test results are shown in Table 4.

3.3 *Circuit interruptions caused by change-over of carrier supply*

Test results are shown in Table 5.

3.4 *Circuit interruptions caused by the interchange of the supergroups of the system*

After the supergroups being used and to be used were connected in parallel at Osaka telephone repeater station, interchange of these supergroups was carried out several times at Osaka telephone repeater station. In this experiment, the lengths of this circuit interruption were 370-870 milliseconds and numbers of error bits were 280-660 bits.

4. *Test results of data transmission*

4.1 *Number of errors*

Number of error bits, error letters and error blocks are tabulated in Table 6.

4.2 *Error rate*

Bit-error rate, letter-error rate and block-error rate are shown in Table 7. Number of error bits per letter and number of error letters per error block are shown in the same table.

TABLE 5

Time length of interruptions and error bits caused by change-over of the carrier supplies

Types of change-over	Number of stations under the test	Time length of interruption	Error bits
Channel carrier supply : normal equipment to stand-by	2		1-3
Channel carrier supply : stand-by equipment to normal	2		0-1
Pre-group carrier supply : normal equipment to stand-by	2	1-4 ms	1-2
Pre-group carrier supply : stand-by equipment to normal	2	1 ms	0-1
Group carrier supply : normal equipment to stand-by .	2	3 ms	2-3
Group carrier supply : stand-by equipment to normal .	2	1 ms	0-1

TABLE 6

Number of total errors

Kinds of modulation system	Modulation rates in bauds	Signal level (dbm)	Causes of errors	Number of error bits	Number of start polarity to stop polarity error bits	Number of stop polarity to start polarity error bits	Number of error letters	Number of error blocks	Number of test bits
A.M. D.S.B.	1500	− 12	Interruption	18	5	13	9	6	6.03×10^7
			Noise and others	19	17	2	8	6	
			Total	37	22	15	17	12	
F.S.	1250	− 16	Interruption	44	8	36	21	10	6.72×10^7
			Noise and others	32	3	29	29	21	
			Total	76	11	65	50	31	
Total			Interruption	62	13	49	30	16	1.275×10^8
			Noise and others	51	20	31	37	27	
			Total	113	33	80	67	43	

TABLE 7

List of error rates

Kinds of modulation system	Modulation rate in bauds	Signal level (dbm)	Bit-error rates	Letter-error rates	Block-error rates	Average error bits per error letter	Average error letter per error block
A.M. D.S.B.	1500	− 12	6.13×10^{-7}	2.26×10^{-6}	5.1×10^{-5}	2.18	1.42
F.S.	1250	− 16	1.13×10^{-6}	5.95×10^{-6}	1.18×10^{-4}	1.52	1.61
Total			8.85×10^{-7}	4.2×10^{-6}	8.63×10^{-5}	1.69	1.56

5. *Summary*

Number of error bits caused by interruption was roughly equal to one of error bits caused by noises.

The average number of circuit interruptions per day was about 5.5 and 80% of the interruptions were shorter than 5 milliseconds. Throughout this test only one interruption longer than 300 milliseconds was observed.

The average bit-error rate during the business hour was about 9×10^{-7}.

The number of error bits per error letter was 1.7 and the number of error letters per error block was 1.6.

Supplement No. 37

I.B.M. (W.T.E.C.) — *(Extract from contribution COM Sp. A/No. 64 — June 1963)*

DATA TRANSMISSION TEST ON A MULTIPOINT TELEPHONE NETWORK IN EUROPE

Introduction

This contribution describes a data transmission test of a multipoint telephone network in Europe. The results obtained are intended to provide an indication of how a typical long-haul, leased, voice-grade facility might be expected to operate at 2000 bits per second.

Test equipment

The test ran from April through June 1962 over a circuit from London to Rome.

Figure 1 shows the layout of the test system. The circuit was a four-wire, full duplex, voice-grade facility from London via Paris and Frankfurt to Rome. Except for a short (approximately 50 km) microwave link across the English channel, the trunk circuit was effected by three carrier

systems in tandem over coaxial cable and symmetric pair cables. Drops were made at London, Paris, Frankfurt and Rome on baseband facilities. These connections were designed to provide the necessary operating levels required for a large multipoint system. There were no equalizers or amplifiers in the drops themselves, and with the exception of the I.B.M. London and the Rome drops, they were all passive and terminated in 600 ohms.

Total length of the circuit, excluding the drops without terminals, was approximately 2200 km one way. Most of the testing consisted of transmitting random bit messages, 729 bits in length, from London to Rome where they were retimed and retransmitted by the repeater back to London. A bit-by-bit comparison was made between each received message and the original message transmitted. If a message was received in error, both the transmitted message and the received message were recorded on magnetic tape. Included in this record were cause of error, real time, message count and number of bits lost due to carrier failure.

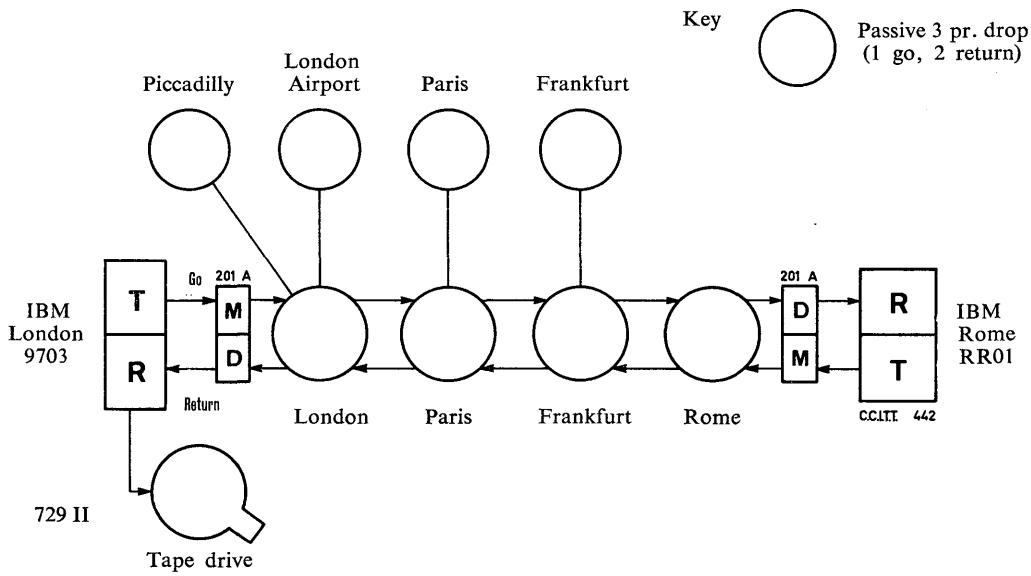


FIGURE 1. — **London-Rome network for 2000 bps test**

Test objectives

1. Measure electrical characteristics of the network.
2. Determine feasibility of 2000 bits per second data transmission over this type of network.
3. Observe the operational behaviour of the line.
4. Obtain data for statistical analysis.
5. Determine what future problems may be associated with the operation of similar networks in proposed reservation systems.

Operation of test

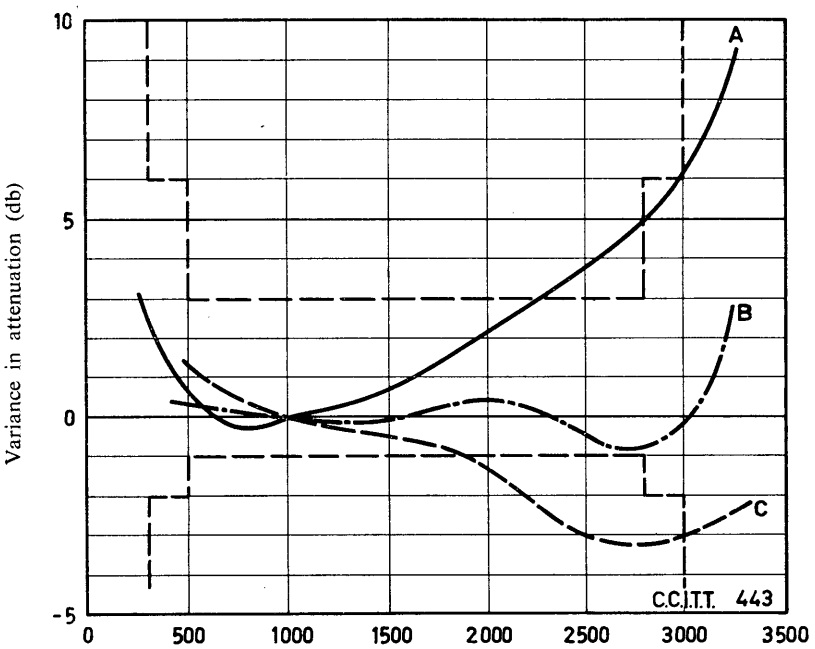
Attenuation and envelope characteristics of the complete circuit (looped) were measured, but signal-to-noise tests were made only on the "return" Rome-to-London leg.

Using the results of the attenuation and delay tests, we designed and constructed two equalizer models. One provided sufficient equalization for use with the PM2 modem while the other

provided the necessary equalization to bring the line characteristics within the limits specified by the FCC schedule 4B[1]. These are referred to in this report as Type 1 and Type 2 respectively.

The attenuation characteristics for one leg are shown in Figure 2. At the time these measurements were made, the results were approximately equal for both legs. Result A shows the frequency response without additional equalization; result B shows the response using the Type 2 equalizer; and result C is obtained with 201A strapped to provide compromise equalization. The line without additional equalization almost meets the limits specified by FCC schedule 4B. With the Type 2 equalizer added, the response falls within the FCC limits; however, the line is overcompensated when the compromise equalizer is used.

Figure 3 shows that similar results were obtained for the delay characteristics. Again, the line without additional equalization almost meets the 4B specifications.



Key	*Attenuation at 1000 c/s (db)*
A. No additional equalization	A. 5.4
B. Type 2	B. 12.0
C. 201A compromise	C. 16.0

———— FCC schedule 4B limit

FIGURE 2. — **Attenuation characteristics**

[1] FCC Tariff 237 "Channels for data transmission", December 1960.

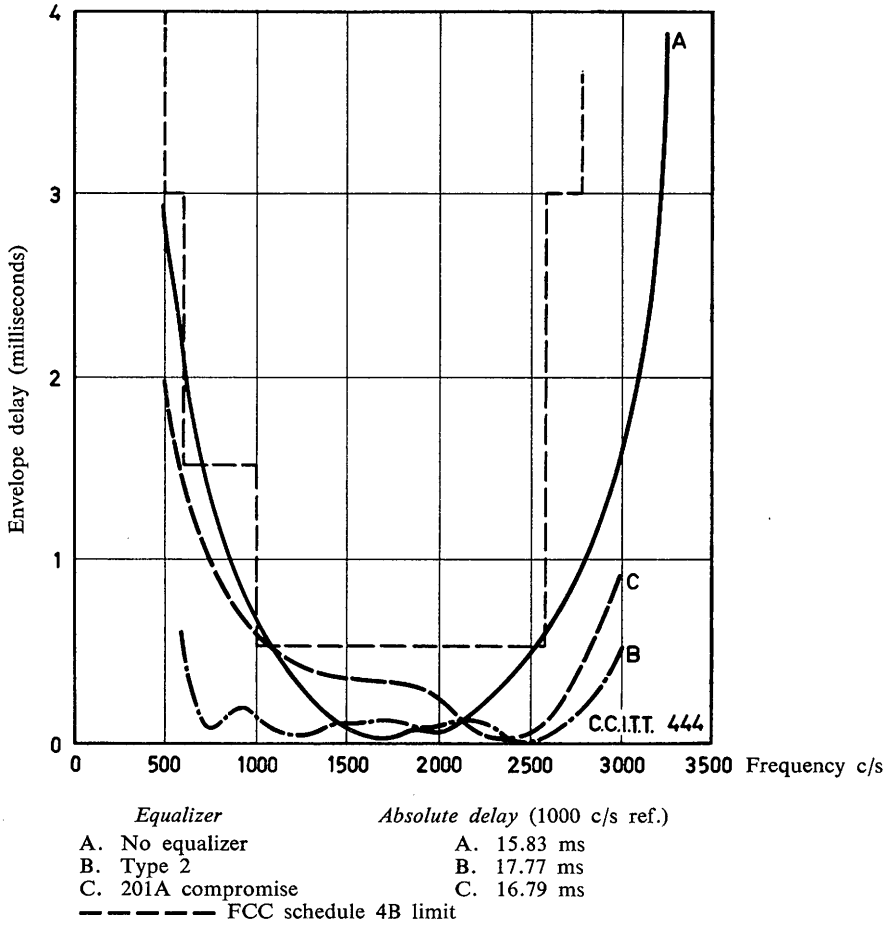


FIGURE 3. — **Delay characteristics**

Figure 4 shows the bit-error rates over a range of signal-to-noise ratios using the 201A modems. These results were obtained by injecting white noise on the line at the transmitting end (Rome) and measuring, at the receiving end (London), the error rate for a given signal-to-noise ratio.

Considering the circuit without additional equalization, it can be said that:

1. The characteristics of the circuit were such that it was suitable for operation with the 201A modem at 2000 bps.
2. The signal-to-noise tests indicated that the 201A was relatively insensitive to the small amounts of frequency and delay distortion that were present.

. . . . . .

The results are divided into two parts, the first dealing with error rates while the second part deals with the outage situation. Figure 5 shows the over-all results in terms of line availability, message-error rates and bit-error rate. An explanation of some of the terms used here might be helpful.

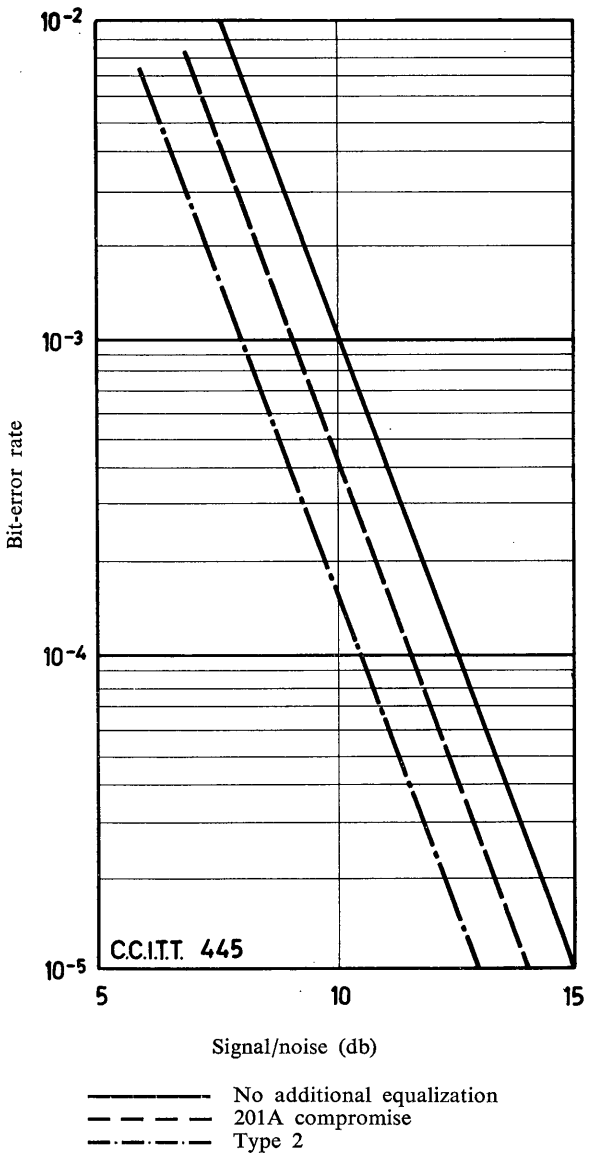


FIGURE 4. — **Bit-error rate versus signal/noise ratio for the 201A modem used on the London-Rome circuit with injected noise**

Test period		21/4/62 to 10/6/62
Test time		822.1 hours
Line availability		94.25 per cent
Data transmission speed		2000 bps
Message length		792 bits
Messages sent		5 766 178
Messages in error due to :		
	Number	*Per cent*
1. Lost sync.	5 219	0.090
1. Carrier off	3 508	0.061
3. Carrier off and lost sync.	934	0.016
4. Noise and drop out	23 589	0.409
Bits sent evaluated for bit-error rate		4 559 161 464
Bit errors		251 460
Mean bit-error rate		5.5×10^{-5}

FIGURE 5. — **Over-all results**

Test time

This is the total time that can be accounted for, either data were being transmitted or a line outage was being monitored. Time lost from equipment changes and scheduled maintenance (I.B.M. or G.P.O.) is not included.

Outage time

The 201A modem has the ability to recognize carrier breaks of more than six milliseconds. Loss of carrier in either direction for a period greater than six milliseconds is classified as a period of outage time.

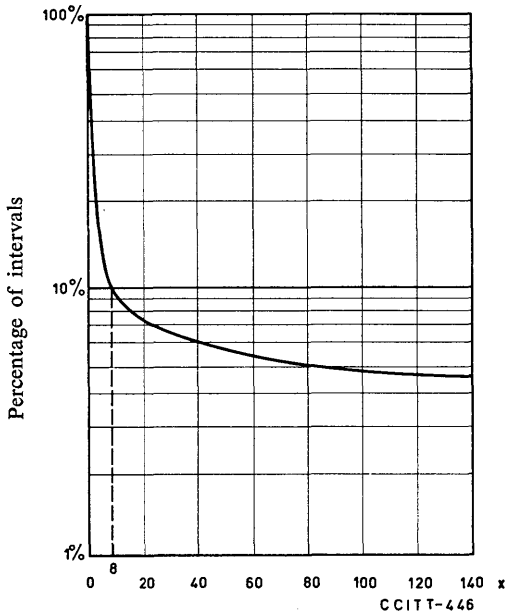
Messages in error[1] *due to :*

1. *Lost sync.* — Messages in this category resulted from the sync. pattern not being received correctly or whenever one-third or more of the received bits of a message were in error. Loss of sync. was caused by noise hits, sub-normal signal levels, or small duration drop-outs. (Drop-out is defined as loss of signal below the normal detection levels for a small duration of time, i.e. less than 6 milliseconds.)

2. *Carrier off.* — As the name implies, these error messages were caused by a carrier break of more than 6 milliseconds.

3. *Carrier off and lost sync.* — Messages in error due to both the effects described in 1 and 2.

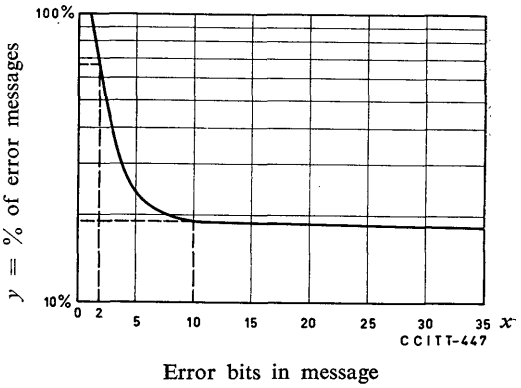
[1] Note that, of the 33 250 erroneous messages, 4442 or 13% involved loss of carrier.



D = length of interval in bits

(y = % of good bit intervals of length $\geqq x$)
Total number of intervals = 221 699

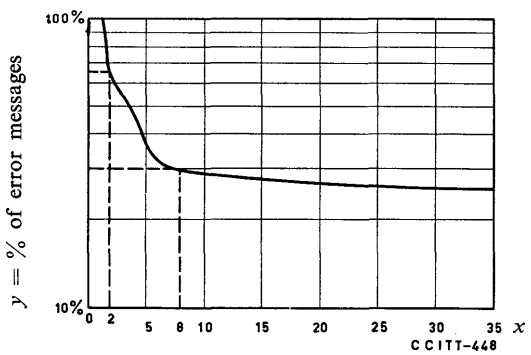
FIGURE 6. — **Distribution of error-free intervals**



Error bits in message

(y = % of messages having x or more error bits)
Error message total = 23 589

FIGURE 7. — **Distribution of error messages versus error bits**



Burst length in bits

(y = % of messages having burst length $\geqq x$)
Error message total = 23 589

FIGURE 8. — **Distribution of error bursts by bit length from first to last error bit in message**

4. *Noise and drop-out.* — Error messages not caused by any of the effects above. It is generally assumed that most of these errors can be attributed to impulse noise hits or drop-outs. Other factors that might cause errors in this category are phase distortion, frequency shift and sudden amplitude shifts.

5. *Bit-error rate.* — Only those error messages in paragraph 4 are analysed for bit-error rates and error-pattern distributions.

Bit-error rates and distributions

A different method was adopted for the evaluation of bit error rates and distributions. These are determined by analysing the intervals of good bits between error bits. (An interval between two adjacent bit errors is counted as 0 bit length.) It is assumed that the channel behaves as a normal voice channel, that is, errors tend to occur in bursts separated by short intervals of good bits. Figure 6 is a distribution of good bit intervals whose length is equal to or greater than b bits. The short-range correlation for most of the intervals indicates that most bit errors occur in small groups or bursts. This graph shows that 90.4% of all the intervals occur within a burst length of 8 or less bits.

Figure 7 provides an indication of the number of error bits occurring within any message. 81% of all error messages had less than 10 error bits while 36% had only one bit in error.

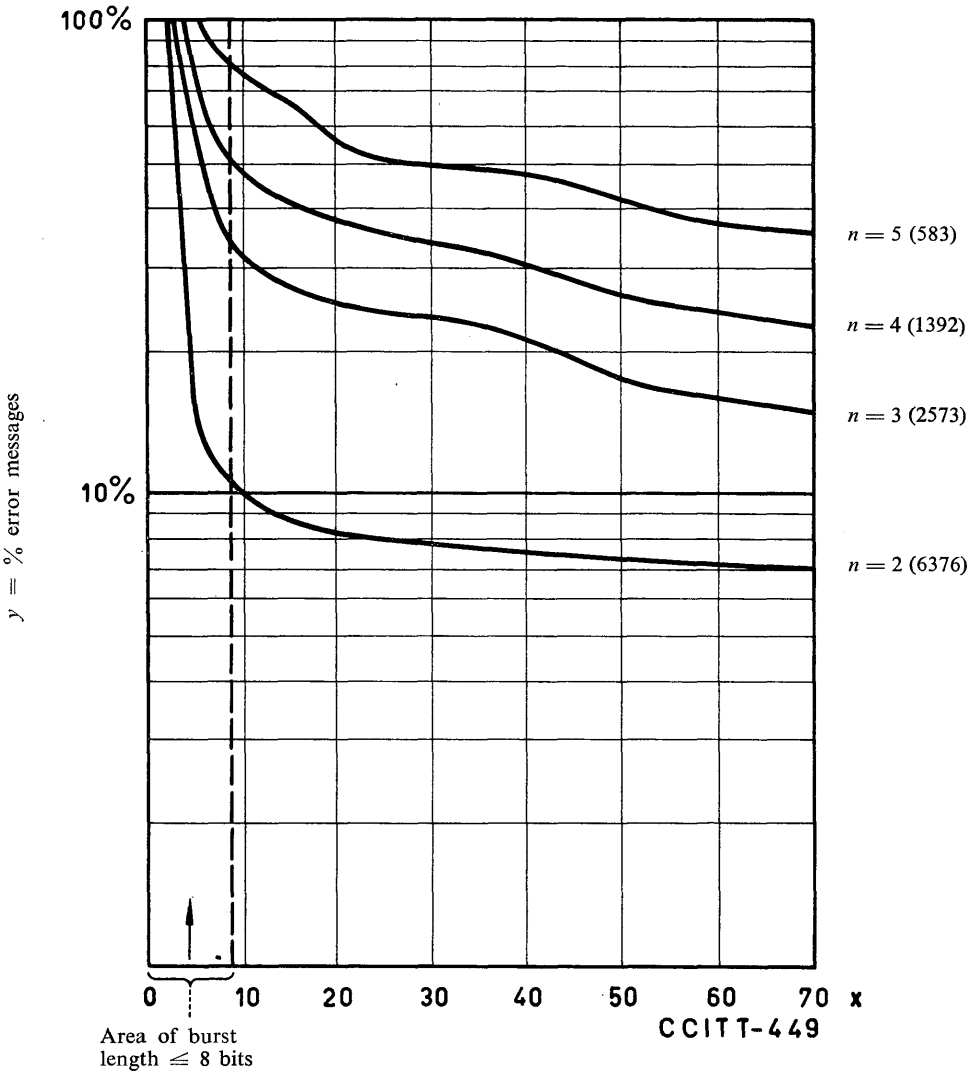
An additional picture of the bit-error patterns within a message is shown in Figure 8. This is a distribution of the burst length or spread of the distance in bits from the first error bit to the last error bit within a message. 34% of the bursts are from 2 to 8 bits in length while 30% are longer than 8 bits. From this it can be seen that at least 70% of all error bursts are less than 8 bits in length. This includes the single bit error case.

In Figure 9, the distribution of error messages having 2 to 5 error bits is shown over the burst length range up to 70 bits. Of the 6376 messages with 2 error bits, 89.5% or 5706 occurred within a burst length of 8 bits or less. Error messages having 3, 4, and 5 error bits are treated in the same manner. The total number of error messages having 2 to 5 error bits within a burst of 8 bits or less is equal to :

$$\begin{aligned}(0.895 \times 6376) + (0.67 \times 2573) + (0.51 \times 1392) + (0.23 \times 583) \\ = 5706 + 1723 + 710 + 134 \\ = 8273\end{aligned}$$

These represent all the error messages having not more than one error burst of 2 to 8 bits in length. For these cases, then, there are 2 to 5 bits in error. Although it is not possible within the framework of the present programmes to analyse those bursts greater than 8 bits, it would be reasonable to assume that most of these bursts consist of several short bursts.

This assumption appears even more valid when the distribution of sequential error bits is considered. Figure 10 shows the distribution of all sequential error-bit sequences whose lengths are equal to or greater than the abscissa. Observe that, of all sequences, 84% are single-bit errors. The predominance of single-bit errors over adjacent bit errors concurs with results obtained from previous tests using 201A modems.



n = error bits in message. Figures in parenthesis are number of error messages
y = % of messages having a burst length of x or more bits

FIGURE 9. — **Distribution of erroneous messages with n error bits versus burst length**

For error sequences of 2 or more adjacent bits, the distribution is an exponential function of the form:

$$p = Ae^{-kx}$$

where p = probability of having b or more adjacent error bits
x = $(b-2)$ adjacent error bits
k = a positive constant
A = a positive constant

This is an application of the well-known exponential law of decay:

$$f(t) = Ae^{-kt}$$

now if $p_1 = Ae^{-k(b-2)}$ is the probability of having b or more adjacent bits in error,
then $p_2 = Ae^{-k(b-2+1)}$ is the probability of having $(b+1)$ or more bits in error,
then $$\frac{p_2}{p_1} = \frac{Ae^{-k(b-1)}}{Ae^{-k(b-2)}}$$
$$= e^{-k}$$
and $$p_2 = p_1 e^{-k}$$

This may be stated in another way. If a probability p_1 exists of having n adjacent bits in error where $n > 1$, then the probability of having $n+1$ bit in error is $p_1 e^{-k}$ (e^{-k} for this test is calculated to be approximately 0.5).

A distribution of error-free intervals for all messages having two or more bit errors is shown in Figure 11, which is an extension of the curve shown in Figure 6. For intervals longer than 150 bits, the distribution is also an exponential function of the form:

$$y = Ae^{-kx} \text{ where } x = (b - 150) \text{ bits}$$

This seems to indicate that at least one error-free state exists between bursts of errors. The formula used to determine the relationship of the probabilities of adjacent error bits within a burst can be applied to the relationship of adjacent error-free bits in this case.

Let p_1 be the probability of having n or more consecutive good bits (where $n > 150$); then the probability of the $(n+1)^{\text{th}}$ bit not being in error is:

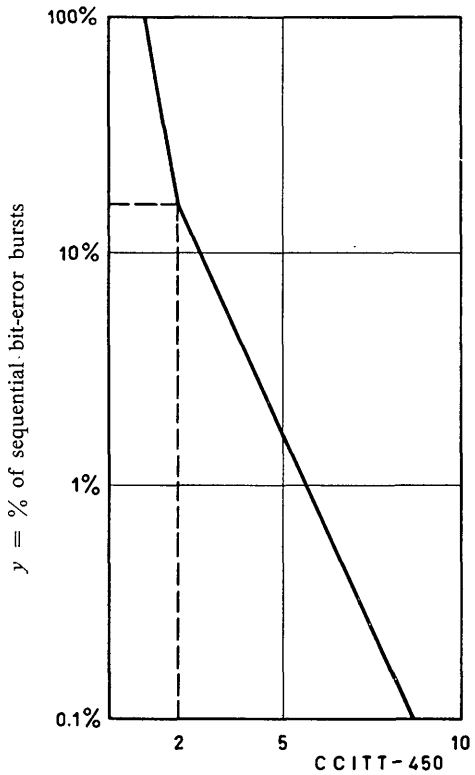
$$p_2 = p_1 e^{-k}$$

Using values from Figure 11, the value of k for any b within the range $150 < b < 792$ is calculated to be 0.0000908. By approximation:

$$e^{-k} \approx 1 - k = 0.9999092$$
$$p_2 = 0.9999092\ p_1$$

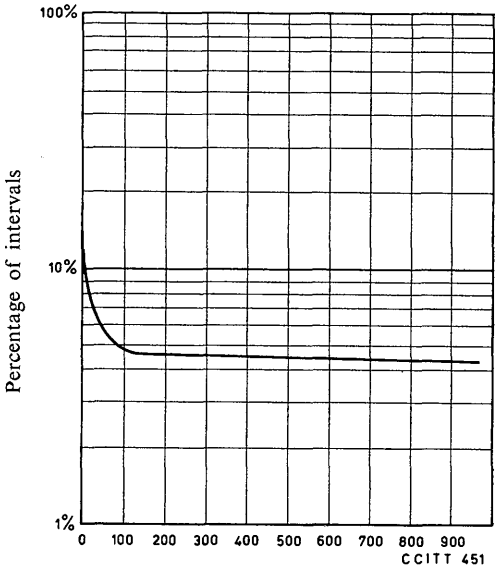
For how large a value of b the exponential function remains constant is not clear, as the reduction programmes do not provide the information required to determine the distribution pattern of error-free intervals greater than one message length (792 bits). As these longer intervals provide the greatest probability of message throughput, the average time of these intervals can be of interest.

Discounting those periods when messages were in error due to conditions other than noise and drop-out, the number of messages transmitted was: 5 756 517 which is equivalent to 43 173.88 minutes at a transmission rate of 8000 messages/hour.



(y = % of bursts having x or more error bits in sequence)
Total bursts = 179 569
b = number of error bits in sequence

FIGURE 10. — **Distribution of bursts of error bits in sequence versus length of burst**



(y = % of intervals having b or more bits)
Total number of intervals = 221 699
b = length of intervals in bits

FIGURE 11. — **Distribution of error-free intervals between two error bits versus length of interval**

If N is the total number of bits representing those intervals whose length is less than 792 bits, then :

$$N = \sum_{i=0}^{792} x_i f_i + \sum_{i=0}^{792} f_i$$

$$= 958\,247 \text{ bits}$$

where x_i is any such interval and f_i is the frequency of that interval.

At a transmission rate of 2000 bps, this is equivalent to 7.98 minutes. Therefore, the sum of all other intervals must be :

$$= 43\,173.88 - 7.98 = 43\,165.90 \text{ minutes.}$$

As there were 9242 intervals greater than one message length, an approximate working value for the average time of such an interval would be :

$$\frac{43\,165.90}{9242} = 4.67 \text{ minutes.}$$

Some allowance should also be made of the fact that no datum was transmitted in the idle period between messages. If it is assumed that the probability of an error occurring within the inter-message gap (50 milliseconds) is equal to the probability of having an error within a message (400 milliseconds), then the average time for a long interval would be:

$$4.67 \times \frac{400}{450} = 4.15 \text{ minutes.}$$

The average number of bits that could be transmitted during this time is:

$$4.15 \times 60 \times 2000 = 498\,000 \text{ bits}$$

which is equivalent to a mean bit-error rate of:

$$\frac{1}{498\,000} \approx 2 \times 10^{-6}$$

The probability of being in this relatively error-free state for intervals greater than 792 bits can be expressed as:

$$\begin{aligned} p &= \frac{\text{total time of intervals}}{\text{transmission time}} \\ &= \frac{43\,165.90}{43\,173.88} \\ &= 0.9998 \end{aligned}$$

This is true only for the time when the line was available and operating at a normal level. These conditions are discussed in the section on outages.

Outages

Probably the most significant results obtained from this test were those that related to outages. The definition of on outage is somewhat arbitrary depending to a great extent on how the line is used. For example, a one-second break in a voice transmission might not be considered too serious. However, if data were being transmitted at 2000 bps, the same break would cause several messages to be lost.

Loss of effective transmission can be caused by complete loss of the carrier or a change in the line characteristics of sufficient magnitude to produce abnormal operating conditions. In this report, an outage will be considered as either:

1. Any period of time that the carrier could not be detected by the receiving modem in either direction of transmission (the minimum loss of carrier time the 201A can detect is 6 milliseconds),

or:

2. Those periods during which a large number of consecutive messages were in error due to a prolonged abnormal line condition. (Such a situation, if it had occurred while the machine was attended, would have caused the test to be stopped and the line condition would have been reported to the telephone company as an outage condition.)

An analysis of the data shows that more than 95% of the lost sync. messages occurred owing to prolonged abnormal line conditions.

A preliminary evaluation of the test results included outages reported during the time the I.B.M. modems were in use, June 11-June 20. A more recent study failed to establish sufficient proof that these suspected outages were caused by line conditions. For this reason, the analysis of outages will be confined to those that occurred during the test period April 21 to June 10.

The line was monitored for outages during this period for 822.1 hours. Table II shows the number and duration of these outages in eight time ranges. From this table, it is apparent that most outages are relatively short and have a negligible effect on total line availability. They do, however, affect line efficiency in terms of error rates. Line availability is mainly determined by the longer outages of which there are relatively few. A measure of the line availability can be given as :

$$\frac{\text{test time} - \text{outage time}}{\text{test time}} \times 100\,\%$$

$$= \frac{822.1 - \frac{2838.40}{60}}{822.1} \times 100\,\%$$

$$= 94{,}25\,\%$$

TABLE II

Outage frequency and time lost by range of duration

Range of outage duration (minutes)	Number of outages	Relative frequency (%)	Time lost (minutes)	Percentage of total time lost
0.0001 to 0.001	685	69.68	0.50	0.017
0.001 to 0.01	83	8.44	0.38	0.013
0.01 to 0.1	105	10.68	2.96	0.104
0.1 to 1	68	6.92	29.59	1.042
1 to 10	23	2.34	62.68	2.208
10 to 100	14	1.42	382.89	13.489
100 to 1000	4	0.41	737.40	25.979
1000 and up	1	0.10	1622.00	57.143
Total	983		2838.40	

It should be kept in mind that the sample period is quite small for long outages and a longer test might produce a more significant figure of line availability.

Most outages were evenly distributed on a day-to-day basis throughout the test period. If the starting times of outages are examined on a two-hourly basis as shown in Figure 12, there is more tendency for outages to occur during the day than at night (the instance between 1 a.m. and 2 a.m., 9 June, is an exception).

178 occurred Saturday 9.6.63

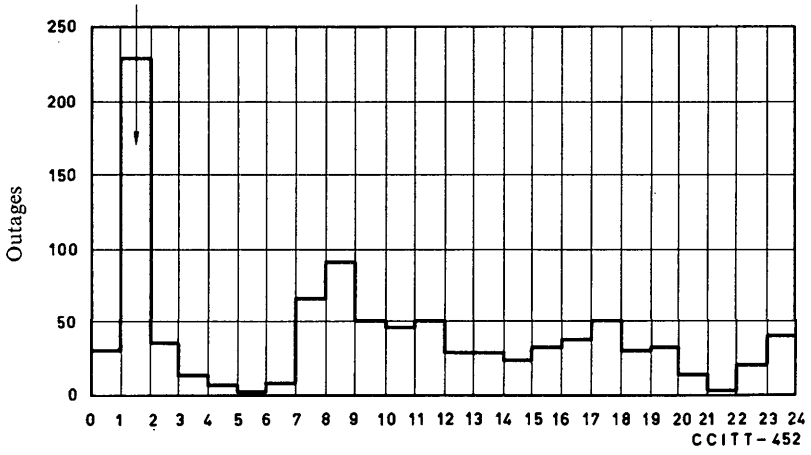


FIGURE 12. — **Distribution of all outages versus hour of day**

TABLE III

Outages greater than 10 minutes

	Time [1] outage started	Date	Length of outage (minutes)
Business hours	0030	8 May	24.2
	0130	25 April	14.5
	0330	29 April	20.0
	0400	23 May	113.4
	0900	8-9 May	1622.0
	0930	5 May	11.1
	1000	26 May	12.6
	1100	1 June	14.0
	1100	3 May	335.0
	1130	10 May	33.0
	1230	28 May	97.0
	1300	27 April	170.0
	1300	7 May	119.0
	1430	25 May	18.2
	1530	25 May	26.0
	1600	1 June	15.4
	1630	26 May	40.7
	1700	1 June	28.1
	1900	24 April	27.9
Total time lost (minutes) : 2742.1			

[1] Time given to nearest half-hour.

The probability of outages occurring during the day rather than at night is more noticeable for outages of more than 10 minutes. This is shown in Table III. Of the 19 outages in this category, 14 occurred during normal European business hours and were reported as line faults to the G.P.O.

Error detection

Several check polynomials representing shift register codes were used to test the effectiveness of cyclic-code error detection for the data obtained from the test. A polynomial of this type will describe the wiring of the shift register and the class of error patterns it will detect. Because errors occur in bursts, and the bursts themselves tend to occur in clusters, it would be very difficult to calculate the theoretical effectiveness of cyclic-code error detection. However, an approximate value of effectiveness can be obtained by assuming that all errors are independent with equal probability of occurrence. If this assumption were true, a code of this type should fail to detect only one out of 2^k erroneous messages where k is the number of check bits or the degree of the check polynomial. It would then follow that the percentage of erroneous messages that a check polynomial can be expected to detect is :

$$\frac{2^{k-1}}{2^k} \times 100\,\%$$

In Table IV, the expected effectiveness of the check polynomials is compared with the actual results obtained with the test data. All erroneous messages are included, regardless of the category in which they fall. It can be seen that, of the seven codes tested, the first four are more effective in detecting the errors than is expected. An examination of the polynomials shows that $x^6 + 1$ is a simple parity check while $x^6 + x^5 + 1$, $x^6 + x + 1$ and $x^7 + x^3 + 1$ are primitive and irreducible. The remaining polynomials are factorable.

TABLE IV

Effectiveness of several check polynomials using cyclic codes

Polynomial	Percentage of erroneous messages detected	
	Expected	*Actual*
$x^6 + 1$	98.44	98.93
$x^6 + x^5 + 1$	98.44	99.25
$x^6 + x + 1$	98.44	99.34
$x^7 + x^3 + 1$	99.22	99.65
$x^{12} + 1$	99.97	99.79
$x^{12} + x^{11} + 1$	99.97	99.97
$x^{12} + x + 1$	99.97	99.96

General observations

Comparison of "go" and "return" legs

Although the received signal level at London seemed relatively stable at $-4 + 1$ dbm with respect to the transmission level at Rome, the same could not be said of the signal level received at Rome. Figure 13 is a frequency distribution of the received signal levels in Rome. Readings were taken on a daily basis and are shown with respect to the level of transmission at London.

Other differences that were observed concerned outages and impulse noise. In the event that an outage affected only one leg, the "go" leg was the one most likely to be out. Impulse noise measurements made at each terminal showed a much higher incidence and level of noise hits on the "go" leg with respect to the "return" leg. It is possible that these problems could have been detected and resolved somewhat if both legs had been fully tested during the initial test phase.

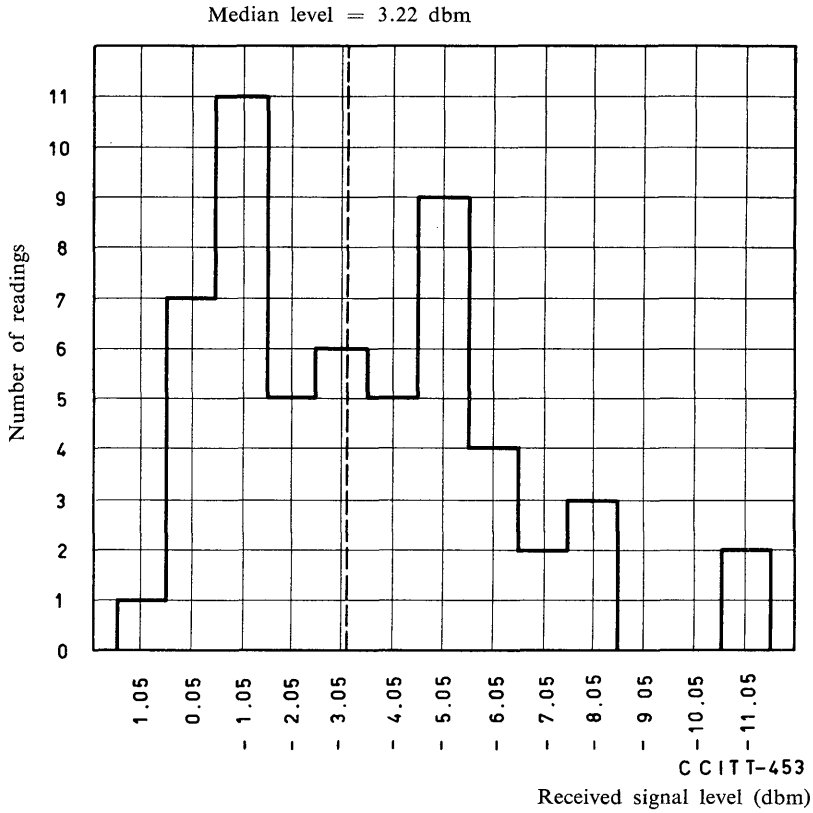


FIGURE 13. — **Distribution of received signal level at Rome April 21 1962 to June 10 1962**

Reliability and availability

The complete test circuit was originally ordered by I.B.M. from four independent common carriers. Operating under the jurisdiction of these four agencies, it soon became apparent that some major problems existed with regard to the reliability and availability of the complete circuit.

As a result of this experience, general agreement was reached between the common carriers to re-order the circuit under the control of one master location. This was accomplished on 9 June. Unfortunately, scheduled operations with the I.B.M. modems after this date and for the remainder of the test did not allow for an accurate comparison of the circuit performance before and after this transfer date.

Summary

The following conclusions are based upon the analysis of the test data along with information obtained from several reliable sources :

1. When the London-Rome circuit was available and operating within its normal limits[1], the quality of data transmission of 2000 bps with the Western Electric 201A modem was acceptable. The error rates and distributions did not differ significantly from what is normally expected with this type of operation over equivalent voice grade facilities within the United States.

2. Most outages were quite short (less than one minute) but the remainder were relatively long. Both short and long outages tended to occur more frequently during the day.

3. The control and operation of the circuit needed a master location for all maintenance problems.

4. A comprehensive set of line specifications (similar to Schedule 4B, FCC Tariff 237) would have helped to determine if and when the circuit was suitable for data transmission. To the best of our knowledge, no such specifications exist for circuits of the London-Rome type in Europe.

5. This test represents an initial effort to determine the performance of 2000 bps data transmission over a long-haul, multipoint network in Europe. The results were obtained on what must be considered a special circuit and therefore should not be used to represent other circuits that now exist, or which may be installed in the future. The quality of performance obtained on the London-Rome circuit, however, may be used as an indication of what might be expected on a similar European international network operating at 2000 bps.

Supplement No. 38

NIPPON TELEGRAPH AND TELEPHONE PUBLIC CORPORATION — *(Extracts from contributions COM Sp. A/No. 3 — July 1961 and COM Sp. A/No. 22 — October 1961)*

DATA TRANSMISSION TESTS OVER POINT-TO-POINT TELEPHONE CIRCUITS ON RADIO-RELAY LINKS

Extract from contribution COM Sp. A/No. 3 — July 1961

1. *Introduction*

This is a report of the point-to-point data transmission test over the telephone circuit on 4000 Mc/s SHF radio-relay links.

[1] Normal limits refer to the limit of signal level variation as suggested by the G.P.O.

2. *Description of the data transmission test*

2.1 *Data transmission test system*

In the data transmission test, it is desirable to investigate the characteristics of the data circuit over telephone lines apart from the problem of synchronization of the input and output machines. For this reason, a timing signal was used to keep the input and output machines (in this case a message pattern generator and a comparator) in complete synchronization.

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2.2 *Measuring equipment*

The specifications of the principal measuring equipment used in this test are as follows:

1) *Envelope delay analyser :*

a) Frequency range	200 c/s-10 kc/s
b) Delay accuracy	± 10 seconds

2) *Message pattern generator :*

a) Code capacity	maximum 128 bits (a combination out of 128 bits can be arbitrarily formed by push-buttons)
b) Pattern length	2-128 bits (arbitrarily changeable)
c) Modulation rate	50-3000 bauds
d) Telegraph distortion	less than 0.5%
e) Types of signal	double current, single current, and dipulse
f) Elements	transistors, 95, and diodes, 320

3) *Comparator* (error detector):

a) Margin	about 50%
b) Items of error check	bit errors, stop polarity to start polarity bit errors, start polarity to stop polarity bit errors, error letters (1 letter = 8 bits), error blocks (1 block = 128 bits or 256 bits), undetected errors by even parity check, undetected errors by 4-out-of-8 check, undetected errors by group check
c) Elements	transistors, 75, and diodes, 180

4) *Signal distortion analyser :*

a) Types of distortion	isochronous distortion and start-stop distortion
b) Number of bits between start and stop signal	1-99 bits

c) Modulation rate	50-3000 bauds
d) Accuracy	1%
e) Items of measurement	maximum distortion average distortion distribution of distortions (range of class : 2.5 and 10%)
f) Elements	transistors, 500, and diodes, 1200

5) *Circuit interruption measuring set :*

Pilot signal	1 kc/s or 2 kc/s
Accuracy of duration measurement	1 millisecond or 0.5 millisecond
Element	decatron

6) *Noise amplitude distribution analyser :*

Type	sampling type (sampling frequency 50 c/s)
Accuracy	± 1 db
Dynamic range	30 db

2.3 *Modulators and demodulators for data transmission*

In this test, three types of modulation system were used :

1) *AM system.* — The modulator is the usual type and the demodulator is an envelope detector with a level variation compensator.

2) *FM system.* — As a modulator, frequency shift keying system by two pulsed oscillators is used and the demodulator is a zero-crossing detector.

3) *PM system.* — The demodulator is a homodyne detector.

2.4 *Test specifications*

Message pattern

In binary signal transmission, each characteristic instant is influenced by the residual response of the preceding bits ; therefore, the characteristic distortion of the particular characteristic instants varies according to the combination of the preceding bits. It has been proved that any combination of 6 bits before the particular transition can be contained in the $(2^7 - 1)$ bits message pattern. The test message pattern was constructed according to this principle but it consists of 128 bits.

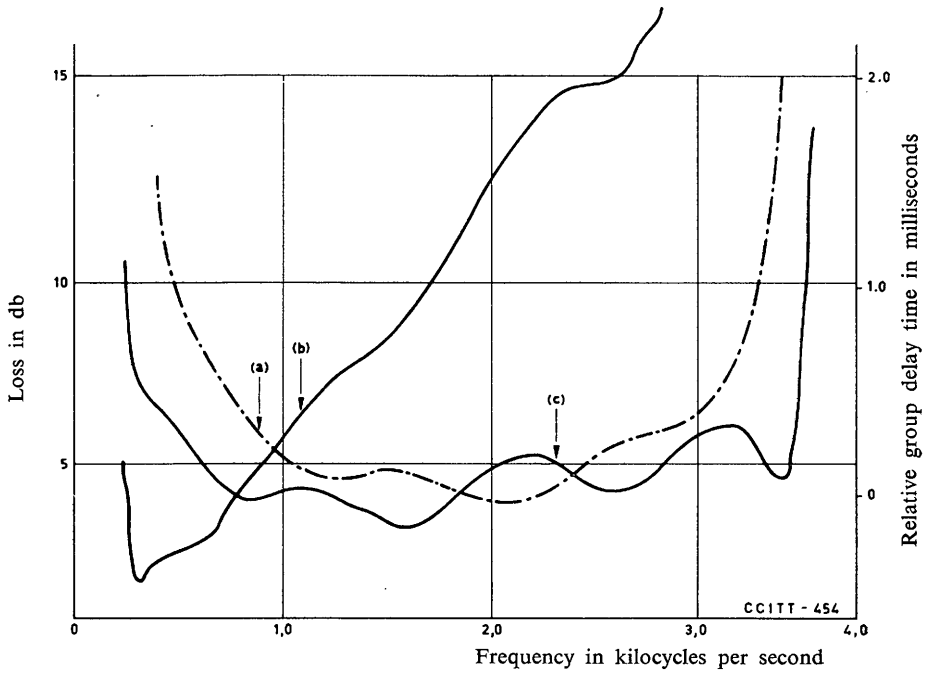
3. *Test circuit* (first series)

3.1 *Arrangement of the test circuit*

A carrier telephone circuit on 4000 Mc/s SHF radio-relay links between Tokyo and Fukuoka (about 1200 km) was used for the test circuit. A carrier telephone circuit on an unloaded cable was employed as the timing circuit ; thus errors according to the failure of timing signal transmission were separated from errors of main transmission.

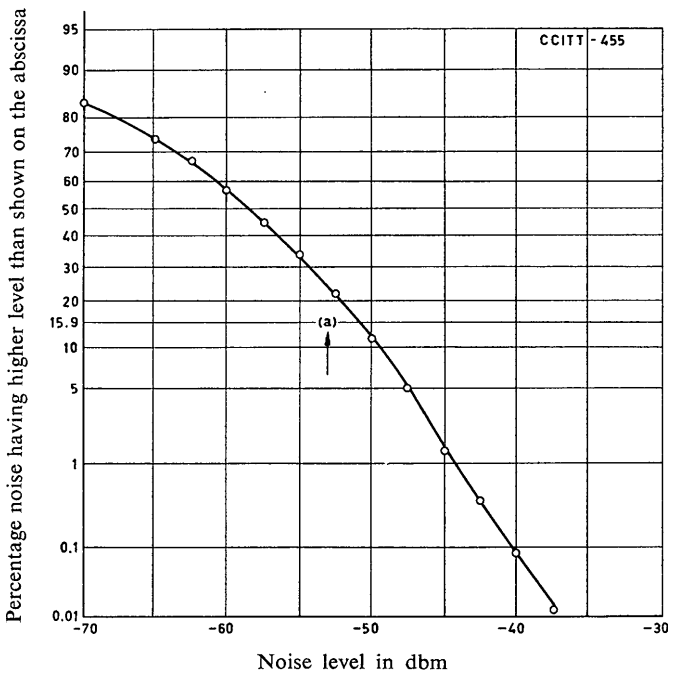
3.2 *Characteristics of the test circuit*

1) *Frequency response of the test circuit* — The net loss characteristics and envelope delay response are shown in Figure 7. The net loss distortion, due to the extension cables between



(a) Relative group delay time
(b) Loss, before equalization
(c) Loss, after equalization

FIGURE 7. — **Loss and phase characteristics of transmission circuit**



(a) = Mean level by level meter —53 dbm

FIGURE 8. — **Level distribution characteristics of noise**

Fukuoka telegraph terminal and Fukuoka-higashi telephone office and those between Tokyo telephone repeater station and Shibuya telephone repeater station (see Figure 6), was equalized but the envelope delay distortion was not equalized.

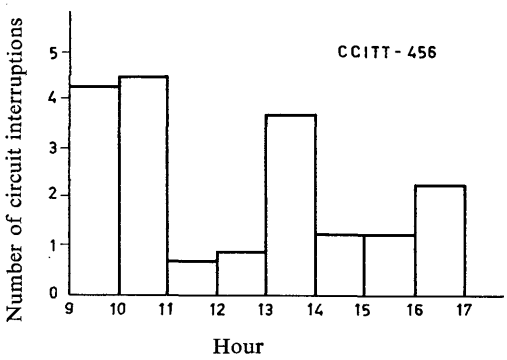


FIGURE 9. — **Time distribution of circuit interruptions**

2) *Noise* — A typical example of the amplitude distribution characteristic of the test circuit noise is shown in Figure 8.

3) *Circuit interruption* — Circuit interruption is one of the most frequent causes of transmission errors. In Figure 9, the number of test circuit interruption per hour is shown. Intervals between successive interruptions and durations of the interruptions are shown in Tables 2 and 3. These data were taken during the business hours of the test period. It is noted that a number of short circuit interruptions, less than 5 milliseconds, which do not influence the quality of 50-baud telegraph transmission was more than half of the total

TABLE 1

Specification of error rate measurement

Types of modulation	Modulation rates in bauds	Signal levels (at relative zero level)	Number of test bits
AM	1500	— 6 dbm (normal)	4×10^7
		— 12 dbm	2×10^7
	1750	— 6 dbm (normal)	4×10^7
		— 12 dbm	2×10^7
FM	1000	— 10 dbm (normal)	4×10^7
		— 16 dbm	2×10^7
	1500	— 10 dbm (normal)	4×10^7
		— 16 dbm	2×10^7

(The size of samples used to measure the telegraph distortion is 3×10^5 bits)

interruptions. Also, there were five long interruptions which were for a few seconds or a few minutes duration. If the errors caused by these long interruptions and those longer than 300 milliseconds were deleted from the results, the efficiency factor of the test circuit during the business hour will be 96-98%. (During the test period, there was one facility out of order for a duration of approximately 30 minutes. This was caused by power supply failure. Without this particular interruption the efficiency factor would be 99%.)

TABLE 2

The distribution of time intervals of circuit interruptions

Time interval between circuit interruptions (in milliseconds)	Percentage of circuit interruptions longer than the time interval shown in the left column
More than 300	0.8
,, 240	1.7
,, 180	1.7
,, 120	4.1
,, 60	13.2
,, 50	15.7
,, 40	20.6
,, 30	24.8
,, 20	31.4
,, 10	47.1
,, 5	53.7
,, 4	61.2
,, 3	66.1
,, 2	71.9
,, 1	81.8
,, 0	100

TABLE 3

The distribution of time length of circuit interruptions

Time length of circuit interruptions (in milliseconds)	Percentage of circuit interruptions shorter than the time length shown in the left column
More than 300	19.8
Less than 300	80.2
,, 200	78.3
,, 100	77.4
,, 70	75.5
,, 50	73.6
,, 40	68.9
,, 30	64.2
,, 20	62.3
,, 10	57.5
,, 5	50.9

4) *Phase variations of the signal transmitted over the test circuit* — There were two kinds of phase variations detected on the test circuit. One was slow phase variation and its maximum value was ± 15° and its repetition frequency was about 50 c/s. The other was a sudden phase variation accompanied by level variation and its maximum value reached 50°. The causes of these phase variations are now under investigation.

4. *Test results*

Error rates and telegraph distortions were measured in the AM and FM systems. Telegraph distortions were measured in the PM system.

TABLE 4

Number of total errors (AM)

Modulation rates in bauds	Signal levels	Causes of error	Number of error bits	Number of start polarity to stop polarity error bits	Number of stop polarity to start polarity error bits	Number of error letters	Number of error blocks	Number of test bits
1500	Normal	Interruption	93	3	90	28	7	4.39×10^7
		Miscellaneous	0	0	0	0	0	
		Total	93	3	90	28	7	
	Normal — 6 db	Interruption	32	3	29	10	2	2.16×10^7
		Miscellaneous	1	1	0	1	1	
		Total	33	4	29	11	3	
	Normal — 12 db	Interruption	58	12	46	16	4	0.59×10^7
		Miscellaneous	0	0	0	0	0	
		Total	58	12	46	16	4	
1750	Normal	Interruption	145	13	125	58	31	4.51×10^7
		Miscellaneous	1	1	0	1	1	
		Total	146	14	125	59	32	
	Normal — 6 db	Interruption	11	7	4	6	5	2.1×10^7
		Miscellaneous	0	0	0	0	0	
		Total	11	7	4	6	5	
	Normal — 12 db	Interruption	0	0	0	0	0	0.67×10^7
		Miscellaneous	0	0	0	0	0	
		Total	0	0	0	0	0	

Errors caused by circuit interruptions longer than 300 milliseconds are excluded.

TABLE 5

List of error rates (AM)

Modulation rates in bauds	Signal levels	Error bit rates	Error letter rates	Error block rates	Average error bits per error letter	Average start polarity to stop polarity error bits per error letter	Average stop polarity to start polarity error bits per error letter	Average error letters per error block
1500	Normal	2.12×10^{-6}	5.11×10^{-6}	4.08×10^{-5}	3.3	0.12	3.2	4.0
	Normal — 6 db	1.53×10^{-6}	4.07×10^{-6}	3.55×10^{-5}	3.0	0.36	2.6	3.7
	Normal — 12 db	9.9×10^{-6}	2.19×10^{-5}	1.75×10^{-4}	3.6	0.75	2.9	4.0
1750	Normal	3.24×10^{-6}	1.05×10^{-4}	1.82×10^{-4}	2.5	0.24	2.6	1.8
	Normal — 6 db	5.26×10^{-7}	2.34×10^{-6}	6.12×10^{-5}	1.8	1.2	0.67	1.2
	Normal — 12 db	0	0	0	0	0	0	0

Errors caused by circuit interruptions longer than 300 milliseconds are excluded.

4.1 *Error rates*

1) *Amplitude modulation* — Amount of errors and error rates at various modulation rates and signal levels are shown in Tables 4 and 5. Most of the errors were caused by circuit interruptions. Average error bits per error letter and average error letters per error block are shown in Table 5.

2) *Frequency modulation* — Same items in the AM system are shown in Tables 6 and 7. Stop polarity to start polarity errors caused by circuit interruptions are more than the start polarity to stop polarity errors in the AM system, while both types of errors are almost the same in quantity in the FM system. This is because the polarity of error depends on the signal polarity when an interruption occurs.

4.2 *Telegraph distortion*

Telegraph distortions generated during the transmission were measured in the AM, FM and PM systems. Distribution curves of the distortion resemble a normal distribution curve without a skirt and with a finite maximum value. When the signal level is high, the curves are not smooth owing to the effect of the regular distortion. For the lower signal level, distribution curves of the distortion become smoother and their standard deviations become greater. That these curves

TABLE 6

Number of total errors (FM)

Modulation rates in bauds	Signal levels	Causes of error	Number of error bits	Number of start polarity to stop polarity error bits	Number of stop polarity to start polarity error bits	Number of error letters	Number of error blocks	Number of test bits
1000	Normal	Interruption	83	39	44	37	18	4.18×10^7
		Noise	1	1	0	1	1	
		Miscellaneous	3	2	1	3	3	
		Total	87	42	45	41	22	
		Grand total	545	87	457	245	36	
	Normal — 6 db	Interruption	54	28	27	16	5	2.65×10^7
		Noise	1	0	1	1	1	
		Miscellaneous	2	1	1	2	2	
		Total	57	29	29	19	8	
		Grand total	584	40	546	152	14	
1500	Normal	Interruption	6	2	4	5	4	4.05×10^7
		Noise	10	4	8	9	5	
		Miscellaneous	5	3	2	3	4	
		Total	21	9	14	17	13	
		Grand total	713	111	602	199	23	
	Normal — 6 db	Interruption and noise	69	40	30	29	14	2.04×10^7
		Miscellaneous	0	0	0	0	0	
		Total	69	40	30	29	14	
		Grand total	302	255	48	97	19	

Grand total includes errors caused by circuit interruptions longer than 300 milliseconds.

TABLE 7

List of error rates (FM)

Modulation rates in bauds	Signal levels	Class	Error bit rates	Error letter rates	Error block rates	Average error bits per error letter	Average error bits per error block	Average error letters per error block
1000	Normal	A	1.35×10^{-5}	4.72×10^{-5}	2.55×10^{-4}	2.22	15.1	6.8
		B	2.12×10^{-6}	8.01×10^{-6}	1.38×10^{-4}	2.12	4.0	1.9
	Normal — 6 db	A	2.72×10^{-5}	5.65×10^{-5}	1.67×10^{-4}	3.84	41.7	11.0
		B	2.65×10^{-6}	7.07×10^{-6}	9.52×10^{-5}	3.00	7.1	2.4
1500	Normal	A	1.75×10^{-6}	3.90×10^{-5}	1.45×10^{-4}	3.58	31.0	8.7
		B	5.20×10^{-7}	3.34×10^{-6}	8.28×10^{-5}	1.23	1.6	1.3
	Normal — 6 db	A	1.54×10^{-5}	4.0×10^{-5}	2.5×10^{-4}	3.11	15.9	5.1
		B	3.52×10^{-6}	1.18×10^{-5}	1.84×10^{-4}	2.38	4.9	2.1

Class A includes errors caused by circuit interruptions longer than 300 milliseconds.
Class B excludes errors caused by circuit interruptions longer than 300 milliseconds.

are not smooth might be due to the repetitive message pattern of 128 bits. In an actual data communication, distribution curves of the distortion could be taken as a normal distribution.

Modulation rate versus maximum isochronous distortion curves in various systems are shown in Figure 10. These curves determine a maximum possible modulation rate when the maximum allowable distortion is given. If we take 5% for the value mentioned above in practical applications, a maximum possible modulation rate would be 1500 bauds in the AM system, 1250 bauds in the FM system and 1600 bauds in the PM system.

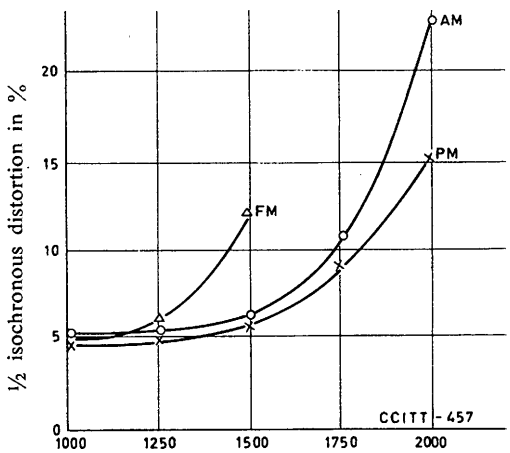


FIGURE 10. — **Isochronous distortion versus modulation rates**

TABLE 8

Standard deviations of half isochronous distortion

Types of modulation	Modulation rates in bauds	Signal levels	Standard deviations of ½ isochronous distortion in %
AM	1500	Normal	2.6
		Normal −6 db	2.6
		Normal −12 db	3.3
	1750	Normal	4.9
		Normal −6 db	5.0
		Normal −12 db	6.6
FM	1000	Normal	2.8
	1250	Normal	3.4
		Normal −6 db	3.4
		Normal −12 db	3.6
		Normal −16 db	3.6
	1500	Normal	6.7
		Normal −6 db	7.2
PM	1000	Normal	1.7
	1250	Normal	2.2
	1500	Normal	2.6
	1750	Normal	3.0
	2000	Normal	5.9

The limit given for the FM system is lower compared with the others. This comes from the particular modulation method employed in the FM system used in this experiment. In this FM system, lower frequency f_1 and higher frequency f_2 should meet the following condition :

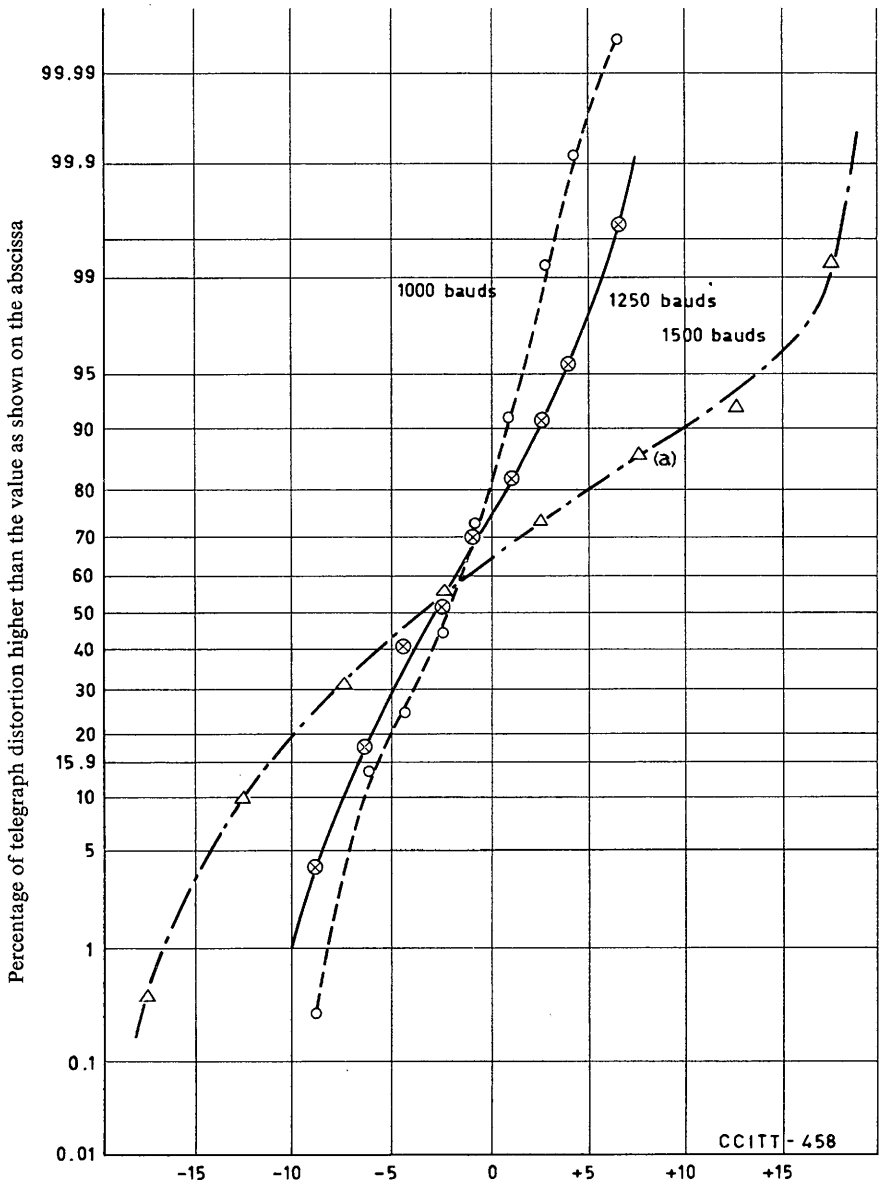
$$2f_1 = f_2$$

In the case of a modulation rate of 1500 bauds :

$$f_1 = 1500 \text{ c/s} \qquad f_2 = 3000 \text{ c/s}$$

A part of the frequency spectrum is limited by the telephone channel filter.

In the other FM system, in which the frequency spectrum can be arranged symmetrically as regards the amplitude characteristic of the transmission line, the modulation rate can be raised more than 1250 bauds. This is now under experiment.



(a) = Normal level −6 db

FIGURE 11. — **Examples of telegraph distortion distribution in the FM system at some modulation rates**

Standard deviations of the distribution of the isochronous distortion for each modulation system at various modulation rates are shown in Table 8. Figure 11 shows curves of the distribution of isochronous distribution at some modulation rates in the FM system as an example.

5. *Summary of the test results*

1. A modulation rate of 1500 bauds is usable without compensation for the phase distortion of the transmission line when one voice link is considered.

2. Most of the errors are caused by circuit interruptions. Half of the circuit interruptions are less than 5 milliseconds.

3. Little changes in the error rate and the telegraph distortion are found when signal levels are made 6 db lower than the standard value recommended in the test specifications "L" of the Study Programme annexed to Question 1/A.

4. It is desirable that the highest usable limit of the modulation rate should be determined from the viewpoint of the telegraph distortion. As to the systems tested in this experiment, a practical speed would be 1500 bauds in the AM system, 1250 bauds in the FM system and 1600 bauds in the PM system.

5. Average error bit rate during the business hours is about $2\text{-}4 \times 10^{-6}$. This rate becomes five to ten times as high when errors caused by the circuit interruptions beyond 300 milliseconds are included.

6. Average error bits per letter are 3 (1 letter = 8 bits) and average error letters per error block also 3.

Extract from contribution COM Sp. A/No. 22 — October 1961

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2. *Test circuit of the second series*

Three telephone circuits were used as data transmission path, circuit interruption monitoring circuit and noise measuring circuit.

2.1 *Arrangement of the test circuit*

The arrangement of the test system is shown in Figure 1 schematically.

2.2 *Characteristics of the test circuit*

1) *Frequency response of the test circuit*

The net loss characteristics and envelope delay response are shown in Figure 2. The envelope delay distortion was not equalized but the net loss distortion due to the extension cables was equalized.

2) *Noise*

The noise level of the noise monitoring circuit was measured throughout the test but S/N during the test period was constant and about 38 db and few errors were brought about by the circuit noise.

3) *Circuit interruption*

The number of circuit interruptions from 24 July to 31 July and their causes are tabulated in Table 1. The time lengths of circuit interruptions are shown in Table 2.

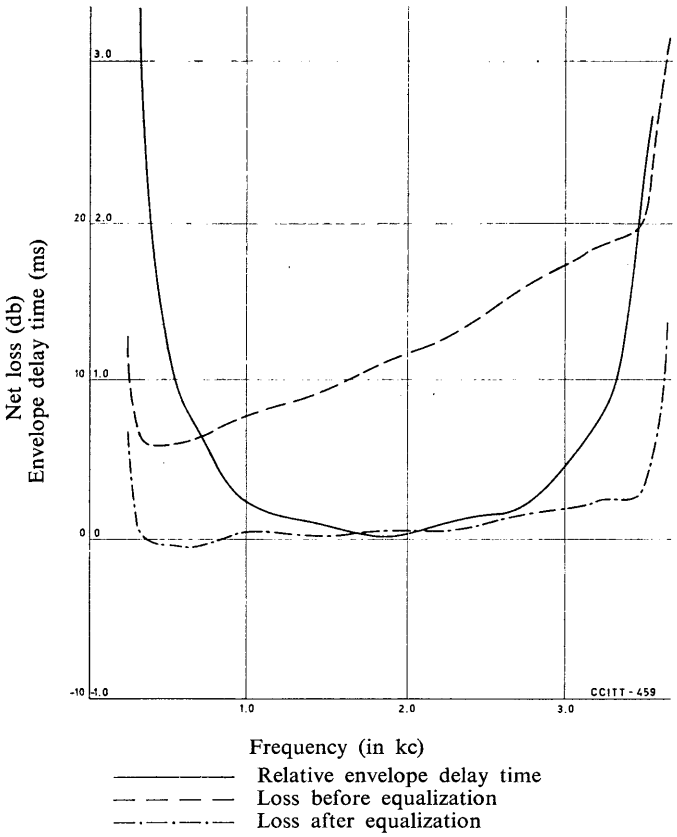


FIGURE 2. — **Frequency characteristics of test circuit**

TABLE 1

The number of the circuit interruptions and their causes

Causes of interruption / Date	Change-over of the power supplies	Change-over of the modulators	Change-over of the demodulators	Change-over of the carrier supplies	Unknown		Total
					Only known is repeater section where interruptions occurred	Unknown where interruption occurred	
24 July	1					5	6
25 July	1				1	1	3
26 July		3	14		3	12	32
27 July			32			6	38
28 July			8	9	3	9	29
29 July	1			3	2	5	11
30 July			2			1	3
31 July		9				4	13
Total	3	12	56	12	9	43	135

TABLE 2

The time length of circuit interruptions

Time length of circuit interruption	Percentage of the number of interruptions to the total number (%)
Longer than 300 ms	7.1
Shorter than 300 ms	92.9
Shorter than 200 ms	92.9
Shorter than 100 ms	90.5
Shorter than 70 ms	89.0
Shorter than 50 ms	82.6
Shorter than 40 ms	81.0
Shorter than 30 ms	77.0
Shorter than 20 ms	73.8
Shorter than 10 ms	70.6
Shorter than 5 ms	68.3

TABLE 3

Circuit interruptions caused by change-over of the power supply system

Types of change-over		Time length of interruption	Number of error bits
Sending station	Three-engine generator to automatic voltage regulator	127 ms	83
	Automatic voltage regulator to three-engine generator	63 ms	40
	Three-engine generator. Normal equipment to stand-by equipment	104 ms	104
	Three-engine generator. Stand-by equipment to normal equipment	1207 ms	783
Receiving station	Three-engine generator to automatic voltage regulator	0	0
	Automatic voltage regulator to three-engine generator	0	0
	Three-engine generator. Normal equipment to stand-by equipment	0	0
	Three-engine generator. Stand-by equipment to normal equipment	0	0

TABLE 4

Circuit interruptions caused by the change-over of the modulator and demodulator

Types of change-over	Time length of interruption	Number of error bits
Mod.: Normal to stand-by	36 ms	30
Mod.: Stand-by to normal	13 ms	8
Dem.: Normal to stand-by	2 ms	2
Dem.: Stand-by to normal	3 ms	2

TABLE 5

Circuit interruptions caused by the change-over of the transmission system

Types of change-over	Time length of interruption	Number of error bits
Automatic change-over: Normal to stand-by.	34-264 ms	26-204
Automatic change-over: Stand-by to normal.	2-3 ms	1-2
Manual change-over: Normal to stand-by	0	0
Manual change-over: Stand-by to normal	2 ms	1

TABLE 6

Number of total errors

Kinds of modulation system	Modulation rates in bauds	Signal levels	Causes of errors	Number of error bits	Number of start polarity to stop polarity error bits	Number of stop polarity to start polarity error bits	Number of error letters	Number of error blocks	Number of test bits
A.M. D.S.B.	1500	— 12 dbm	Interruption	660	375	303	360	116	1.57×10^8
			Miscellaneous	8	3	5	8	4	
			Total	668	378	308	368	120	
F.S.	1250	— 16 dbm	Interruption	412			127	30	1.0×10^8
			Miscellaneous	16			11	8	
			Total	428			138	38	
Total			Interruption	1072			487	146	2.57×10^8
			Miscellaneous	24			19	12	
			Total	1096			506	158	

TABLE 7

List of error rates

Kinds of modulation system	Modulation rates in bauds	Signal levels	Error bit rates	Error letter rates	Error block rates	Average error bits per error letters	Average error letters per error block
A.M. D.S.B.	1500	− 12 dbm0	4.26×10^{-6}	1.88×10^{-5}	1.96×10^{-4}	1.82	3.07
F.S.	1250	− 16 dbm0	4.25×10^{-6}	1.10×10^{-5}	9.6×10^{-5}	3.1	3.62
Total			4.26×10^{-6}	1.57×10^{-5}	1.57×10^{-4}	2.16	3.2

3. *Circuit interruptions caused by maintenance operations and their effects on data transmission*

In advance of the data transmission test, the time length of interruptions caused by maintenance operations and their effects on transmission performance were measured, and are as shown in Table 3, Table 4 and Table 5.

4. *Test results of data transmission*

4.1 *Number of errors*

Numbers of error bits, error letters and error blocks are tabulated in Table 6.

4.2 *Error rate*

Bit-error rate, letter-error rate and block-error rate are shown in Table 7. Number of error bits per error letter and number of error letters per error block are shown in the table too.

5. *Summary*

1. 98% of the error bits were caused by circuit interruption.

2. The average number of circuit interruption per day was about 17 and 68% of the interruptions were shorter than 5 milliseconds.

3. The average bit-error rate during the business hour was about 4×10^{-6}.

4. If errors caused by the long circuit interruptions (longer than 300 milliseconds) were included in the results, the error-bit rate increases to twice.

5. The number of error bits per error letter was 2.16 and the number of error letters per error block was 3.2.

Supplement No. 39

MASSACHUSETTS INSTITUTE OF TECHNOLOGY — LINCOLN LABORATORY —
(Contribution COM Sp. A/No. 70 — June 1963)

TRANSMISSION TESTING ON ALASKAN FORWARD-PROPAGATION TROPOSPHERIC SCATTER CIRCUITS

Abstract

During the period 7 July through 9 August 1961, digital data were transmitted over a forward-propagation tropospheric scatter circuit in Alaska. The COded BIphase (COBI) was the modem used. The circuit, routed from Ketchikan to Anchorage, was 858.7 miles in length.

Error rates were established. Occurrence of errors is related, in so far as possible, to the records of equipment failures, drop-outs, and circuit failures contained in the daily log.

Description of circuit

Digital data were transmitted from Ketchikan (KTN 8) to Smugglers Cove (SCO 1), a distance of 18.9 miles, over TD-2 facilities. At Smugglers Cove the FPTS was used to transmit the data

to Neklason Lake (KNV 1), a distance of 806 miles. From Neklason Lake to Anchorage (ANC 8), a distance of 33.8 miles, digital data were again transmitted via TD-2 facilities.

The FPTS system has a bandwidth of approximately 600 kc/s and can accommodate 72 voice-frequency channels. The carrier equipment utilizes single side-band operation and uses frequency-division multiplexing. The circuit is shown in Figure 1.

Attenuation and delay measurements, made daily, are shown in Figures 2 through 24.

Test conditions

The output of the COBI transmitter was set at 0 dbm. The receiver level was set at −12 dbm to −15 dbm. Operation was at a rate of 1200 bits per second with 16 bits per block during the test. Bit patterns, which included odd, even, and random choices, were changed from day to day.

The error data recorded by the automatic digital data error recorder (ADDER) included:

1) The 16 bits of each incorrect block;
2) Time of occurrence of error;
3) An excess count, i.e., the total number of blocks in error occurrence during intervals when the storage of the ADDER was full;
4) The loss or gain of block-start signals in the output of the COBI receiver; and
5) Additional information about the performance of external error detectors, such as parity.

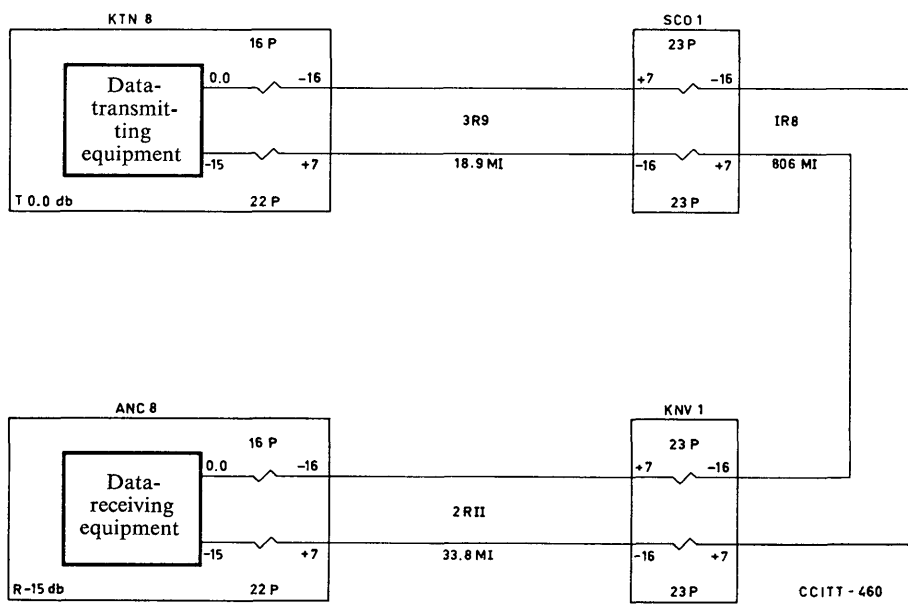
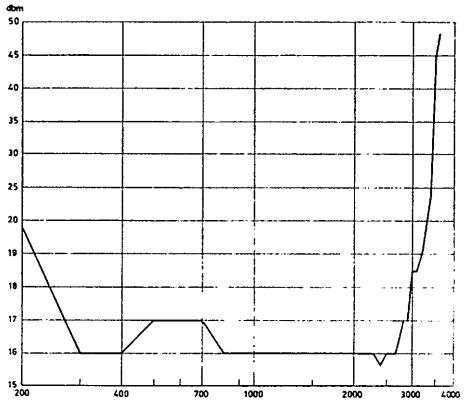
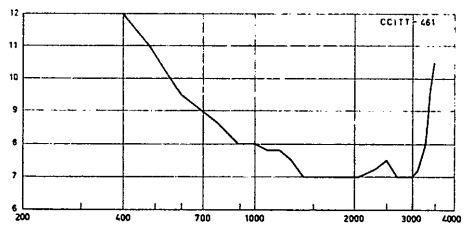


FIGURE 1. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1025, 7 July 1961



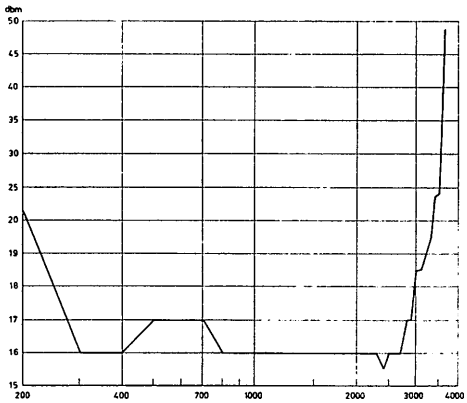
Frequency vs. delay
1235, 7 July 1961



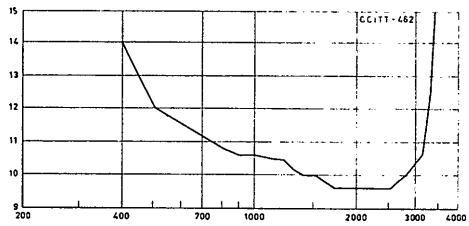
Frequency c/s

FIGURE 2. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
0955, 8 July 1961



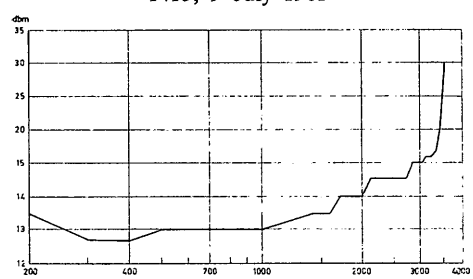
Frequency vs. delay
1010, 8 July 1961



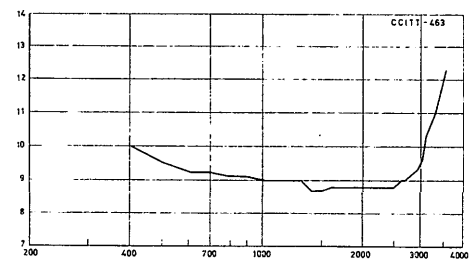
Frequency c/s

FIGURE 3. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1415, 9 July 1961



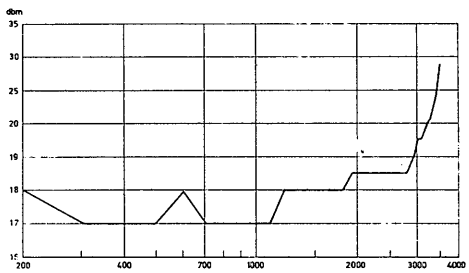
Frequency vs. delay
1545, 9 July 1961



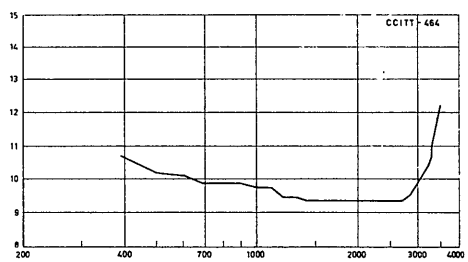
Frequency c/s

FIGURE 4. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1000, 10 July 1961



Frequency vs. delay
1117, 10 July 1961



Frequency c/s

FIGURE 5. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
0944, 11 July 1961

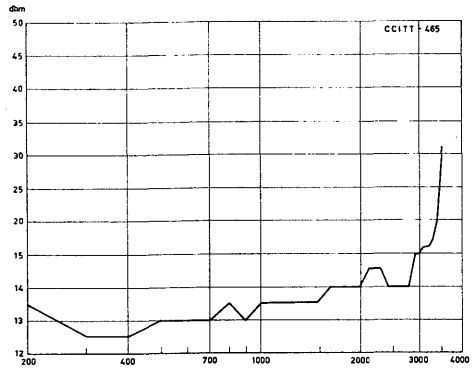
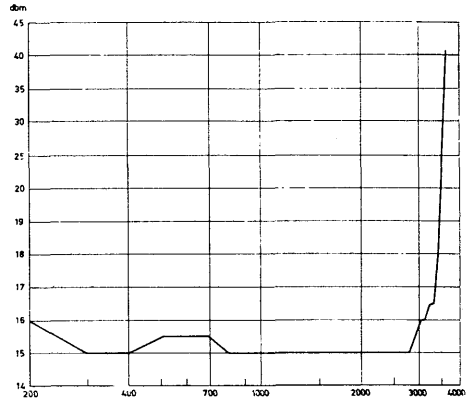


FIGURE 6. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1243, 12 July 1961



Frequency vs. delay
1255, 12 July 1961

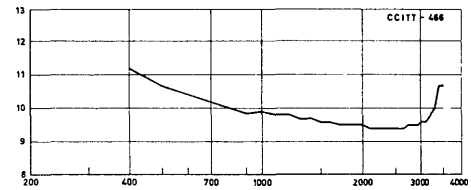
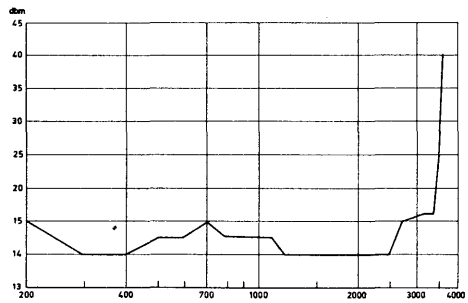


FIGURE 7. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1050, 15 July 1961



Frequency vs. delay
1135, 15 July 1961

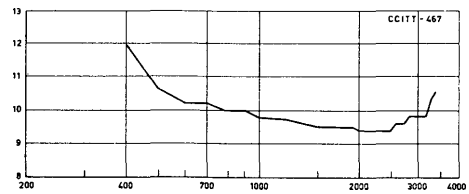


FIGURE 8. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1400, 16 July 1961

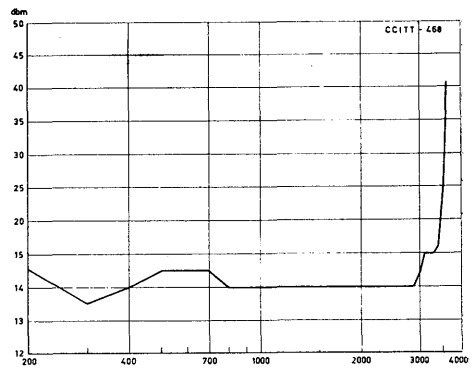
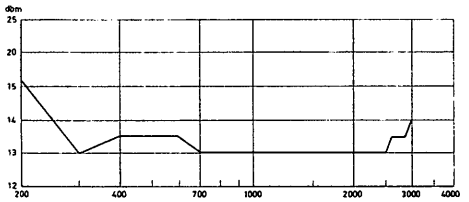
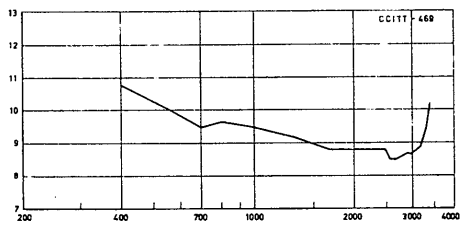


FIGURE 9. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1258, 17 July 1961



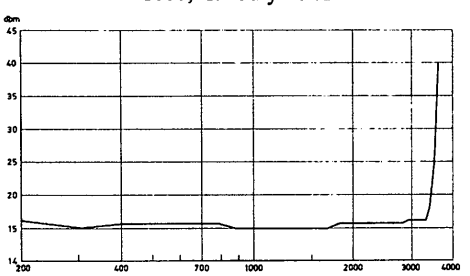
Frequency vs. delay
1328, 17 July 1961



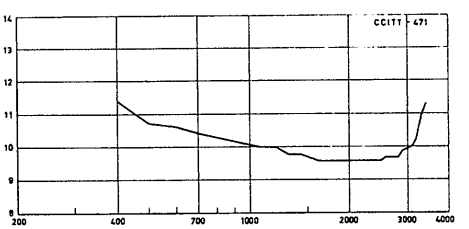
Frequency c/s

FIGURE 10. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1000, 19 July 1961



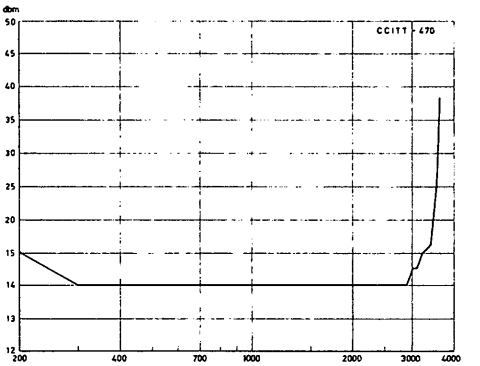
Frequency vs. delay
1016, 19 July 1961



Frequency c/s

FIGURE 12. — **TD 1845 Liaison Ketchikan–Anchorage**

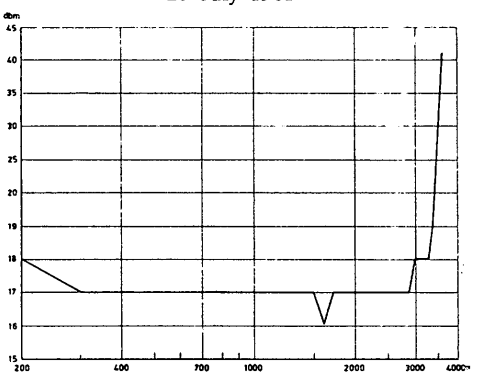
Frequency vs. dbm
0937, 18 July 1961



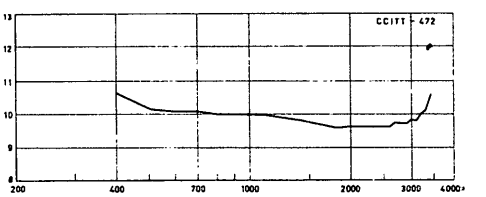
Frequency c/s

FIGURE 11. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
20 July 1961



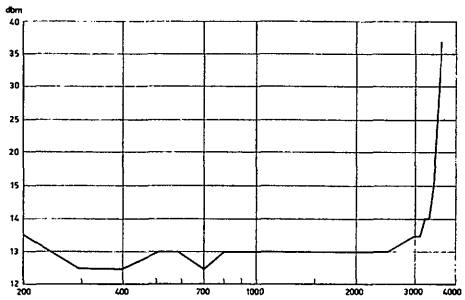
Frequency vs. delay
20 July 1961



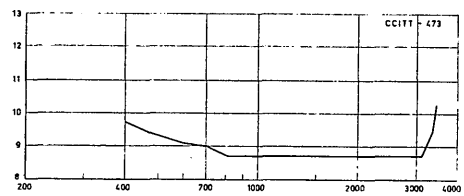
Frequency c/s

FIGURE 13. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
21 July 1961



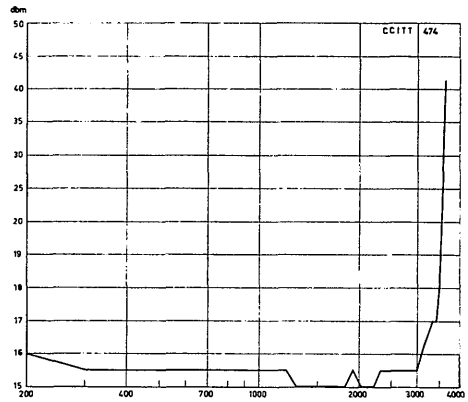
Frequency vs. delay
21 July 1961



Frequency c/s

FIGURE 14. — **TD 1845 Liaison Ketchikan–Anchorage**

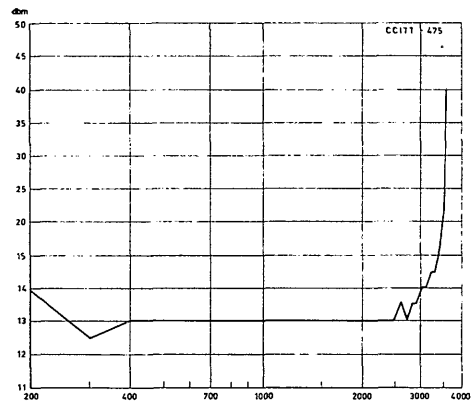
Frequency vs. dbm
22 July 1961



Frequency c/s

FIGURE 15. — **TD 1845 Liaison Ketchikan–Anchorage**

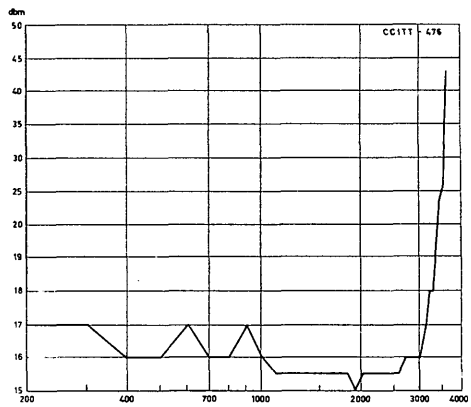
Frequency vs. dbm
1220, 23 July 1961



Frequency c/s

FIGURE 16. — **TD 1845 Liaison Ketchikan–Anchorage**

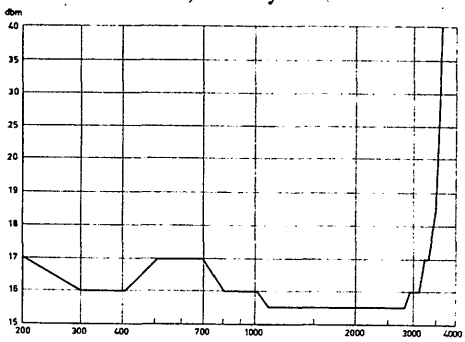
Frequency vs. dbm
0915, 26 July 1961



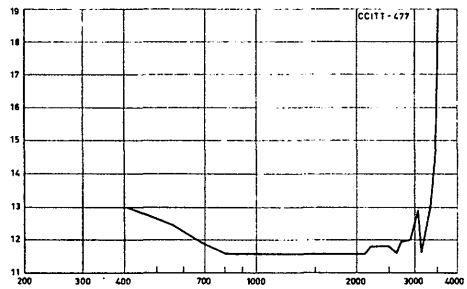
Frequency c/s

FIGURE 17. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1000, 27 July 1961



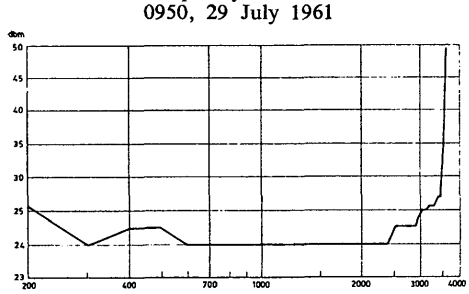
Frequency vs. delay
1020, 27 July 1961



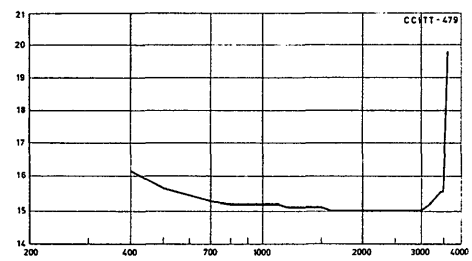
Frequency c/s

FIGURE 18. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
0950, 29 July 1961



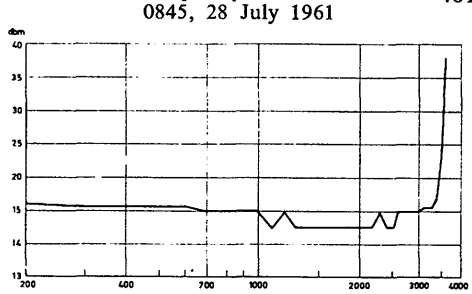
Frequency vs. delay
1000, 29 July 1961



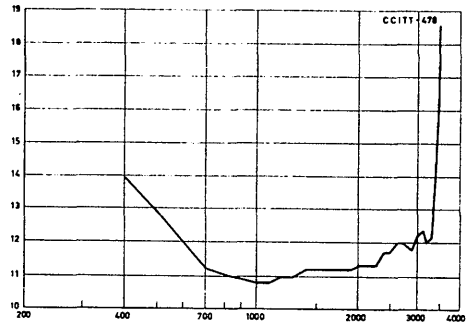
Frequency c/s

FIGURE 20. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
0845, 28 July 1961



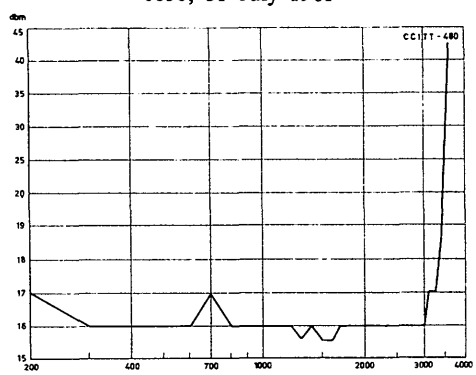
Frequency vs. delay
0848, 28 July 1961



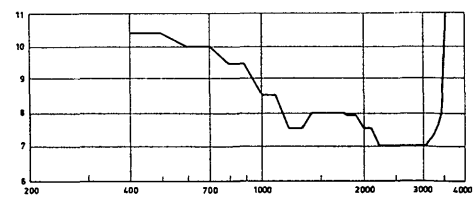
Frequency c/s

FIGURE 19. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
0830, 31 July 1961



Frequency vs. delay
0835, 31 July 1961

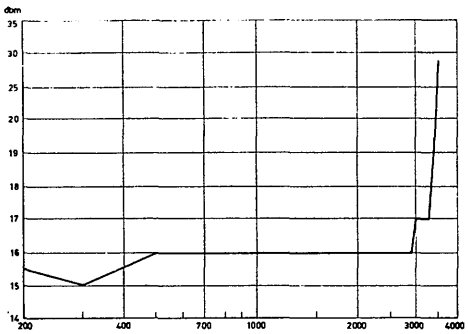


Frequency c/s

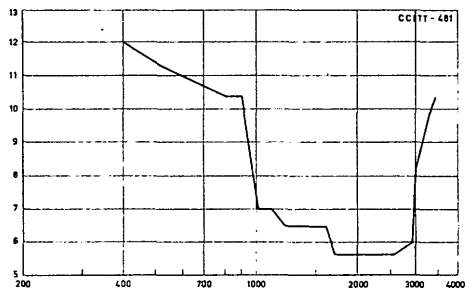
FIGURE 21. — **TD 1845 Liaison Ketchikan–Anchorage**

(Suppl. 39)

Frequency vs. dbm
0935, 2 August 1961



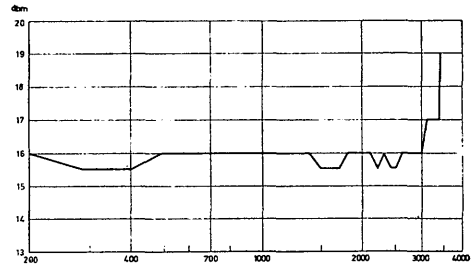
Frequency vs. delay
0940, 2 August 1961



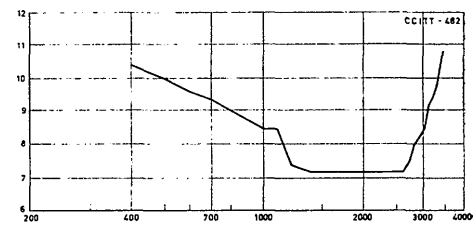
Frequency c/s

FIGURE 22. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
1154, 4 August 1961



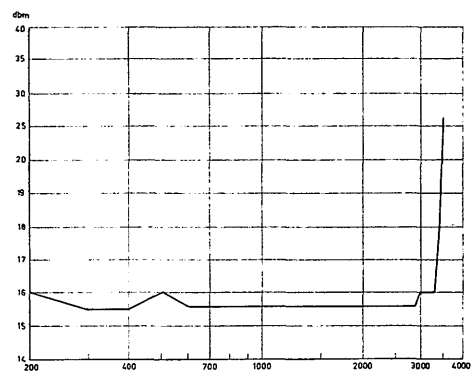
Frequency vs. delay
1158, 4 August 1961



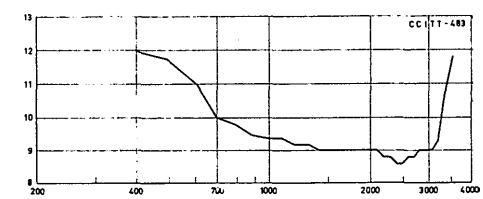
Frequency c/s

FIGURE 23. — **TD 1845 Liaison Ketchikan–Anchorage**

Frequency vs. dbm
0955, 6 August 1961



Frequency vs. delay
1004, 6 August 1961



Frequency c/s

FIGURE 24. — **TD 1845 Liaison Ketchikan–Anchorage**

Drop-out measurements

Drop-outs were measured throughout the test. Level, signal, time of drop-out occurrence, and error data obtained from the ADDER were recorded on a Brush oscillograph.

Relationships between errors recorded by the ADDER and the number of errors possible during the drop-out may be determined from Figure 26, in which the length of drop-outs, as well as their amplitude and time of occurrence, are presented.

By using the information on block errors obtained from the 7090 analysis, the following formula was derived :

$$\text{Percentage of ADDER errors due to drop-out phenomena} = \frac{\text{Three} \times \text{total time that ADDER errors were recorded on drop-out paper}}{\text{True number of block errors recorded by the ADDER}}$$

Error rates, shown in errors per minute, indicate that high daily error rates were caused primarily by drop-outs.

Analysis

Processing

Error data were accumulated over a month for a total of 706.4 hours. The resulting punched paper tape from the ADDER was processed by the IBM 7090 computer to obtain totals, averages, and time distributions of the errors.

Error rates

Figure 27 shows the error rates obtained on a daily basis. Included are :

1) The dates on which measurements were taken ;
2) The total time of operation ;
3) The block-error rate in blocks in error per minute ;
4) The bit-error rate in bits in error per 10^5 bits transmitted ;
5) The average number of bits in error in a block in error, and
6) The grand averages of totals of the foregoing quantities.

Daily and weekly distribution of errors

Error rate

Both the bit and block error rates tabulated in Figure 26 are replotted as bar graphs in Figures 28 and 29. The error patterns revealed are interesting. Peaks, which correspond to high-error days, appear to be random on all curves. The curves representing bit-error rates are similar in shape to corresponding curves for block-error rates.

Bit error per block error

Figure 30 shows the percentage (obtained from daily averages) of blocks errors as a function of bit errors per block for each circuit. The averages represented are weighted. It will be noted that a single bit error per block error occurs most frequently.

Temporal analysis

Figure 31 shows the variation of block-error rate with the day of the week. The minimum and maximum curves represent the lowest and highest values, respectively.

Date	Number of drop-outs	Total time of drop-outs in seconds	Number of times ADDER errors and drop-outs occurred simultaneously	Total time ADDER errors were recorded in seconds	Predicted number of block errors which may have occurred during drop-outs (includes excess errors)	Number of block errors actually recorded by ADDER (not including excess errors)	Percentage of ADDER errors due to drop-outs	True error rate in blocks in error per minute
1	2	3	4	5	6	7	8	9
7/7 -7/8	40	1042	16	317	951	691	100	4.3
7/8 -7/9	32	603	11	125	375	525	71	4.3
7/9 -7/10	88	551	16	82	246	283	79	2.8
7/10-7/11	73	254	47	51	153	180	85	0.37
7/11-7/12	53	2356	15	320	960	759	100	7.9
7/12-7/12	5	65.1	0	0	0	0	0	0
7/12-7/14	35	1153	8	74	222	1721	12.8	4.9
7/14-7/15	41	1474	0	0	0	23	0	0.14
7/15-7/16	40	1628	2	20	60	455	13.1	0.73
7/16-7/17	13	546	1	5.0	15	128	11.7	0.39
7/17-7/18	12	289	1	9	27	132	20.45	0.5
7/18-7/19	11	1214	4	887	2661	1656	100	11.9
7/19-7/20	20	1050	3	19	57	158	29	0.3
7/20-7/21	21	1314	2	33	99	386	25.6	1.5
7/21-7/22	43	1126	1	38	114	154	74	0.25
7/22-7/23	17	425	1	321	963	1050	91.7	10.3
7/23-7/24	27	1599	0	0	0	62	0	0.065
7/24-7/26	43	2648	7	269	807	1801	44.8	7.1
7/28-7/29	264	595	116	208	624	281	100	0.68
7/29-7/30	12	466	1	380	1140	528	100	1.05
7/30-7/31	55	1183	19	68	204	105	100	0.098
7/31-8/1	47	290	22	205	615	458	100	1.58
8/1 -8/2	73	303	20	41	123	1574	0.078	15.3
8/2 -8/3	44	500	13	51	153	69	100	0.064
8/3 -8/4	63	343	16	55	165	64	100	0.091
8/4 -8/5	90	1329	20	211	633	330	100	1.65
8/5 -8/6	27	351	6	220	660	0	0	0.351
8/6 -8/7	18	428	5	31	95	91	100	3.308
8/7 -8/9	62	293	32	56	168	158	100	0.112

FIGURE 26. — **Drop-out measurements**

Date	Time hours	Block errors	Lost block start	Gained block start	Average bits in error per block in error	Average bits error rate in parts in 10^5
1	2	3	4	5	6	7
7/7 -8	17.6	4.325	0.435	0.034	4.956	29.770
7/8 -9	21.9	4.324	0.163	0.109	6.421	38.724
7/9 -10	17.6	2.840	0.055	0.174	6.337	24.991
7/10-11	18.8	0.371	0.027	0.035	3.200	1.650
7/11-12	23.4	7.952	0.129	0.192	5.787	63.913
7/12-14	55.2	4.914	0.252	0.078	6.276	42.832
7/14-15	4.3	0.146	0.027	0.023	2.478	0.503
7/15-16	24.2	0.728	0.022	0.017	2.028	2.050
7/16-17	22.9	0.3866	0.026	0.013	3.543	1.900
7/17-18	18.8	0.501	0.043	0.019	3.440	2.395
7/18-19	14.3	11.897	0.270	0.462	2.537	41.916
7/19-20	21.5	0.298	0.033	0.019	2.435	1.009
7/20-21	18.7	1.528	0.071	0.046	2.646	5.613
7/21-22	23.0	0.249	0.022	0.020	3.192	1.105
7/22-23	23.1	10.348	0.061	0.092	2.327	33.439
7/23-24	19.0	0.065	0.012	0.007	1.961	0.177
7/24-26	47.2	7.061	0.189	0.099	5.761	56.501
7/26-27	23.5	0.245	0.013	0.009	2.397	0.816
7/27-28	21.6	0.022	0.003	0.003	1.959	0.059
7/28-29	25.5	0.682	0.065	0.033	3.618	3.427
7/29-30	22.1	1.057	0.057	0.038	1.643	2.413
7/30-31	23.5	0.098	0.010	0.007	2.019	0.275
7/31-8/1	24.2	1.584	0.248	0.008	6.454	14.199
8/1 -2	23.9	15.367	0.423	0.235	7.410	158.162
8/2 -3	22.8	0.064	0.010	0.009	2.169	0.192
8/3 -4	23.3	0.091	0.011	0.014	3.500	0.442
8/4 -5	20.5	1.653	0.099	0.120	6.015	13.812
8/5 -6	24.9	0.351	0.098	0.092	6.312	3.075
8/6 -7	20.7	3.308	0.010	0.002	2.723	1.163
8/7 -8	19.4	0.189	0.028	0.016	3.101	0.816
8/8 -9	21.3	0.036	0.009	0.004	3.424	0.171
Grand averages		2.855	0.105	0.064	5.263	20.870
Total time 706.4 hours						

FIGURE 27. — **Error rates**

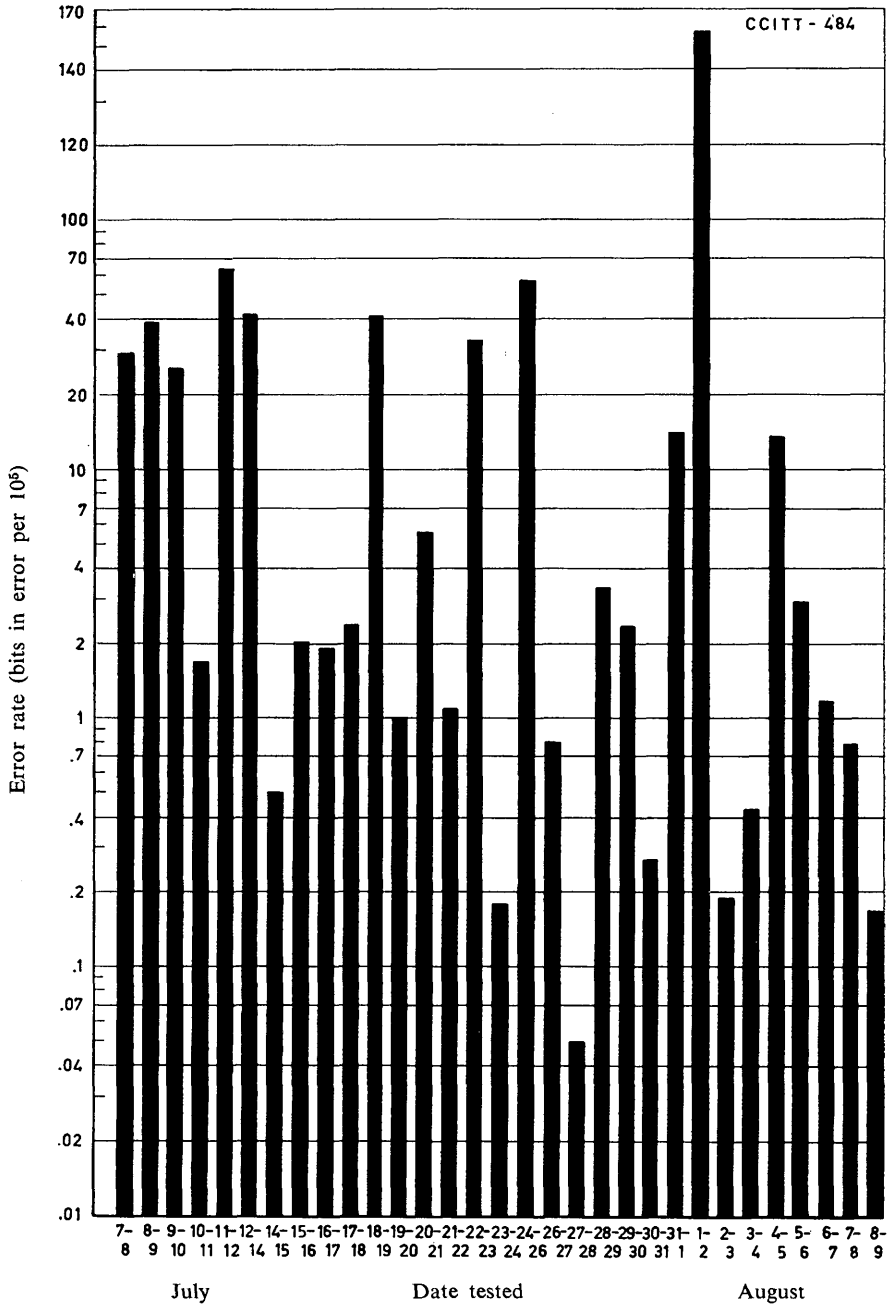


FIGURE 28. — **Error rate (bits in error per 10⁵) on a run basis TD 1845**

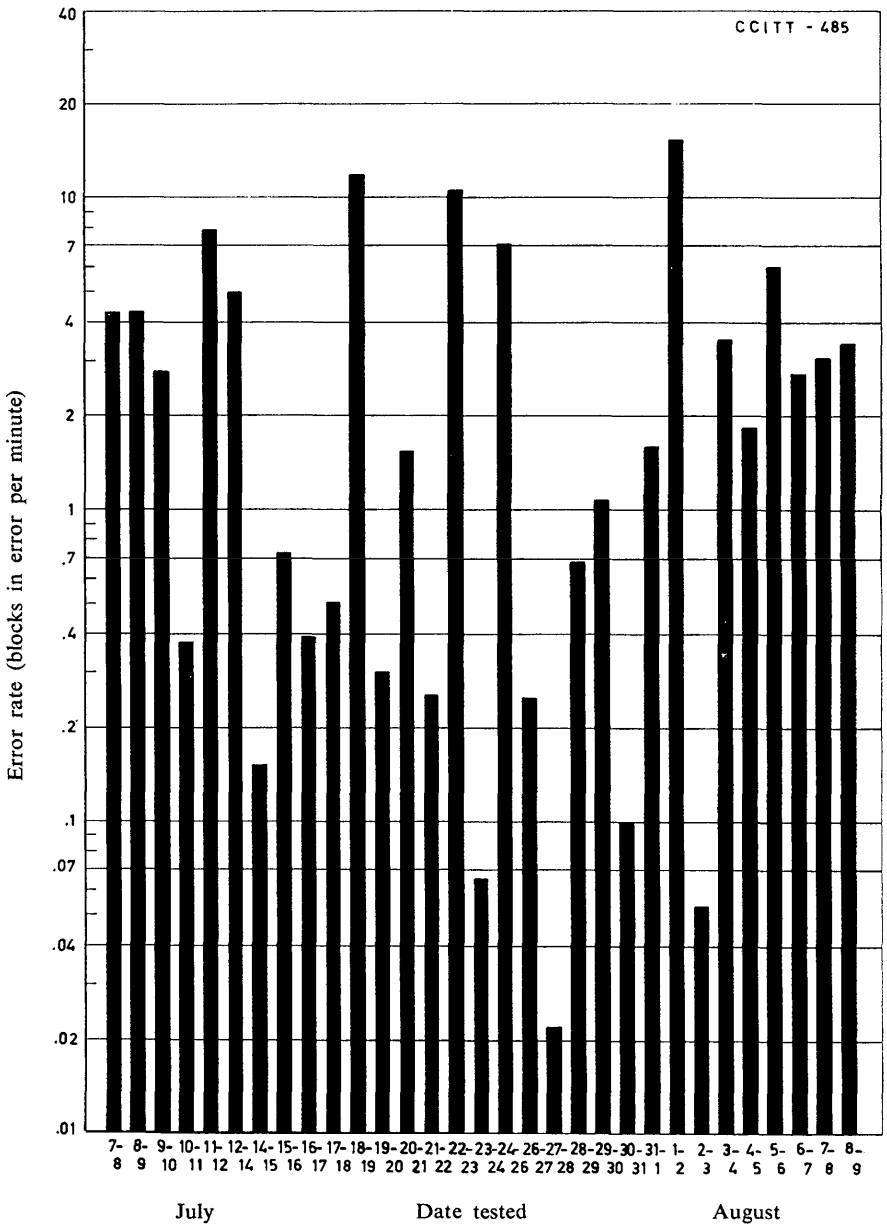


FIGURE 29. — **Error rate (blocks in error per minute) on a run basis TD 1845**

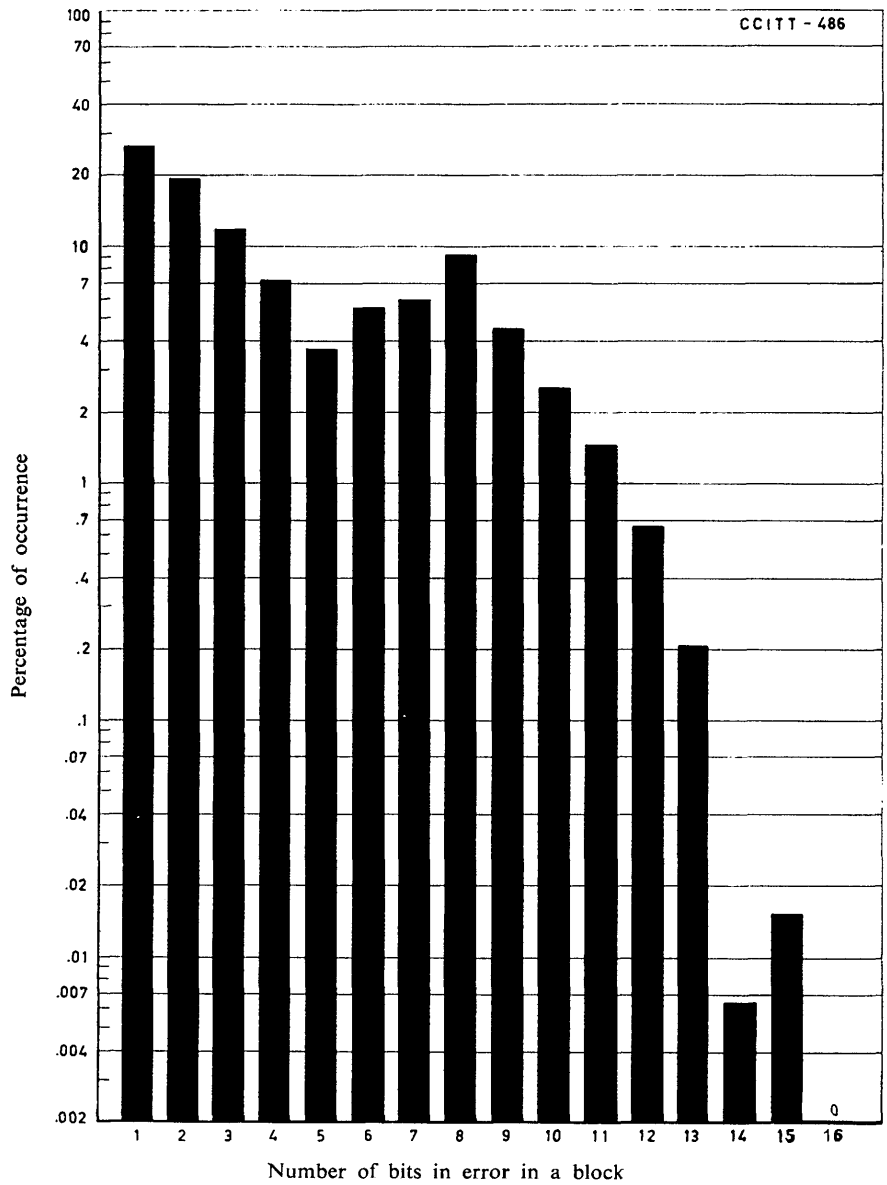


FIGURE 30. — **Distribution of block errors by the number of bit errors per block (non-cumulative) TD 1845**

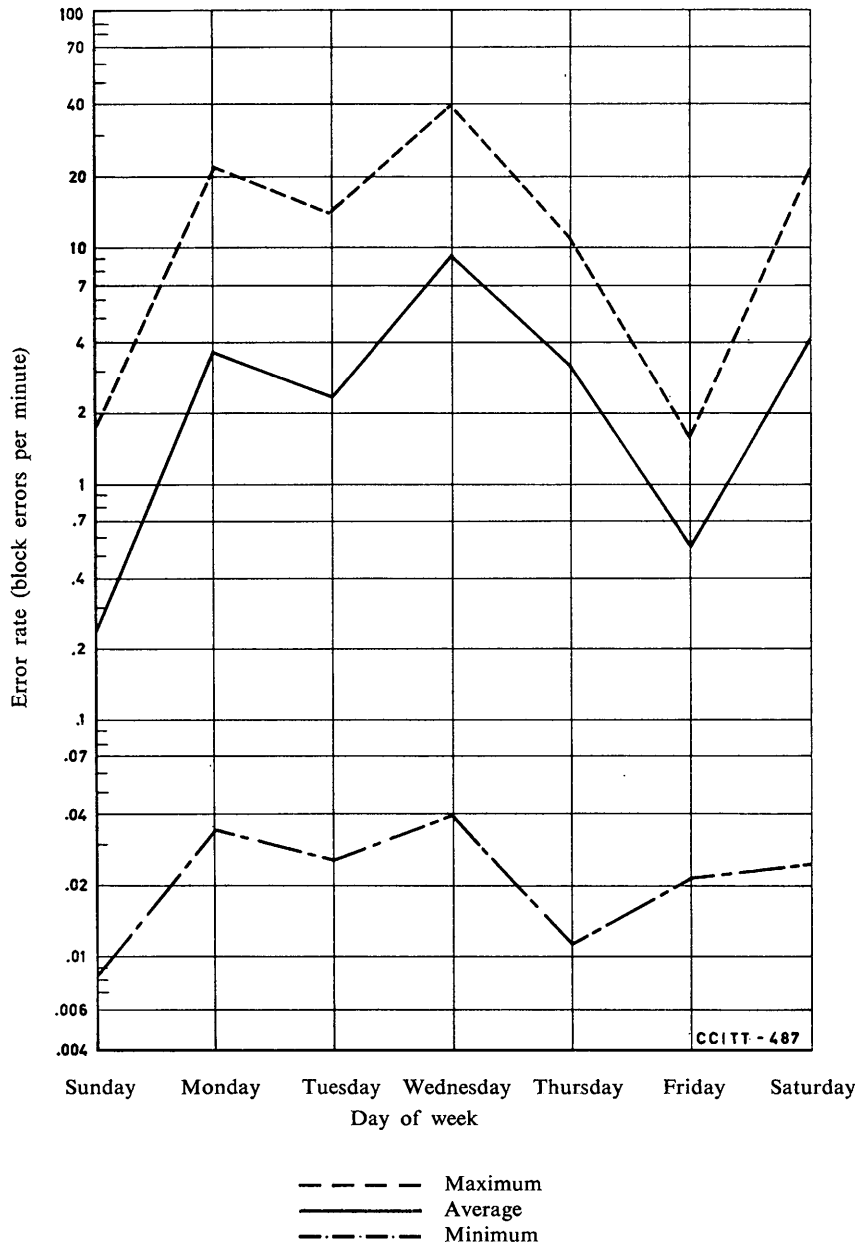


FIGURE 31. — **Variation of block error rate with day of week TD 1845**

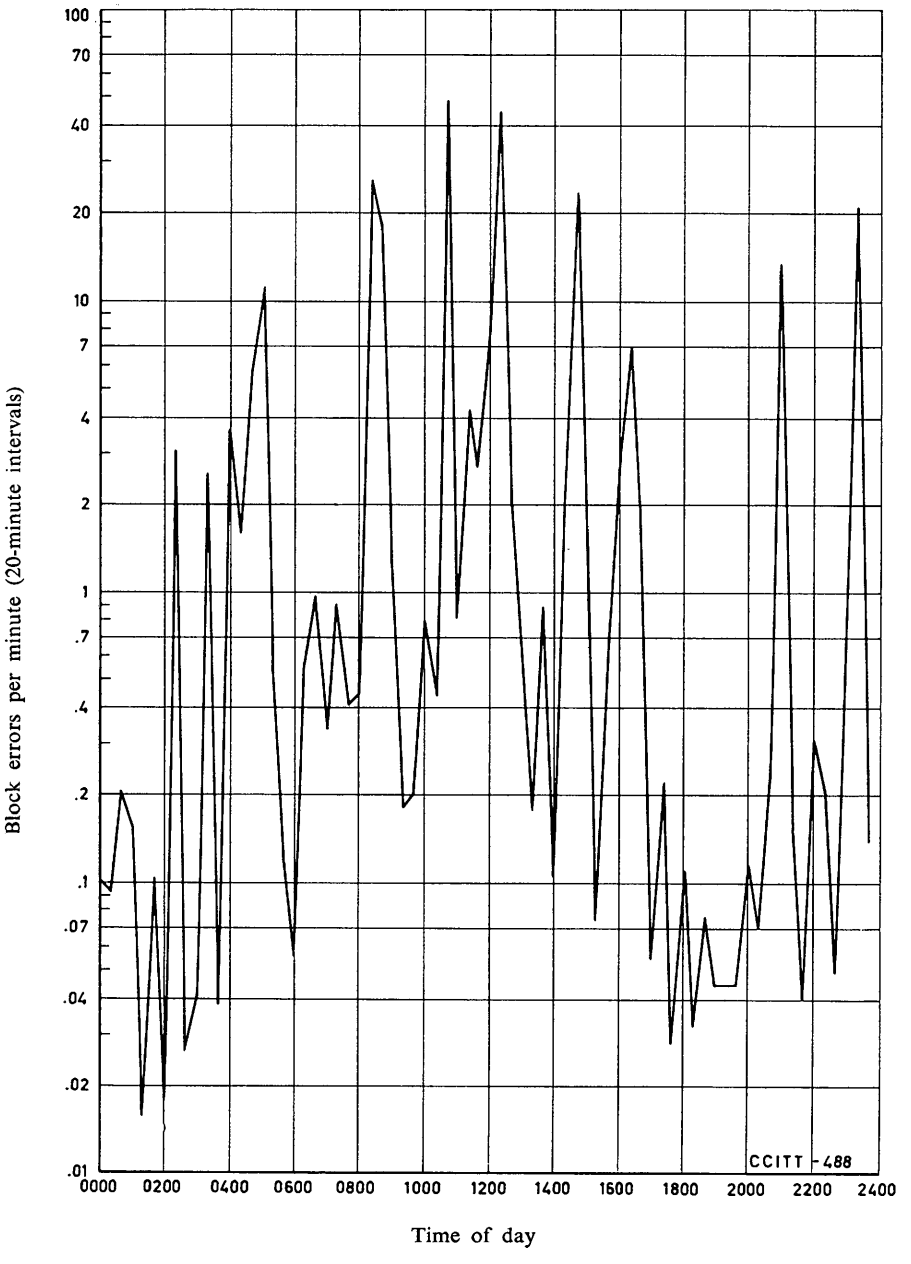


FIGURE 32. — **Error rate (block errors per minute) versus time of day TD 1845**

Minute-by-minute distribution

In order to describe the characteristics of a burst, the data were analysed in intervals of one minute. Figure 33 shows the percentage of run time as a function of the block-error rate. Figure 34 shows the probability of occurrence of various error-free periods between error bursts. The average number of bit errors per block error is shown as a function of error rate in Figures 36 and 37.

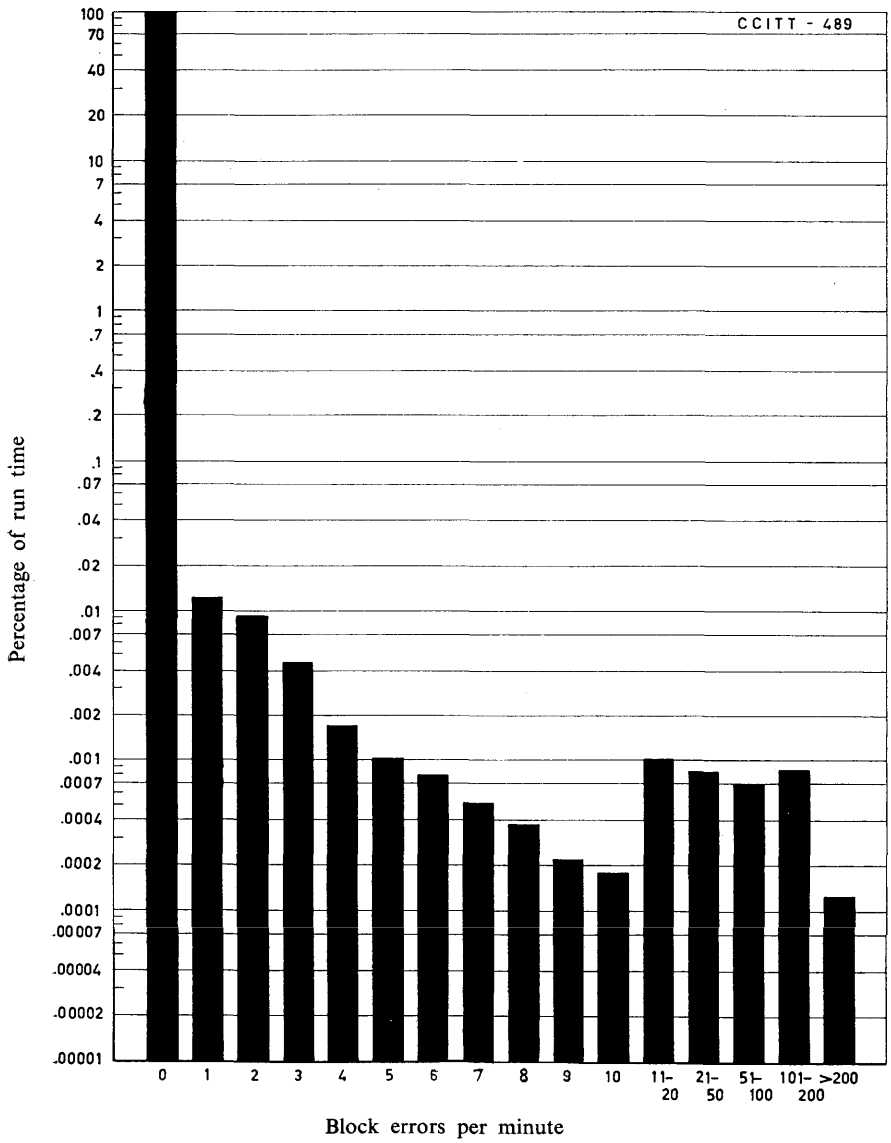


FIGURE 33. — **Percentage of run times versus block error rate TD 1845**

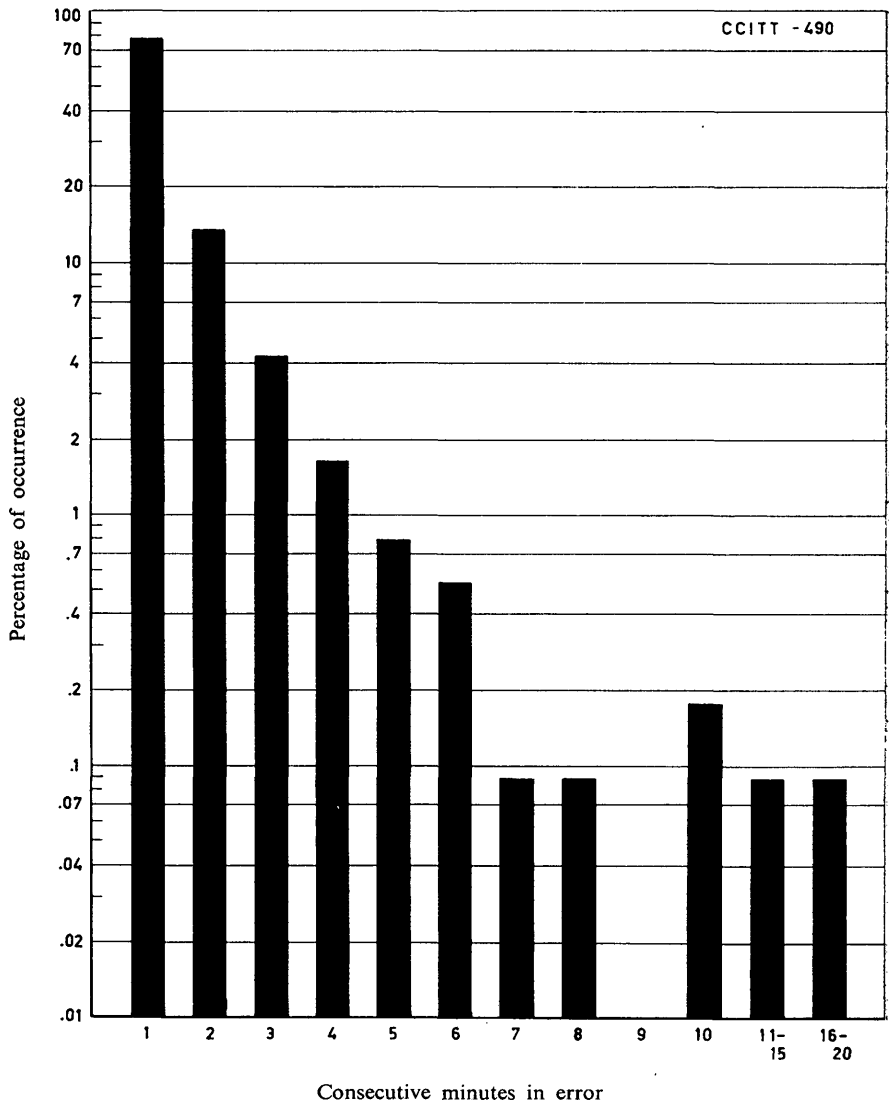


FIGURE 34. — **Occurrence of consecutive minutes in error (non-cumulative) TD 1845**

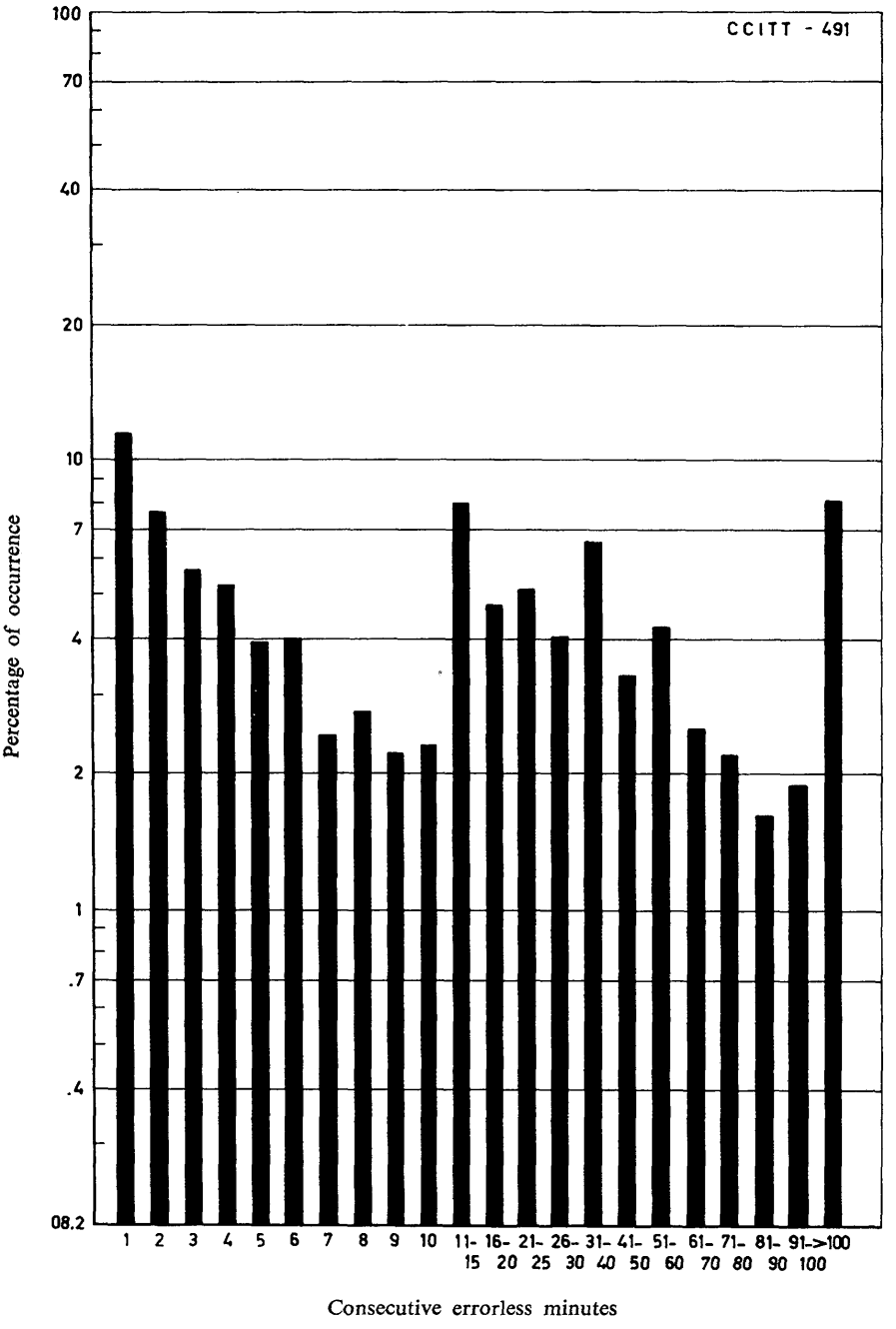


FIGURE 35. — **Percentage of occurrence of consecutive errorless minutes during transmission (non-cumulative) TD 1845**

Consecutive block-error analysis

The distribution of consecutive blocks in error is shown in Figure 36. The occurrence of such a large percentage of the block errors in a burst greater than 26 suggests that the high-error-rate periods are due to drop-outs or loading conditions.

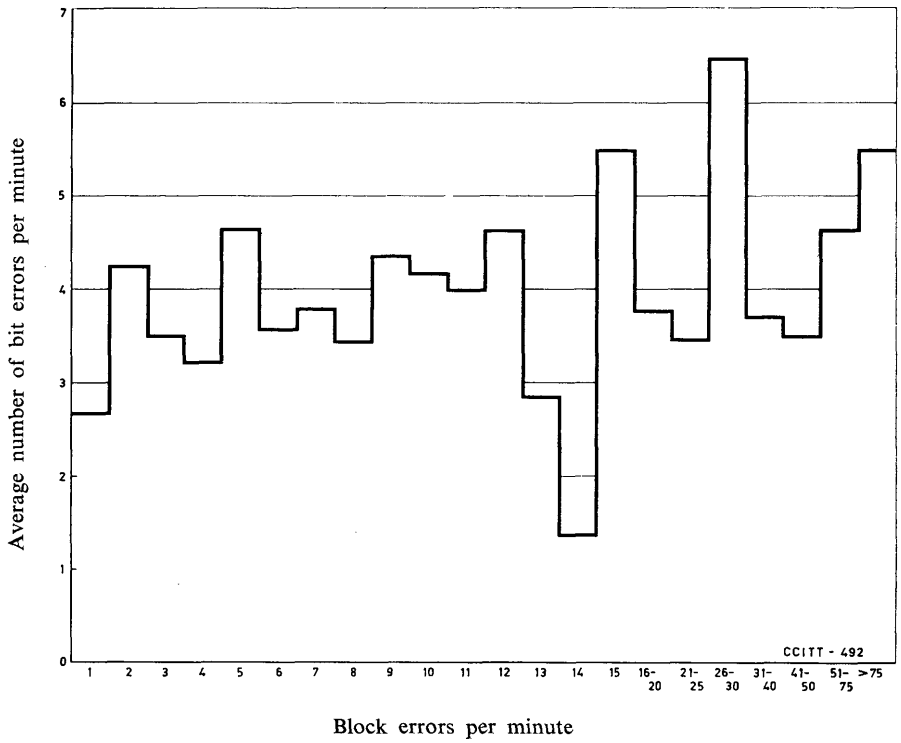


FIGURE 36. — **Average number of bit errors per block error rate TD 1845**

Bit-error distribution

The percentage of occurrence of consecutive bits in error within a block in error is plotted in Figure 37.

Figure 38 shows the probabilities of a bit lost or a bit gained as a function of bit position. A gained bit is defined as a conversion from a space ("zero") to a mark ("one"). A lost bit is the converse. The probability of a bit being lost or gained depends on the position of the bit block being transmitted. These probabilities for lost or gained bits are reflected in the graphs.

Block-start errors

Figure 39 shows the variations of block-start error day by day. A lost block start is defined as a missing block start. A gained block start is a spurious or additional block-start signal.

Parity analysis

An analysis of the blocks in error was made in order to determine the effectiveness of a simple parity check (Figure 40). For this test, a simple parity check would have been 60 per cent effective in detecting block errors.

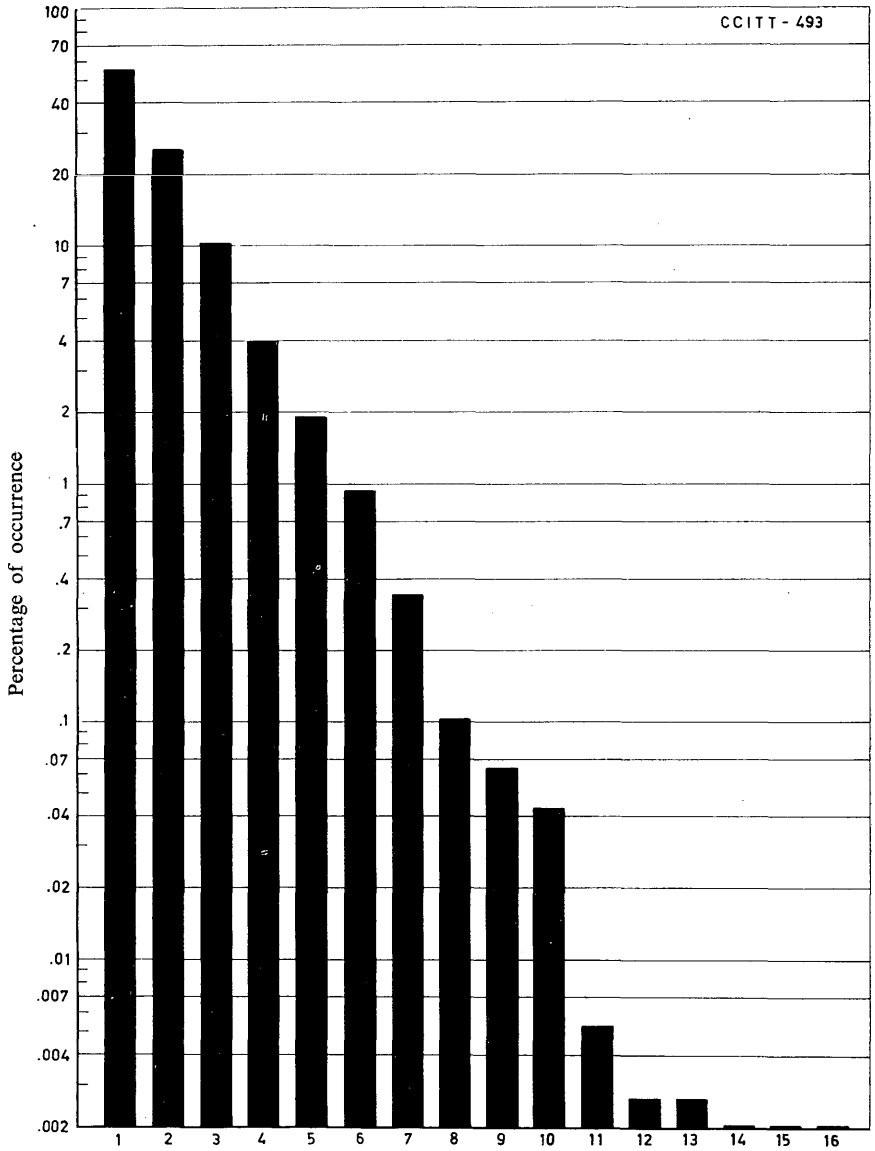
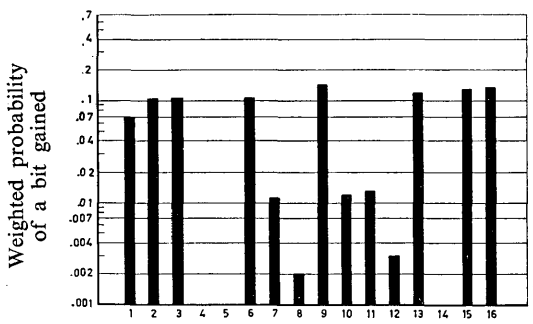
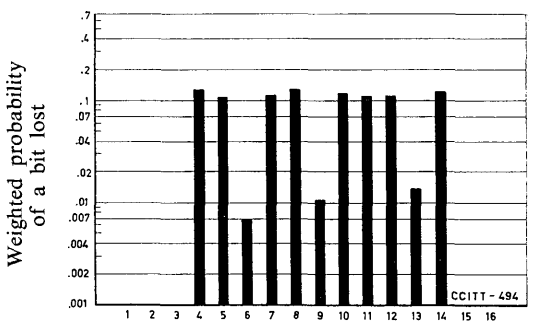


FIGURE 37. — **Consecutive bit error analysis TD 1845**



TD 1845 weighted probability of a bit gained as a function of bit position



Weighted probability of a bit lost as a function of bit position

FIGURE 38

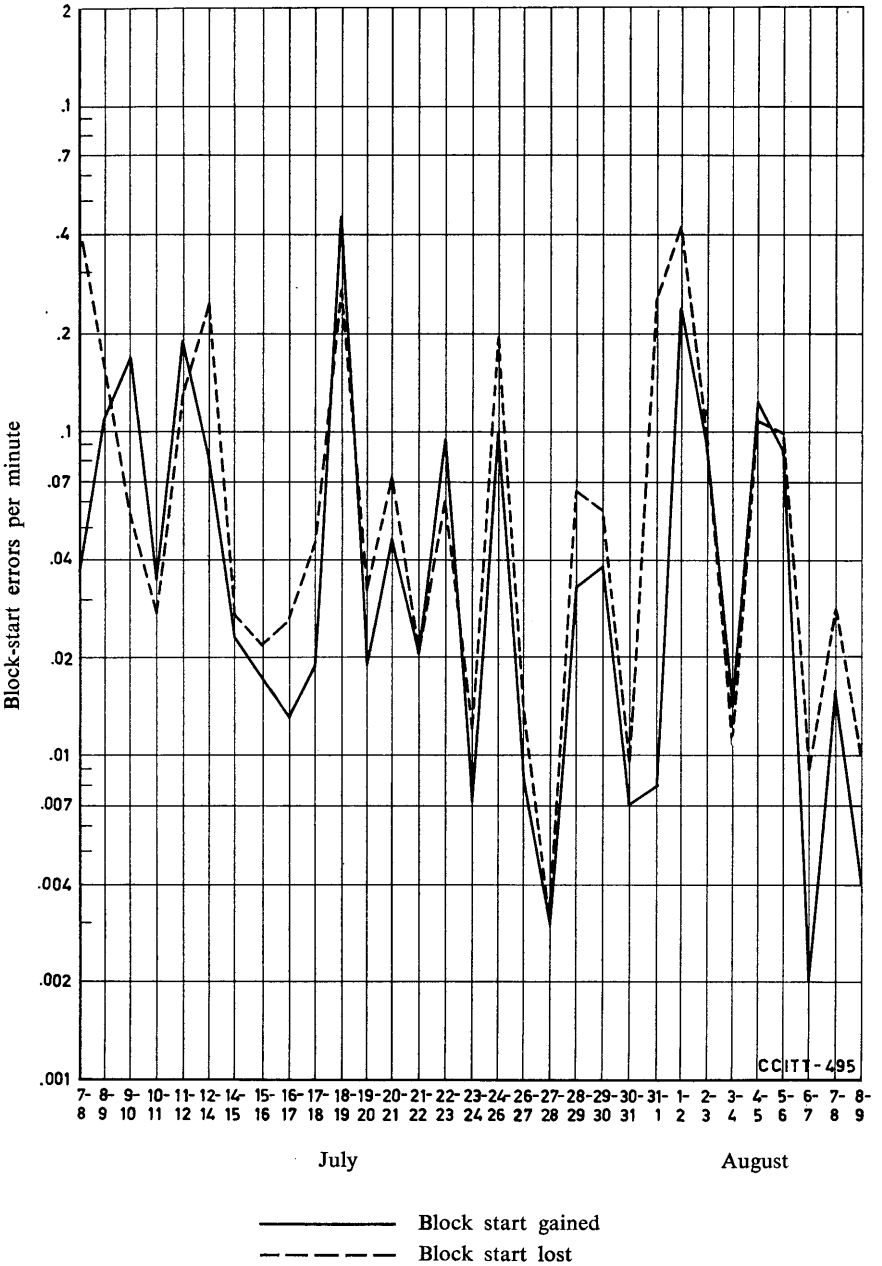


FIGURE 39. — **Variation of block-start error rate by day TD 1845**

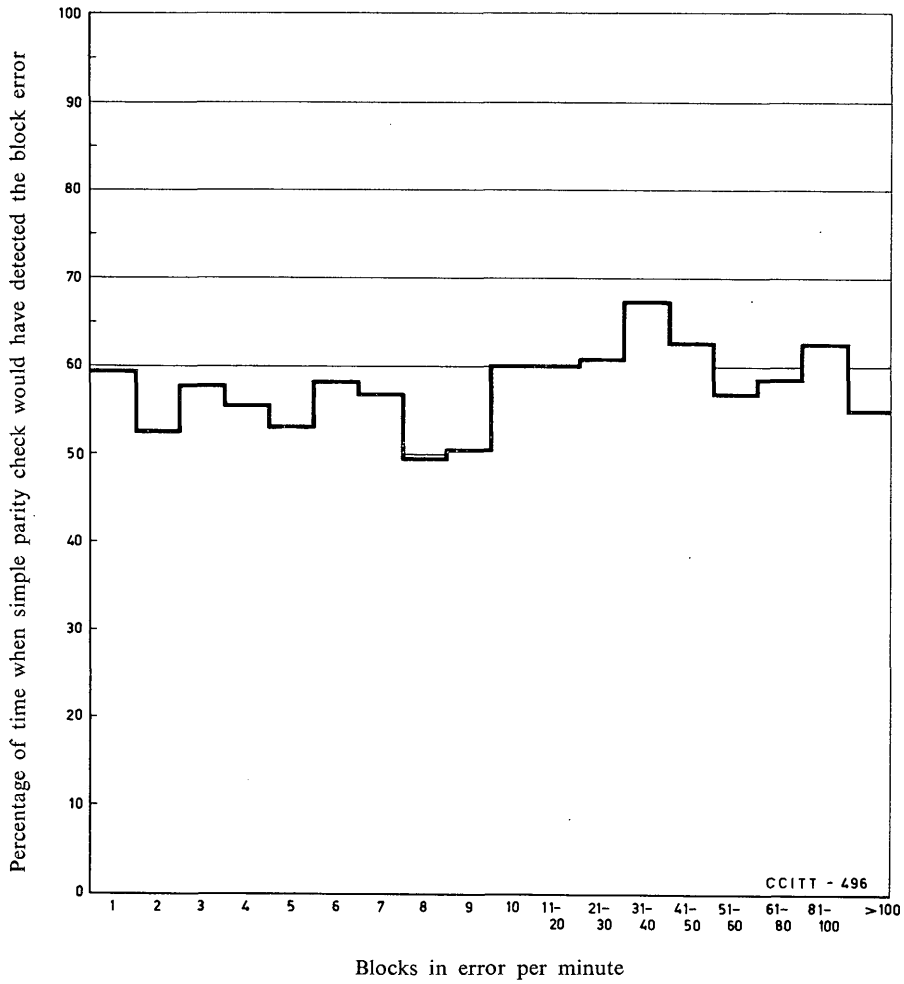


FIGURE 40. — **Parity check as a function of the block-error rate TD 1845**

REFERENCES

HOFMANN: E. J. "Automatic Digital Data Error Recorder", presented at the I.R.E. Instrumentation Conference, 10 November 1959. Published in *I.R.E. Trans. on Instrumentation*, Vol. 1-10, No. 1, pp. 27-31, June 1961.

HOFMANN, E. J. : "Specifications for COBI Data Transmission Modem", 25G-0035 (U), Lincoln Laboratory, M.I.T. (18 July 1961), ASTIA 261546, H-327.

Supplement No. 40

CHILE TELEPHONE COMPANY — *(Extract from contribution COM Sp. A/No. 90 — September 1963)*

ERROR MEASUREMENTS AT 250 BAUDS

Telephone type circuits (Statistics relevant to information feedback data transmission systems)

1. The results of error measurements over switched telephone connections appear as a supplement in Volume VII of the *Red Book*, pp. 413-420. In that contribution the analysis of the measured data assumed a division of the continuous data stream into blocks of 50, 100, 200 and 500 bits so as to facilitate their application to the design of error-control systems using error detection and block retransmission.

2. When a full-duplex channel is available for data transmission, the proposed 200 bauds system—for example, an error control system based on information feedback is feasible. In such a system any disparity between the data transmitted by the sending station and the data received back by the sending station would automatically initiate a repetition cycle. It is necessary, therefore, to consider the detailed distribution of errors within some defined number of elements following the first error of an error burst, this number being sufficient to span a complete repetition cycle, rather than on a block basis. If a single error burst exceeds this number of elements, then—as regards information feedback systems—it is appropriate to consider this burst as consisting of two (or more) separate bursts.

3. The raw error data measured at 250 bauds modulation rate which formed the basis of the supplement in Volume VII of the *Red Book*, pp. 413-420, have therefore been analysed again in a form suitable as a basis of design for information feedback systems. All errors recorded in the tests A1 to FB2 have been included in this analysis, excluding only those error bursts which began too late in the measurement cycle of 1375 bits to be completely observed. The total number of element errors available for analysis was 4601, derived from about 6×10^7 transmitted elements, after allowing for the exclusion of certain recorded errors. As certain of the tests used depressed signal levels to induce errors, this average element-error rate is worse than might normally be expected. The analysis was performed for three separate maximum lengths of error bursts, i.e. 82 bits (corresponding to a particular repetition procedure), 100 bits and 200 bits. For each maximum burst length the following information was derived :

i) distribution of errors according to positions in error bursts ;
ii) distribution of burst lengths within the prescribed maximum ;
iii) distribution of error bursts according to the total number of errors within the burst ;
iv) probability for each bit within a burst being in error.

A large proportion of error bursts consist of only a single bit, or two consecutive bits, in error. It was considered useful to analyse longer bursts separately as well as in an over-all analysis which included these very short bursts.

The detailed analysis for maximum burst lengths of 82, 100 and 200 bits are given in Appendices 1, 2 and 3 respectively of this Supplement.

4. The preceding statistics relate to one-way transmission over a telephone connection. The burst characteristics would also be applicable to transmission in both directions of an information feedback system in which the Go and Return channels are entirely independent. However, when these channels are derived by frequency division of a single speech connection many of the disturbances are likely to affect both channels.

Appendix 4 gives the analysis of errors in a stream of data which has been transmitted in both directions over a speech connection, this analysis having been derived from the statistics for one-way transmission given in Appendix 1. Certain assumptions had to be made as regards the manner in which the errors occurring during transmission in the two directions would combine and these are given below :

i) of the disturbances causing single error bursts in one direction of transmission, 10% would simultaneously cause error bursts in the other direction ;

ii) of the disturbances causing double (consecutive) error bursts in one direction, 90% would simultaneously cause error bursts in the other direction ;

iii) of the disturbances causing longer bursts in one direction, 95% would simultaneously cause error bursts in the other direction.

5. *Undetected-error rate*

In an information feedback system there is a risk that an error in the forward direction is again changed during transmission over the return channel, the error remaining undetected. This undetected-error rate can be estimated on the basis of the statistics in the Appendices and the assumptions as regards coincident errors in the Go and Return channels.

It is possible to forecast that not more than one error burst in 1000 will result in an undetected error. Taking account of the burst incidence this corresponds to about one undetected character in 10^8 transmitted.

Undetected errors will almost invariably be due to single element changes only. Where there are multiple element changes in a character it is very unlikely that transmission over the return channel will cause all the "Go" channel element errors to be "corrected" and cause no residual errors which would be detected. The number of single element undetected errors could be reduced, and the over-all undetected-error rate improved perhaps by an order, if a parity element could be reversed by the receiving station before retransmission in each character where the parity check at the receiving station fails. This would help ensure that disturbed characters are detected by the sending station.

The derivation of the estimated undetected-error rate given above involved the determination of the probability that an error burst, the second element of which is known to be unaffected, would cause only one error in an 11-element character. As the table of probabilities may be of more general interest, it is here reproduced.

TABLE 1

First error of burst falls on character element number	Chance that last element in character is error	Chance that last element in character is correct	Chance of single error in character	Chance of more than one error in character
(1)	(2)	(3)	(4)	(5)
11	1	0	1	0
10	0	1	1	0
9	0.23	0.77	0.77	0.23
8	0.07	0.93	0.72	0.28
7	0.04	0.96	0.69	0.31
6	0.03	0.97	0.67	0.33
5	0.025	0.975	0.65	0.35
4	0.02	0.98	0.64	0.36
3	0.02	0.985	0.62	0.38
2	0.015	0.985	0.61	0.39
1	0.015	0.985	0.6	0.4
			Total 7.97	3.03
			% 0.72	0.28

This table is based on the burst characteristics given in Appendix 1.

6. *Cancel signal imitation and mutilation*

On the detection of an error it will be necessary for the sending station to initiate a cancellation and repetition at a time when transmission is disturbed. The failures to which the repetition cycle procedure should be made as immune as possible are :

i) cancel signal imitation, causing reception of a cancel signal when none was transmitted or a genuine cancel signal too early ;

ii) cancel signal mutilation, causing failure to recognize the cancel signal.

That one of these failures has taken place will normally be apparent at the sending station.

APPENDIX 1

Analysis of error bursts (250 bauds one-way transmission over switched telephone connections)
Maximum burst length = 82 elements

Analysis	Ordinal bit numbers in burst or numbers of bits in error as appropriate								Totals, etc.
	1	2	3/5	6/10	11/15	16/25	26/50	51/82	
A) *All recorded bursts*									
1) Error distribution .	1387 (30.1 %)	442 (9.62 %)	309 (6.72 %)	231 (5.02 %)	137 (3.00 %)	313 (6.8 %)	843 (18.32 %)	939 (20.42 %)	4601
2) Burst lengths . . .	671 (48.4 %)	249 (17.95 %)	(..... 467 .....)						1387
3) Probability of error for individual bits .	1	0.318	0.0741	0.0332	0.0196	0.0225	0.0241	0.0226	Average errors per burst = 3.34
B) *Excluding all single and double (consecutive bit) error bursts*									
1) Error distribution .	274	0	167	118	84	179	498	512	1832
(i) 2nd bit correct .	193	193	142	113	53	134	345	427	1600
(ii) 2nd bit wrong .	467	193	309	231	137	313	843	939	3452
(iii) total	(13.69 %)	(5.61 %)	(8.96 %)	(6.71 %)	(4.00 %)	(9.09 %)	(24.56 %)	(27.38 %)	
2) Probability of error for individual bits .	1	0.413	0.22	0.099	0.0586	0.067	0.0722	0.0628	Average errors per burst = 7.36

APPENDIX 2

Analysis of error bursts (250 bauds one-way transmission over switched telephone connections)
Maximum burst length = 100 elements

Analysis	Ordinal bit numbers in burst or numbers of bits in errors as appropriate									Totals, etc.
	1	2	3/5	6/10	11/15	16/25	26/50	51/100	101/200	
A) *All recorded bursts*										
1) Error distribution . . .	1348 (29.3 %)	432 (9.4 %)	284 (6.17 %)	211 (4.6 %)	122 (2.65 %)	286 (6.22 %)	762 (16.54 %)	1156 (25.12 %)		4601
2) Burst lengths	640 (47.5 %)	248 (18.4 %)	71 (5.27 %)	36 (2.66 %)	21 (1.55 %)	36 (2.66 %)	88 (6.52 %)	208 (15.44 %)		1348
3) Errors per burst . . .	640 (47,5 %)	353 (26.2 %)	186 (13.8 %)	52 (3.84 %)	41 (3.01 %)	61 (4.54 %)	15 (1.11 %)	0 0		1348
4) Probability of error for individual bits . . .	1	0.32	0.070	0.313	0.0181	0.0212	0.0223	0.0171		Average errors per burst = 3.41
B) *Excluding all single and double (consecutive bit) error bursts*										
1) Error distribution										
(i) 2nd bit correct . .	276	0	149	103	71	157	434	675		1865
(ii) 2nd bit wrong . .	184	184	135	108	51	129	328	481		1600
(iii) total	460 (13.25 %)	184 (5.37 %)	284 (8.26 %)	211 (6.16 %)	122 (3.55 %)	286 (8.36 %)	762 (21.86 %)	1156 (33.22 %)		3465
2) Burst lengths			71 (15.44 %)	36 (7.83 %)	21 (4.56 %)	36 (7.83 %)	88 (19.14 %)	208 (45.2 %)		460
3) Errors per burst. . . .		105 (22.82 %)	186 (40.45 %)	52 (11.3 %)	41 (8.92 %)	61 (13.25 %)	15 (3.26 %)	0 0		460
4) Probability of error for individual bits . . .	1	0.4	0.206	0.0919	0.053	0.0621	0.0663	0.053		Average errors per burst = 7.53

APPENDIX 3

Analysis of error bursts (250 bauds one-way transmission over switched telephone connections)
Maximum burst length = 200 elements

Analysis	Ordinal bit numbers in burst or numbers of bits in error as appropriate									Totals, etc.
	1	2	3/5	6/10	11/15	16/25	26/50	51/100	101/200	
A) *All recorded bursts*										
1) Error distribution . . .	1192 (25.9 %)	379 (8.24 %)	189 (4.11 %)	132 (2.87 %)	62 (1.35 %)	182 (3.95 %)	443 (9.62 %)	820 (17.82 %)	1202 (26.14 %)	4601
2) Burst lengths	549 (46.1 %)	225 (18.85 %)	59 (4.95 %)	28 (2.35 %)	17 (1.425 %)	17 (1.425 %)	52 (4.36 %)	68 (5.7 %)	177 (14.84 %)	1192
3) Probability of error for individual bits . . .	1	0.318	0.0529	0.0222	0.0104	0.01525	0.0148	0.0137	0.0101	Average errors per burst = 3.86
B) *Excluding all single and double (consecutive bit) error bursts*										
1) Error distribution . . .	418 (11.6 %)	154 (4.27 %)	189 (5.25 %)	132 (3.66 %)	62 (1.72 %)	182 (5.05 %)	443 (12.3 %)	820 (22.76 %)	1202 (33.39 %)	3602
2) Burst lengths			59 (14.12 %)	28 (6.7 %)	17 (4.07 %)	17 (4.07 %)	52 (12.45 %)	68 (16.27 %)	177 (42.32 %)	418
3) Probability of error for individual bits . . .	1	0.37	0.15	0.06	0.03	0.04	0.04	0.039	0.029	Average errors per burst = 8.61

APPENDIX 4

Analysis of error bursts (250 bauds two-way transmission over switched telephone connections)
Maximum burst length = 82 elements

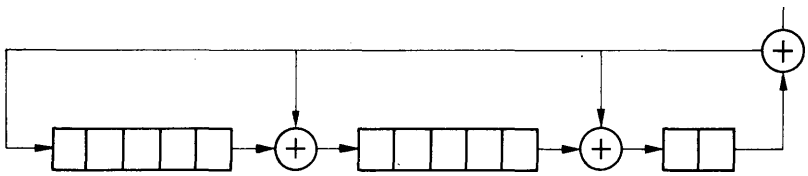
Analysis	Ordinal bit numbers in burst or numbers of bits in error as appropriate								Totals, etc.
	1	2	3/5	6/10	11/15	16/25	26/50	51/82	
A) *All recorded bursts*									
1) Error distribution .	2116 (23.65 %)	524 (5.86 %)	491 (5.5 %)	425 (4.76 %)	283 (3.17 %)	884 (9.88 %)	2209 (24.75 %)	2005 (22.43 %)	8937
2) Burst lengths . . .	1222 (57.9 %)	74 (3.5 %)	(..... 820 .....)						2116
3) Probability of error for individual bits .	1	0.248	0.0775	0.0402	0.0268	0.0418	0.0418	0.0297	Average errors per burst = 4.22
B) *Excluding all single and double (consecutive bit) error bursts*									
1) Error distribution .	820 (10.81 %)	450 (5.95 %)	491 (6.50 %)	425 (5.62 %)	283 (3.73 %)	884 (11.68 %)	2209 (29.18 %)	2005 (26.53 %)	7567
2) Probability of error for individual bits .	1	0.549	0.200	0.504	0.0690	0.108	0.108	0.0765	Average errors per burst = 9.23

Assessment tests of cyclic error detection codes

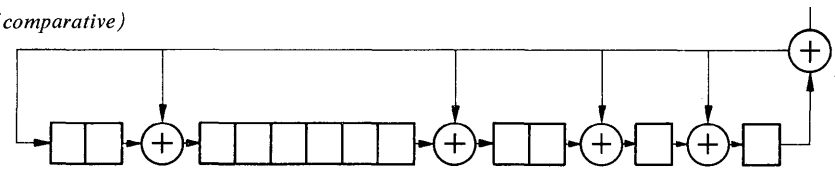
The error-detection capability of the 12-bit cyclic code has been assessed using computer generated errors. The computer was so programmed that the errors generated had a separation characteristic similar to that measured in practice. The block size used was 256 bits and only those containing even numbers of errors of 4 or more were analysed.

For comparison purposes the computer simultaneously assessed a variation of this 12-bit code, using more feedback points. The two codes are illustrated below.

Code 1



Code 2 (comparative)



The detailed results are :

Blocks analysed		Undetected blocks	
Errors/block	Number of blocks	Code 1	Code 2
4	43 376	27	12
6	23 976	16	7
8	13 841	12	5
10	7 910	5	0
12	4 497	0	1
14	2 699	5	2
16	1 530	0	0
18	777	0	0
20	479	1	0
22	254	0	0
24	159	1	0
26	98	0	0
28	43	0	0
30	26	0	0
32	10	0	0
34	4	0	0
36	4	0	0
38	0	0	0
40	2	0	0
42	1	0	0
44	1	0	0
46	0	0	0
48	1	0	0
Totals	99 688	67	27

Allowing for the exclusion of blocks containing only two or an odd number of errors, all of which errors would have been detected, one may estimate that only 67 blocks out of 300 000 faulty blocks would remain undetected by Code 1 and that the element-error rate reduction factor attributable to the detection code alone would be 5×10^3. This error rate improvement, allowing also for improvements due to other factors in a practical system, is considered quite adequate.

Additional feedback arrangements such as indicated in Code 2 would give somewhat better error detection, but the necessity for these is questionable.

Supplement No. 41

SPECIAL STUDY GROUP A — *(Extract from report COM Sp. A/No. 34 — October 1961)*

REPORT ON TEST RESULTS OF DATA TRANSMISSION OVER NATIONAL SWITCHED TELEPHONE CIRCUITS

1. *Introduction*

The test results show unequivocally that data transmission is practicable with acceptable and, in many cases, good efficiency over the national switched telephone network at speeds of 600 and 1200 bauds. A wide range of error performance has been found, but error-detecting schemes exist and new ones are being devised which permit essentially error-free results after correction. As high a degree of reliability in this respect can be achieved as desired ; variations in transmission performance then only change the effective speed but not the accuracy of transmitted data.

Although detailed mathematical analysis is not included in this report, it is recommended that the results of all the tests, as reported, be assembled and be made available to all who may need them. An index (not necessarily complete) of these contributions is associated with this report.

The tests results do not of themselves display an obvious recommendation but form a basis upon which can be made comparative evaluations of specific types of terminal arrangements (viz., error detection, modulation, etc.).

2. *Purpose of tests — Questions to be answered*

a) Can the switched network be used for data transmission ?

b) What are the principal causes of errors ?

c) What compatibility problems arise with echo suppressors ?

d) What compatibility problems arise with signalling ?

e) What are the uncorrected error rates ?

f) Can recommendations be made for optimum standards on the evidence of the results ?

g) Can features of error control arrangements be suggested which could form a solid basis for design ?

h) Can any other general conclusions be drawn ?

3. *Conclusions*

a) *Can the switched network be used for data transmission ?*

Although the telephone network was designed for voice usage, the test results show that telephone circuits in the switched network can be used for data transmission but will need good terminal device design. There is some correlation between good voice communication requirements and good data performance ; for example, a good received signal level, especially at the final selector stage, is highly desirable both for voice and for data communication. Similarly, the broader the available bandwidth, the better the data transmission.

Specifically, using existing types of frequency or simple phase modulation, in the large majority of cases in the switched network, 1200 bauds transmission is possible, and 600 bauds is possible in almost all the remaining cases. As the voice communication quality of lines will, on the average, improve with the normal growth of the network, so the proportion of poor connections for data transmission will diminish.

b) *What are the principal causes of errors ?*

Causes of errors have been considered under the following headings :

i) Impulse noise and interruptions ;

ii) Systematic distortion (i.e. binary or "telegraphic" distortion) ;

iii) Echoes, etc.

i) White noise of levels likely to be encountered is of little importance. Impulse noise and interruptions are considered to be a major cause of errors. The impulse noise problem varies in severity with the kind of switching arrangements. Selectors using pressure contacts with little relative motion give better performance than some other designs but this is very dependent on the detailed design of the system. In one instance, it was found that the quality of transmission over a route depended on which end set up the call, owing to differences in the details of the routes set-up. Metering operations have also been associated with errors, possibly by the introduction of impulse noise into the data circuit. There is much evidence of correlation between interruptions and maintenance activity ; Administrations might like to review some maintenance procedures especially in repeater stations. Operator intervention is also a frequent source of interruptions. Change-over to stand-by channels and fading on microwave relays, and poor contacts are all common sources of interruptions. Most are of short duration but no distributions are available since information on this was not requested. Long breaks due to complete failures of switched circuits are not considered as important as they would be on a private line since the call can be re-established.

ii) Systematic distortion is likely to be associated with frequency offset, phase distortion and attenuation distortion. The frequency offset encountered has not been sufficient to be a serious factor with carefully designed modulator/demodulators but its existence must be borne in mind in future design of modems. Phase distortion, on the other hand, is a controlling factor in data transmission and attenuation distortion may have some influence. For use on the switched network a phase or attenuation equalizer can only be a compromise, but note should be made of any circumstances which allow the best compromise to be reached. This could be the application of prescribed equalizers where transmission standards over the routes which a particular customer is likely to use are reasonably uniform—i.e., the selection of the most appropriate "standard" equalizer for application to customers' equipments.

iii) Test results suggest that echoes are not a problem while data are being transmitted. However, end-of-message signal errors have been found to be a result of talker-ends echoes causing repeated transmission. Further, listener-end echoes have caused the addition of spurious digits. Both effects have been eliminated by redesign of terminal equipments to ensure correct sequence of send/receive transition.

The presence of compandors in the circuit was not found to be a source of errors. It should be noted that all the systems tested were of a kind which would keep a fairly constant power on the line; the same might not be true of an on-off or amplitude-modulated system.

c) *What compatibility problems arise with echo suppressors?*

Echo suppressors of existing types would prevent simultaneous transmission of control signals in the backward direction when data are being sent in the forward direction. An examination of the test results on block-error rate suggests that, if it is not possible to arrange simultaneous two-way operation of this kind, the user will have to accept a lower effective speed of data transmission owing to the need to wait after each block for echo suppressor release and an acknowledgement signal, then to wait again for the same interval before repeating an erroneous block or continuing to transmit the next block.

This suggests that either the echo suppressors should be modified to allow simultaneous two-way operation on data calls, or arranging not to switch-in echo suppressors for data calls on circuits where they would normally have been included.

d) *What compatibility problems arise with signalling?*

Tests were not specifically pointed at determining how far the various systems were compatible with signalling systems, but results do not indicate that any problems arose. More testing is required on this point as modulators become more closely standardized. Most of this testing should probably be done on a laboratory basis, in order to examine the greatest variety of conditions in the shortest time.

e) *What are the uncorrected error rates?*

This report makes no attempt to summarize error rates. The contributions by the various Administrations contain a large body of valuable data. Only a statistical treatment of this subject is appropriate. Almost all the error rates quoted in the detailed results require qualification in the light of their applicable circumstances, and it would be misleading to over-simplify the reported results. The question also arises of the extent to which the frequently used "average element-

error rate" has any significance. For these reasons no attempt has been made to do anything other than index the results under the following headings :

i) Block error rate for given block size (s)

ii) Number of erroneous bits within erroneous blocks

iii) Error burst length

iv) Structure of error bursts or error distribution

v) "Average element-error rate"

These indices are given separately (and may not yet be complete).

f) *Can recommendation be made for optimum standards on the evidence of the results ?*

The test results do indicate that satisfactory performance can be had at speeds of 600 and 1200 bauds. It is to be understood that there is no evidence for the choice of precise speeds but merely for speeds in the regions quoted, which also include the start-stop case. The test results do not enable a definite recommendation to be made for a modulation system.

g) *Can features of error control arrangements be suggested which could form a solid basis for design ?*

The detailed statistics can be used as the basis for design of appropriate error control systems. For the transmission of batch data, error correction by means of re-transmission is appropriate. For most applications it would be possible to select block size solely on a transmission basis, but the economics of terminal arrangements may influence block length more, as might also considerations of procedures for working the data transmission system.

h) *General conclusions*

The test results give the raw error performance to be expected and hence indicate how powerful the error-detecting code must be for a given over-all performance. The data given on the structure of error bursts and the distribution of errors can be used to construct mathematical models of error patterns which might be used for comparative testing of various codes, generally performed on computers. The results might also be used to assist with the composition of "message start" patterns. Some of the tests have been recorded in the form of detailed tapes and use of these could reduce field testing time. Sufficient field tests to verify theoretical and laboratory work should always be performed on every new system.

4. *Future testing — Future work*

There does not appear to be a need for tests comparable in extent to those already submitted to be undertaken again. The testing that has been done has to a large extent defined the data transmission parameters associated with the switched network. Further tests will nevertheless be of value, especially in those countries which have not assessed the data transmission qualities of their network.

Future tests, of a somewhat different nature, will be needed to perform comparative evaluations of new types of modulation systems operating at their nominal modulation rates. With the

knowledge now available of the network capabilities, however, much of the future comparative testing could be done in the laboratory and merely confirmed in the field.

Although there is no recommendation for further large-scale testing as a result of the series of tests now completed, additional information on these topics is of value. To maximize its value it is recommended that any future tests should have their fine-grain results recorded in as detailed and as unprocessed form as possible, e.g. on punched tape or magnetic tape, so that the information can subsequently be processed in the most appropriate way for the use to be made of it.

It would be very helpful to the designers of data modems to have statistical information on over-all subscriber-to-subscriber phase distortion. With the exception of test results on the U.S. network and included in Volume VII of the *Red Book* (starting on page 439), the test results give little assistance on this point, and because of the magnitude of the task, Study Group A hesitates to ask for a formal programme of tests. However, the results of any tests which the Administrations are able to make would be of help in the future study of this problem. In the absence of test results on this topic, designers can be referred to the data on phototelegraphy contained in Volume VII of the *Red Book*, at pages 104-105 and 328-335. Additionally various Administrations might provide estimates of any circuit routing of special interest.

As the installation and use of data transmission terminal equipments grows, it will become increasingly necessary to be able to make a quick assessment of the quality of the available line and circuits. The Study Group wishes to draw attention to the need to consider future work on this point. (One method of performing this assessment is described in Supplement No. 83.)

Supplement No. 42

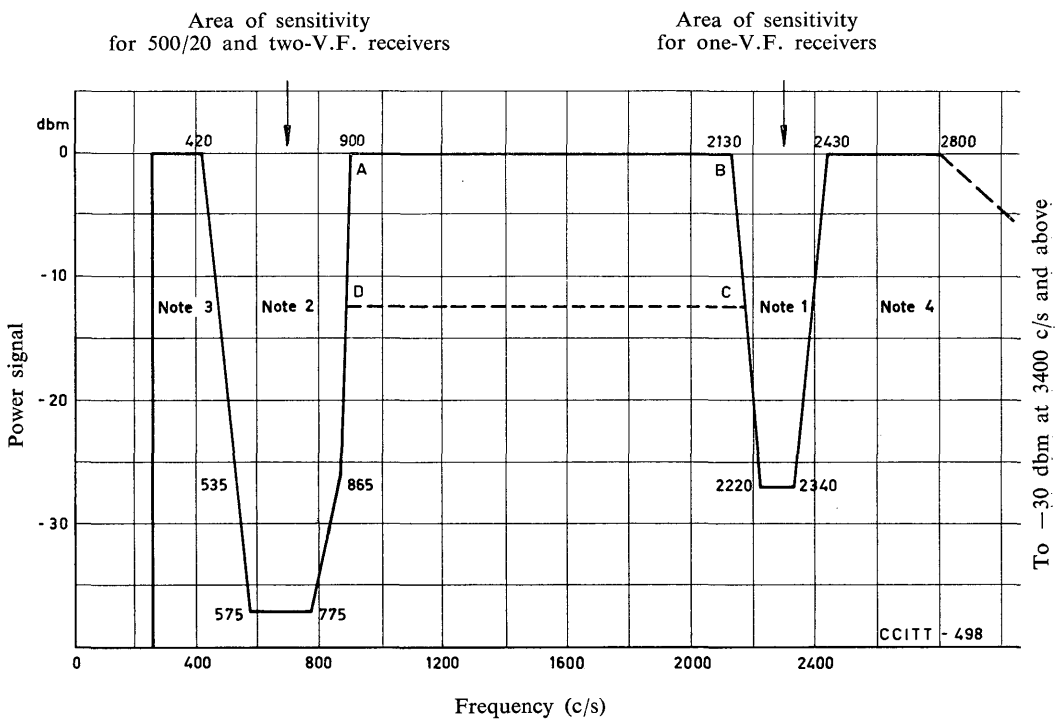
UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 5 — August 1961)*

COMPATIBILITY OF MODEMS AND THE TELEPHONE SIGNALLING SYSTEMS

Voice-frequency telephone signalling systems must be unresponsive to signalling frequencies occurring in speech. This is achieved by using signal receivers which respond only to a relatively narrow band of frequencies around the nominal signalling frequency (or frequencies), by incorporating receiver guard circuits designed to inhibit response when frequencies outside the signalling band are present, and by including in the system, delays which prevent any circuit functions taking place until the receivers have been operated for a pre-determined time. The earliest circuit function which takes place on receipt of a signal is, usually, "splitting" of the transmission path, the purpose of which is to limit the duration of the signal which may spill over into other links in the same connection. False splitting during data transmission would result in mutilation of the data signals and could not be tolerated.

Complete signal recognition requires longer delays than the splitting delay, and it is unlikely that such signals would be imitated by the data signals. Should imitation of a complete signal occur, a false circuit condition would result.

A variety of voice-frequency signalling systems are in use which employ different signalling frequencies and different splitting times. Further, systems using the same frequencies use receiver designs which differ in respect of signal response bandwidth, the frequencies to which the guard circuits respond, and the relative sensitivities of the signal and guard circuits. Different designs of receivers have different response times and account needs to be taken of the possibility that a succession of short bursts of signalling frequency from data transmissions may be integrated by the receivers and result in one long operation which may exceed the splitting time. Having regard to the great diversity of signalling systems in use in the world, it is apparent that there will be considerable difficulty in attempting to define a data transmission bandwidth for universal



Note 1. — Signals up to 0 dbm within this area may be permitted if always accompanied by signals in area ABCD.

Note 2. — Signals having components lying within this area may be permitted if their characteristics preclude false operation of trunk signalling equipment.

Note 3. — Signals within this area must not be capable of confusion with or by exchange tones.

Note 4. — Signals within this area may not be transmitted satisfactorily.

FIGURE 1. — **Permissible signal level (as a function of frequency) at subscribers' premises assumes at least 3-db loss sub-exchange**

use which would be compatible with all telephone signalling systems, although the specification of such bandwidths could be very useful over limited areas.

National telephone signalling systems

Two in-band voice-frequency signalling systems are in use in the United Kingdom national network :

a) a 2 V.F. system operating on frequencies 600 c/s and 750 c/s, and

b) a 1 V.F. system operating on 2280 c/s.

In regard to a), different designs of receivers, using different guarding arrangements, are employed. In earlier designs, guarding is obtained solely from a band in the vicinity of 300 c/s and 375 c/s, whereas in later designs guarding is provided by an aperiodic circuit which responds to frequencies over the whole speech bandwidth. The absolute and relative sensitivities of the guard and signal circuits also differ considerably in the various designs of receivers. In the 1V.F. system, the receiver design is such that guarding is obtained from a wide frequency band below the signalling band, and a limited amount of guarding from frequencies above the signalling band.

It has proved practicable in the United Kingdom, taking account of both these systems, to define a frequency band between 900 c/s and 2100 c/s which may be used for data transmission without the risk of misoperation of the signalling equipment (see Figure 1). Data systems which generate frequencies outside this bandwidth will not necessarily misoperate the signalling equipment, since guarding frequencies of sufficient amplitude to prevent the receivers from responding may be present, or any false operations which do occur may not be of sufficient duration to cause false splitting. Such systems are not therefore completely prohibited, but are subjected to special tests to establish their compatibility with the telephone signalling systems in the network before approval is given to their use.

International telephone signalling systems

At the present time two international voice-frequency in-band signalling systems are in use in the United Kingdom :

a) a two-V.F. system operating on frequencies 2040 c/s and 2400 c/s, and

b) a one-V.F. system operating on 2280 c/s.

Both systems operate the European international service and both are C.C.I.T.T. type.

The two-V.F. receiver guarding is provided by an aperiodic guard circuit which responds over the whole speech bandwidth. The splitting time is 35 ms and is on the compound frequency. In the one-V.F. system, guarding is provided by a wide frequency band below, and a limited frequency band above, the signalling band ; the splitting time is 55 ms.

Since in the two-V.F. system it requires both 2040 c/s and 2400 c/s to persist together for 35 ms to cause the line split, it is considered reasonable to assume that the frequency band 900-2100 c/s which may be used for data transmission without the risk of misoperation of the United Kingdom national signalling systems can also apply to the United Kingdom-Europe international signalling systems inter-working with the United Kingdom national systems. Here again, data systems which generate frequencies outside this 900-2100 c/s band will not necessarily misoperate

the international signalling equipment as guard frequencies may be present, or the imitation may not persist for the 35 ms.

A transoceanic international signalling system will be in service in the near future. As far as line signalling is concerned, this system is a two-V.F. 2400/2600 c/s system with a splitting time of 35 ms on either, or both, of the signalling frequencies. This signalling receiver is guarded by a wide frequency band below 2400 c/s and a limited band above 2600 c/s. It is considered that the comments made regarding the application of a data transmission frequency band 900-2100 c/s to the European international signalling systems apply equally to the transoceanic system inter-working with the United Kingdom national systems.

A future standard intercontinental voice-frequency signalling system(s) is now under consideration by the C.C.I.T.T. The line signalling frequencies are not yet determined, but it is thought unlikely that the lowest will be below 2100 c/s. At this time it is considered reasonable to assume that the comments made above regarding data transmission on the present, and immediate future, international circuits inter-working with the United Kingdom national network will also apply to the projected C.C.I.T.T. intercontinental system.

Supplement No. 43

THE CHAIRMAN OF STUDY GROUP XI — *(Contribution COM Sp. A/No. 9 — August 1961)*

COMPATIBILITY OF DATA TRANSMISSION METHODS AND TELEPHONE SIGNALLING SYSTEMS

1. *General*

Data transmission over switched telephone connections is liable to introduce difficulties in that the frequencies used for the data transmission signalling may fall within the signalling response bandwidth of the signalling receivers of in-band voice-frequency telephone signalling systems and cause misoperation of such systems. The problems arise both in the national and in the international switched networks. In international connections, the data transmission could be extended to the national networks and thus the characteristics of both the national and international telephone signalling systems need be taken account of.

2. *Problems arising*

In-band voice-frequency telephone signalling systems are designed to be, as far as possible, unresponsive to signalling frequencies occurring in speech, noise, or other interference. This is achieved by:

a) using signalling receivers which respond only to a relatively narrow band of frequencies centred around the nominal signalling frequency (or frequencies);

b) incorporating guard features in the receivers for the purpose of inhibiting signal response when frequencies outside the signalling band are present ; and

c) including delays in the system, which delays prevent any circuit function from taking place until the receiver has been operated for a pre-determined time.

The earliest circuit function which takes place on receipt of a signal is, usually, "splitting" of the transmission path. The purpose of this split is to limit the duration of the signal which may spill over into other links in the same connection. False splitting during data transmission would result in mutilation of the received data signals and could not be tolerated. This splitting is the main problem arising in the compatibility of data transmission and telephone signalling systems.

Complete recognition of a telephone signal requires longer recognition times, and thus delays, than the line splitting delay, and it is very much less likely that such signals would be imitated by data signals. Should imitation of a complete signal occur, a false circuit condition would result and, at worst, the connection would be released falsely.

A variety of in-band voice-frequency telephone signalling systems are in use in the world. These systems employ different signalling frequencies and different splitting times. Further, systems using the same signalling frequencies use signalling receiver designs which differ in respect of signal response bandwidth, the frequencies to which the receiver guard circuit responds, and the relative and absolute sensitivities of the signal and guard circuits. Different designs of receivers have different response times and account needs to be taken of the possibility that a succession of short bursts of signalling from data transmission may be integrated by the receivers and result in one long operation of the receiver which may exceed the splitting time. Having regard to the great diversity of signalling systems in use in the world, it is apparent that there will be considerable difficulty in attempting to define a data transmission bandwidth for universal use which would be compatible with all telephone signalling systems, although the specification of such bandwidths could be very useful over limited areas.

3. *General characteristics of telephone signalling systems having influence on the problem*

The following are the main factors :

a) the signalling frequency (or frequencies) ;

b) signal receiver guard arrangements ;

c) splitting time.

Account must be taken of these factors for both the national and international signalling systems.

3.1 *Signalling frequencies*

Volume VI of the *Red Book* (pp. 40 and 41) details the signalling frequencies of the various national signalling systems in the world. Excluding generator signalling the frequencies are :

600, 750, 1200, 1600, 1900, 2000, 2040, 2100, 2280, 2300, 2400, 2500, 2600, 3000 c/s.

The two present C.C.I.T.T. international signalling systems have signalling frequencies 2280 c/s and 2040/2400 c/s.

Transoceanic intercontinental signalling systems having line signalling frequencies 2400 c/s and 2600 c/s will be in service in the near future.

A future standard intercontinental signalling system(s) is now under consideration by the C.C.I.T.T. The line signalling frequencies are not yet determined, but it is thought unlikely that the lowest will be below 2100 c/s.

3.2 *Signalling receiver guard arrangements*

Usually, these fall into three main categories :

a) Guarding provided by an aperiodic guard circuit which responds to frequencies (including the signalling frequencies) over the whole speech bandwidth. Usually, this guard is not very sensitive.

b) Guarding obtained from wide frequency bands below and above the signalling band. This is a tuned guard circuit and is usually more sensitive than the aperiodic guard, particularly below the signalling band.

c) Guard obtained from a relatively narrow band of frequencies, usually below the signalling band, and broadly tuned to the sub-multiple of the signalling frequency.

Usually, guard circuits are "difference" type, where the output of the receiver guard circuit is subtracted (or compared with) the output of the receiver signal circuit, and the response of the receiver at the final detection stage made to depend on the difference. Arrangements a), b) and c) above are usually of this type. Exceptionally, a guard circuit may be "stop" type, where the output of the receiver guard circuit completely negatives the effect of the receiver signal circuit regardless of the magnitudes of the guard and signal outputs.

It is clear that, for data transmission purposes, the "stop" guard has considerable advantage in that, providing guard frequencies are present, signal response is inhibited with greater certainty than with the "difference" type guard. Unfortunately, this merit for data transmission is a considerable disadvantage for normal telephone signalling as the top guard gives rise to considerable signal interference and for this reason stop guards are not incorporated in modern techniques.

3.3 *Splitting times*

Volume VI of the *Red Book* (pp. 40 and 41) details the splitting times of the various national signalling systems. These times vary between 8 and 400 ms. The C.C.I.T.T. has recommended 35 ms maximum splitting time for simple, and 55 ms maximum for compound, international signalling systems.

In view of the extreme importance of the splitting time in relation to data transmission, more particularly should the data signalling frequencies include the telephone signalling frequencies, it is considered wise to assume a short, rather than a long, splitting time, and a time of the order 25 ms is suggested for data transmission considerations.

4. *Possible approach considerations to ensure compatibility between data transmission and telephone signalling systems*

There appear to be three main lines of approach which could be considered :

4.1 *Restricted bandwidth for data transmission*

The basis of this proposal is to adopt a restricted bandwidth for data transmission such that the data signalling frequencies do not include the telephone signalling frequencies, and thus line

splitting should not occur on data transmission. If all the telephone signalling frequencies listed in section 3.1 are to be taken into account, this would not be possible. A practical approach, however, could be to assume a band of, say, 900-2100 c/s for data transmission. This would give rise to possible difficulties with the 1200, 1600, 1900, 2000, 2040 and 2100 c/s national signalling systems as listed in Volume VI of the *Red Book* (pp. 40 and 41) and would require further study. It could well be that some of the data signalling frequencies would be guard frequencies of these systems and thus the systems guarded against false line splits. No difficulty would arise with the C.C.I.T.T. 2-V.F. 2040/2400 c/s international signalling system with the 900-2100 c/s data bandwidth as this system splits on compound signals only.

4.2 *Data transmission using whole speech bandwidth*

Here it is assumed that, while the data signalling frequencies include the telephone signalling frequencies, they also include (as data signalling frequencies) guard frequencies of the various telephone signalling systems, and of sufficient amplitude to prevent receiver signal response, or any false operations which do occur may not be of sufficient duration to cause false splitting.

This is an attractive line of approach from the data point of view, but to reach preliminary conclusions would require the characteristics of the data transmission technique to be established in relation to the guard characteristics of the various receivers in use in the various in-band voice-frequency telephone signalling systems. Confirmatory tests would then need to be carried out on switched connections. The logical approach for the data transmission technique in this case would be to include as many data signalling frequencies as possible with the aim of these including telephone signalling guard frequencies, and for the signal element durations to be as short as possible relative to the splitting time.

It is thought that this line of approach would be difficult to resolve if the considerations of all the national and international signalling systems are to be taken into account.

4.3 *Data technique applying a continuous guard frequency*

The basis of this proposal is that the data transmission system transmits a continuous guard frequency while data signalling. This frequency would not be used for data signalling, but used to energize the signalling receiver guard circuit to inhibit signal response. This technique would allow the data signalling frequencies to include the telephone signalling frequencies and would permit the data system to use the whole speech band. The problem is to select the frequency, and the level, of the guard tone.

The guard frequency must not be a telephone signalling frequency, and if all the national and international signalling systems are to be taken into account (see section 3.1) it is clear that the guard frequency would require to be below 600 c/s, or above 3000 c/s to avoid the telephone signalling frequencies. Unfortunately not all signalling receivers have guard action at these frequency regions. Further, even if they have, the sensitivity of the guard circuit tends to be low at these regions as :

a) in modern telephone signalling systems, deliberate steps are taken to make the guard circuit insensitive at about 400 c/s and below to avoid signal interference due to low frequency noise and surges, and

b) tuned guard circuits are generally designed to have maximum sensitivity at the sub-multiple of the signalling frequency (or frequencies). As this peak response would be at a frequency much less than 3000 c/s, in practical receiver design the guard response above 3000 c/s is low.

A further factor is that the attenuation/frequency distortion of the various circuits, particularly some national circuits, is high and rising rapidly at below 600 c/s, and above 3000 c/s (with the possible danger of frequency cut-off at around 3000 c/s on some national circuits). Also, as the guard frequency would require to be applied, and received, at subscribers' premises, account would require to be taken of the loss of local lines.

All the above factors point to the requirement for a particularly high level of transmitted guard frequency below 600 c/s, or above 3000 c/s, and, as end-to-end reception of data transmission over the connection would be involved, it would not always be possible to ensure satisfactory working.

A possible alternative would be to consider a guard frequency within a range (say 900-2100 c/s) which would be less subject to line attenuation/frequency distortion and receiver guard insensitivity problems ; which range may be suitable for some, or most, of the telephone signalling systems. It could well be, however, that this alternative would put an undesirable restriction on the bandwidth availability for data signalling frequencies.

Supplement No. 44

SWEDEN — *(Contribution COM Sp. A/No. 60 — June 1963)*

ADVANTAGES AND DISADVANTAGES OF THE VARIOUS FORMS OF MODULATION FOR THE TRANSMISSION OF DATA

1. *General*

Laboratory tests were undertaken on four different types of modem equipment intended for use over the commercial telephone network, and the results compared. Two-phase shift type modem equipments were tested, Ph 1 and Ph 2, and two frequency shift type modem equipments, FS 1 and FS 2. Ph 1 used differential coding and was intended for synchronous operation with a modulation rate of 1500 bauds and a carrier frequency of 1700 c/s. Ph 2 was intended for a synchronous operation with a modulation rate of 1200 bauds and a carrier frequency of 1800 c/s. The frequency shift modems used for the tests, FS 1 and FS 2, were group modulation systems designed to be in accordance with the recommendations of the C.C.I.T.T. In the FS 1 modem provision is made for a 75-baud backward channel with mid-frequency of 420 c/s and frequency shift of ± 30 c/s.

Type of modem	Modulation rate (bauds)	Lower side frequency (c/s)	Mid-frequency (c/s)	Upper side frequency (c/s)
Ph 1	1500	—	1700	—
Ph 2	1200	—	1800	—
FS 1	1200	1300	1700	2100
FS 2	1200	1300	1700	2100

2. *Measurements*

2.1 *General*

Measurements were carried out over an ideal artificial line with each modem connected back to back, and with the following types of disturbance injected :

a) White noise ;

b) Concentrated telephone line disturbance injected from a tape recording ;

c) Interruption type disturbance.

The modems were also tested for phase distortion sensitivity. The results of this particular test indicated that both types of modem put similar requirements on the line characteristics. Special tests were carried out at different receive levels. All modems proved to have sufficient level range. In order to carry out the measurements, a digital programme was transmitted over the system being investigated. The programme consisted of a number of blocks containing all combinations obtained with an 8-unit code (2048 bits).

Disturbances of different types and levels were injected and the element error rate was recorded on a special measuring instrument. As the measurements were mainly intended as comparative measurements great care was taken to keep the injected disturbances as constant as possible during the measuring period. Each type of measurement was repeated several times in order to ensure reliable test results.

2.2 *Measurements with white noise*

The error rate, as a function of the injected white noise, was measured with five different noise levels for each modem. Each modem was investigated by sending a total of approximately 14×10^6 bits. Naturally the greatest number of bits were used with the lower error rates.

The results of the measurements are shown in Figure 1.

2.3 *Measurements with impulse disturbances*

The modem equipments were investigated with concentrated disturbance injected from a tape recording. This recording contained disturbances from both local and long-distance lines and the disturbance level was sufficiently high to produce a suitable error rate during the measurements. For each measurement a total of approx. 3.6×10^6 elements were transmitted. The results of the measurements are shown in Figure 2.

2.4 *Measurements with interruption disturbances*

An interrupter, adjustable for different interruption times, was connected between the modulator and demodulator. Tests were made with interruption times of approx. 1.2, 1.6, 2, 4, 8, 16, 32 and 64 ms.

The interruption-free periods between the interruptions were approximately ten times as long as the interruption length. Tests were carried out for a period of approx. 5 minutes for each of the interruption lengths. The system Ph 2 fell out of phase with interruption lengths greater than approx. 16 ms and Ph 1 produced double errors with short interruptions. The results are shown in Figure 3.

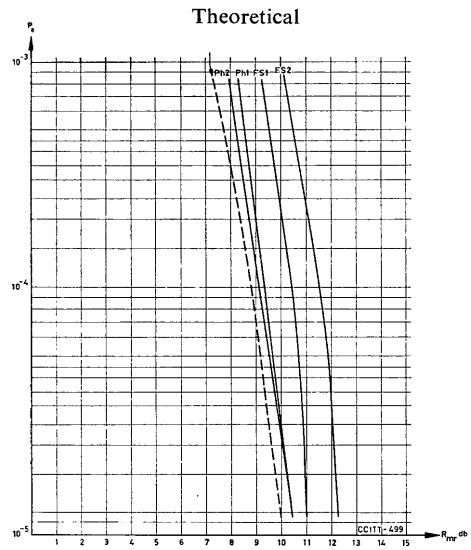


FIGURE 1. — **Element-error rate plotted against the normalized signal-to-noise ratio**

Dotted curve is the theoretical optimum for phase modulation with ideal phase reference

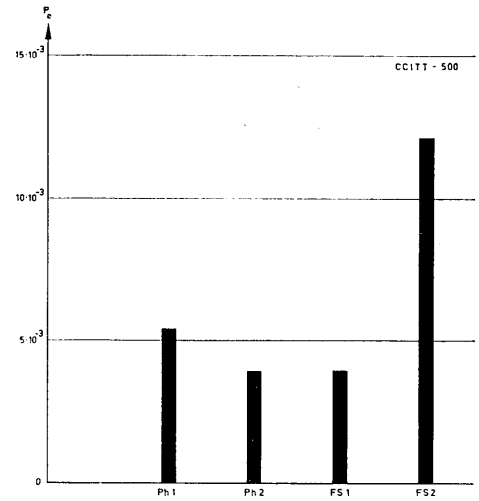


FIGURE 2. — **Element-error rate at measurements with concentrated telephone line disturbance injected from a tape recording**

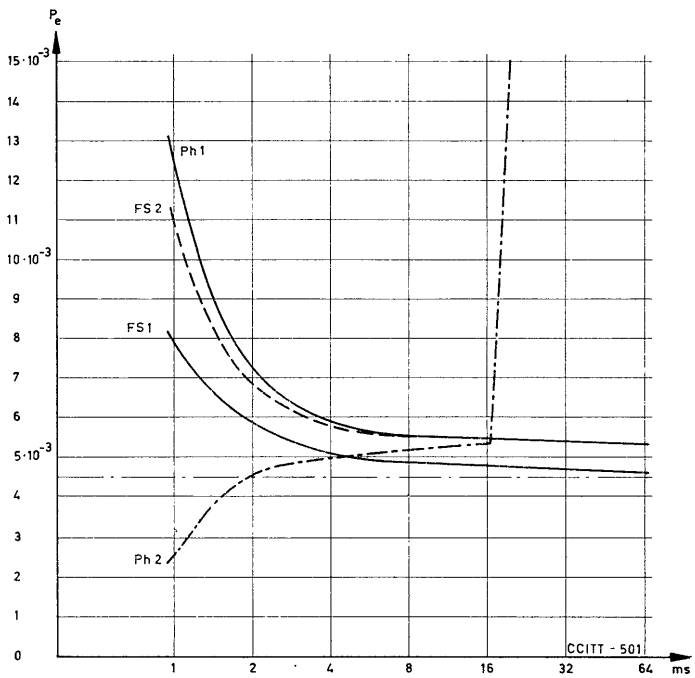


FIGURE 3. — **Element-error rate plotted against line interruption time**

3. *Conclusions*

The tests carried out indicate that all the modems tested are suitable for data transmission at modulation rates up to approximately 1200 bauds. Each modem has its particular advantages and disadvantages. The phase modems, as expected, are somewhat less sensitive to disturbances in the form of white noise (see Figure 1). However, the difference between the phase modems, and the frequency modulation modem FS 1 is only of the order of 0.5 to 1 db for comparable error rates. On the other hand a comparison of the results in Figure 3 indicates that the synchronized phase shift modem Ph 1 always causes at least twice the errors produced by the frequency shift modem FS 1. The FS 1 modem is in general superior to the FS 2 system. Results of the tests of the asynchronous phase shift modem, Ph. 2, do not show this double error characteristic, but on the other hand the modem has the fundamental disadvantage that it falls out of phase with interruption lengths greater than approximately 16 ms. With impulse disturbances the FS 1 modem appears to be at least as good as the phase shift modems.

It is possible to conclude from the above that the phase shift modems do not have any decided advantages compared with the frequency shift modems. The slight advantage in signal-noise ratio with white noise should not be decisive, when the disadvantages of double error, with consequent increased error detection difficulties, and interruption of the system synchronism, demanding manual adjustment in order to restart the data transmission, are taken into consideration. In view of their simpler construction, the frequency shift modems should also prove to be preferable when maintenance work is taken into consideration.

Supplement No. 45

SPECIAL STUDY GROUP A — *(Extract from report COM Sp. A/No. 92 — October 1963)*

SPECIFICATION TO TRANSMIT DATA AT MODULATION RATES ABOVE 1200 BAUDS

With regard to the need for leased telephone-type circuits of quality higher than that normally recommended by the C.C.I.T.T., there was general agreement that a requirement for such circuits exists.

It was further agreed that the steady state part of the schedule 4B specifications for leased telephone circuits given in the A.T. & T. contribution (Supplement No. 74) were a suitable standard for consideration. Special Study Group A will pursue, with the Study Groups IV and XV of the C.C.I.T.T., the possibility of providing and maintaining such circuits.

The Study Group was unable to decide on a target specification for impulse noise but noted the A.T. & T. contribution in Supplement No. 74.

Modulation methods for data transmission when data-signalling rates at the interface exceed 1200 bits per second

The following tables summarize these proposals. The modulation methods indicated in columns 1 and 2 are, of course, applicable to the type of circuits mentioned in columns 2 and 3, respectively.

Abbreviations are explained at the end of the tables.

a) *Interface data-signalling rate : 1800 bits/second*

Modulation method to be used on :		
Switched circuits (1)	Normal-type leased circuits (2)	High-quality leased circuits (3)
180° Ph.M./D.S.B. F_0 = 1800 c/s F.M. biternary or C.D.C. Fixed rate Subject to bandwidth availability : Synch. Ph.M. system (switchable 1200-1800 bauds) Binary D.S.B. Differentially coherent detection Fixed rate F_0 = 1800 c/s Ph.M./V.S.B. F_0 = 1800 c/s Fixed rate Coherent F.M. binary coherent or not coherent	As for switched circuits but without switching to 1200 bauds at reduced bandwidth	

b) *Interface data-signalling rate : 2000 bits/second* [1]

Modulation method to be used on :		
Switched circuits (1)	Normal-type leased circuits (2)	High-quality leased circuits (3)
V.S.B./A.M. system Transparent		F.M., transparent
Four-phase fixed rate		180° Ph.M./D.S.B. F_0 = 1800 c/s
Quaternary Ph.M./V.S.B. F_0 = 1800 c/s Non-coherent		

[1] This 2000 bits/second data-signalling rate is not proposed by the C.C.I.T.T. as a preferred value.

c) *Interface data-signalling rate : 2400 bits/second*

Modulation method to be used on :		
Switched circuits (1)	Normal-type leased circuits (2)	High-quality leased circuits (3)
Ph.M./S.S.B. Carrier frequency at lower end of the transmission band. Transparent at asynchronous mode		180° Ph.M./D.S.B. F_0 = 1800 c/s
A.M./V.S.B. with synchronous homodyne detection, transparent		Binary/ternary F.M. Fixed rate
Four-phase synchronized Fixed rate	Four-phase fixed rate	
Twin channel, phase reversed with phase reference Fixed rate		
Subject to bandwidth availability ; synch. Ph.M. (switchable 1800-2400) ; quaternary D.S.B. system ; differentially coherent detection F_0 = 1800 c/s	As for switched circuits but without switching to 1800 at reduced bandwidth	
Three-phase diff. mod. F_0 = 1700 c/s	V.S.B./A.M. Transparent	
Four-phase diff. mod. F_0 = 1800 c/s		
Four-phase/V.S.B. F_0 = 1800 c/s Coherent	Binary Ph.M./V.S.B. F_0 = 2400 c/s Coherent	

d) *Interface data-signalling rate : 3000 bits/second*

Modulation method to be used on :		
Switched circuits (1)	Normal-type leased circuits (2)	High-quality leased circuits (3)
Quaternary Ph.M./V.S.B. F_0 = 1800 c/s Non-coherent	180° Ph.M./S.S.B. Carrier frequency on the lower end of the transmission band, and variable between 600 and 1600 c/s. Transparent at asynchronous mode	V.S.B./A.M. ; transparent
	Synchronous Ph.M. Quaternary D.S.B., Differentially coherent detection F_0 = 1500 c/s	Quaternary Ph.M./V.S.B. F_0 = 2400 c/s Non-coherent

e) *Interface data-signalling rate : 3600 bits/second*

Modulation method to be used on :		
Switched circuits (1)	Normal-type leased circuits (2)	High-quality leased circuits (3)
Quaternary Ph.M. binary A.M. modulation V.S.B. F_0 = 1800 c/s Coherent	Quaternary ; and 2 level A.M. 180° Ph.M./V.S.B.	Ph.M./S.S.B. Carrier frequency on lower end of the transmission band and variable between 600 c/s and 1600 c/s ; transparent at synchronous mode
	Six-phase diff. mod. with individual equalization F_0 = 1700 c/s	
	Eight-phase diff. mod. with individual equalization F_0 = 1800 c/s	Synchronous quaternary Ph.M./D.S.B. system Differentially coherent detection F_0 = 1800 c/s
	Quaternary Ph.M./V.S.B. F_0 = 1800 c/s Coherent	

f) *Interface data-signalling rate : 4800 bits/second*

Modulation method to be used on :		
Switched circuits (1)	Normal-type leased circuits (2)	High-quality leased circuits (3)
Quaternary Ph.M. Quaternary amplitude modulation/V.S.B. F_0 = 1800 c/s Coherent		Six-phase diff. modulation with individual equalization F_0 = 1700 c/s Eight-phase diff. modulation with individual equalization F_0 = 1800 c/s Quaternary 180° Ph.M./V.S.B. and 2 levels A.M. Quaternary Ph.M./V.S.B. F_0 = 2400 c/s Coherent

Abbreviations :

A.M. = amplitude modulation
F.M. = frequency modulation
Ph.M. = phase modulation
F_0 = carrier frequency or centre frequency
C.D.C. = characteristic distortion correction

Supplement No. 46

I.B.M. (UNITED STATES OF AMERICA) — *(Extract from contribution COM Sp. A/No. 13 — September 1961)*

AN EXPERIMENTAL PHASE REVERSAL SYSTEM FOR 600- TO 2400-BAUD OPERATION

A. Factors affecting the choice of modulation system

Reasonable cost of data transmission is essential to the success of many communications-based computer systems ; therefore, simplicity and efficiency are important considerations in choosing a standard modulation method. A flexible modulation system, adaptable to transmission

channel characteristics, would also be very desirable. The ability to operate as reliably as possible in the presence of impulse noise and sudden amplitude and frequency shifts is another critical factor.

Simple implementation and greatest reliability in the presence of noise can be achieved if data is transmitted in a binary mode. Theoretical and experimental comparisons can be made of the relative reliability of different modulation systems in the presence of white Gaussian noise, which is easy to treat mathematically. Such studies show that a properly designed 180-degree phase modulation and synchronous detection system promises to be most reliable in the presence of white noise, as well as simplest to implement. A well-designed system of this type will not be sensitive to the amplitude variations and frequency spectrum drifts commonly encountered on telephone lines.

But reliability in the presence of impulse noise is our main concern. Extensive laboratory tests, under controlled conditions, with injected prerecorded impulse noise from actual telephone lines, have shown that the systems which are more reliable in the presence of white Gaussian noise have the tendency to be also more reliable in the presence of impulse noise. The relative reliability depends on the particular sample of impulse noise used in the tests. Tests with a distribution of impulse noise typical of many telephone lines have shown that an improvement of 3 db in the signal-to-Gaussian-noise figure can result in a fourfold increase in reliability in the presence of impulse noise. This is, of course, only true for a given distribution of the number of impulse noise bursts as a function of impulse noise levels and will differ from case to case.

Thus, it appears that close to optimum reliability of operation can be achieved with simple phase reversal systems. However, such systems have, until now, been limited in speed, primarily because of their inability to operate with less than one cycle of the carrier frequency per bit of information. This limitation has been overcome in a recent experimental system in which classical modulation techniques have been used to achieve speeds close to the predicted achievable limit. With these techniques, it is possible to place the transmitted information spectrum at an optimum frequency allocation, as indicated by statistical information on the available spectrum of the transmission channel for different types of services. In addition, bit-synchronous operation is no longer required and therefore start-stop operation is permissible up to the highest reliably operable speed (in this case, 2400 bauds).

In the following sections, an operating laboratory model of this system is described and performance results given. Extensive tests will be required to determine the relative merits of this and other proposed systems.

B. Transmitter

The experimental system to be described is capable of operating at 600, 1200, 1800 and 2400 bauds with corresponding carrier frequencies of 1200, 1500 and 1800 c/s, and is easily switched from one speed and carrier frequency to another. Only the 2400-baud operation with an 1800-c/s carrier frequency will be described in detail. In order to generate a phase reversal signal from an 1800-c/s carrier at a rate of 2400 bauds (with the carrier frequency and the bit rate not synchronized) a linear product modulator is used as shown in Figure 1. Before the squared data signal is fed into the modulator, it must be optimized for linear operation by being passed through a low-pass filter (LP I) having a cut-off frequency of 1500 c/s. The modulation product thus

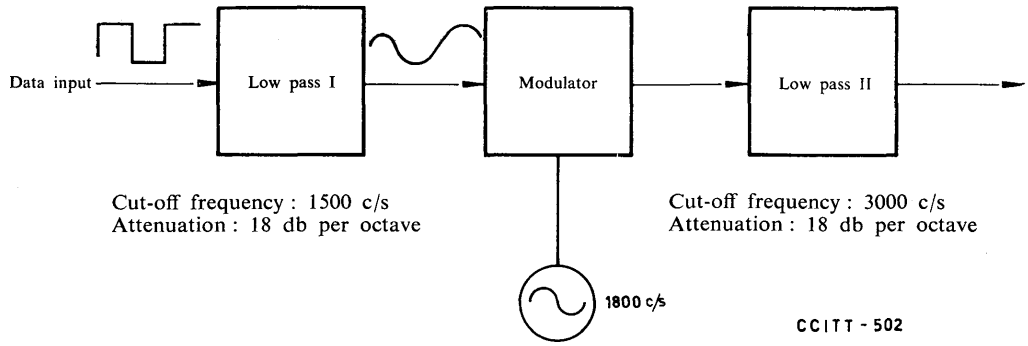


FIGURE 1. — **2400-baud transmitter**

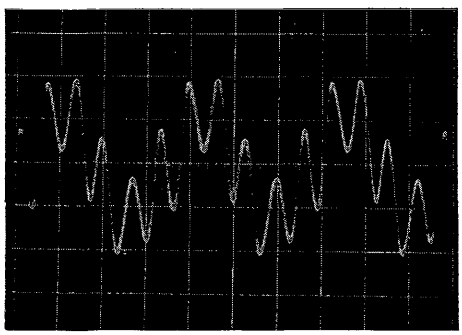


FIGURE 2. — **Line signal for the 10101010 binary combination transmitted at 2400 bauds**

(Scale : 500 µs/div)

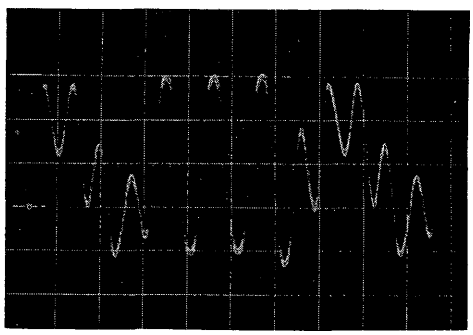


FIGURE 3. — **Line signal for the 00001010 binary combination transmitted at 2400 bauds**

(Scale : 500 µs/div)

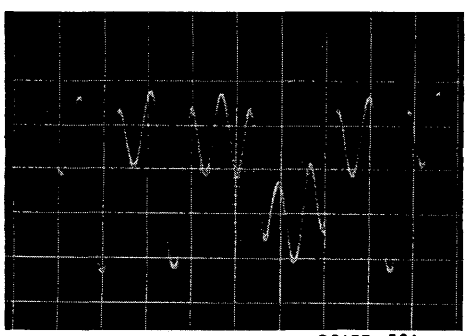


FIGURE 4. — **Line signal of a 10101100 binary combination transmitted at 2400 bauds**

(Scale : 500 µs/div)

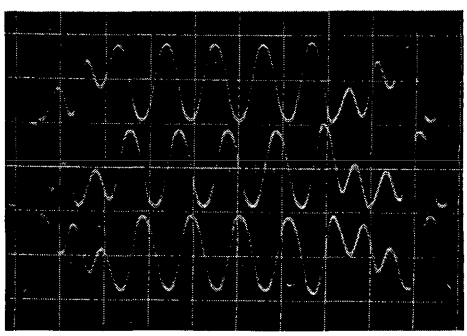


FIGURE 5. — **Transmitted single-bit signal with different carrier phase relations**

(Scale : 500 µs/div)

obtained is ready for transmission after the higher order products have been eliminated by a low-pass filter (LP II).

To illustrate the operation of the transmitter, the line signal waveforms will be shown for different bit combinations. If a 1 0 1 0 ... pattern (up-level = 1, down-level = 0) is transmitted at a rate of 2400 bauds, for example, the low-pass filter will eliminate all the harmonic components from the incoming square wave and let only a sine wave of 1200 c/s through. The product modulator will produce the upper and lower sideband frequencies of 600 and 3000 c/s only (1800 c/s $\pm$ 1200 c/s). This pattern will therefore result in a line signal of two frequencies, 600 and 3000 c/s. If, on the other hand, a series of ones or a series of zeros is transmitted, the line signal will consist of the 1800-c/s carrier frequency, the phase of which is determined by the polarity of the d.c. signal.

C. Receiver

The receiver block diagram (Figure 6) consists basically of two parts: carrier generation and synchronous detection. A practically noise-free synchronous detection carrier is derived from the received signal by a full-wave rectifier followed by a circuit tuned to twice the carrier frequency. The effect of full-wave rectification is the multiplication of sidebands:

$$\cos 2\pi (f_c - f_j)\, t \times \cos 2\pi (f_c + f_j)\, t = \frac{1}{2} \cos 2\pi (2 f_c)\, t + \frac{1}{2} \cos 2\pi (2 f_j)\, t$$

Thus, $f_c = 1800$ c/s; $f_j = 1200$ c/s in the example of reversal transmission at 2400 bauds.

A circuit tuned to $2f_c$ will thus recover this frequency independently of the modulation. The frequency divider following the tuned stage provides a practically noise-free reference carrier f_c for signal detection. The received data signal is first passed through a high-pass filter having a cut-off frequency at 2.5 kc/s with a slope of 18 db per octave. This high-pass filter considerably reduces interbit interferences. By eliminating the lower sideband of the signal, it minimizes the sensitivity of the data receiver to phase distortion introduced by the lower portion of the available voice-frequency band. The effect of the high-pass filtering can be seen in Figure 10. The following low-pass filter eliminates a portion of the available upper frequency band, which is also subject to considerable phase distortion. The high-pass, low-pass filter combination, in effect, limits the received signal to the good portion of the available frequency band and makes the receiver operate on a vestigial sideband basis. The data is recovered by multiplying the upper sideband of the signal by the reference carrier, in a synchronous detector:

$$\sum_{j=1}^{h} a_j \cos 2\pi (f_c + f_j)\, t \times \cos 2\pi f_c t = \frac{1}{2} \sum_{j=1}^{h} a_j \cos 2\pi f_j t + \frac{1}{2} \sum_{j=1}^{h} a_j \cos 2\pi (2f_c + f_j)\, t$$

The first term, $\frac{1}{2} \sum_{j=1}^{h} a_j \cos 2\pi f_j t$, is the recovered data signal. The second term is eliminated as an unwanted modulation product by a low-pass filter, following the detector. The squaring circuits following the low-pass filter (LP II) restore the data signal into its usual square waveform.

To illustrate the operation of the receiver, consider a line signal consisting of two sideband signals, 600 c/s and 3000 c/s, corresponding to the bit pattern 1 0 1 0 ... (Figure 7a). The full-

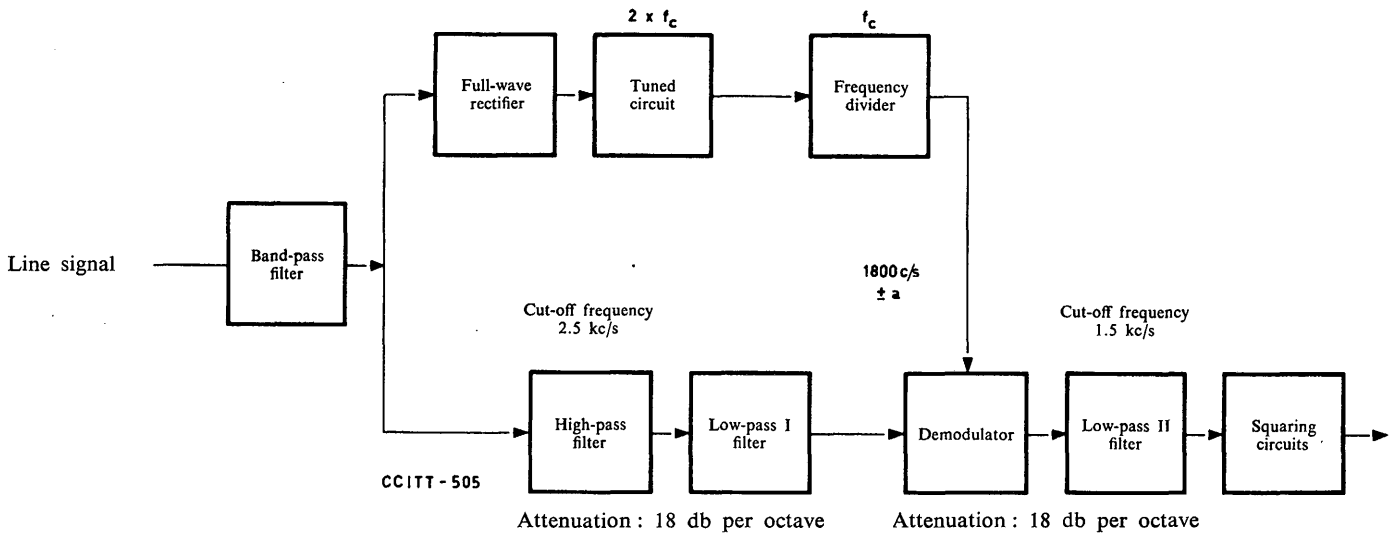


FIGURE 6. — **2400-baud receiver**

wave rectification produces a component at 3600 c/s (the sum of the sideband frequencies) which is filtered out and divided by 2 to yield the 1800-c/s carrier (Figure 7c). Before detection, the line signal passes through a high-pass filter to remove the lower sideband. The output of that filter shown in Figure 7b consists essentially of a 3000-c/s sine wave. The output of the synchronous detector, the product of (b) and (c), is shown in Figure 7d. The 1200 c/s component, Figure 7e, is obtained by feeding (d) through a low-pass filter. The clipped low-pass filter waveform, reconstructing the binary 1 0 1 0 signal, appears in Figure 7f.

Corresponding waveforms are shown in Figure 8 for an 0 1 0 1 0 0 0 0 bit pattern. In addition to the sideband frequencies, a strong carrier frequency component of 1800 c/s is present in the line signal (Figure 8a).

The low-pass high-pass filter combination will have to be optimized for each speed. Extensive field tests (still to be performed) may yield the optimum combination of high-pass and low-pass filters (Figure 6) in conjunction with compromise or variable phase equalization, for each speed and type of service.

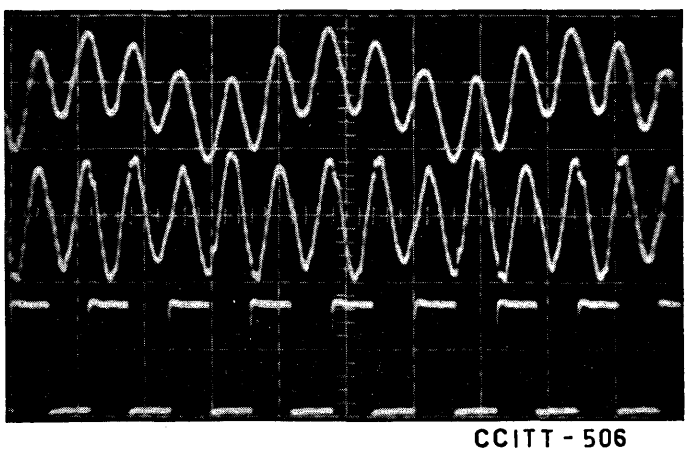
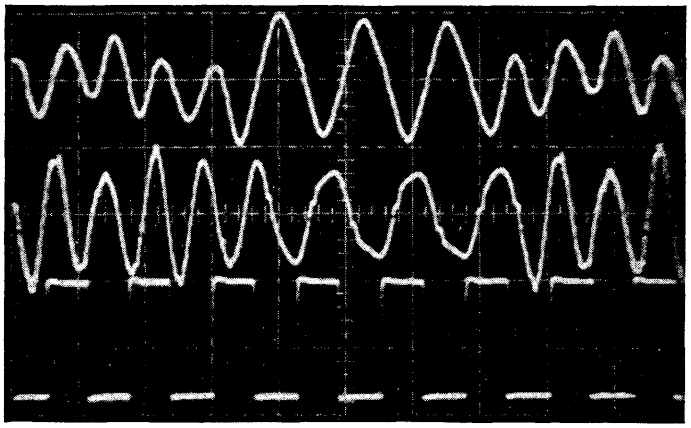


FIGURE 7. — **Receiver waveforms (10101010 bit pattern) at 2400 bauds**

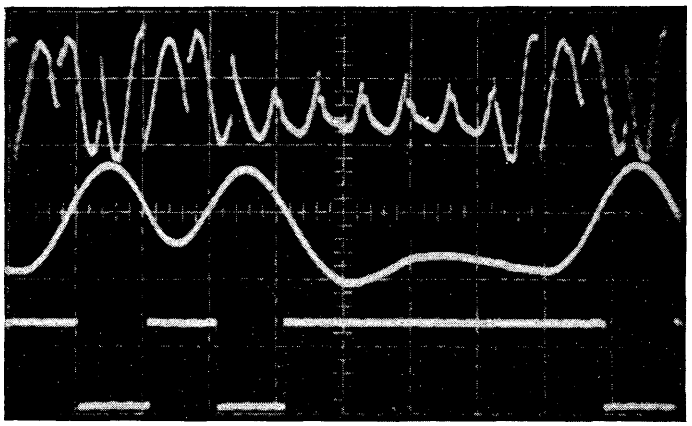
(Scale : 1 bit/div)



a. Line signal

b. Upper vestigial sideband

c. Squared recovered carrier



d. Detector output

e. Recovered data

f. Squared data

FIGURE 8. — **Receiver waveforms (01010000 bit pattern) at 2400 bauds**

(Scale : 1 bit/div)

D. TEST RESULTS

1. *2400-baud operation over a carrier quality line*

A 100-mile L-carrier loop (the frequency response of which is shown in Figure 9) was used for functional tests of the 2400-baud phase reversal system shown in Figures 1 and 6. The total bias distortion and jitter was approximately ± 10% of the bit duration. With LP I in Figure 6 removed, the receiver requires phase equalization for 2400-baud operation. Figure 10 shows the characteristic receiver waveforms with and without simple equalization. A RIXON EN-776-2 phase equalizer was used in the tests. The effect of high-pass filtering before detection is shown by comparison of Figures 10a and 10c ; with high-pass filtering, the clipping level can be set halfway between the extreme excursions of the data ; without high-pass filtering, the waveform

is not symmetric about the clipping level. In addition, the interbit interference is noticeably higher, as can be seen by observing the separation between the two zeros in the 1 1 1 1 1 0 1 0 pattern, which is much more distinct when the high-pass filter is used. One can observe that practically no interbit interference exists after post detection low-pass filtering (Figure 10c).

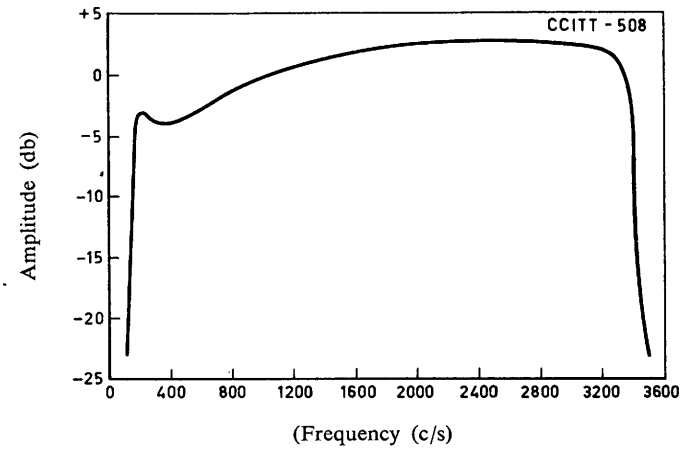
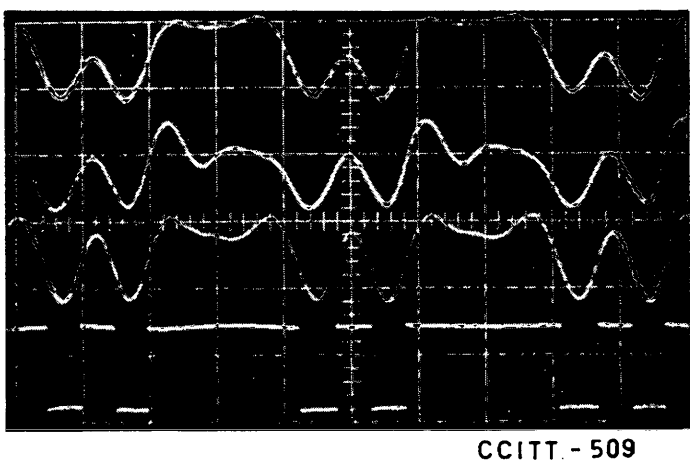


FIGURE 9. — **Frequency response of the 100-mile carrier quality loop**



a. LP II output with phase equalization but without high-pass filtering

b. LP II output without phase equalization but with high-pass filtering

c. LP II output with phase equalization and high-pass filtering

d. Squared data with phase equalization and high-pass filtering

FIGURE 10. — **Effect of phase equalization and high-pass filtering on 2400-baud data signals received over a 100-mile L carrier loop having a frequency response as in Figure 9**

(Scale : 2 bit/div)

2. *Tests over a combined loaded cable and carrier quality line*

Figure 11 shows the frequency response of a combined loaded cable and carrier quality line. With a carrier frequency of 1500 c/s and a high-pass, low-pass filter combination with a cut-off

frequency of 2 kc used in the receiver shown in Figure 6, 1800-baud operation was achieved with no phase equalization. Figure 12 shows characteristic receiver waveforms of the received signal at 1800 bauds using the line shown in Figure 11 and no phase equalization. The total bias distortion and jitter was again approximately ± 10%. Only 1200-baud operation has previously been possible over such a line, with no phase equalization.

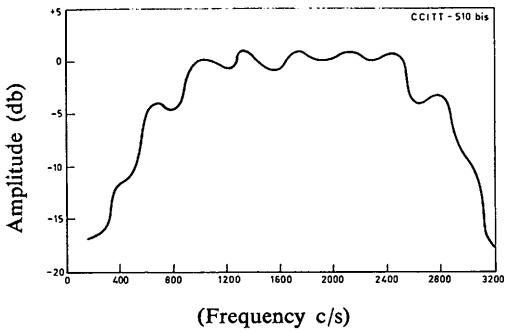


FIGURE 11. — **Frequency response of a combined loaded cable and carrier quality line**

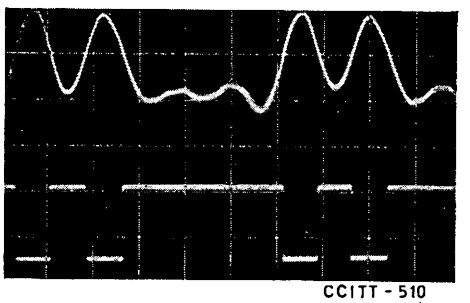
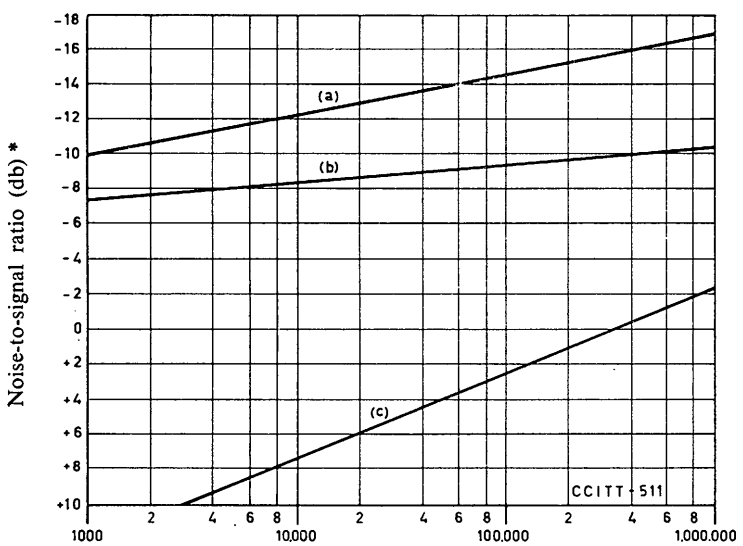


FIGURE 12. — **1800-baud data signal received over a line having the frequency response shown in Figure 11**
(Scale : 600 µs/div ; no phase equalization used)



(a) = Random noise without equalization
(b) = Random noise with equalization
(c) = Impulse noise with equalization

* Zero db noise-to-signal ratio for white noise means a flat noise-to-signal ratio on the 3 kc/s band, whereas zero db noise-to-signal ratio for impulse noise represents normal operating conditions under the circumstances prevailing during the recording of the noise tape.

FIGURE 13. — **Reliability tests at 2400 bauds**

3. *Reliability tests*

Figure 13 shows the results of laboratory tests performed under white noise and impulse noise conditions at 2400 bauds with the test set-up shown in Figure 14. If one compares these with results obtained for a similar phase reversal system, one can see that the field experience gained from reliability tests with the phase reversal system can be safely applied to the system described in this Supplement. However, extensive field experiments and comparative tests with other systems are necessary for a final evaluation of the described system.

4. *Dynamic range*

The experimental system is capable of accepting line signals having amplitude variations of as much as 40 db.

E. Carrier frequency considerations

Not only is it necessary to have a set of standard speeds for a universal data transmission service, but it is also mandatory that the line signal be standardized. In order to place the transmitted information band within the most suitable frequency spectrum, the following carrier frequencies for a phase reversal system could be considered : an 1800-c/s carrier for 2400-baud operation, a 1500-c/s carrier for 1200- and 1800-baud operation, and a 1200-cycle carrier for 600-baud operation. The 1800-cycle carrier places the minimum required transmission band for a 2400-baud service between 600 and 3000 c/s, which seems to be a reasonable requirement for private lines. A 1500-c/s carrier for 1200-baud operation would put the frequency spectrum at a near optimum location for the great majority of switched network links. A 600-baud service might be achieved over heavily loaded cables and half-channels by using a carrier frequency of 1200 c/s with a minimum transmission band between 900 and 1500 c/s. These carrier frequencies seem reasonable on the basis of present knowledge but might not be the optimum choice. A 1200-baud service with a 1000-c/s carrier might prove to be feasible under all line conditions.

Supplement No. 47

I.B.M. (UNITED STATES OF AMERICA) — *(Contribution COM Sp. A/No. 86 — September 1963)*

A VESTIGIAL SIDEBAND PHASE REVERSAL MODEM

I. *Introduction*

The experimental system described below has some basic features which are generally recognized to be desirable : it is basically simple, reliable and efficient. This basic modem allows synchronous and asynchronous operation and permits the addition of a variety of clocking circuits and an increase of speed by the addition of an extra channel if required. In its simplest form (binary

modulation), it might operate at 1000 bauds over almost any telephone line using compromise equalization, and at up to 5000 bauds if very good voice lines are available and variable equalization is used.

Experimental operation at up to 8000 bauds has been achieved over high-quality carrier channels by encoding the data into four levels before modulation—i.e., by using an amplitude-modulated phase reversal system.

II. *Vestigial sideband operation and carrier recovery*

Double-sideband (DSB) data transmission systems employing phase reversal modulation and synchronous detection can provide near-optimum noise rejection with simple means.

Maximum reliability as it relates to the modulation and detection system itself thus can be achieved and, with the addition of error detection and correction schemes, the reliability required for communications-based computer systems can be provided.

Since, in double-sideband transmission, the same information is carried in both of the two sidebands, the transmission speed or efficiency could be almost doubled by single-sideband or vestigial sideband (VSB) transmission.

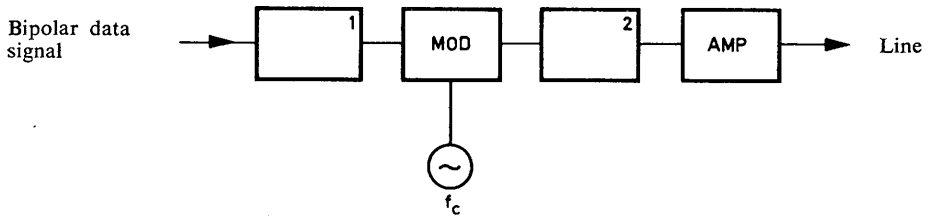
The use of VSB for data transmission using synchronous detection to eliminate quadrature distortion was discussed by Nyquist [1] in 1928. The noise advantage of suppressed carrier transmission or phase reversal modulation has also been recognized for some time, and indeed, has been utilized quite successfully in DSB data systems. However, suppressed carrier VSB transmission has not been utilized owing to the lack of a suitable means of recovering the synchronous detection carrier from the received data signal. A solution to this problem has been found, making practical the system described in this paper.

The VSB phase reversal system (shown in block diagram in Figure 1) is identical to a DSB phase reversal system, except for the addition of the VSB and the complementary VSB filters whose amplitude responses are shown in Figure 2. The transmitted signal spectra for DSB and VSB are illustrated in Figure 3. If the signal is sent on the line as DSB, at least two components symmetrically spaced above and below the carrier are always present, assuming that the data signal does not rest at zero for any appreciable length of time (e.g., it may be bipolar non-return-to-zero (NRZ) swinging between + V and −V). The carrier can be retrieved by beating the sidebands together. This generates components at twice the carrier frequency which can be recovered by filtering. Division by two, as shown in the block diagram of Figure 1b, results in the desired carrier. As an illustration, if the baseband component is f_m, the sidebands are at $f_c + f_m$ and $f_c - f_m$. After passing through a channel with a frequency translation a, the sidebands are $f_c \pm f_m + a$. The square law detector (in practice, a full-wave rectifier) output contains components at $2f_c + 2a$ which, when divided by two, gives the desired carrier at $f_c + a$.

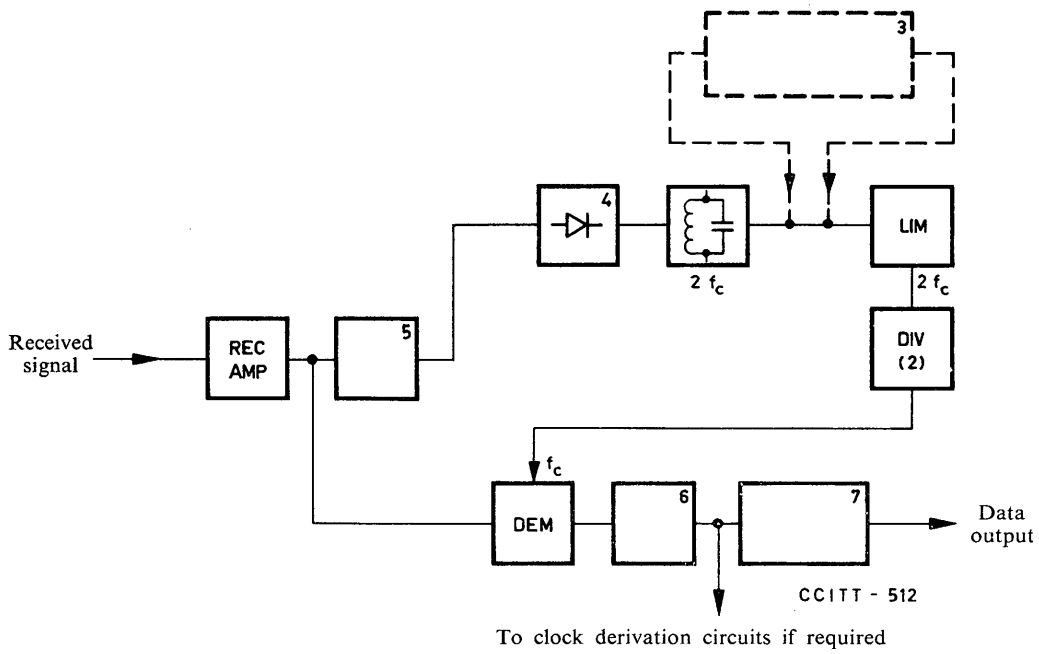
In cases where the data signal contains low-frequency components (including d.c.) which generate pairs of sidebands within the passband of the VSB filter, this basic carrier recovery scheme can be used, with the addition of the complementary VSB filter to make the spectrum

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[1] NYQUIST, H.: "Certain Topics in Telegraph Transmission Theory", *A.I.E.E. Transactions*, Vol. 42, April 1928.



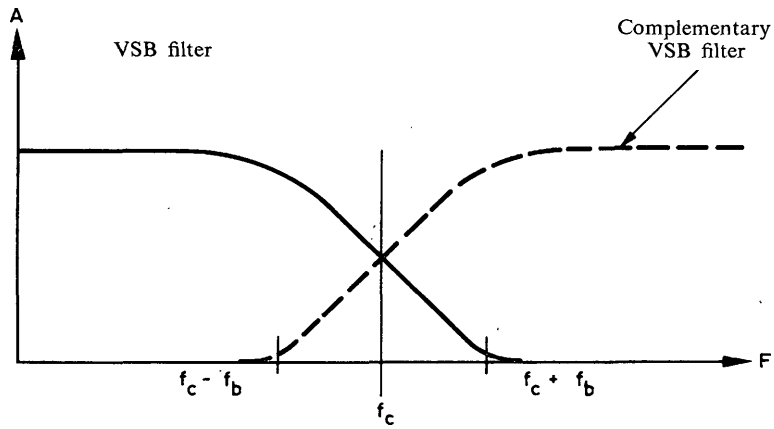
(a) *Transmitter*



(b) *Receiver*

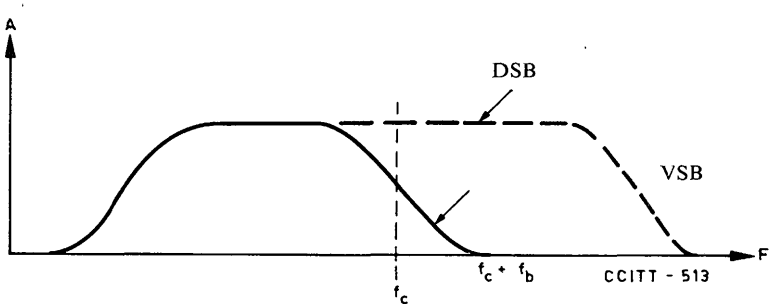
1. LP I
2. VSB
3. Phase-locked oscillator
4. Full-wave rectifier
5. Complementary VSB filter
6. LP II
7. Squaring circuits

FIGURE 1. — **Vestigial sideband (VSB) data transmission system**
(f_c = Carrier frequency)



(f_b = baud-rate frequency per channel
f_c = carrier frequency)

FIGURE 2. — **VSB and complementary VSB filtering**



(f_b = baud-rate frequency per channel
f_c = carrier frequency)

FIGURE 3. — **DSB and VSB line signal spectrum**

at the input to the full-wave rectifier be symmetric about the carrier frequency and thus eliminate undesirable phase modulation of the carrier. However, this type of carrier recovery will fail in cases where components are not present in both sidebands. For example, in Figure 3, if the lowest baseband component is sufficiently greater than f_b, only one sideband is transmitted and carrier recovery failure occurs.

One solution to the problem is to transmit two pilot tones outside the data spectrum with their frequencies chosen so that the carrier can be derived from them by a process of mixing, multiplication, and division. This approach is feasible, but relatively complex to implement. The two pilot signals create difficult filtering problems; moreover, the required frequency space makes the vestigial sidebands system less efficient in spectrum utilization than is theoretically possible.

Carrier recovery in single-sideband systems presents a different problem. The d.c. components of the baseband signal are eliminated at the transmitter by means of a high-pass filter and restored at the receiver by d.c. restoration circuits [1]. A single-sideband modulation system is then formed by filtering the double-sideband suppressed carrier signals. The data cause no modulation products at the carrier frequency, and allow an injected carrier to be transmitted without interference. Thus, the synchronous detection carrier can be recovered at the receiver by a linear filtering process. This system requires an accurate automatic gain control (AGC) for d.c. restoration. The major limitation, however, is that the elimination of the d.c. component in the data signal causes a 6-db increase in the peak line signal. Consider an NRZ pattern with a long string of "0's" followed by a "1". If the baseband signal were d.c. coupled, the input to the modulator would swing between $+$ V and $-$V. However, when the d.c. component is removed, the signal shifts so that in this pattern it swings between approximately 0 and 2 V. Since the envelope of the carrier is proportional to the baseband signal, the peak line signal is doubled; thus, if the peak voltage determines the transmission level, the average transmitted power must be reduced.

In the VSB system described here, the carrier recovery problem is solved by operating on the baseband signal at the transmitter in such a way as to guarantee the presence of low-frequency (including d.c.) components in all data patterns. Coding restrictions can accomplish this, but are costly and reduce the effective bit rate. Several other methods of guaranteeing low-frequency components in the data have been investigated, of which the simplest and apparently the best technique is to add a small d.c. component to the data signal (or to achieve the same effect by adding a small carrier to the modulated signal).

It would seem that the average d.c. level of a repetitive NRZ pattern might be anywhere between its positive and negative extremes, and that, to make sure the injected d.c. level is not cancelled out, it would have to be larger than the maximum signal. However, careful analysis has shown that the patterns for which the carrier fails due to lack of low-frequency components can take on only a relatively small number of average d.c. values. For example, if only the repetitive patterns of 5 bits or less fall outside the VSB filter, the possible d.c. averages are 0, $\pm$ 1/5, $\pm$ 1/3, $\pm$ 1/2, $\pm$ 3/5, and $\pm$ 1 volt, assuming the signal swings between $+$ 1 and $-$1 volt. Therefore, a d.c. signal of, say 1/10 volt can be added to the data to avoid any d.c. cancellation in these patterns: a carrier can then be recovered for all patterns. The analysis above concerns a bit-

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[1] BECKER, F. K., DAVEY, J. R., and SALTZBERG, B. R.: "An AM Vestigial Sideband Data Transmission Set Using Synchronous Detection for Serial Transmission Up to 3000 Bits per Second", paper presented at the Fall General Meeting of the A.I.E.E., October 15-20, 1961.

synchronous system, i.e., one in which the data are synchronized with a continuous clock. However, analysis indicates that the carrier retrieval will also operate successfully on start-stop transmission using the standard teletype code, for example; therefore, synchronous transmission by character is possible.

The level of the injected d.c. must be controlled rather precisely over long periods of time. It has been found that a conventional active switching balanced modulator achieves quite satisfactory results. The more classical suppressed carrier linear ring modulator, which suffers from poor d.c. stability, was not found to be satisfactory.

On the basis of the authors' limited experience with the experimental system, it seems reasonable that a speed of 2000 bauds could be achieved on the majority of lines of the switched network with suitable compromise phase and amplitude equalization. A reasonable choice of carrier frequency seems to be around 2000 cycles per second. For heavily loaded cables and other severely band-limited circuits, a carrier frequency of around 1500 c/s seems to be indicated, and a speed of 1000 bauds could be expected over almost any kind of voice line.

On private lines, where good phase and amplitude line equalization can be provided, speed can be increased to 3000-5000 bauds, depending on the quality of the available line. A reasonable carrier frequency for 3000-baud operation seems to be 2500 c/s. For 5000-baud operation, the carrier frequency would have to be increased to 3000 c/s and a line permitting the increased frequency spectrum would be required.

III. *Four-level system*

In the switched network one has no control over the noise conditions of lines; therefore, the most reliable modulation and detection system achievable with simple means and reasonable efficiency should be chosen. On the other hand, in private line systems there is usually time to study the route and to control, to a certain extent, the noise conditions and quality of lines. Therefore, the noise conditions and the available bandwidths in private line systems as a rule are considerably better than in switched networks, and transmission efficiency can be considerably increased. A signal-to-noise (S/N) ratio of at least 30 db can be reasonably expected. The available bandwidths might extend to the range of 2800 to 3400 cycles, depending on the lines which can be made available.

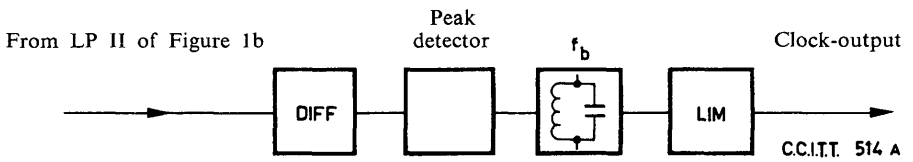
One way to take advantage of improved line conditions and to provide higher transmission efficiency with simple means is to encode the binary data into a four-level waveform. For example, two binary channels may be transmitted simultaneously by encoding the information of one channel into the polarity of the data signal fed into the low-pass filter of the transmitter of Figure 1a, and the information of the second channel into the amplitude of the data signal. This approach is not novel [1] but deserves further investigation in order to explore the practical speed limitations of available transmission facilities, using the previously described vestigial sideband modulation techniques. There would be no changes required in the modulator itself, except to reduce the d.c. injection level by a factor of three. This is necessary because the four-level waveform has more possible average d.c. levels than a binary waveform.

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[1] GROFF, W. M., and POWERS, R. C.: "A High Speed Synchronous Digital Data Transmission Modem", *Proceedings of the MIL-E-CON*, June 1961.

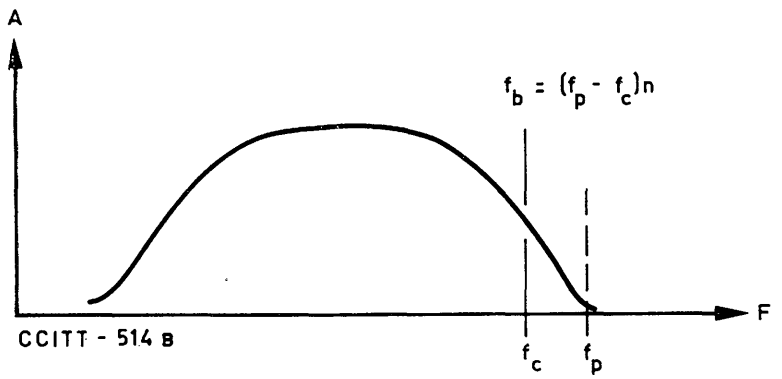
At the output of the low-pass filter (LP II of the data receiver in Figure 1b), the polarity and amplitude of the received data signal would be sampled at each bit time. This way one could derive the two binary output channels at the receiver. Additional circuits are required at the receiving end for sampling the information. For a complete quaternary system, one would need a data clock in the receiver for sampling as well as for automatic gain control of the amplitude channel (both discussed in the following sections). In addition, quaternary operation requires lines with well-equalized phase and amplitude responses.

In order to establish optimum threshold levels, the receiver must be designed to allow for the amplitude and polarity of the injected d.c. (although it is a small quantity). A starting procedure is thus necessary to establish proper carrier phase (of the two possible phases in the frequency-dividing circuits) so as to obtain the desired polarity of the demodulator output.



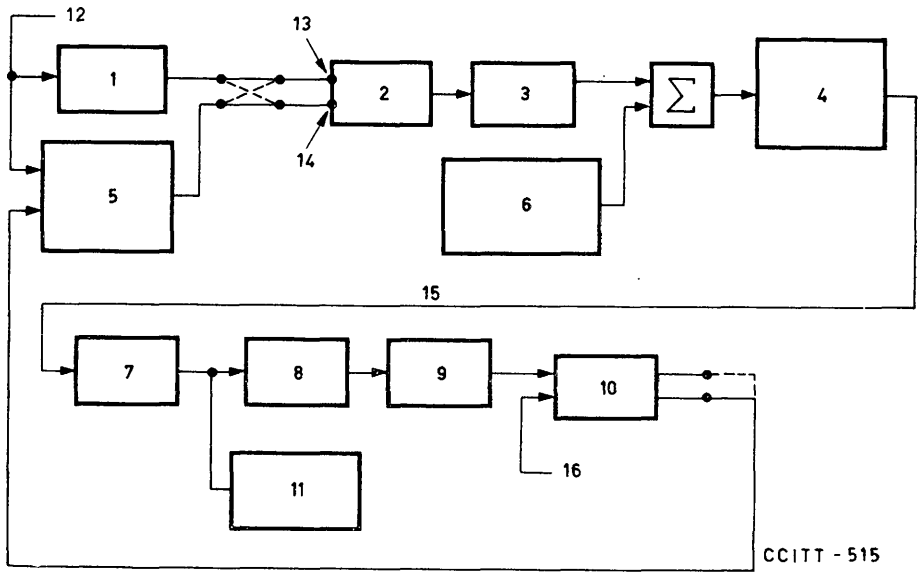
(f_b = baud-rate frequency per channel)

FIGURE 4. — **Clock derivation circuits**



(f_b = baud-rate frequency per channel
f_c = carrier frequency
f_p = pilot frequency)

FIGURE 5. — **Clock derivation from pilot and carrier signals**



1. Data generator
2. Four-level encoder
3. Transmitter
4. Carrier channel bank (3300 c/s bandwidth)
5. Data generator and error detector
6. Random noise generator-General Radio Model 1390B
7. Line equalizer
8. Receiver
9. Level detectors
10. Decoder
11. Voltmeter (H/P Model 400 D)
12. Clock source
13. Amplitude channel
14. Phase channel
15. Line signal
16. Clock source

FIGURE 6. — **Experimental set-up for tests of reliability of four-level modem in the presence of random noise**

Modem noise test results
(clock patched in all cases)

	S/N ratio in db for 10^{-5} error rate					
	Four-level [1]				Binary	
	6000 bauds		8000 bauds		3000 bauds	4000 bauds
	Phase channel	Amplitude channel	Phase channel	Amplitude channel		
Carrier patched	19	20	20	24	13	14
Carrier recovered	22	23	27	30		

[1] Phase and amplitude channels of a four-level system measured separately.

From limited experience with the experimental system and the statistical information on available private lines, it seems reasonable to expect a speed range for the quaternary system from 4000 to 8000 bauds, depending on the quality of the lines. A 4800-baud experimental system which could serve as an example of bandwidth requirements had a data signal spectrum covering the frequency range from 500 to 3000 c/s at the 20-db points, with a carrier frequency of 2400 c/s. A carrier frequency of 3000 c/s was chosen for 8000-baud operation, with a frequency spectrum extending up to 3400 c/s.

The reduced reliability of the four-level system as compared with the basic two-phase system is discussed in section VI.

IV. *Clocking*

The selection of a clocking method for a VSB phase reversal transmission system depends on the mode of operation desired. Several possibilities under different operating conditions are discussed below.

A. *Start-stop operation*

If the basic VSB system is used on a start-stop basis, the classical clocking methods used in low-speed telegraph and data transmission systems can be used. A Hartley oscillator or similar devices may be started with the start bit of the received character and can provide the sampling time for the bits of information of the received character.

B. *Synchronous operation*

1. *Full-wave rectification*

In synchronous transmission, a continuously operating clock must be derived from the incoming data signal. This can be done in several ways. In binary transmission, slight code restrictions or fixed-ratio codes, such as the IBM 4-out-of-8 code, can provide at least one transition for each character transmitted. The sampling time can be derived from the demodulated signal by full-wave rectification (after LP II in Figure 1b), which provides sufficient energy for pulsing of a tuned circuit at the bit rate frequency. The output of this tuned circuit is normally fed into limiters which provide clocking information in the zero crossings of the output square-wave signal.

2. *Maximum slope detection*

Since the time at which a band-limited multilevel signal crosses a given threshold depends on which two levels are involved in the transition, the time jitter in such a signal (Figure 8) is considerably larger than the time jitter in the zero crossings of a binary signal (Figure 7). However, the time at which maximum slope occurs is independent of type of transition (except for interbit distortion). Thus, in multilevel transmission, a clock retrieval scheme based upon differentiation of the received data waveform (see Figure 4) is preferred.

3. *Amplitude modulation*

If at least one transition or pulse per character cannot be guaranteed, the clocking information for synchronous transmission can be transmitted via an amplitude channel similar to that discussed in section III. The clocking information is sent by a redundant signal consisting of a continuous repetition of alternate 1's and 0's from which the bit sampling time can be derived. Since narrow

band circuits are used in the clock retrieval, this channel can tolerate much more noise than the data signal. The clocking channel level can, therefore, be considerably (in the order of 20 db) lower than that of the data. The interference and channel loading caused by clocking information transmission would therefore be negligible. This approach can be extended to a multilevel system. For example, if transitions cannot be guaranteed in either the phase or amplitude channel, a low-level clock signal or tone can be superimposed over the four-level data stream at the input of the modulator.

4. *Pilot transmission*

A convenient method for deriving a clock signal which is independent of the data signal can be provided by transmitting a pilot tone outside of the data signal spectrum as shown in Figure 5. The frequency difference between the tone and the carrier is a submultiple, n, of the baud rate (f_b) per channel.

The clocking information is derived at the receiving end from the difference frequency between the pilot frequency, f_p, and the reconstructed synchronous detection carrier frequency, f_c, followed by a process of multiplication by n, giving

$$f_b = (f_p - f_c)\, n.$$

Frequency spectrum shifts of the line do not affect the clock signal since both the carrier and pilot frequency would shift by the same amount and the difference would remain constant. As an example, for 4800-baud four-level (two-channel) operation, a carrier of 2400 c/s and an n of 4 could be chosen, resulting in a pilot tone, f_p, at 3000 c/s. This pilot signal can also very conveniently be used for automatic gain control, which is of special interest in a quaternary system and is discussed in the next section.

V. *Automatic gain or threshold control*

The described quaternary data transmission system must have means for automatic gain control since one of its channels carries the information in the amplitude of the demodulated received signal. Since the amplitude channel of the system generally has no constraint in its information content, the magnitude of the transmitted signal is not constant as it is in binary phase reversal modulation. Therefore, the automatic gain control must be independent of the data signal. A pilot frequency such as described above can very conveniently provide such control, if the bandwidths and speed permit. In the experimental system, it was found convenient to use the pilot for detection threshold and volume control.

If a pilot signal cannot be provided for either clocking or threshold and volume control of a quaternary system, the following method can be used:

At the beginning of every block of information transmitted, a sample sine-wave signal is transmitted, in order to permit the threshold detection and volume control circuits to set the proper levels, which remain fixed during the transmission of the entire block of information. Since the amplitude variations of the line signal are usually rather slow and blocks of information are usually in the order of 100 to 1000 characters, this method is suitable for satisfactory operation.

VI. *Reliability and functional laboratory tests*

A. *Reliability tests*

Reliability tests performed in presence of random noise under laboratory conditions do not give a full measure of the reliability to be expected in operation over actual telephone lines. Nevertheless, such tests may be indicative of relative reliability as compared with other systems tested under similar conditions. Figure 6 shows the block diagram of the test set-up used to obtain the results listed in the table. In order to evaluate the modem performance in presence of white noise alone and because of the variety of clocking circuits available, the transmitter clock, rather than the one recovered from the received signal, was used in the tests. The S/N ratio was measured with a Hewlett-Packard Model 400 D voltmeter with appropriate corrections to give true RMS values. The data source was a 511-bit quasi-random character generator which was also used for error detection.

The carrier recovery was performed as shown in Figure 1b with a single tuned circuit with a Q of 30. The system performance at 8000 bauds in presence of noise can be improved if a phase-locked oscillator is inserted between the $2f_c$ tuned circuit and the limiter as shown by the dashed lines in Figure 1b. Several decibels of improvement in the S/N ratio in the presence of white noise can be expected. The two sets of figures in the table of results showing the required S/N ratio for a given reliability are obtained with a patched-through carrier and a single tuned circuit recovered carrier, in order to indicate the two extremes of performance that can be expected. Insertion of a phase-locked oscillator increased the S/N ratio by about 2 db at 6000 bauds.

Supplement No. 48

UNITED STATES OF AMERICA — *(Contribution COM Sp. A/No. 31 — October 1961)*

AN AM VESTIGIAL SIDEBAND DATA TRANSMISSION SET USING SYNCHRONOUS DETECTION FOR SERIAL TRANSMISSION UP TO 3000 BITS PER SECOND — BY MESSRS. F. K. BECKER, J. R. DAVEY AND B. R. SALTZBERG

Summary

A data transmission system has been designed which uses vestigial sideband, suppressed carrier, amplitude modulation. Carrier recovery and synchronous detection are performed in the receiver.

The system is asynchronous and will operate at any bit speed up to its maximum. Satisfactory operation has been achieved at 2000 bits per second over the switched telephone network, and at 3000 bits per second over equalized voice band circuits. This data system appears economically favourable when compared with other systems of the same speed capability.

Introduction

Virtually all forms and combinations of forms of modulation and detection have been explored for the transmission of digital data. Vestigial sideband transmission gives spectrum utility. Synchronous detection together with carrier suppression gives ideal signal-to-noise performance. The advantages of combining these methods have been pointed out [1, 2, 3]. The same advantages are also obtainable by using two double sideband channels at quadrature carrier phases which is equivalent to quaternary phase modulation. The use of a vestigial sideband, however, has the added advantage of the simplicity of being a single binary channel. In general such a system has not been used because of the difficulty in establishing the demodulating carrier with vestigial sideband transmission. Vestigial sideband systems have usually used envelope detection together with a modulation index of about 50% to avoid quadrature distortion [4, 5]. This results in a system that has about one-quarter the tolerance to noise obtainable with a suppressed carrier signal of the same peak amplitude. Such systems are also very sensitive to changes in level of the received signal. Also the use of synchronous detection eliminates the quadrature distortion associated with envelope detection. The advantage to be gained by a system of carrier recovery permitting synchronous detection of a suppressed carrier vestigial sideband signal is thus great.

Synchronous detection has been used with double sideband suppressed carrier or phase reversal modulation with various methods of carrier recovery [6, 7]. One method partially suppresses the carrier and places suitable restrictions on the intervals of reversed carrier phase to permit carrier recovery by a narrow band-pass filter or phase locked oscillator. Usually it has been found more attractive to recover the carrier by modulating the received signal by the detected baseband wave or by a sequence of frequency doubling and halving. These methods, however, lead to ambiguity of the carrier phase and in turn the polarity of the data wave. Encoding the data into signal transitions solves this problem but adds complexity and requires the data to be transmitted synchronously. In a vestigial sideband system where there are always low modulating frequencies placing pairs of side frequencies in the vestigial cut-off region, carrier may be recovered from this double sideband portion of the signal. This method has been used for television signals [8], but is not attractive for data because of the restrictions on the data sequences required to insure low frequency component and also because of the carrier phase ambiguity mentioned above. With carrier suppression and with a single sideband, as for the higher modulating frequencies in a vestigial sideband system, no carrier frequency information exists in the received signal.

This paper discusses another approach or further modification which approximates the desirable features of the optimum system. A low level reference carrier is added at the transmitter. This signal is provided with spectrum isolation by the removal of the low frequency components of the data wave. Suppressed carrier amplitude modulation and vestigial sideband transmission convey the altered data signal to the receiver. A phase-locked oscillator extracts the reference carrier for the synchronous detector. After suitable filtering, the discarded low frequency components of the data wave are reinserted by a regenerative feedback circuit which utilizes *a priori* knowledge that the data wave was originally binary. This regenerative action performs a similar function to the keyed-clamping circuits of television systems.

Circuit description

Block schematics of data communication terminal equipment which will handle up to 2000 bits/s are shown in Figures 1 and 2.

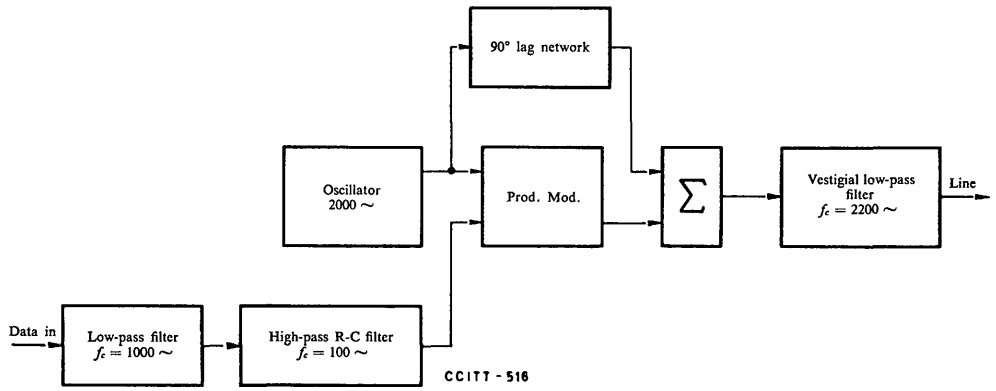


FIGURE 1. — **Transmitter block diagram**

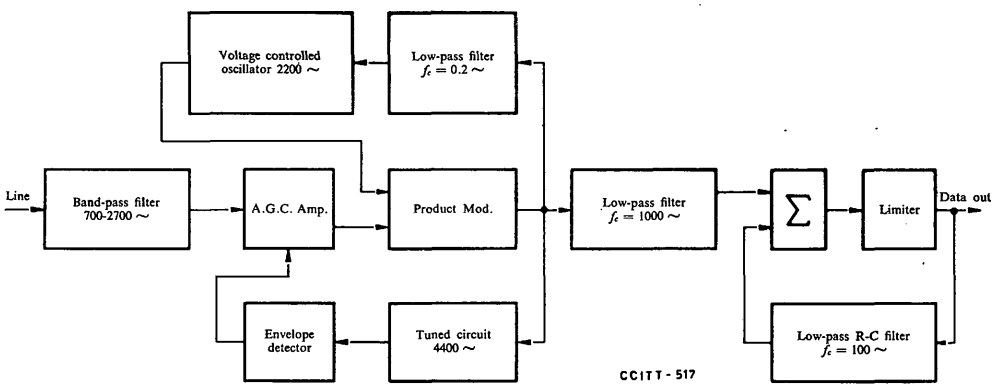


FIGURE 2. — **Receiver block diagram**

Any serial data input signal which meets E.I.A. standards [9] is accepted by the transmitter. The amplitude is standardized in a limiter and the signal is then passed through a low-pass filter. The filter must provide proper shaping as well as elimination of high frequency components which would cause foldover upon modulation.

The signal is next passed through a simple RC high-pass network whose 3 db point is at 100 c/s. The output of this network is the input to the modulator.

A Hartley oscillator generates the 2200-c/s carrier. A carrier frequency square wave drives the product modulator. The output of the modulator consists of a double sideband, suppressed carrier, AM signal. The spectrum in the vicinity of the carrier is empty because of the high-pass filtering of the data signal. Sidebands about odd multiples of the carrier frequency are also present at this point.

A carrier frequency tone in quadrature phase to the modulating carrier is added. The amplitude of this tone is such that its power on the line is 7 db below the average signal power.

A vestigial low-pass filter eliminates most of the upper sideband and all higher modulation products. The output of this filter is amplified and coupled to the line.

The input to the receiver is passed through a band-pass filter which limits the noise bandwidth. The band-limited signal is amplified and coupled to a product modulator. The modulator performs three distinct functions, as described in the following paragraphs.

First, the modulator serves as part of a phase-locked servo loop [10]. It is driven by a free-running multivibrator whose natural frequency is dependent on its base supply potential. If the phase of this local oscillator is not in quadrature with the incoming carrier frequency pilot tone, the output of the product modulator includes a d.c. error signal. This signal is strongly filtered and adjusts the frequency of the multivibrator. Because of the guard space about the carrier frequency, very little of the signal other than the pilot tone has any effect on the phase on the local oscillator. For the worst data sequences, less than $\pm$ 15° jitter results. Typically this jitter is approximately $\pm$ 3°. The phase error is less than 6° in the presence of 20 c/s frequency offset. The loop will lock in over a $\pm$ 45-c/s range.

Second, the modulator output has a component whose frequency is twice that of the carrier. Its amplitude is proportional only to that of the pilot tone input to the modulator. This component is selected by a tuned circuit, rectified and filtered. The resultant signal is used to control the gain of the amplifier. This automatic gain control system is sensitive only to the level of the carrier frequency pilot tone and not to the data sequence. The variations in the amplifier output level are held to less than 1 db in the presence of 30-db input level variations.

Third, demodulation of the signal is accomplished by the same product modulator. Since the pilot tone is in quadrature with the original modulating carrier, and the local carrier is in quadrature with the pilot tone, the local carrier therefore has the correct phase for demodulating the signal. No ambiguity is present since only one of the two quadrature states results in stable equilibrium of the phase-locked servo loop.

Post-detection filtering is performed by a low-pass filter. This provides both noise reduction and elimination of higher frequency modulation products. In particular, a lower sideband of the double carrier frequency is present and must be removed.

The low frequency data components are restored by a quantized regenerative feedback circuit. The limiter output is $\pm$ 6 volts, according to whether its input is positive or negative. The output is passed through an RC low-pass network which is the exact complement of the high-pass network in the transmitters; that is, the step responses of the two networks add up to a step function. For simple RC networks, it is sufficient that their time constants be equal.

The output of the RC network is added to the demodulated data, and the sum is presented to the limiter. The two signals must have the correct relative amplitude. It is for this reason that the output level of the AGC amplifier is held to an exact value.

The limiter does not switch states until the data signal crosses the threshold value. Because of the data shaping filters in both the transmitter and the receiver, this occurs about one-half a bit interval after the start of the transition. As a result, the output step and the feedback transient are delayed. This effect would lead to some inter-symbol interference if not corrected. Since the feedback transient is a simple exponential, it can be made to appear to have started at an earlier instant by reducing its amplitude by the appropriate factor. It is shown that if appropriate high-frequency shaping is used, then low-frequency data components can be removed and restored as described here, without any inter-symbol interference.

Because of the use of positive feedback, any bias in the limiter tends to be magnified. Another penalty of this scheme is a slight loss of noise margin.

The binary data signals are regenerated in amplitude and the output meets E.I.A. standards [9]. No retiming is employed as a basic part of the terminal. This allows the system to be

used asynchronously and at any bit rate up to its maximum. Retiming may be achieved by any bit synchronization method which operates by averaging data transition times.

The terminals used for transmission of up to 3000 bits per second are identical to those described above except for appropriate frequency scaling of the data filters and a change of carrier frequency to 2500 c/s.

Results

The data terminals have been tested extensively at speeds of 2000 and 3000 bits per second under simulated line conditions [11]. Various types of channel distortion were introduced and Gaussian and impulsive noise added. Distortion of the detected signal was noted and errors were counted.

In all cases a quasi-random data sequence, 63 bits in length, was transmitted repetitively. Error rates were based on runs of 500 000 bits. Both Gaussian and impulsive noise limited to a bandwidth of approximately 3000 c/s.

In the presence of additive Gaussian noise and no other impairment, an error rate of one in 10^4 bits occurs at a signal-to-noise ratio of about 12 db at a bit speed of 2000. If the bit rate is reduced to 1600, the 10^{-4} error rate does not occur until the signal-to-noise ratio is reduced to 9.5 db. The 3000 bits per second system also showed a noise performance of 12 db.

In order to determine the sources of degradation, tests were performed on a system in which no low frequency attenuation of the signal was employed, and in which a perfect carrier was provided for demodulation. An immediate 0.8-db improvement was obtained by eliminating the pilot signal. The additional improvement measured was only 0.5 db. This indicates that the penalty suffered for using the carrier recovery method described previously is very small.

No serious degradation of performance was noted when carrier frequency offsets within the lock-in range of the phase-locked loop were introduced.

The effect of channel attenuation slope characteristics on the performance of the system is also small. At 2000 bits per second, poor results were not obtained until the difference in loss between 1000 and 2600 c/s reached 14 db. This limiting characteristic is worse than that of 90% of the switched network [12]. The introduction of a slight fixed equalization should allow good operation in the presence of virtually any amplitude characteristic which may be encountered. The 3000-bits-per-second system operates in the presence of slopes of up to 12 db.

The receiving terminal is sensitive to sudden decreases in input level. There is a high probability of error when losses greater than about 3 db are rapidly introduced. Sudden increases in level up to 15 db present no problem.

The most severe impairments are those due to delay distortion. It is the delay characteristics of the switched telephone network that limits the bit rate of the system. The carrier frequency was chosen so as to place the signal spectrum in the most favourable region of the expected channel delay characteristics. Care was taken to avoid introducing additional phase distortion in the terminals.

Figure 3 illustrates three limiting cases of delay distortion through which the system will operate at 2000 bits per second. The error rates for these delay characteristics are shown in Figure 4. In each of these cases, the performance with additive noise is degraded by 10 to 15 db. A slight increase in delay prevents any reception. The limiting one-sided and symmetrical curves were determined. The carrier frequency and the carrier frequency minus half the bit rate are shown as dotted lines. The principal portion of the signal spectrum lies between these two frequencies.

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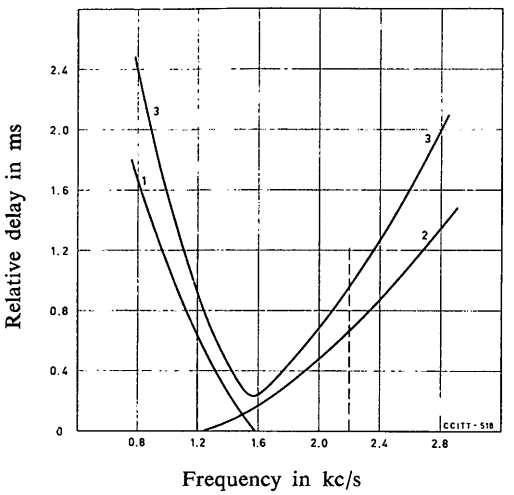
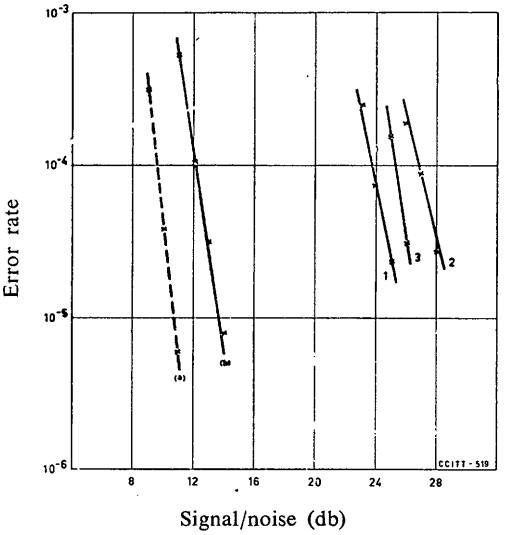


FIGURE 3. — **Maximum delay characteristics for transmission of 2000 bits per second**



1 - 2 - 3 = Delay distortion
(a) = 1600 bauds/bps
No impairment
(b) = No impairment

FIGURE 4. — **Error rates at 2000 bits per second**

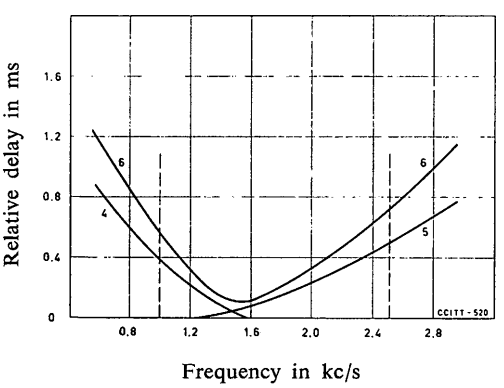
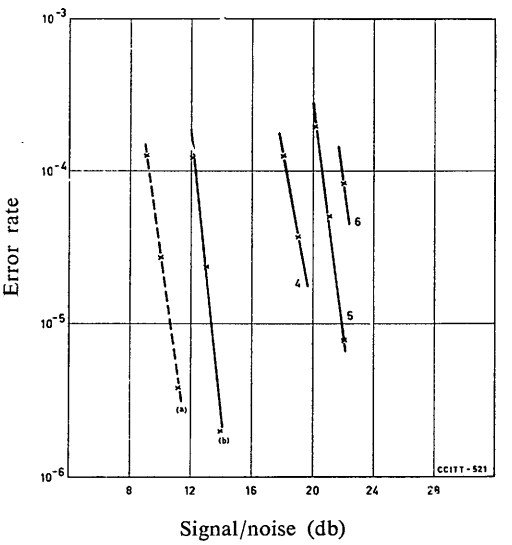


FIGURE 6. — **Maximum delay characteristics for transmission of 3000 bits per second**



(a) = 2600 bauds/bps
No impairment
(b) = No impairment
4 - 5 - 6 = Delay distortion

FIGURE 7. — **Error rates at 3000 bits per second**

The limiting delay characteristics for transmission of 3000 bits per second are shown in Figure 6, and the corresponding error rates in Figure 7. An approximate generalization which may be drawn from the two sets of curves is that the system will tolerate up to 1.3 bit lengths of delay distortion at the carrier frequency and at the carrier frequency minus half the bit rate for typical delay shapes.

Approximately 90% of all long-haul telephone circuits have less delay distortion than that shown in Figure 3 [12]. With a suitable fixed compromise delay equalization, the 2000-bits-per-second system may be expected to operate satisfactorily over almost any circuit which may be encountered in the switched telephone network.

An average long-haul circuit has a delay characteristic which is slightly worse than the limiting curves of Figure 6 [12]. In addition, the distribution of delay characteristics is too wide to permit transmission of 3000 bits per second using a single compromise equalization. However, private lines may be equalized without much difficulty to allow good transmission at this bit speed.

Tests were run in the presence of impulsive noise, both artificially generated and recorded from a noisy telephone line. The performance of the system under these conditions was comparable to that of other serial data transmission systems.

The use of vestigial sideband transmission and synchronous detection provides a relatively inexpensive means of providing high speed serial data transmission over voice-band circuits. The transmitting and receiving terminals described here have been implemented using only 18 transistors.

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Supplement No. 49

BELL TELEPHONE LABORATORIES — *(Contribution COM Sp. A/No. 32 — October 1961)*

PM DATA SETS FOR SERIAL TRANSMISSION AT 2000 AND 2400 BITS PER SECOND

1. *Introduction*

The 201A and 201B transmitter-receivers transmit and receive binary data at a fixed rate over switched voice-frequency telephone circuits or leased lines. Phase modulation and differential phase detection are employed. This data transmission system achieves a one-bit-per-second data rate per cycle of bandwidth due to a unique method of line signal generation which differs from those used in other PM systems. Continuous timing synchronization recovery at the receiver is made possible by a method proposed by M. A. Logan. The operation of the timing recovery circuits is not affected by message content or line frequency shift.

The transmission rate is fixed at 2000 or 2400 bits per second and is determined by a crystal oscillator in the transmitter. The carrier frequency for the 2000 bits per second set (201A) is 1750 c/s and the principal line signal spectrum covers the range 750 c/s to 2750 c/s. The carrier frequency for the 2400 bps set (201B) is 1800 c/s and the spectrum covers the range 600 c/s to 3000 c/s.

The set may be connected for two-wire or four-wire operation. In two-wire operation the receiver is normally on and the transmitter is normally off. In four-wire operation both transmitter and receiver are always on.

To transmit, a customer must change the polarity on the "request to send" wire from negative to positive. When the set is ready to transmit data, it will return a "clear to send" signal to the customer. When carrier is present at the receiver, a "carrier on" signal and timing signals are supplied to the customer.

The customer must provide beginning and end of message codes and any desired error detecting or correcting means.

A general description of operation is given in the next section. Circuit descriptions follow and in the final section we give test results. Both laboratory data and some results from private line and switched network tests are given.

2. *General description of operation*

Transmitter

The data are encoded as a succession of signal element carrier phase shifts which are restricted to 1, 3, 5, or 7 times $\frac{\pi}{4}$ radians (45°) with respect to the previous signal element carrier phase [1]. Serial data, as received by the transmitter, are examined in pairs. Each of the four possible combinations of binary digits (called dibits), 00, 01, 11 and 10 causes the generation of a signal element embodying one of the four possible carrier phase shifts.

[1] This method of encoding was suggested by H. A. Henning.

Timing at the bit rate is necessary for customer use. However, timing at half the bit rate is fundamental to set operation because, as mentioned above, the data sets encode on a two-bit-per-signal element basis. Table I gives the bit and dibit rates for the 201A and the 201B.

TABLE I

	201A	*201B*
Bit rate	2000 bits/s	2400 bits/s
Dibit or signalling rate	1000 symbols/s	1200 symbols/s
Carrier frequency — ω_C	1750 c/s	1800 c/s
Envelope frequency — ω_E	500 c/s	600 c/s

To conserve bandwidth, a limiting factor for operation over telephone lines, the data set line signal is composed of overlapping signal elements. The envelope rate is half the dibit rate and the corresponding frequency is given as the fourth item in Table I.

Receiver

At the receiver an incoming line signal element is demodulated by modulation with the previous element which is delayed by a period of time corresponding to the duration of a signal element. This delayed element is made available with both 0° and 90° carrier phase shift. See the block diagram in Figure 4. After demodulation the product signal is applied to a low-pass filter which attenuates high frequency components and passes the baseband data pulse. The data pulse is then sampled at peak amplitude.

Pairs of bits are identified simultaneously, but delivered serially to the receiver data output circuit under control of the recovered timing synchronization signal. This timing is supplied to the customer.

Line signal characteristics

Line signal characteristics provide a basis for a more detailed explanation of the demodulation process. The diagram in Figure 1 shows three successive amplitude modulated, phase modulated, carrier pulses in sequence. The envelope period, dibit or signalling period, and carrier frequency are shown. A modified raised cosine pulse envelope is used in an attempt to minimize the required line signal bandwidth and, at the same time, detect baseband pulses with raised cosine shaping. Transmit and receive band-pass filters are not used in this system. Pulse shaping is effected by envelope modulation at the transmitter and post-demodulator filtering at the receiver.

Figure 1 illustrates pulse overlap in the line signal. Note that only the centre portion of each pulse is completely free of interference due to adjacent signal elements. Even though this interference is present in both the line signal and baseband data pulse train, we are able to produce approximate raised cosine baseband pulses with negligible inter-symbol interference by proper selection of line signal envelope and post-demodulator low-pass filter characteristic.

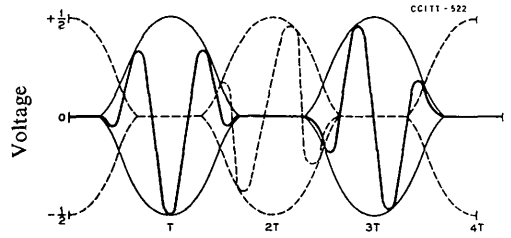


FIGURE 1. — **Line signal characteristics of the 201 B**

The demodulation process

Differential phase detection of the phase-modulated line signal element is described below. An isolated incoming signal element is represented by equation 1 where $A(t)$ is an envelope function and φ_2 is the carrier phase angle.

$$f_2(t) = A(t) \cos (\omega_c t + \varphi_2) \tag{1}$$

The delayed signal elements are given by equations (2) and (3). Equation (2) represents the 90° output of the delay line while equation (3) represents the 0° output.

$$f_{1(90)}(t) = A(t) \cos \left(\omega_c t + \varphi_1 - \frac{\pi}{2}\right) \tag{2}$$

$$f_{1(0)}(t) = A(t) \cos (\omega_c t + \varphi_1) \tag{3}$$

The functions remaining after product demodulating (1) with (2) and (3) and filtering (baseband pulses) are represented by equations (4) and (5). We are interested only in the polarity of these pulses and not in the time functions $B(t)$. Equation (4) is obtained from the product $\{ f_2(t) \times f_{1(90)}(t) \}$ while equation (5) is obtained from the product $\{ f_2(t) \times f_{1(0)}(t) \}$.

$$E_A = B(t) \cos \left(\varphi_1 - \varphi_2 - \frac{\pi}{2}\right) = B(t) \sin (\varphi_1 - \varphi_2) \tag{4}$$

$$E_B = B(t) \cos (\varphi_1 - \varphi_2) \tag{5}$$

Table II gives baseband pulse polarity as a function of the phase angle difference $(\varphi_1 - \varphi_2)$ given in equation (6). The restriction of $(\varphi_1 - \varphi_2)$ to these particular values results in identical

TABLE II

$(\varphi_1 - \varphi_2)$	Polarity of E_A	Polarity of E_B
$\frac{\pi}{4}$	+ ≡ " 0 "	+ ≡ " 0 "
$3\frac{\pi}{4}$	+ ≡ " 0 "	− ≡ " 1 "
$5\frac{\pi}{4}$	− ≡ " 1 "	− ≡ " 1 "
$7\frac{\pi}{4}$	− ≡ " 1 "	+ ≡ " 0 "

demodulated pulse amplitudes and makes synchronization recovery independent of message content.

$$(\varphi_1 - \varphi_2) = n\frac{\pi}{4}, \quad n = 1, 3, 5, 7 \tag{6}$$

The arbitrary but consistent identities in Table II define binary 0's and 1's.

Synchronization recovery

A unique feature of the 201A and 201B is the method of obtaining bit synchronization timing in the receiver.

An elementary, non-rigorous derivation of the recovered timing wave is given below. Without loss of generality we may assume that a full raised cosine envelope is used. Equation (7) represents the received line signal over a period of time, T, extending from the peak of one pulse to the peak of the next.

$$f(t) = \frac{1}{2}(1 + \cos\ \omega_E t)\cos\left[\ \omega_c(t + T) + \varphi_1\ \right] + \frac{1}{2}(1 - \cos\ \omega_E t)\cos\left[\ \omega_c t + \varphi_2\ \right] \tag{7}$$

$$0 < t < T$$

In equation (7) the time origin is the time of peak amplitude of one of the pulses and the phase angle difference, $(\varphi_1 - \varphi_2)$, is the data phase shift discussed above.

The following manipulations are performed on $f(t)$ to recover a timing wave:

1. Suppress terms which are functions only of the carrier frequency ω_c.
2. Product modulate terms which are functions of $(\omega_c - \omega_E)$ by those which are functions of $(\omega_c + \omega_E)$.

Note that by this operation any line frequency shift, which equally shifts all frequencies, disappears when differences are taken. Hence the recovered synchronization frequency is independent of line frequency shift.

3. Again suppress terms which are functions only of ω_c. The resulting expression is given by equation (8).

$$f_T(t) = \frac{1}{8}\cos\ 2\omega_E t\left\{1 - \cos\left[\ (\varphi_1 - \varphi_2) + \omega_c T\ \right]\right\} \tag{8}$$

Noting that: ω_D (the frequency corresponding to the dibit rate) $= 2\omega_E$; $T = \dfrac{2\pi}{\omega_D}$; and that, by equation (6), $(\varphi_1 - \varphi_2) = n\dfrac{\pi}{4}$, we rewrite equation (8) to give equation (9).

$$f_T(t) = \frac{1}{8}\cos\ \omega_D t\left\{1 - \cos\left[n\frac{\pi}{4} + \frac{2\pi\omega_c}{\omega_D}\right]\right\} \quad n = 1, 3, 5, 7 \tag{9}$$

An examination of the amplitude term in brackets in equation (9) shows that the expression is always positive. It is non-zero if ratios of $\dfrac{\omega_c}{\omega_D}$ are chosen such that $\dfrac{\omega_c}{\omega_D}$ does not equal $\dfrac{N}{8}$, where

N is an odd integer[1]. The amplitude of $f_T(t)$ is a function of the data through the dependence on n. However, amplitude limiting is used to eliminate this variation.

3. *Circuit description*

Transmitter

From Table II we can determine the rules for transmitter encoding. If the pair of bits to be encoded is "00", shift the carrier phase of the line signal element in process by $\frac{\pi}{4}$ radians with respect to the previous element carrier phase. The element will then be decoded in the receiver as "00". Similar statements apply to the other dibits and phase shifts. The implied logical conversion of binary dibits to phase shift is given in the following table. Note that the bit pair is ordered and that the assumption is made that a given phase shift can be obtained as a sum of smaller shifts.

TABLE III

Phase shift required

Dibit	$\frac{\pi}{4}$	$\frac{\pi}{2}$	π	Total
A B				
0 0	yes	no	no	$\frac{\pi}{4}$
0 1	yes	yes	no	$3\frac{\pi}{4}$
1 1	yes	no	yes	$5\frac{\pi}{4}$
1 0	yes	yes	yes	$7\frac{\pi}{4}$

A simple circuit capable of generating a phase modulated wave according to the table is given in Figure 2. This circuit illustrates the principle of operation of the data set transmitter. Neglecting the modification necessary to allow the generation of overlapping amplitude modulated signal elements the figure and explanation do, in fact, describe the actual circuit. An integral number of cycles of carrier per dibit interval is assumed in this explanation.

A chain of three binary counters, each with double count inputs, divides the frequency $8\omega_c$ by 8 to derive the carrier frequency ω_c. At intervals of time determined by the dibit rate—the rate at which line signal elements are generated—and at an instance of time when the $8\omega_c$ pulse is not present, a pulse is applied to the phase set lead. This pulse causes a chance of state of binary counter B_1 and in doing so generates a $\frac{\pi}{4}$ radian (45°) phase shift of the output wave. The phase

[1] The values of N used are 14 and 12 for the 201A and the 201B, respectively. This requirement is equivalent to the restriction of the carrier frequency to an integral number of quarter cycles per dibit interval.

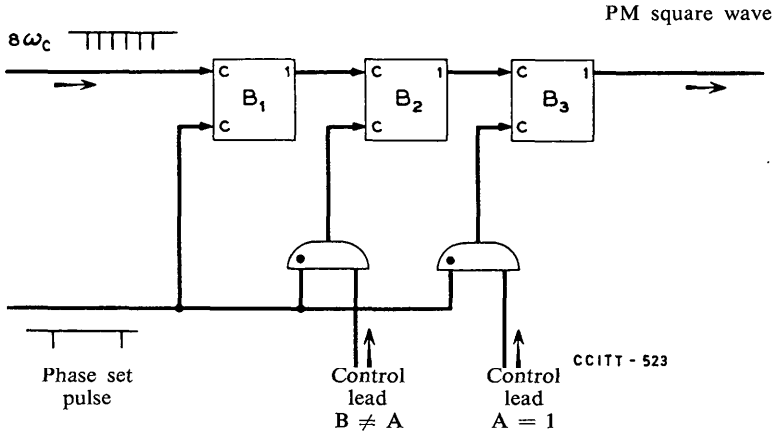
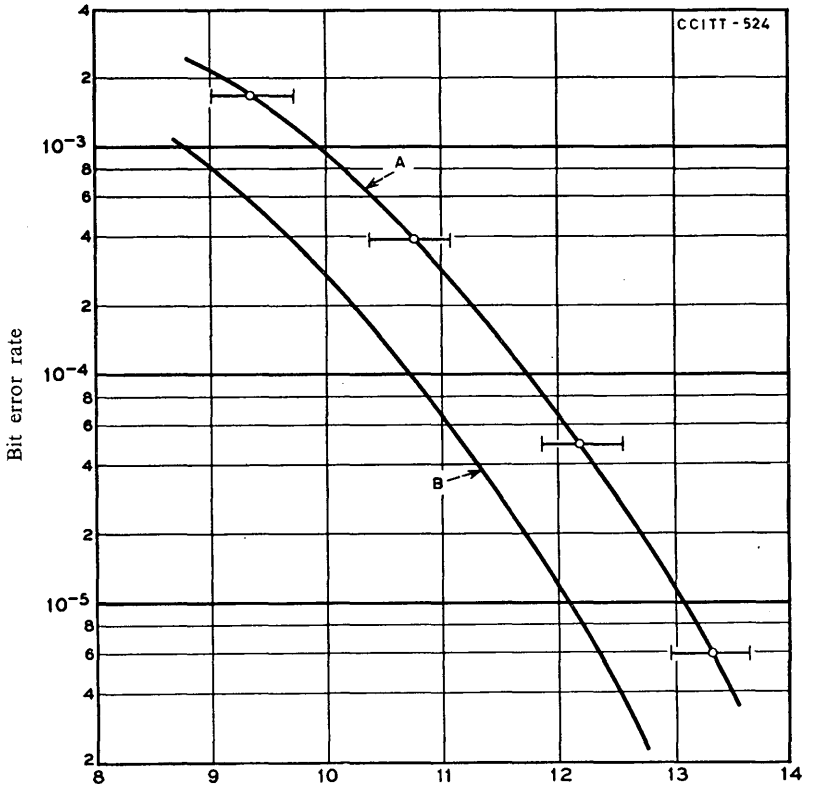


FIGURE 2. — **Simple circuit for generating a phase-modulated wave**



Curve A — 201A and 201B data sets at 2000, 2400 bits per second
Curve B — Theoretical error rate assuming 2.3 db degradation due to differential phase distortion

FIGURE 3. — **Bit error rate as a function of signal-to-noise ratio**

set pulse, after a very short delay to prevent interference, is applied to B_2 or B_3 if the access gate to either is open. The access gates are controlled by the pair of bits to be transmitted. The pulse, if applied to B_2 or B_3, generates a $\frac{\pi}{2}$ (90°) or π (180°) radian phase shift respectively of the output wave.

If the first bit of the pair (A) is a "1" the phase set pulse causes a change of state of binary counter B_3. If the second bit of the pair (B) is unlike the A bit, that is, if the dibit to be encoded is "10" or "01", the phase set pulse causes a change of state of binary counter B_2.

Two separate waveforms are generated and then summed to give the actual line signal. Data set timing circuits include an oscillator, a divider chain, and binary counters with outputs at the dibit and half dibit (or envelope) rates. The phase logic circuit operates at the dibit rate to compute the required carrier phase angle and is similar to that discussed earlier, but here phase logic binary counter states are transferred to slaved binary counter countdown chains. This transfer occurs at the dibit rate and each channel is set alternately. The channel output—a phase-modulated square wave—is then modulated by a modified raised cosine envelope obtained from the envelope low-pass filter. The carrier pulse train shown in Figure 2 provides an illustration. Pulses peaking at $1T$ and $3T$ are generated in channel I, while those peaking at 0, $2T$ and $4T$ are generated in channel II. The carrier phase angle of the pulse peaking at $3T$ was set at $2T$ when the envelope amplitude was a minimum.

Before summing, each channel output is filtered to remove frequency components above the band resulting from the modulation process. In the 201A and 201B this filter cut-off frequency is 3300 c/s.

Receiver

The demodulation process and the synchronization recovery process have been described earlier.

The automatic volume control circuit limits the output signal to a 3-db change for an input signal variation from −50 dbm to −20 dbm. Total harmonic distortion is less than 3% for any input signal level above −50 dbm.

A circuit for carrier detection is provided. Line signal levels above −50 dbm produce an output signal less than 9 ms after line signal is received.

Synchronization recovery circuits

The bit synchronization timing recovery circuits are driven by the received line signal and produce two pulse trains 180° out of phase. One of these pulse trains is used to sample the received baseband data pulse. The other train is used to transfer data in the B data register to the A data register at a time halfway through the dibit cycle. The two pulse trains trigger a monopulser to generate a timing wave at the bit rate.

The line signal is split about the carrier frequency by high and low band-pass filters. These filters are designed so that the difference between the two phase characteristics is essentially constant for frequencies separated by the dibit frequency. The outputs of the two filters operate a modulator.

In principle the band-splitting filters are not essential—practically the filters are necessary to reduce the number and amplitude of undesired frequency components produced by the modulator to amplitudes no higher than the wanted signal—and thereby reduce phase jitter in the recovered timing wave.

4. *Laboratory and field test results*

Laboratory tests

Figure 5 gives the number of bits in error per bit transmitted for both the 201A and 201B as a function of normalized received signal-to-noise ratio (SNR) for band-limited random noise and back-to-back operation. Experimental error limits reflect expected variations between sets and different test equipments. The theoretical error rate curve was obtained from articles by Sunde [1] and Cahn [2]. To the best of our knowledge the theoretical and actual error rates for a given SNR differ as a result of imperfect, i.e. non-ideal, circuit instrumentation in the data sets and not as a result of basic conflict with the assumptions made by Sunde and Cahn.

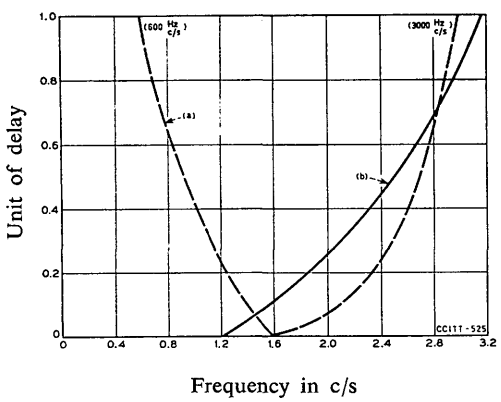


FIGURE 4. — **Differential delay distortion characteristics**

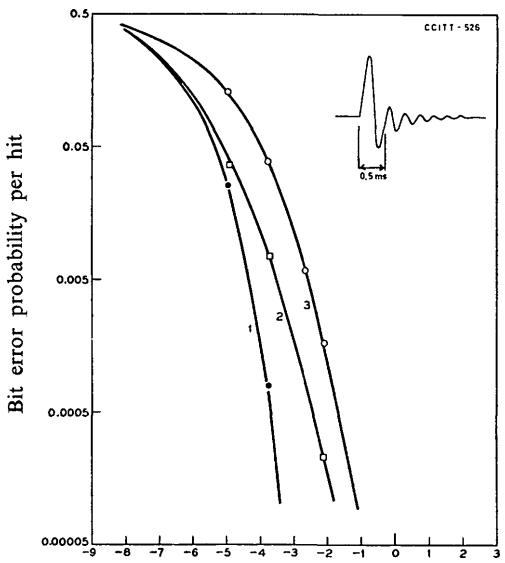


FIGURE 5. — **Probability of an error per hit generated impulse noise**

Transmission impairments due to various degrees of differential delay distortion for the 201B are given in Table IV. These data were obtained by G. R. Schwarz. Band-limited random noise was used and the magnitude in db of the increase in SNR required to maintain a fixed error rate of 10^{-4} is the measure of transmission impairment. Differential delay distortion characteristics are shown in Figure 6. In the case of symmetrical delay distortion the delays given in Table IV are measured at 600 c/s and 3000 c/s. Non-symmetrical delay distortion (high end only) is measured at 3000 c/s. Impairments in Table IV apply to the 201A data set for differential delays in ms equal to the ratio D/T given in column 2 at 750 c/s and/or 2750 c/s.

Figure 7 gives the probability of a bit error per impulse noise hit as a function of the received RMS signal to peak impulse noise ratio in db for back-to-back operation. The noise was generated in the laboratory and the data given in Figure 7 are not normalized, i.e. the difference in bit rate between the 201A and 201B is ignored. Curve 1 was obtained with the 201A, curve 2 with the

TABLE IV

Transmission impairments in db for 201B (2400 bits/s) data set with various degrees of differential delay distortion and a fixed bit error rate of 10^{-4}

1 Differential delay at 600 c/s and/or 3000 c/s (ms)	2 Ratio D/T T = dibit period = 0.833 ms	3 Symmetrical delay impairment db	4 High end only delay impairment db
0	0	0	0
0.4	0.48	1.1	0.6
0.8	0.96	1.8	1.8
1.2	1.44	2.8	4.7
1.6	1.92	4.6	13.2
2.0	2.40	7.2	Fail
2.4	2.88	10.6	Fail

201B and curve 3 with an early model of the 201A for which we give field test results in the next section. The generated impulse waveform shown in Figure 7 is not necessarily representative of telephone line impulse noise and therefore these data do not represent actual on line performance. On a comparative basis, however, the results are meaningful and reflect a real improvement in the 201A and 201B with respect to the impulse noise performance of the early model data set.

From the data given in Figure 7 it is seen that for a −3.0 db signal-to-peak impulse noise ratio the error probability per hit for the early model data set is 0.01 (one in one hundred hits would cause an error) while the 201A is error-free.

Switched network test

During the summer of 1960 a Bell Laboratories group made a series of point-to-point data transmission tests over the switched network with an early model of the 201A. In order to gain more information about the switched network potentially marginal (for data transmission) facilities in geographical areas not covered in previous tests were used. The results of the test are given in Table V.

Private line tests

A private line test for which an early model of the 201A was used as part of the test equipment is described in a paper by J. Kelly, J. Mercurio, and D. Willard to be given at the Fall General Meeting of the A.I.E.E. [3]. This test was conducted to investigate impulse noise and drop-outs — the data set was used to tag those impulse noise hits or drop-outs which would cause errors in data transmission.

TABLE V

Distribution of error rates for 92 10-minute switched network calls

Percentage of calls with error rate less than or equal to that given in next column	Bit error rate
20	0
30	1.0×10^{-6}
40	2.0×10^{-6}
50	5.0×10^{-6}
60	1.0×10^{-5}
70	2.5×10^{-5}
80	6.7×10^{-5}
90	2.5×10^{-4}
98	1.0×10^{-3}

A large multi-point network test using an early model of the 201A is described in a series of four papers to be given at the Fall General Meeting of the A.I.E.E. [4, 5, 6, 7]. We give data obtained from this test, see Table VI, concerning the percentage of "working" days (i.e. periods of supervised operation—approximately 8 hours a day on weekdays) with a given number of error messages per minute. A summary of the main features of the test follows :

TABLE VI

Distribution of number of error messages per minute per "working" day for multiple terminal network test Equivalent 792-bit messages

1 Percentage of working days with error rate less than or equal to column 2	2 Number of error messages per minute (125 messages per minute)	3 Efficiency loss in per cent due to message errors	4 Minimum equivalent bit error rate
10	0.08	0.06	1.6×10^{-6}
20	0.13	0.10	2.6×10^{-6}
30	0.18	0.14	3.6×10^{-6}
40	0.28	0.22	5.6×10^{-6}
50	0.37	0.30	7.4×10^{-6}
60	0.48	0.38	9.6×10^{-6}
70	0.76	0.61	1.6×10^{-5}
80	1.45	1.16	2.9×10^{-5}
90	3.46	2.76	6.9×10^{-5}
95	5.62	4.50	1.1×10^{-4}
97.5	8.30	6.60	1.7×10^{-4}
100	41.70	33.30	8.3×10^{-4}

1. Messages of varying lengths were transmitted—108, 396, and 792 bits.
2. The test period covered 11 months and 174 "working" days.
3. In addition to 3000 miles of "backbone" circuit—8 links—there were 24 voice-frequency drops on the circuit.

In order to include data from all "working" days on an equivalent basis, error rates for the shorter, less efficient, message lengths were multiplied by appropriate factors to bring them to equivalent 792-bit message efficiency. The data given in Table VI include error messages due to all causes. The most important of these were impulse noise hits, drop-outs, and line circuit malfunctions.

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Supplement No. 50

THE MASSACHUSETTS INSTITUTE OF TECHNOLOGY — LINCOLN LABORATORY
(Contribution COM Sp. A/No. 69 — July 1963)

APPLICATION OF SEQUENTIAL DECODING TO HIGH-RATE DATA COMMUNICATION ON A TELEPHONE LINE

Conventional telephone-line data transmission systems over comparable channels communicate about 1600-2400 data bits per second, with a probability of error in the order of 10^{-4}-10^{-5} [2]. The errors that occur in these systems are attributable primarily to relatively-infrequent impulsive noise bursts and channel drop-outs. For data rates in this (conventional) range, the error rate can be reduced virtually to zero (without substantial rate reduction) by the simple expedient of using simple "detect and repeat" coding techniques [3), 4].

For substantially higher data rates, however, the situation changes radically. Random noise and intersymbol interference due to delay distortion are no longer negligible and introduce signalling anomalies that call for error correction as well as error detection. In the experiment described here, a high-speed modulator and demodulator are combined with a variable data rate sequential coder-decoder. The decoder, in the course of its operation, monitors the signal disturbance (from all sources). If the perturbation level is within a range such that all transmitted signals can be reconstructed, the decoder proceeds in its operation at the current data rate ; if the perturbation level becomes too high, the decoder requests retransmission at a lower rate. Finally, if the perturbation becomes very low, the decoder requests a rate increase. The net result is a system transmitting as much information as the state of the channel and terminal equipment permits, with a vanishingly small probability of error.

SECO

The structure of the encoder-decoder (referred to as SECO), as well as some experimental results obtained with its use on a binary-symmetric channel, have been reported previously [6]. In brief, the encoder takes an infinite sequence of data bits and inserts an average of $\left(\frac{1}{r} - 1\right)$ parity check digits into the sequence for each data bit. Each check digit is a function of the 60 preceding data bits. The parameter r can be any one of the 8 values 3/4, 3/5, 1/2, 2/5, 1/3, 1/4, 1/5, 1/6. However, when modified for phone-line operation, the system uses only the three rates 3/5, 2/5 and 1/5.

The SECO decoder contains within it an encoder which generates replicas of possible transmitted sequences. It uses these sequences as hypotheses, compares them with the received sequence, and accepts or rejects each hypothesis on a digit-by-digit basis according to whether or not the hypothesis "agrees sufficiently well" with the received sequence. More precisely, a given bit, say x_i, of a hypothesis is accepted if the $60/r$ digits of the hypothesis beginning with x_i are "sufficiently close" to the corresponding $60/r$ received digits.

In the experiment reported here an effort is made to measure "agreement" in terms of a metric that is reasonably adapted to the nature of the channel disturbance. The optimum metric of course, would have to be a function of the statistics of this disturbance : for example, an optimum measure of similarity ("distance") between a transmitted sequence x and a received sequence y would be $-\log p_r\,(x/y)$ [7].

In the case of a binary symmetric channel, the Hamming distance is optimum. However, this is not generally true. The phone line modulation system described below uses a 5-bit alphabet whose symbols are represented by one out of 32 amplitudes, as shown in Figure 2. These amplitudes are determined by taking the binary output sequence from the encoder, framing it in groups of 5 bits, and converting each resulting 5-bit number into a decimal voltage. The only *a priori* fact known about the behaviour of the phone line with this modulation system is that symbols which are close together in amplitude are more likely to be confused than are symbols which are far apart in amplitude. Accordingly if symbol a_i is hypothesized and a_j is received, the distance between them is defined in SECO to be $|i - j|$, where the index i (or j) denotes the amplitude of the letter a_i (or a_j). Thus a sequence of $60/r$ binary digits corresponds to a sequence of $12/r$ letters, and the distance between two such sequences is the sum of $12/r$ elementary distances of the form $|i - j|$.

In the SECO decoder the decoding time per digit is a random variable. To average out computational peaks, a buffer memory of 3000 bits is inserted between the demodulator and the

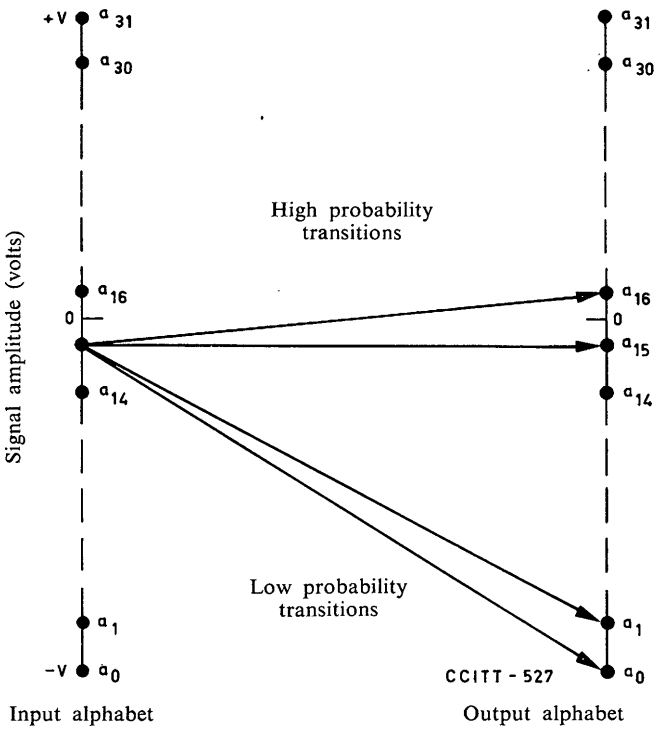


FIGURE 2

decoder. The decoding distance described above and the behaviour of the waiting line in the buffer memory provide the information which enables the decoder to monitor the channel and initiate feedback requests. The decoder stops and automatically requests a repeat at a lower value of r if either the buffer fills completely or the decoding distance remains "too large" for a "sufficiently long" time. The decoder requests an increase in r if the distance remains "sufficiently small" for a "sufficiently long" time. Appropriate numerical values for the criteria in quotation marks above have been obtained experimentally for both the binary symmetric channel and the phone line.

The modulation system

The modulator generates a stream of Gaussian-shaped pulses at a rate of approximately 3000 per second. Thirty-two level amplitude modulation is used to form the 32-symbol alphabet, resulting in an alphabet capacity of approximately 15 000 binary digits per second. The pulse sequence is generated at d.c. and then used to vestigial-sideband modulate a 2500-c/s carrier. The received signal is demodulated with a 2500-c/s tone that is synchronous with the received carrier. The resulting signal is low-pass filtered and then sampled and converted to 5-bit numbers at points of time corresponding to the occurrences of the maxima of the Gaussian pulses.

The system as described thus far would result in an excessive amount of inter-symbol interference. This interference is reduced in two ways. First, a compromise equalizing filter is inserted in series with the system. This has the effect of increasing the width of the flat ($\pm$ 500 µs) portion of the path delay characteristic from 1900 to 2600 c/s. Second, a compensator is used to cancel

out the contributions of adjacent pulses to the amplitude of the waveform at the sampling times. More precisely, as shown in Figure 3, the compensator output is a pulse whose amplitude is forced close to zero at several sampling times both preceding and following the maximum of the pulse (at most 13 zeros can be forced in this way). In the existing experimental equipment, the adjustment of the compensator controls is performed by observing the received waveform at the demodulator; if the transmitting and receiving terminals were not adjacent, some other method of adjustment would be necessary. However, frequent readjustment does not appear to be required.

The phase crawl introduced by the phone-line carrier system is tracked by the introduction of a low-level pilot tone at 3000 c/s and the subsequent derivation of the demodulating 2500 carrier wave from the received pilot tone. The phase of the demodulating signal requires adjustment at intervals of about 10 minutes, apparently because of changes in the electrical length of the phone line. Equipment for performing this adjustment automatically is being constructed.

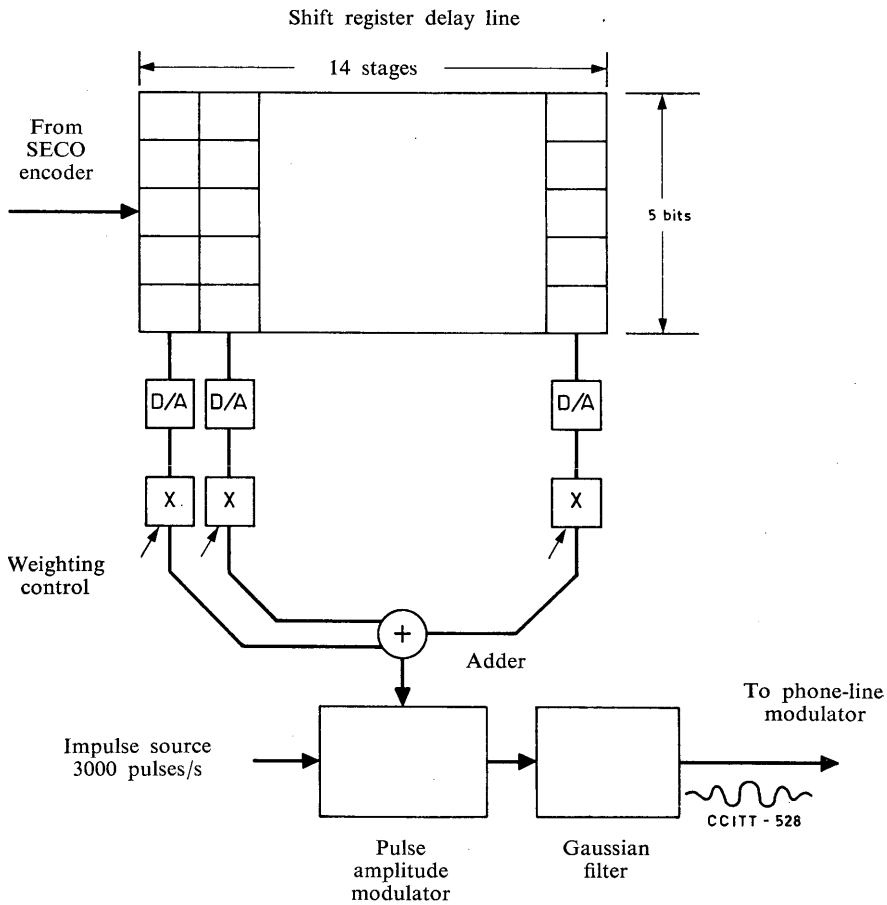


FIGURE 3

Results

With the modulation system operating at 3000 symbols/s the probability that the received symbol differs from that transmitted varies from 0.06 to 0.3 depending upon the phone-line noise and the precision of adjustment of the equipment. A rough average figure is 0.1, and under these conditions SECO is observed to spend roughly half of its time with $r = 2/5$ and half with $r = 3/5$, yielding an over-all decoded data rate of 7500 data bits/s. When the channel error probability drops below 0.1, a greater percentage of the time is spent at $r = 3/5$, with the decoded data rate correspondingly closer to 9000 bits/s. Thus far, in some 15 hours of operation, only 1 erroneous data bit has been released from the decoder.

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Supplement No. 51

SIEMENS & HALSKE — *(Extract from contribution COM Sp. A/No. 78 — July 1963)*

MODULATION METHODS

1. *Introduction*

The most convenient approach to the transmission of binary information is the employment of binary, that is two-state, modulation methods. As each information-bearing bit is then directly associated with one modulation state, use can be made of particularly simple transmission facilities which do not necessitate a definite modulation rate or a definite clock pulse rate, that is a rhythmic succession of the signal elements. C.C.I.T.T. Special Study Group A has already worked out a preliminary recommendation specifying the most essential characteristics of such methods with frequency or phase modulation for modulation rates up to 1200 bauds.

What many users of data transmission systems desire, however, is an increase of the transmission speed, especially the doubling of this speed. There are different methods which are in principle suitable for increasing the bit rate (bits/second) without actually increasing the modulation rate.

These are :

1. Multi-state modulation in which each state has the duration T of one signal element or a whole multiple thereof.

2. Modulation in which also other than whole multiples of the signal element duration T, e.g. T, 1.5 T, 2 T, 2.5 T, etc., are included, while modulation states shorter than T are not admitted. This approach does not involve an increase of the modulation rate nor the provision of a wider frequency band.

These two methods can also be employed jointly, which has been assumed in the procedure discussed here below.

2. *Ternary phase differential modulation, description of procedure*

1. The modulation rate amounts to 1200 bauds, hence the duration of one signal unit to $T = 0.833$ milliseconds. The modulation states may have the durations T, 1.5 T, 2 T, 2.5 T, etc.

2. Use is made of ternary phase modulation with the phase angles of 0°, + 120° and −120°. The reference angle is defined by the preceding state (differential phase modulation).

3. The co-ordination between the binary signals appearing at the input with a modulation rate of 2400 bauds and the phase displacement angles is shown in Table 2.1.

TABLE 2.1

Binary signals (2400 bauds)	$\Delta\varphi$	Brief designation	Duration ($1/T$ = 1200 bauds)
01	+ 120°	+	T
00	− 120°	−	T
1 unless covered by (01)	0	0	$T/2$

Note. — A modulation state of duration $T/2$ does not occur, since it is invariably preceded by the same modulation state for a minimum duration of T.

The coding process can best be explained by referring to the example laid down in Table 2.2. Line 1 shows a sequence of binary signal elements. Starting at the left side, the first "zero" element is merged with the succeeding element to form a pair (dibit), the next "zero" element is merged with the following one in the same way, and so forth. These pairs may thus read either (01) or (00). The "one" elements, which are not paired, remain isolated.

TABLE 2.2

1	01	00	00	1	01	1	1	1	01	1	00	1	01	1	1	00
2	+	−	−	0	+	0	0	0	+	0	−	0	+	0	0	−
3 u	v	u	w		u				v		w		v			u

Line 2 of the table designates in terms of the symbols of Table 2.1 (see above) the phase displacement angles associated with the individual modulation states, "+" designating a displacement by +120°, "−" a displacement by −120°, and "0" the retention of the carrier wave in the phase position last held. If we now denote the three phase *states* by *u*, *v*, and *w*, a displacement by +120° marks one of the transitions from *u* to *v*, or from *v* to *w*, or from *w* to *u*, while a displacement by −120° marks a transition in the reverse direction. What state the system was in initially is irrelevant to differential modulation. In our example, we have assumed the system to be in state *u* at the beginning of the train of signal elements, and entered in line 3 the respective modulation states. None of these prevails for a time shorter than *T*, and there is neither an intermission nor a congestion in the flow of information.

Note. — Lines 1, 2 and 3 of Table 2.2 merely demonstrate the interrelations. To obtain a timing diagram with a common time base, it would be necessary to shift lines 2 and 3 by the duration *T* of a pair (2 binary elements) to the right, because it is only *after* the arrival of, say, pair (01) before a decision can be made on the associated modulation process.

3. *Comparison between ternary and quaternary phase differential modulation*

Let "ternary modulation" be the abbreviated term for the above-described method modulation. Since "ternary modulation" entails the same increase in the bit rate as the usual quaternary modulation with only whole multiples of the signal unit duration, it is well to compare the two modulation methods in respect of their other characteristics.

It should be noted initially that when using quaternary phase differential modulation the elements of the binary signals are continuously paired. The resulting element pairs may be co-ordinated with the four-phase displacement angles of 0°, 90°, 180°, and 270° as exemplified in the following Table 3.1.

TABLE 3.1

Binary signals (2400 bauds)	$\Delta\varphi$	Duration ($1/T$ = 1200 bauds)
11	0	T
01	90°	T
00	180°	T
10	270°	T

1. Ternary phase differential modulation has the following disadvantage: since the duration of the modulation states, as far as they exceed value *T*, have to be distinguished from each other at intervals of half a signal element duration, the tolerable margin of distortion will only be half that of quaternary modulation. However, this disadvantage will come into play only in the presence of relatively heavy distortion.

2. The disadvantage as per 1 above is partly made up for by the fact that two contiguous modulation states are here spaced by 120° as against 90° with quaternary modulation, and becomes less significant in cases where the transmission errors are essentially due to pulse-type noise going beyond the signal amplitude.

3. Ternary phase differential modulation, as compared with quaternary modulation, carries momentous advantages as far as equipment requirements are concerned. These advantages result from the fact that only three instead of four modulation states have to be accommodated.

With the quaternary method, it is necessary to evaluate all of the four possible bit-pair combinations of the binary signal before the transmitter can decide on whether a phase displacement has to be made or not, and if so, of what magnitude. With the ternary method, on the other hand, the situation is such that after each pairing, as explained in paragraph 2, the next "zero" element of the binary signal initiates the decision that a phase displacement has to be made, while the succeeding element controls the displacement to either $+120°$ or $-120°$. This, in turn, means that the coder has to discriminate between only two pairing alternatives. Between the processes mentioned above the modulator must not receive new instructions, that is, the carrier wave is transmitted in the phase position last held. Thus, the task of the coder with the ternary mode is much simpler, and realizable with much less equipment complication, than with the quaternary mode.

What has here been said with respect to the send side applies analogously to the decoding process at the receiving end.

Over and above this, it should be borne in mind that the demodulator functions are much easier to accomplish with the ternary mode than they are with the quaternary mode. In the latter, the demodulator must be able to discriminate, in addition to the retention of the phase, between three phase displacements. In the ternary demodulation, however, there are only two types of phase displacements, namely $+120°$ and $-120°$. This implies that when a phase displacement occurs all the demodulator has to do is to recognize the sense of this displacement.

Without enlarging upon the numerous possible variants of actual realization, attention is invited to the fact that the functions of the demodulator can, with the ternary mode, be accomplished by an element as simple as a frequency discriminator which, in the presence of a phase displacement by $+120°$ may emit, say, a positive pulse, and in response to a $-120°$ displacement a negative pulse. The decoder merely has to generate, in the former case, one signal element, and in the latter case two successive signal elements of binary signal element duration and in the polarity assigned to "zero". Outside these processes, it has to emit the polarity associated with "one".

The advantages resulting from the fact that with the ternary mode only three modulation states have to be accommodated are considered significant. The characteristics laid down in the next paragraph may form the basis of a recommendation.

4. *Characteristics of the method using ternary phase differential modulation*

1. The phase displacements $\Delta\varphi$ are assigned to the bit train of the binary signal as indicated in the following Table 4.1 :

TABLE 4.1

Binary signal units	$\Delta\varphi$
01	$+120°$
00	$-120°$
1 unless covered by 01	0

2. Considering the frequencies already recommended for binary phase modulation, the characteristics laid down in Table 4.2 are recommended :

TABLE 4.2

Bit rate (bits/second)	2400	1200 [a]
Modulation rate (bauds)	1200	600
Carrier frequency (c/s)	1700	1500

[a] Recommendable only by way of exception in cases where the line characteristics do not accommodate a modulation rate of 1200 bauds but a bit rate of 1200 bits/second is desirable.

3. Signals should pass the interfaces between modem and data terminal equipment in serial binary form. The coder and decoder of the multi-state signal consequently are components of the modem. More interface lines than hitherto specified are not required.

4. As a consequence of the multi-state character of the modulation mode (in contrast to only binary modulation), the frequency clock forms part of the modem. In exceptional cases, it should be possible for the clock pulse rate at the send side to be generated by the data terminal equipment.

5. If it is deemed necessary in special cases to use the same equipment for transmission of binary signals at a non-increased bit rate, the co-ordination is as follows (see Table 4.3) :

TABLE 4.3

Binary signal	$\Delta\varphi$
1	+ 120°
0	− 120°

In this case, the clock rate is inherent in the signal element proper.

6. If a return channel is required, it should conform to recommendations to be worked out elsewhere.

5. *Conclusion*

The modulation method outlined in paragraph 1.2 working also with other than whole multiples of the signal element duration can, in principle, also be used in conjunction with quaternary and higher-state modulation modes, whereby the bit rate is increased. Not for all number of modulation states will this lead to simple methods. Quaternary modulation, for instance, would not be easily manageable by this approach, while 5-state (and 9-state) modulation, to quote just a couple of examples, would.

Supplement No. 52

TELEFUNKEN — *(Extract from contribution COM Sp. A/No. 78 — July 1963)*

QUATERNARY PHASE MODULATION

Only two-condition modulation methods (two frequencies, two phase positions) have been recommended for the time being, and preferred characteristics for modulation rates of 600 and 1200 bauds have been established. The results of experiments made so far show that 600-baud operation is possible on all circuits and 1200-baud operation on modern C.C.I. circuits without additional delay equalization both by frequency modulation and by phase modulation. Furthermore, a supervisory channel can be operated.

Recent performance tests on the various circuits have shown that the quaternary phase modulation offers considerable advantages with regard to the increase of data transmission speed both on standard C.C.I. circuits and on older, heavily loaded circuits. A report on the measurements carried out by Telefunken in co-operation with the German P.T.T. is contained in the comments on the study of Question 1/A, point L. From this report the conclusion may be drawn that the quaternary phase modulation is superior to the quaternary frequency modulation, particularly on circuits with limited transmission properties. Since sudden phase deviations of the carriers are to be expected on carrier circuits, we are giving preference to the differential phase modulation over the phase modulation with a fixed reference phase.

After it has now become possible through suitable circuit techniques to build uncomplicated equipment for quaternary differential phase modulation, this method appears very useful also from the economical point of view. In particular, a suitably designed equipment can be simply switched over from quaternary to binary phase modulation. In the following, therefore, a proposal is submitted for the recommendation of a quaternary DPhM Modem. Our proposal also considers the objective of covering the full range of applications (point-to-point connections and switched connections consisting of circuits with excellent and limited transmission properties) by as few different types of modems as possible. In order to enable in particular the transmission at a speed of 1200 bits/s over all (also over heavily loaded) circuits, we propose for this modulation rate a carrier frequency of 1500 c/s and quaternary phase modulation. This proposal permits a supervisory channel in spite of the low carrier frequency.

The carrier generator contained in these modems can also be advantageously used as clock generator. Therefore, it is in the case of this type of equipment advisable to integrate the clock in the modem. However, in order to ensure the universal use of the modem, the possibility of the clock being supplied by the data-processing terminal equipment should also be provided for.

Proposal for the recommendation regarding DPhM modems :

TABLE 1

Type of circuit	Any type of circuit (also heavily loaded circuits) without additional delay equalization		Carrier circuits and medium loaded VF circuits without individual delay equalization
	1	2	3
Data transmission speed, bits/s	600	1200	2400
Carrier frequency, c/s . .	1500	1500	1800
Type of phase modulation	binary	quaternary	quaternary

It is possible without appreciable equipment complication to include in a modem switchover circuits so that it is suited for all three speeds. For instance, a modem having switchover facilities for conversion from 2 to 3 could be used on all circuits for transmission of 1200 to 2400 bits/s.

A further specification to be established for a modem with four modulation conditions is the assignment of the phase shifts to the binary digits transmitted or received by the data-processing terminal equipment.

In Recommendation V.1 the assignment of the binary digits to the phase shifts for the binary differential phase modulation is laid down (see Table 2).

TABLE 2

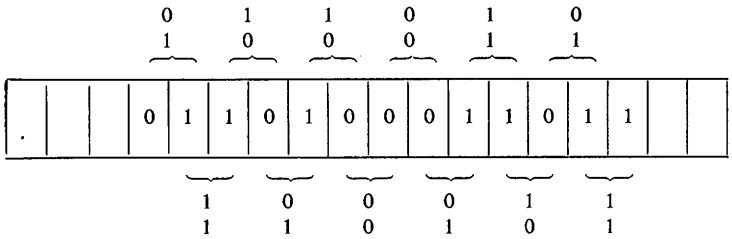
Binary digit	Phase difference $\Delta\varphi$
1	0°
0	180°

No specifications have so far been issued by the C.C.I.T.T. for the quaternary differential phase modulation.

The following is a proposal with regard to such a specification :

1) At the interface between the data-processing terminal equipment and the data transmission equipment (modem), the data shall be transmitted in series in the form of binary digits, like in the binary modulation methods. The conditions at the interface so far specified shall be acceptable also for modems with quaternary phase modulation (no additional requirements to be met for the modem in the data-processing equipment, no additional interface lines).

2) The combination of the binary digits into pairs shall be accomplished in the modem by the grouping of two subsequent binary digits each into one pair. The dividing of the sequence of binary digits into pairs can be arbitrarily determined by the modem when the transmission starts (see sketch below), but it should be retained throughout the transmission.



A typical sequence of binary digits and the two possible arrangements into pairs

3) The code table.

TABLE 3

Digit pair		Phase difference $\Delta\varphi$
1	1	0
0	1	+ 90°
0	0	+ 180°
1	0	+ 270°

Definition of the sign of the phase change:

Carrier phase prior to phase change $s = s_0 \cos(\omega_0 t + \varphi)$

Carrier phase after phase change: $s = s_0 \cos(\omega_0 t + \varphi + \Delta\varphi)$

Motivation of the proposal

Items 1) and 2) are in line with present technical standards and need not be explained further.

Item 3):

The determination $11 \longleftrightarrow \Delta\omega = 0$ suggests itself in order to meet the C.C.I.T.T. recommendation (see Table 2).

With regard to the equipment design, this proposal brings the advantage that the switchover from quaternary to binary phase modulation (switchover from high to low transmission speed) is no difficult procedure.

It is appropriate to use a cyclic code (Gray code). It will be noted that all equipments known so far use a cyclic code. When not considering initially the sense of the cycle, the relation

$$00 \longleftrightarrow \Delta\varphi = 180°$$
$$01 \longleftrightarrow \Delta\varphi = + 90°$$
$$10 \longleftrightarrow \Delta\varphi = + 270°$$

The two relations can as well be interchanged crosswise. We chose this assignment since thereby simplifications in the circuitry of our equipment are possible.

Supplement No. 53

LENKURT ELECTRIC CO. (U.S.A.) — *(Contribution COM Sp. A/No. 85 — September 1963)*

A SYSTEM WITH DUOBINARY CODING FOR DATA TRANSMISSION OVER TELEPHONE CIRCUIT

Introduction

A novel coding technique is used in what is believed to be the most effective method for transmitting more information through a given channel than specified by the Nyquist relationship. This technique is called duobinary coding. It is used in a data transmission system for providing data communication service over voice facilities.

Duobinary coding

Figure 1 shows one way of achieving duobinary coding. Binary data pulses are applied to one input of a simple AND gate, and a stream of binary "clock" pulses of the same repetition rate as the data pulses is applied to the other input. Whenever a data space occurs, the AND gate yields an output pulse which is applied to a "flip-flop" circuit or bi-stable multivibrator. The multivibrator has the characteristic of changing its output voltage from zero to plus or plus to zero, whenever it is triggered by the AND gate. Thus, for every data space, the output of the flip-flop changes state. Whenever a data mark is received, no change occurs.

The binary waveform from the flip-flop could be amplified and applied directly to the line for transmission, but this would merely waste some of the transmission power and increase crossfire into other circuits. Instead, it is normally passed through a low-pass shaping filter. It should be noted that the output signal from the flip-flop is a binary waveform which, although of a different pattern, has a direct correlation with the original data, and from which the original data can be recovered.

Because of the encoding process, the signal now has several unique properties. As it passes through the channel or shaping filter (which must have a cut-off frequency at quarter the data rate), the waveform is altered so that it tends to occupy three amplitude levels. Superficially, this three-level signal resembles a so-called "ternary" (three-level) signal or a bipolar signal, both of which have been used in telegraphy for many years. However, there are significant differences : ternary and other conventional multilevel signals must be able to go from any given level to any other level in order to transmit information at a rate faster than binary systems. By contrast, the duobinary signal has the effect of forbidding the signal from going directly from one extreme level to the opposite extreme within one bit-interval.

A continuous series of input data spaces causes the flip-flop to change state repeatedly, thus creating a series of binary pulses at the data rate. This is too fast to pass through the filter, and a "zero" level signal results. A continuous series of marks results in the flip-flop remaining in whichever state it was in at the beginning of the sequence. The single voltage, tone, or phase by which this marking condition is transmitted will pass through the channel or filter. Since either

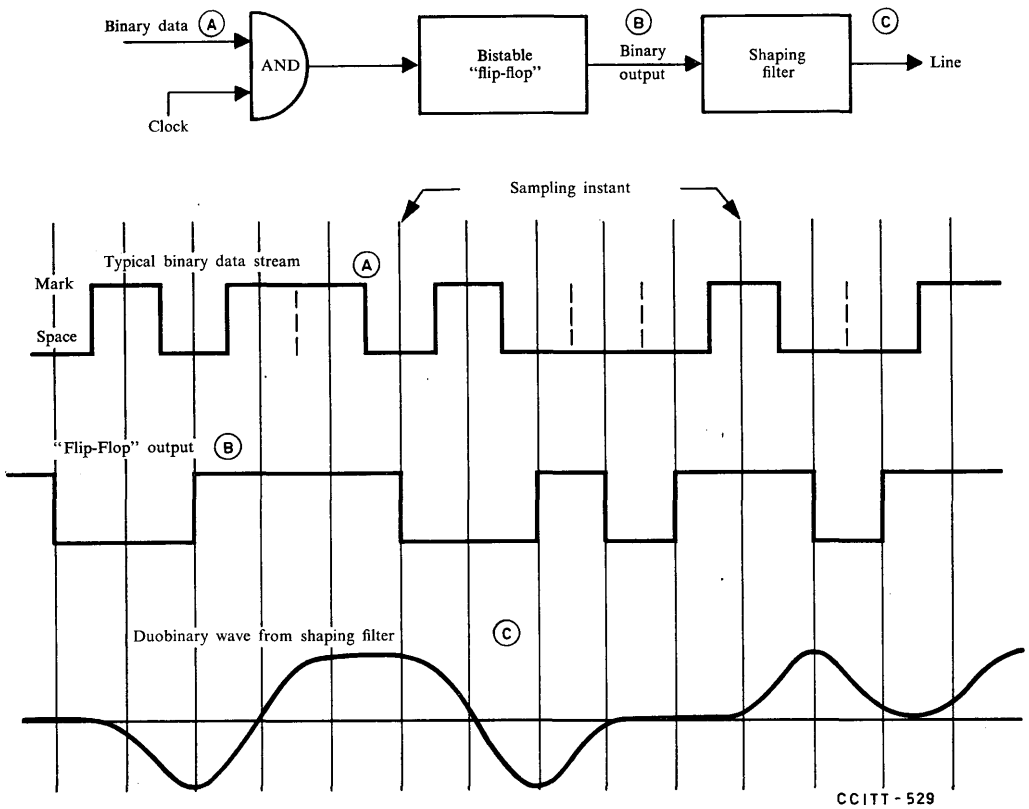
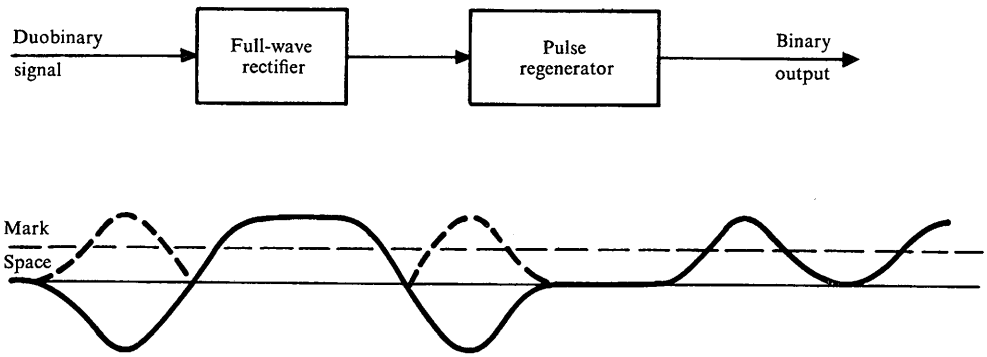


FIGURE 1. — **One form of duobinary encoder and its resulting waveforms**

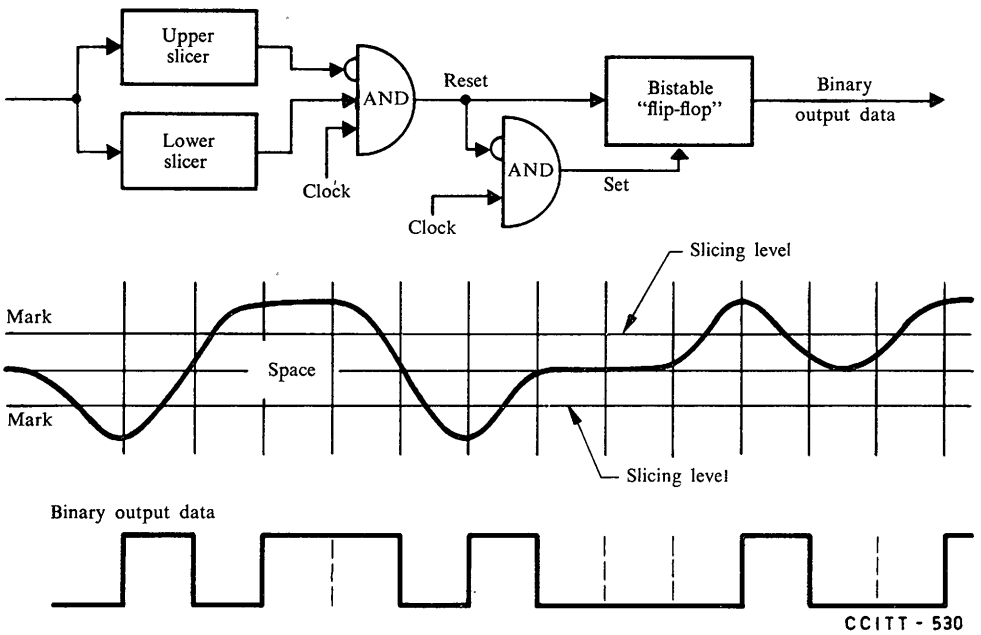
Coincidence of "clock" pulse and data space causes bistable multivibrator ("flip-flop") to change state. When passed through transmitter wave-shaping filter the three-level duobinary wave results

plus or minus represents mark, the signal will not change for consecutive marks, but only when a space follows a mark. A second space holds the output in the zero condition, but a mark following the space returns the signal to a mark condition, as shown in Figure 1. Synchronization of receiver and transmitter, an absolute necessity in multiphase transmission methods, may be used but is not required.

Another method of decoding duobinary signals uses a synchronized "clock" signal at the receiver to sample the waveform at the centre of each bit interval, thus reducing the likelihood of noise impulses causing transmission errors. Two binary "slicers" determine at the sampling moment whether the signal is in the central or "neutral" region, thus indicating a space or whether it is outside the central region in either the positive or negative region, thus indicating a mark.



Method 1



Method 2

FIGURE 2. — **Two forms of duobinary decoders**

Simplest and most versatile consists only of full-wave rectifier and pulse regenerator. Pulse retiming is optional. More elaborate version below uses "slicers" to determine whether duobinary wave is in "mark" or "space" condition at sampling instant

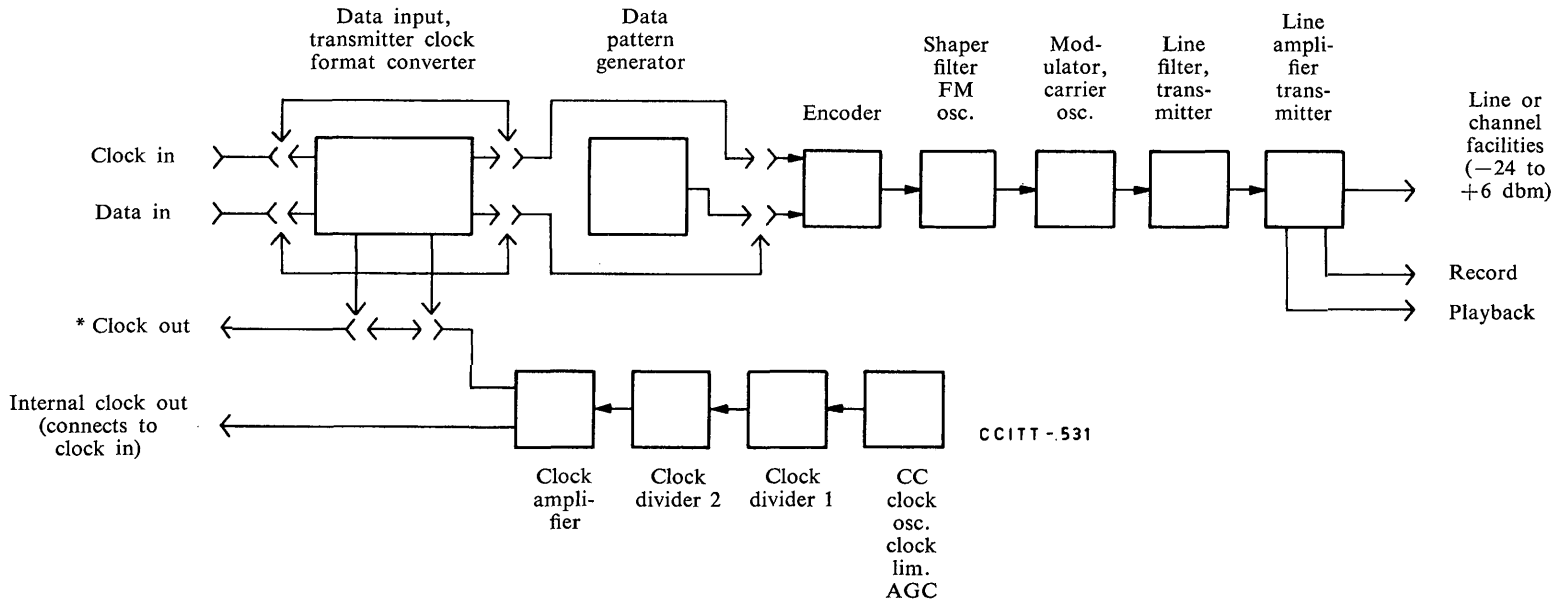
Description of system

The system accepts serial binary data at discrete (fixed) rates of 2400, 1200, or 600 bits per second, or continuous rates at any speed up to a maximum rate of 1200 bits per second. For an input digital data rate of 2400 bits/s, a duobinary mode of transmission is used, while a straight binary mode is used for the other data rates. At the transmitting end, clock signals are required only for the 2400 bits/s data speed, and are not necessary for any continuous rate either at or below 1200 bits per second. Clock signals are used at the receiving end only if retiming of the received data is desired. Internal clocks that have switch-selected outputs of 2400, 1200 and 600 cycles per second are available if retiming of either the input or the output data is desired. To change from one mode of transmission to the other requires only selection of the appropriate mode by means of a switch on the front of the equipment and selection of the appropriate clock speed, if a clock is used. Both the transmit and the receive directions of transmission require a data channel having a nominal bandwidth of 300 to 3000 cycles per second.

The system can be used either in simplex (one-way) or in duplex (two-way) applications. Figures 3 and 4 show block diagrams of the transmitting and receiving sections, which are, respectively, contained in the transmitter shelf and receiver shelf.

Serialized binary data are accepted by the transmitting section and converted to a binary pattern that is best suited for either the duobinary or the binary transmission process. The encoding operation is accomplished by changing the state of the binary digits whenever the binary level "0" is present. If retiming is used, the incoming data are sampled at a rate controlled by either an external clock or an integral clock. The encoded binary data pulses, after passing through a low-pass shaper filter to remove the harmonic frequencies, control the frequency of an FM oscillator. In the binary mode of transmission, a binary level "1" is represented by the lower line frequency and the binary level "0" is represented by the higher line frequency. In the duobinary mode, either the lowest or the highest line frequency represents binary level "1" depending on the previous data pattern. Binary level "0" is represented by the average value of the two frequencies. A modulation stage, followed by a lowpass filter, translates the FM oscillator output tones to the 1.1 and 2.3 kc/s voice-channel range. This band is then amplified and passed through an impedance-matching transformer to the line facilities. Terminals are provided at the output for bridging on a high-input impedance tape recorder and, by means of a selecting switch, the line signals can be played back.

In the receiving section, the band-limited FM signals are passed through delay-and-amplitude correcting equalizers (optional) and a bandpass filter to an amplifier. Terminals are provided at the output of the amplifier for bridging on a tape recorder to record or play back the received line signals. The amplified signal is then demodulated to the original FM spectrum. After being filtered, the received signals are passed through a two-stage limiter that has a 40-db dynamic range. Amplitude variations in the filtered signal are eliminated by the limiter, which passes the constant amplitude signal to a discriminator. The discriminator changes the FM signals into analog signals for presentation to the level selectors. In the duobinary mode of transmission, two level selectors are used to determine the state of the duobinary signal. In the binary mode, only one of the level selectors is used. The signals are then reconverted to their original form by a binary decoding process, both in the duobinary and the binary techniques. Use of a clock is optional in the receive



* For external use.

FIGURE 3. — **Block diagram of transmitter shelf. Optional high-stability clock is shown**

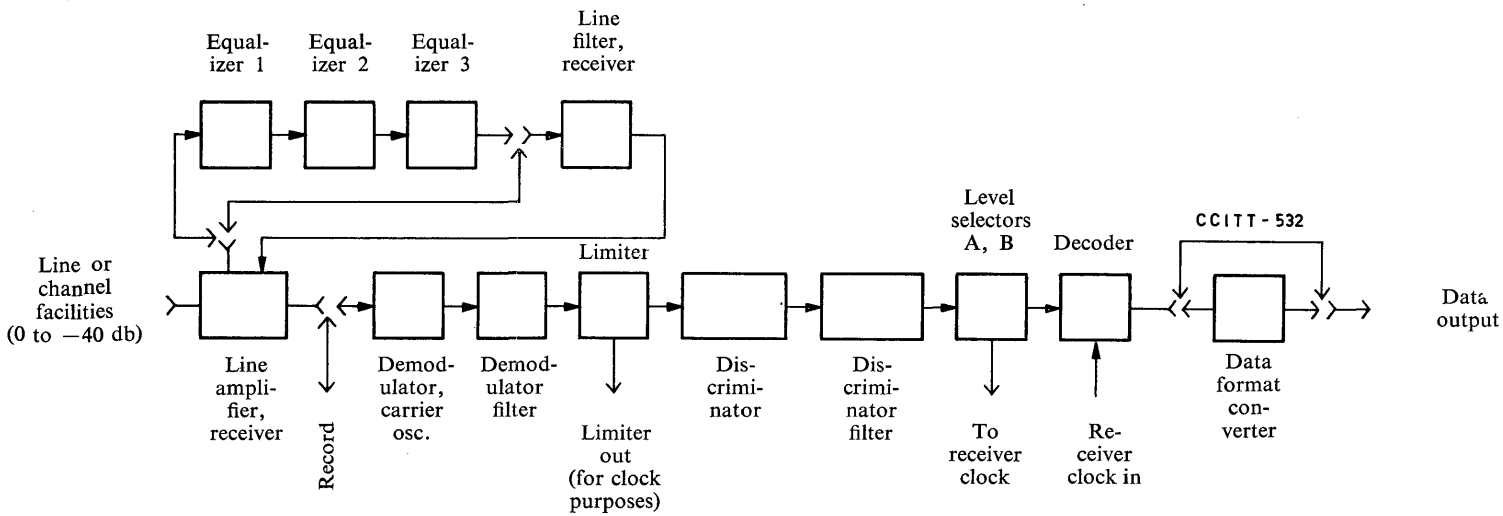


FIGURE 4. — **Block diagram of receiver shell**

section, and if one is employed, it will sample the data bits to provide retimed data. If a clock is not used, the output from the level selectors is amplified to provide the proper data output levels.

Error detection

An outstanding feature of the duobinary technique is its ability to provide "free" error detection ; that is, error detection which is not dependent on the addition of redundant bits or data code symbols to the transmitted data. Error detection is very important in most digital data transmission because the message may frequently consist of unrelated numbers or characters suitable for direct use by business machines or computers. Single errors may have the effect of invalidating or "spoiling" a much larger block of information. A good example of this might be in the transmission of information from punched cards. The information in such transmission has little or no inherent redundancy which might help reveal errors. By contrast, written text or spoken words are very redundant ; occasional errors are usually very conspicuous and thus easily corrected by the recipient.

Heretofore, all digital transmission methods have been able to achieve error detection by the deliberate addition of redundant symbols to the transmitted information, thus "diluting" it and lowering the effective transmission rate.

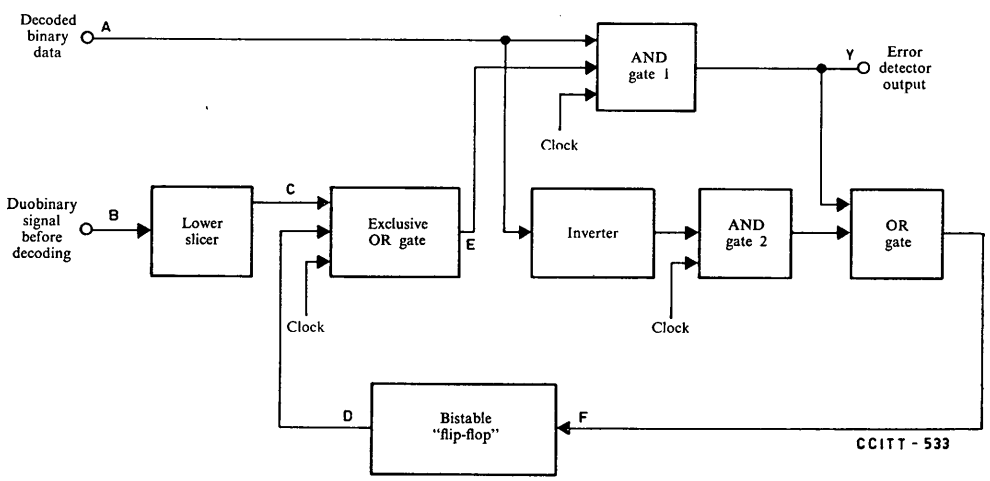


FIGURE 5. — **One form of automatic error detector**

In the absence of transmission errors, proper polarity of data marks is determined by the number of intervening spaces. The flip-flop acts as memory for polarity of preceding mark and counts intervening spaces. Points C and D should have opposite polarity, causing output at E, only during data spaces. If output at E coincides with mark at point A, an error has occurred, causing indication at point Y. Error signal resets flip-flop to "correct" polarity to avoid subsequent false error indications

Duobinary coding permits error detection without redundancy because the encoding process follows a systematic pattern which can be constantly monitored by error-detecting circuits. Any violation of the pattern results in an error indication. This pattern can be summarized as follows:

Space is always represented by "zero" or neutral.

Mark is represented by either + or −, depending on the number of spaces between marks.

The polarity of a mark remains the same as the preceding mark if the number of intervening spaces is even, but is reversed if the number of intervening spaces is odd.

Since any single error changes the number of spaces between marks, the encoding plan would be violated, and the error can be detected automatically. The error detector diagrammed in Figure 5 follows the pattern of the received signal and indicates any violation of the encoding rules. As a result, all single errors, all odd numbers of errors, and those even numbers of errors which also result in a violation of the encoding rules can be detected without the addition of redundancy. In order to achieve this same capability with conventional parity-check methods, a large portion of the transmitted data would have to consist of redundant check bits.

Facility requirements

The system is capable of operating over any voice-frequency facility meeting the line requirements listed below, and it was engineered to conform to applicable portions of Ela Standard RS-232. Normal subscriber station protection devices can be used with the equipment.

Line requirements

1. Frequency response with optional receiver shelf equalizers	900-2300 c/s ± 3 db 400-2700 c/s ± 6 db
Without equalization	900-2500 c/s ± 1 db
2. Envelope delay with optional receiver shelf equalizers	Within 1 ms from 800 to 2350 c/s Within 1.7 ms from 600 to 2700 c/s
Without equalization	Within 0.3 ms from 900 to 2500 c/s
3. Line impedance	600 ohms nominal at 1000 c/s
4. Line S/N ratio [1]	16 db at 2400 bits/s 10 db at 1200 bits/s

[1] For a bit-error rate of 1×10^{-5} or better.
S — rms signal power.
N — rms white noise power in 3-kc band, flat weighting.

Performance

Systems employing duobinary coding have been used over various facilities including over 3000 miles of submarine cable (on 3-kc/s-spaced channels). The results obtained have been excellent.

Figure 6 shows a comparison of error rate and signal-to-noise ratio with thermal noise. The difference between conventional binary and duobinary is about 3 db on a normalized basis (signal

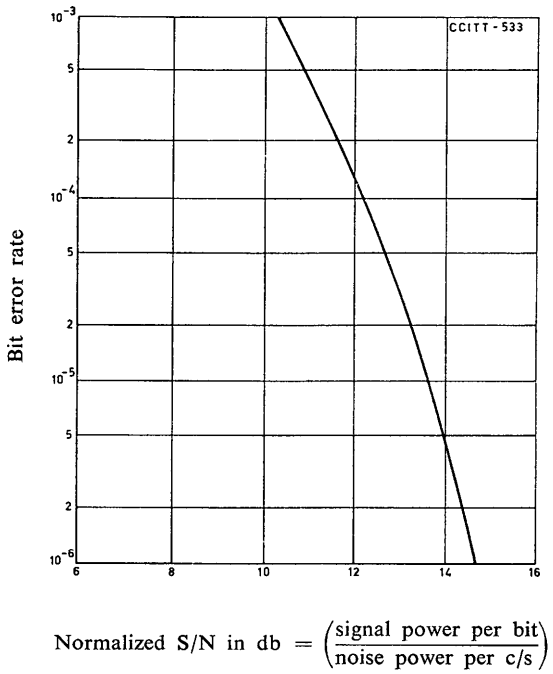


FIGURE 6. — **Typical error performance at 2400 bits per second**

power per bit divided by noise power per cycle of bandwidth). Theoretical considerations show that :

a) The noise bandwidths are identical for each system.

b) The duobinary system employs three levels rather than two as in the binary system, giving the binary system a 6-db advantage over the duobinary system.

c) The speed of the duobinary system is twice that of the binary, giving the duobinary system an advantage of 3 db over the binary on a normalized basis.

Thus the net difference is 3 db in favour of the binary system. Experimental results also confirm this.

REFERENCE

A vigorous and detailed treatment of duobinary technique has been published as :

LENDER, A. : "The Duobinary Technique for High Speed Data Transmission". Paper 63-283, *I.E.E.E.* Winter General Meeting, February 1963.

Supplement No. 54

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — July 1963)*

THE NEED FOR A PARALLEL DATA TRANSMISSION SYSTEM ON SWITCHED TELEPHONE CIRCUITS

The justification for a parallel as against a serial binary transmission system for the transmission of data signals depends upon :

1. Whether there is an advantage economically (more efficient utilization of available bandwidth, or a reduction of errors) when compared with the serial system over the same transmission media.

2. Whether, quite apart from the merits of the serial or parallel method used for the transmission media, there is also an economical advantage in connecting the data-processing devices (e.g. tape readers, punches, card readers, card punches and computer inputs and outputs) to a parallel transmission because both are in a parallel form.

Considering

a) That if we compare the two methods (serial and parallel transmission) on the basis that in both cases the binary data signals are presented and received from the transmission terminals in a parallel form, then the difference in costs of the two may not be great. This is because, generally speaking, the parallel/serial and serial/parallel data signal translation required at each end of the serial system of sets counterbalances the costs of the frequency division techniques required for the parallel system. However, this is only likely to approach a balance for a particular combination of information rate and available bandwidth of the system. For higher information rates with a smaller bandwidth the frequency division techniques of the parallel system would become increasingly onerous and consequently more expensive that the serial methods. For lower information rates with the same or greater bandwidth both methods become cheaper and the serial case approaches the cost of one channel of the parallel method; however, the cost of the data signal serial/parallel conversion begins to preponderate and makes the serial arrangement more expensive over-all at the lower speeds.

If we include from these cost considerations the data signal conversion whether to a parallel or serial presentation, it appears that the serial method of transmission is likely to be cheaper in the range of practical information transmission rates.

b) That we may consider multifrequency parallel systems under two headings : those that always send a fixed number of tones, say N out of a set of M tones, and those that transmit up to a maximum of M tones out of a set of M tones :

"N out of M" system

Where there is no restriction in the choice of the N in the coding—i.e. there are $\frac{M!}{N!(M-N)!}$ allowable combinations—this is just a system of M binary channels in parallel with redundancy in the code and can give protection as in any fixed ratio code.

Another form where one tone is sent from each of N equal sub-groups of M is virtually a frequency exchange system of N channels with each operating to a code of base $\frac{M}{N}$ giving a total number of combinations equal to $\left(\frac{M}{N}\right)^N$.

In each form the maximum power allowable in each tone, in order to avoid overloading for N tones at a time parallel system would be $1/N$ of that for a serial system where N is less than eight. White noise interference on each tone of the parallel system would be reduced by $1/M$ where M is the total number of tones used, thus the signal-to-noise power ratio on the parallel system would be M/N times better than for a serial on/off system. In addition to the lower modulation rate ($1/N$ of that of the serial system) the system would be still less affected by noise and also delay distortion. Short breaks and noise bursts would have less effects due to the narrow band filtration of the parallel system but single-tone interference would give rise to a signal-to-noise ratio N times worse than for the serial case. By virtue of its limited selection of "N out of M" both the parallel and serial here compared allow the possibility of error detection.

"M out of M" system (i.e. M binary channels in parallel)

This system possesses some of the advantages of an "N out of M" system but it has similar performance with the serial system in respect of white noise (provided that the power limitation is an average and not peak power basis) and is inferior to single-tone interference by a factor of 1 over M. The method of modulation in itself carries no error-detecting properties.

Both the systems considered are inferior to serial methods in respect of non-linear distortion (inter-modulation products). In addition they are restricted in the form of coding—particularly the "N out of M" arrangement.

c) For maximum information transfer rates both the "N out of M" and "M out of M" systems should work synchronously. However, in the case of asynchronous working it is necessary to indicate when one character has finished and another is being transmitted. It is only really essential to send such an indication when the same character is sent successively. Every other character change would be indicated by the character transition itself. For simplicity, however, it is possible to reserve one of the code combinations, or one of the channels to indicate the end or beginning of every character transmitted.

Considering

that as most forms of data reading and recording devices associated with computers depend upon a parallel signal, it is clear that any system of transmission which allows the signals to be transmitted directly without any re-coding or time rearrangement would save costs in the terminal

data reading and recording devices. This would be of most significance where the capital cost of the device is low.

Conclusions

Parallel transmission systems may have an advantage over serial systems:

1. Where the required information rate is low.
2. Where the total costs of the terminal are low and the costs of the parallel/serial conversion would be significant.
3. Where a restricted code "*N* out of *M*" is used, as this may produce a lower error rate than a corresponding serial system.
4. At the higher modulation rates within a restricted bandwidth the parallel system of transmission is likely to be more expensive and therefore not recommended.

Supplement No. 55

SIEMENS & HALSKE — *(Extract from contribution COM Sp. A/No. 78 — July 1963)*

MODEMS FOR PARALLEL TRANSMISSION OVER TELEPHONE CIRCUITS

Introduction

The modem serves as an instrument for the transmission over telephone circuits of data originating from, say, a punched-card reader, punched-tape reader, or a keyboard. The most general application envisaged is the (non-simultaneous) transmission of data from a great number of out-stations to a central station. This application makes it desirable that the out-station equipment be as simple in design as possible.

The following paragraphs describe a parallel transmission method using a three-times-one-out-of-four code (3-unit quaternary code). Provision has also been made for a return channel.

In drawing up this proposal, it has been assumed that, generally, only small quantities of data will be transmitted and, therefore, no stringent requirements need be imposed on the transmission speed.

Description of method

The proposal provides for the use of three frequency groups, each of which is made up of four alternative frequencies. These are given in the table below.

As long as the circuit is ready for transmission, but no data are actually being transmitted, it is in the "idle-circuit state", characterized by the steady transmission of only one frequency, viz. the 780-c/s frequency of Group I. The frequencies of Group I additionally serve as pilots, in which capacity they control the gain of a central-station receive amplifier which is common to all frequencies.

If data come up for transmission, Group I is cut over in no-break fashion to that frequency which corresponds to the code combination to be transmitted (this may also be the 780-c/s idle-circuit frequency), while the frequencies of Groups II and III for the code combination in question

TABLE 1

Frequency group I	Frequency group II	Frequency group III
420 c/s	1020 c/s	1500 c/s
540 c/s	1140 c/s	1620 c/s
660 c/s	1260 c/s	1740 c/s
780 c/s	1380 c/s	1860 c/s

are keyed ON. The ON state should be maintained for some 25 milliseconds. Before the next data signal is transmitted, the idle-circuit state should be restored for a minimum of 25 milliseconds, optionally for a longer duration. The on/off keying cadence of the frequencies of Groups II and III doubles as a means of conveying the basic pulse rate from the out-station to the central station. The maximum transmission speed amounts to 20 characters/second.

As long as the idle-circuit state prevails, the frequency band occupied by Frequency Groups II and III is not utilized for transmission by the out-station, so that it is available for transmission in the opposite direction (return signals).

The return channel is an F.M. channel using the frequencies 1620 c/s and 1740 c/s.

Preference should be given to the following three return signals :

1. *Go-ahead-signal*, characterized by the 50-ms transmission of the 1620-c/s frequency. It signalizes to the out-station the ready-to-operate condition of the central-station receiver and, if sent at the end of a data message or a data block, serves at the same time as an acknowledgement signal attesting to error-free reception.

2. *Debarment signal*, characterized by the transmission of the 1740-c/s frequency for at least 100 ms. It blocks the transmission of data from the out-station and is placed on the line for such a time as the central-station receiver is not ready to receive. The debarment signal is followed by the go-ahead signal. If the debarment signal is placed on the line at the end of a message or a block, this is evidence to the out-station that this message or block has been received in an errored condition and has to be repeated.

3. *Ringback signal*, characterized by the alternate transmission of the frequencies 1620 c/s and 1740 c/s in a 25-ms rhythm. This signal is to invite the out-station operator to cut over from the data mode to the voice mode.

When transmitting data from, say, punched cards, punched tapes, or from the keyboard, the modulator will have to be preceded by suitable coders.

Motivation of the proposals

An essential portion of the modem circuitry is devoted to the generation and selection of the individual frequencies. The frequency spacing should be as wide as possible to allow great tolerance with respect to frequency stability and frequency accuracy. The number of frequencies used should be kept to the minimum consistent with the three-times-one-out-of-four code pattern, i.e. to 12 frequencies. On the other hand, due regard must be had to the signalling frequencies employed by the telephone switching facilities. The use of the C.C.I.T.T.-recommended frequencies for multi-channel V.F. telegraph transmission systems would appear to be a sound approach. The gap between the highest frequency of Group I and the lowest one of Group II facilitates the selection

of the level pilot frequency in the central station ; such a gap is not required between Groups II and III. The relatively wide frequency spacing permits the use of low-priced filter elements in the central station, whose transients are appreciably shorter than the signal duration of 25 ms. The requirement of providing filter elements for each discrete frequency (12 send filters) in the out-station can therefore be waived, which makes for the desired simplicity of the out-station equipment.

The pilot-controlled sensitivity reduction of an amplifier common to all of the 12 frequencies tends to diminish degrading echo effects on long circuits. If the residual level variations are evened out in the single-channel receivers through level controls featuring relatively long time constants, a virtually full suppression of echo effects is possible.

Supplement No. 56

STANDARD ELEKTRIK LORENZ AG. — *(Extract from contribution COM Sp. A/No. 78 — July 1963)*

PARALLEL TRANSMISSION ON SWITCHED TELEPHONE CIRCUITS

1. *General*

The use of parallel transmission is economical in cases where relatively small amounts of data are to be gathered from various out-stations. In this application there is no need for an extremely efficient use of the bandwidth available, as speed requirements are rather low. Cheapness of the out-station, however, is mandatory. This implies the use of mechanical peripheral equipment. Very often it is required to add data manually by operating a keyboard.

Parallel transmission offers mainly two advantages in this case :

i) no parallel-series converters are required at the out-station ;

ii) no accurate time-base is required, as the timing information can be transmitted separately.

The necessity to use more filters does not outweigh these advantages, as, especially in the out-station, very simple filters will do. This is true as the bandwidth need not be used very efficiently and as all parallel channels are keyed simultaneously.

The main characteristic which must be standardized in order to allow international co-operation are the following :

I) transmission scheme, i.e. the correspondence between the signals representing a character and the line frequencies ;

II) value of the line frequencies ;

III) maximum modulation rate ;

IV) interface between modulator respectively demodulator and data terminal equipment.

The following proposals are made for these characteristics.

2. *Transmission scheme*

The input equipment will consist in most cases of either a punched-card reader or of a punched-tape reader, perhaps with an additional keyboard. The requirement for economy excludes the use of complicated code-converters, which could convert the card-code or the tape-code into a trans-

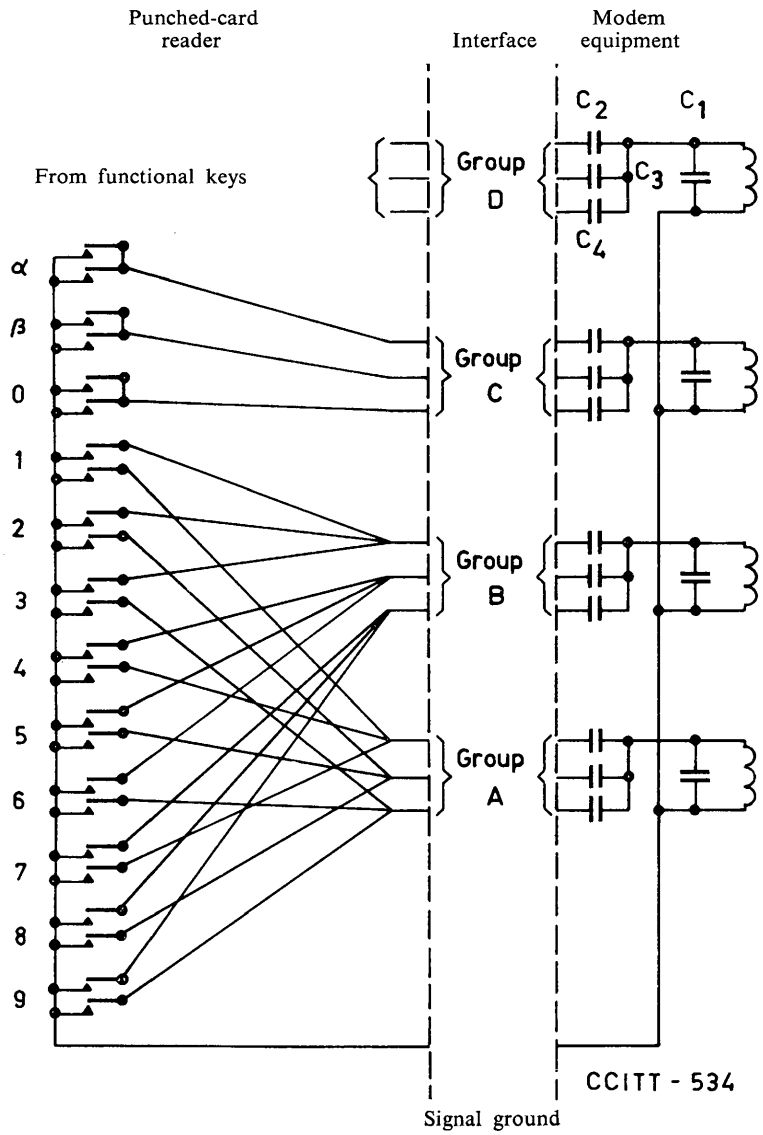


DIAGRAM 1. — **Out-station equipment for transmission of punched-card information**

mission-code to be used for both types of input equipment. On the other hand it is undesirable to provide different types of parallel modems for different input equipment.

Bearing this in mind in the following, a transmission scheme is proposed which allows the design of modulators providing a universal means for parallel mode data transmission, the data source being a reader for punched cards, a reader for tape or a keyboard.

The transmission frequencies are arranged in four groups. A character is represented by the transmission of one-out-of-four frequencies in each group.

The modulator receives the data signals on twelve leads, three for each group of frequencies, when operated from a card reader. Only one lead in each group is marked to represent a character. When no lead is operated, the highest frequency of the respective group, the rest frequency, will be transmitted. (See Diagram 1.)

When operated from a tape reader the modulator is controlled by only 8 leads, namely two for each frequency group. (See Diagram 2.) The four frequencies corresponding to the four possible

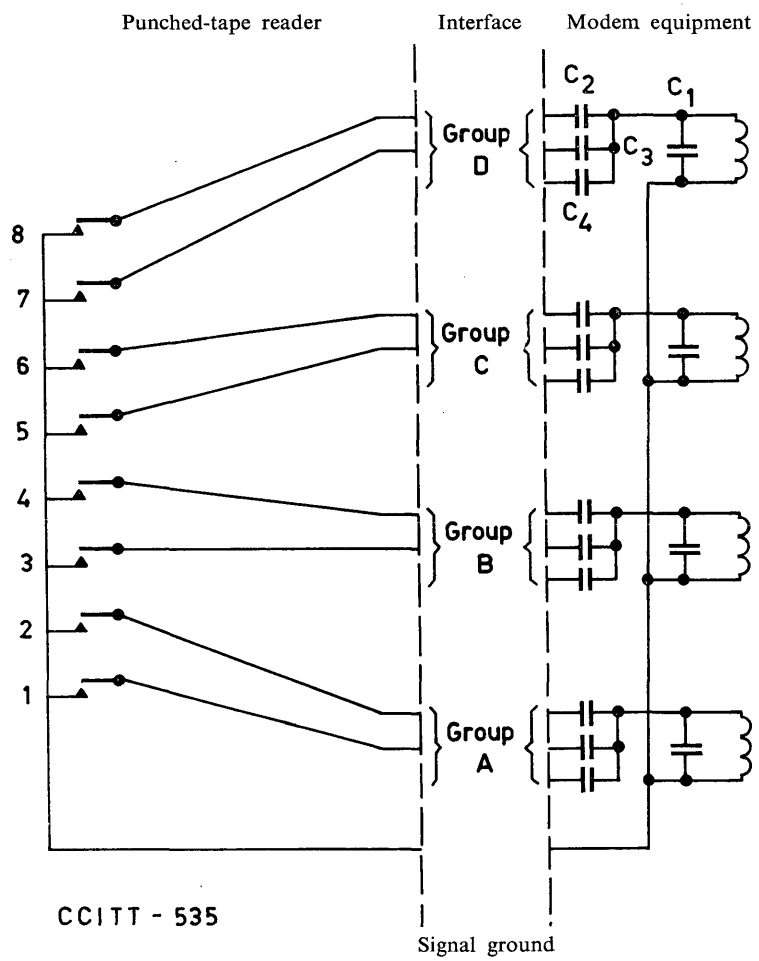


DIAGRAM 2. — **Out-station equipment for transmission of punched-tape information**

states of the two leads are determined by the following capacities :

00 : C_1
01 : C_1, C_2 parallel
10 : C_1, C_3 parallel
11 : C_1, C_2, C_3 parallel

The capacitor C_4 is not used in this case. The frequency error due to the fact that the lowest frequency is now determined by $C_1 + C_2 + C_3$ (with punched tape operation) rather than by $C_1 + C_4$ (with punched-card operation) can be kept in the range of $\pm$ 5 c/s.

Diagram 3 shows as an example the simplified block schematic of a demodulator for punched-tape operation. The frequency groups are separated by filters and amplified separately in limiter-amplifiers. In the discriminator the frequency transmitted in the respective group is detected and the corresponding output is marked. The discriminator outputs are combined by a simple logic which controls eight output relays.

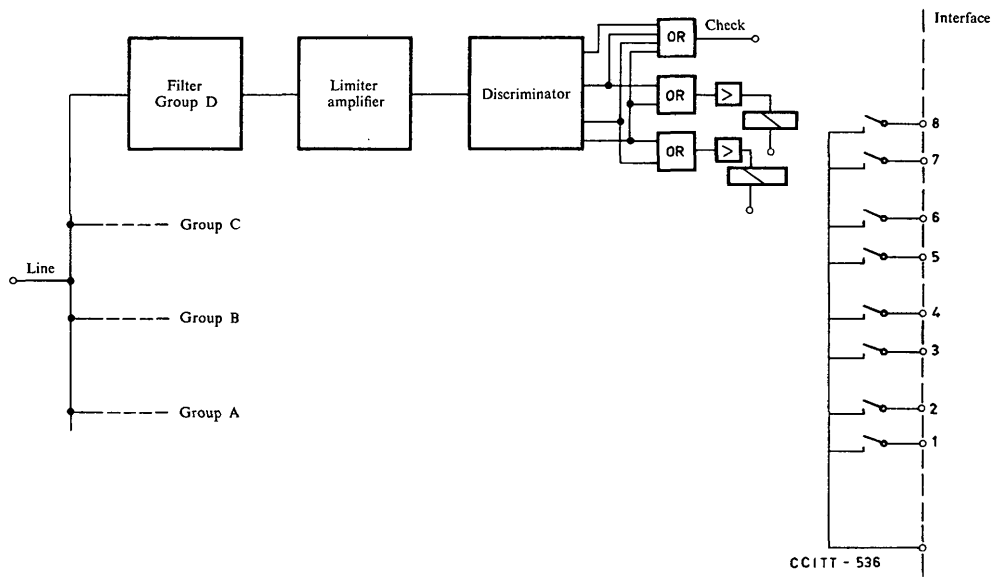


DIAGRAM 3. — **Centre-station equipment for punched-tape information**

With punched-card operation only the logic circuits after the discriminator outputs and the number of relays have to be changed.

The simplest way to achieve timing information in the case of card transmission is by return to rest frequency in all groups between characters which will allow for a maximum modulation rate of about 25 characters/second. For punched-tape transmission an optional timing channel can be provided in the middle of the voice band between two of the frequency groups. In this case the maximum modulation rate is about 50 characters/second, because no pause need be left between characters.

As an option a frequency shift channel is foreseen as return channel signalling at a maximum speed of 150 bauds. A frequency shift of $\pm$ 60 c/s is proposed at a mid-frequency of 480 c/s.

3. *Line frequencies*

The limited modulation rate makes possible that all 16 frequencies are contained within the range 900-2100 c/s which is desirable with regard to signalling receivers used on telephone circuits in the United Kingdom. The following frequency groups are proposed :

Group A	Group B	Group C	Group D	Optional timing channel
930 c/s	1230 c/s	1590 c/s	1890 c/s	
990 c/s	1290 c/s	1650 c/s	1950 c/s	
1050 c/s	1350 c/s	1710 c/s	2010 c/s	1470 c/s
1110 c/s	1410 c/s	1770 c/s	2070 c/s	1530 c/s

Tone is present in all groups as long as the out-station is in the transmitting mode.

4. *Interface*

The data signals are accepted at the out-station in the form of contact closures between respective data-leads and a common lead for all groups. At the centre station signals are delivered in a similar manner. Offering contact closures rather than voltage potentials allows for most economical solutions, which is the predominant factor in the application of data gathering.

Supplement No. 57

AMERICAN TELEPHONE AND TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A/No. 75 — July 1963)*

RECENT DEVELOPMENTS IN LOW-SPEED PARALLEL DATA SETS

Introduction

The "two-out-of-eight" and "three-out-of-twelve" parallel data transmission systems, described in the *Red Book* (Vol. VII, p. 372), have evolved into the present "three-out-of-fourteen" system. This system, exemplified by data sets 401E (transmitter) and 401J (receiver), is commonly used for the transmission of information read from punched cards.

Frequency assignment

The system uses 14 frequencies within the voice band. They are divided into three groups, two groups of five frequencies each and one group of four frequencies. The lowest frequency of

each group is designated a "rest" (or "null") frequency. The lowest frequencies are designated "data" frequencies. The frequencies and their designations are as follows :

Frequency	*Designation*	*Group*	*Type*
600	A_0	A	REST
697 [a]	A_1	A	DATA
770 [a]	A_2	A	»
852 [a]	A_3	A	»
941 [a]	A_4	A	»
1098	B_0	B	REST
1209 [a]	B_1	B	DATA
1336 [a]	B_2	B	»
1477 [a]	B_3	B	»
1633 [a]	B_4	B	»
1950	C_0	C	REST
2050	C_1	C	DATA
2150	C_2	C	»
2250	C_3	C	»

[a] Indicates frequencies which are the same as those used in touch-tone calling, a system of multifrequency push-button "dialling" from telephone and teletypewriter stations.

The transmitter, as in earlier systems, has one oscillator per group, in this case three. There is an interface lead for each *data* frequency. When one data lead in a group is closed to the *data common* lead, the oscillator of that group is tuned to the corresponding *data* frequency. If no data lead is closed in a group, the corresponding oscillator is residually tuned to the rest frequency of that group.

Since the interface leads are connected directly to the oscillator-tuned circuit, the capacitance of the encoder-to-data set cable is significant. The tolerances, however, are quite broad and a cable length of 6 feet is generally used.

An additional pair of leads, designated "keying", energizes all three oscillators, resulting in the simultaneous transmission of three frequencies. This circuit is closed at the beginning of the transmission of a series of digits, or at the beginning of each digit, as appropriate.

The receiver has detectors for all 14 frequencies, including the "rests". No retiming of the signal is provided ; each detector closes a relay contact for the duration of the associated frequency.

Coding systems

Two basic coding methods have been used :

A. In this method the "all rest" combination (A_0, B_0, C_0) ,with all data leads open, is used as an intercharacter separator, or "space". Each character is represented by closing *one* data lead in *each* group, sending a frequency combination such as A_2, B_1, C_3. This gives an alphabet of $4 \times 4 \times 3$, or 48 characters. Any received character not having a single output in each group is non-valid. A transmission error will generally result in one or two groups having no output, or perhaps multiple outputs. Any failure of a coding contact to close will result in the corresponding group having a "rest" output at the receiver. Thus, coding equipment errors may be distinguished from transmission errors.

B. In this method, the "all rest" combination (A_0, B_0, C_0) is also used as an intercharacter separator. A character, however, is formed by closing a data lead in one, each of two, or each of three groups. Thus, one or two "rest" frequencies may remain in a character. Typical characters under this system might be :

$$A_1,\ B_1,\ C_2\ ;\ A_2,\ B_2,\ C_0\ ;\ A_0,\ B_0,\ C_1.$$

This system provides 99 distinct combinations : the use of all 14 frequencies gives $5 \times 5 \times 4 = 100$ combinations, less one for the intercharacter separator A_0, B_0, C_0.
Note that transmission errors may still be detected, since any valid character will have one and only one output per group, a "data" frequency in at least one group.

Coding contact failures will often go undetected, however, as the resulting false character, containing a "rest" frequency, will appear valid.

This *second* method is often used for transmitting information read from I.B.M. "Hollerith" alpha-numeric tabulating cards. These cards represent a letter by two punchings, one called the "zone" and one the "numeric" portion. A numeric digit is represented by a single hole in the "numeric" area of the card. A single punch in the "zone" area of the card is used to represent certain "special" characters.

Each hole in the numeric portion of the card closes two contacts, one to an "A" data lead and one to a "B" data lead. Each A-B pair thus represents a "numeric" hole position. Each zone hole closes a "C" data lead contact. An alphabetic character is thus represented by three "data" frequencies. A numeric character is represented by a pair of data frequencies in groups A and B accompanied by C_0. The C_0 represents "no zone", the mark of a numeric character, and results from the lack of a contact-making hole in the zone area of the card.

A "zone-only" special character is represented by A_0, B_0, and a C-group data frequency, since no contacts are closed in the numeric area of the card.

If the system is used with numeric-only card equipment, the C-group circuits are unused at the transmitter and receiver. Frequency C_0, continuously on the line, has no effect.

Echo protection

Any receiving-end echoes are rendered impotent by the presence of the "rest" frequencies after each character. Since the echoes are reduced in amplitude, they are unable to "capture" the receiver from the rest frequencies, and hence cannot produce false multiple outputs. It must, however, be ensured that the transmitting oscillators keep running for at least 100 ms after the data contacts of the coding device open. When a manual-entry keyboard is used, the keying contacts operate with each keystroke, and it is difficult to hold them closed for the required interval after each key is released. For this reason, an R-C network in the transmitter keying circuit is provided which causes the oscillators to persist in operation for at least 100 ms after the keying circuit is open. Thus, "data" and "keying" contacts may be opened simultaneously, yet an interval of A_0, B_0, C_0 will follow to suppress the effect of any receiving-end echoes.

Data-set features

The maximum speed of the system is 20 characters per second.

The receivers are equipped to answer automatically, allowing unattended operation at the receiving location.

The data set 401E transmitter is powered entirely by power from the telephone line. The receivers operate from 110-125 V a.c., 60 c/s power.

Each frequency is transmitted at a power of approximately −9 dbm, a combined power of approximately −4 dbm, slightly less than ½ mW.

As in the earlier systems, the receiver can return either a 1017- or 1785-c/s tone for acknowledgement. Presently, these tones are heard audibly in a loudspeaker in the card reader, driven by a limiting amplifier in the transmitter. Simple time codes are used to convey various signals to the transmitting attendant. This is referred to as "audible answer-back".

Transmitters are now under development which will receive these two "answer-back tones" and close contacts to control mechanized card reading equipment, a feature known as "electrical answer-back".

Data sets 401A transmitter and 401B receiver were the first commercial sets in this family, and are still in extensive use for the transmission of numeric data. The 401A transmitter was essentially identical to the 401E except that no "C" group oscillator was used. The 401B receiver was similar to the 401F, but had detectors for only the eight data frequencies in group A and B. No "rest" frequency detectors or C-group detectors were provided.

The 401B receiver included a character re-timing circuit. This circuit sampled the output of the detectors 10 ms after the arrival of either of the two data tones which comprise a character. Output trigger circuits were set at this time and reset 25 ms later. Thus, output pulses are delivered whose duration is fixed, regardless of the length of the transmitted character. The "A" and "B" outputs are also coincident regardless of any difference in arrival times of the "A" and "B" frequency. Such difference could arise either from a time stagger, or "skew", of the "A" and "B" encoding contacts, or from propagation delay differences.

This timing was abandoned in the later receivers 401F and 401J. When a simple alpha-numeric card reader, operating at 10-12 characters per second, was used, the "skew" between A and B groups and the C groups can become large, due to the fact that two different card holes are acting. A 10-ms receiver sample delay might be insufficient to accommodate the received skew, yet a longer sample delay would not permit a more sophisticated reader to utilize the system at 20 characters per second.

The sample delay is now set by the customer's receiving equipment, where it may be optimized to the skew and speed which are applicable.

Data sets 401E transmitter and 401F and 401J receivers are equipped with arrangements for remote testing. In the case of the 401E, two non-locking keys are provided. The customer operates these two keys in succession when in contact with a data test centre. During the process the lowest and highest frequency in each group is sent. The test centre can thus verify the correct operation of all three oscillators.

In the receivers, a test key is provided which "primes" the set for a test condition. A test call placed from the data test centre is automatically answered, and an interval of one answer-back tone returned for measurement. The test centre may then send each of the 14 frequencies. Operation of the detectors is verified by signals returned as bursts of the other answer-back tone. The test condition is automatically cancelled when the test call is released.

Future programme

Experimental work has been done on the use of a "spoken" reply to inquiries presented to a computer by 3-out-of-14 coding. The computer selects or assembles the spoken reply from a magnetic recording device. The data set receiver is equipped with a "coupler" which applies the reply to the telephone connection at a controlled level. The data set transmitter is arranged to present the reply to the caller either through a loudspeaker or through the telephone handset receiver.

This "data inquiry-voice answer" ("DIVA") type of operation allows complex replies to be received from a computer without the expense of a data receiver or display device at the inquiring station.

It is planned that in the future it will be possible to use a touch-tone telephone set to transmit data. After establishing the call by means of the touch-tone "dial", data can then be sent to a receiver similar to the 401B using the same push-button "dial". The compatibility between 401 and touch-tone frequencies will permit much flexibility in the use of data set transmitters and touch-tone dials in the same system.

Economics of application

The various low-speed parallel data transmission systems have found wide use in data-gathering systems, particularly those having a multiplicity of sending stations. The extremely low cost of the 3-out-of-14 transmitter, and the simplicity which is possible in the associated encoding device, produce a highly economical system, even in the face of the moderate complexity required of the receiver. The "contact-type" interface, while not possessing the universality of the "standard EIA", or bipolar voltage, interface, contributes to this over-all economy of the sending station.

The error performance of the 401-type system is extremely good, with most of the errors which do occur being detectable as described above.

Supplement No. 58

AMERICAN TELEPHONE AND TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A/No. 75 — July, 1963)*

A DATA SYSTEM FOR UP TO EIGHT-CHANNEL PARALLEL OPERATION AT RATES UP TO 75 CHARACTERS PER SECOND ON VOICE BAND CIRCUITS

1. *The system*

Data sets have been developed to provide a binary, parallel data transmission system for use in service over the switched telephone network or in leased line service. The system will transmit any number of data levels up to 8 at any speed up to 75 characters per second. Timing signals are

Channel	1	2	3	4	T	5	6	7	8
Mark	730	900	1070	1240	1410	1580	1750	1920	2090
Space	800	970	1140	1310	1480	1650	1820	1990	2160

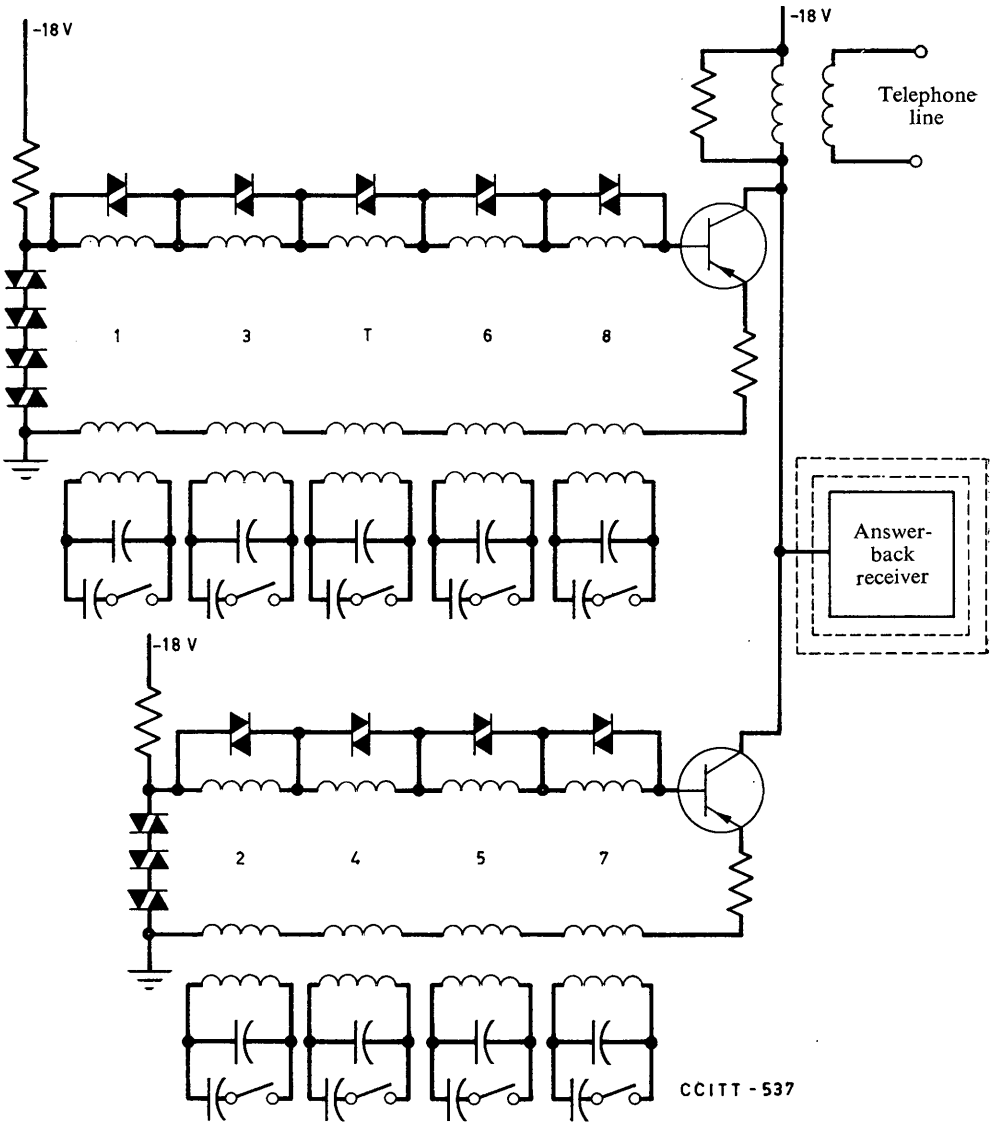


FIGURE 1. — **Data transmitter**

also transmitted. A non-simultaneous answer-back feature is provided at a maximum speed of 20 two-bit characters per second.

The parallel system offers those advantages arising from the facts that parallel-serial conversion is not required and that a timing output is supplied to the receiving business machine. The simplicity and resultant low cost of the transmitter makes the system particularly attractive for data-gathering applications.

All of the signals interchanged between the business machines and the data sets are in the form of contact closures.

An arrangement can be made for using a transmitter and a receiver alternately on the same line.

This system does not provide eight independent channels such as would be available through eight channels of voice-frequency carrier telegraph equipment. The economy of the receiver system is derived through use of the knowledge that condition changes of the data channels have a definite relationship to changes of condition of the timing channel, thereby permitting the data channels to be sampled at an optimum time.

The contact type interface is employed in the system because of its simplicity, economy, and ease of application for relatively unsophisticated business machines.

The parallel family of data sets includes very cost-sensitive systems designed for the transmission at 20 characters per second of two out of eight or three out of fourteen possible character inputs. This system lends itself to economical transmission of data that are in Hollerith Code such as found on tab cards. The 2-out-of-8 and 3-out-of-14 systems are discussed in Supplement No. 56. The medium-speed system is intended to meet the requirements of very cost-sensitive systems that require no format limitation (such as 2 out of 8 or 3 out of 14) on the transmission of up to 8 levels of parallel data or where rates of greater than 20 characters per second are needed. These requirements result when the data to be transmitted are in paper-tape form or from direct keyboard outputs and for transmission of data from tab cards at the higher rates.

2. *Equipment and circuit description*

2.1 *Transmitter*

The transmitter data set generates nine frequency shift channels (see Fig. 1). It is composed of two multifrequency oscillators, one generating five frequencies and the other four. Two oscillators are used to allow wider spacing between channels in the same oscillator and reduce the tendency of one channel to pull another. The frequency allocation is the same as that which has been used in 43A1 voice-frequency carrier telegraph systems for some time. The assignments are as follows :

Data transmission mode

Channel	Frequency	
	Mark	Space
1	730	800
2	900	970
3	1070	1140
4	1240	1310
Timing	1410	1480
5	1580	1650
6	1750	1820
7	1920	1990
8	2090	2160

Answer-back mode

Condition	Frequency
A	1017
B	2025
A B	1785
Neither	1152

The transmitter data set contains an answer-back receiver to receive a signal chosen on a one-out-of-four basis. One of the four frequencies will be a rest frequency which saturates the limiter without actuating any relays. Two of the remaining three frequencies will operate one relay each and the fourth frequency will operate both relays.

2.2 *Receiver*

2.2.1 *General*

A block diagram of the receiver data set receiver is shown in Figure 2. It controls eight output relays on the basis of the data on the frequency shift channels. After the signal from the telephone line passes through an automatic volume control circuit, the channels are separated by channel separation filters. A discriminator in each data channel detects whether mark or space was sent. The timing signal also passes to a discriminator and the timing channel discriminator output is sliced to give a rectangular wave. A pulse is generated at each transition of the timing signal and delayed. The delay is set so that a pulse occurs at the centre of the received character. This delayed pulse is used to sample the data channel discriminators and set the appropriate output flip-flops which operate the output relays. The answer-back transmitter generates one-of-four frequencies under control of two binary inputs.

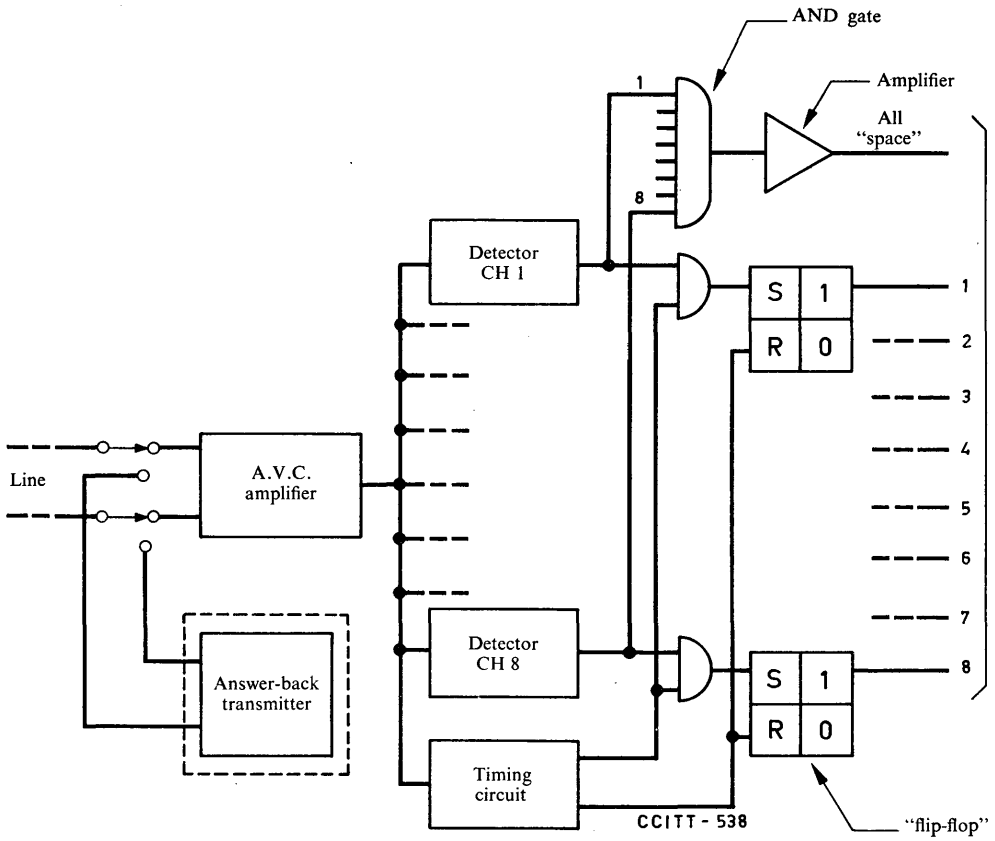


FIGURE 2. — **Data receiver**

2.2.2 *Automatic volume control circuit*

The received level of the data signals can vary over a large range. However, an attenuation of 35 db with respect to the transmitted level is rarely exceeded.

Even for a given connection between the transmitter and the receiver, the levels are variable with time, although this variation does not exceed a few db. An automatic volume control circuit (AVC) has been designed for the system which can correct for the expected level variations.

The AVC circuit is capable of maintaining an output constant within 3 db for an input varying from −6 dbm to −36 dbm.

The output amplifier of the AVC has a very low output impedance and drives the eight data channels and the timing channel.

2.2.3 *Timing channel circuit*

The timing channel receives a signal which alternates between mark and space. The transitions in the timing input correspond to the time at which data characters begin. The function of the timing channel is to generate the pulses used to sample the data channels. Since the system is insensitive to transmission rate, the timing channel generates appropriately time pulses for all speeds providing only that transitions are separated by at least 13.3 milliseconds (corresponding to the 75-character-per-second maximum speed).

The timing channel (Fig. 3) is driven by the AVC. The input circuit is a band-pass filter with high attenuation for all frequencies except those of the timing channel. The filtered signal is amplified, limited, and applied to a discriminator channel.

The discriminator gives a d.c. output voltage centred around ground when the timing frequencies are correct. A carrier shift in the transmission medium will cause a d.c. shift of the discriminator voltage. Satisfactory operation of the bistable slicer can be achieved by not operating on the basis of absolute d.c. voltage, but on the (a.c. coupled) voltage shift caused by the transition of the timing channel.

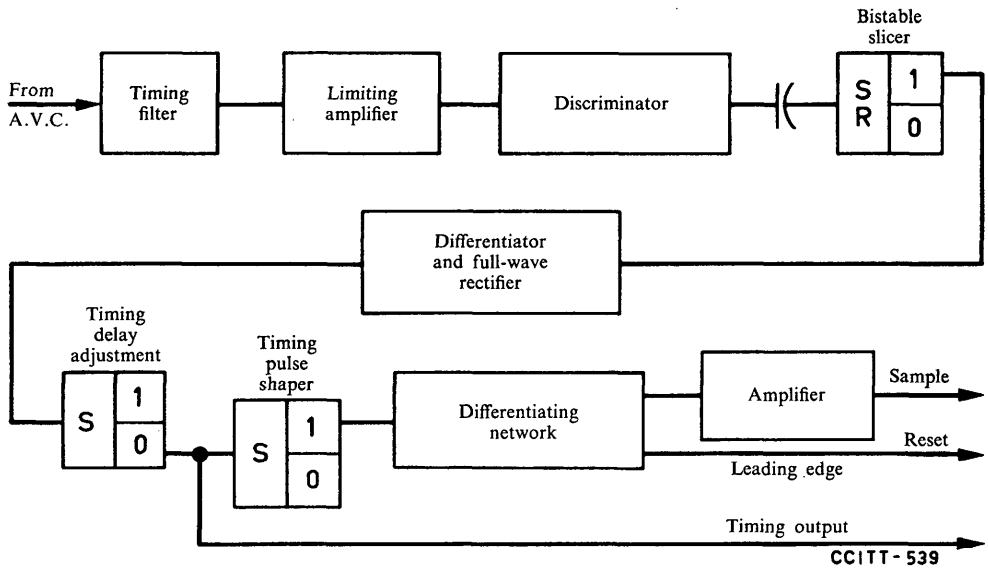


FIGURE 3. — **Timing channel block diagram**

Since a sampling pulse at every transition of the timing channel is required (Fig. 4) the rectangular wave from the bistable slicer is differentiated and full wave rectified. The pulses obtained do not necessarily appear at the optimum time for the sampling of the data channels. A monostable flip-flop produces the necessary delay to cause sampling at the centre of the character interval. Another monostable flip-flop generates the proper timing pulse wave shape.

The timing pulse has to perform a double function in the data channels. The timing pulse is differentiated, and the leading edge resets output flip-flops which clears the preceding character from storage. The lagging edge is used to sample the data channels and store the new character in the output flip-flops. The time interval between leading and lagging edges is approximately 100 microseconds.

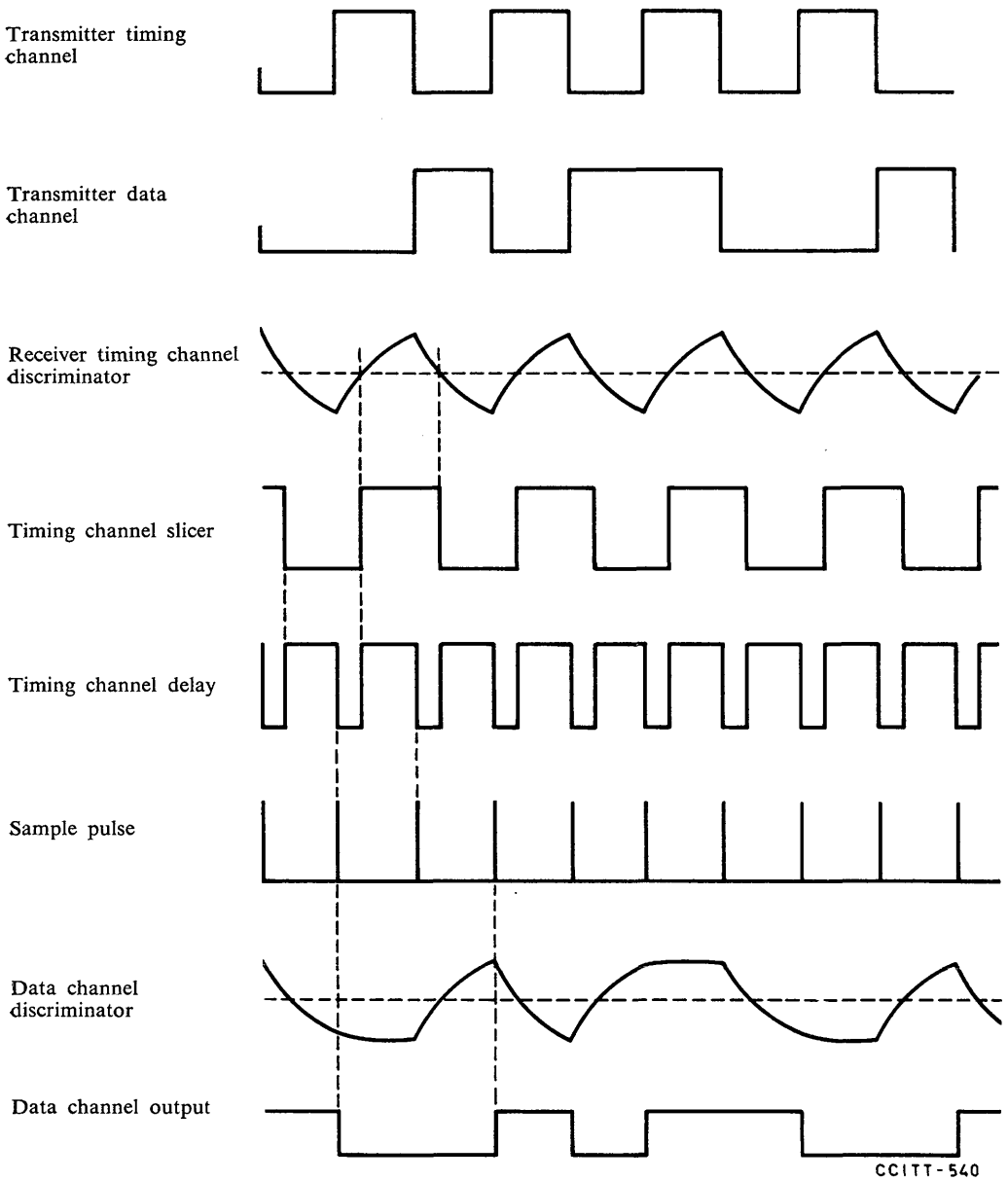


FIGURE 4. — **Waveforms and time relationships**

2.2.4 *Data channel circuits*

The channel filtering of the data channel circuits is simpler than that of the timing channel. Attenuation of the next adjacent channels is obtained by two parallel tuned circuits in the emitter

of the data channel amplifier. Attenuation of the channels separated further in frequency is obtained by the bandpass characteristics of the discriminator tuned circuits in the collector of the data channel amplifier.

The discriminator output voltage is AND gated with the sampling pulse from the timing circuits. If the discriminator voltage is negative, the AND gate output will set the flip-flop. If the discriminator voltage is positive the flip-flop will not be set. The flip-flop data store is clearly by the reset pulse from the timing circuit.

A relay is directly operated by the flip-flop. When two successive mark signals are received in the data channel, the arrangement of reset and sample pulses will cause the flip-flop to be reset for a duration of 100 microseconds between the two set conditions. This time is too short to cause action of the relay.

2.2.5 *All-space detector*

When signals are absent at the input of the data receiver, noise may cause operation of the data machine. This problem can be solved by disabling the data machine until a special start signal has been given.

This start signal consists of a space signal in all data channels persisting for several character intervals. This signal will activate the all-space detector. This information can be used by the customer to reactivate the data receiver.

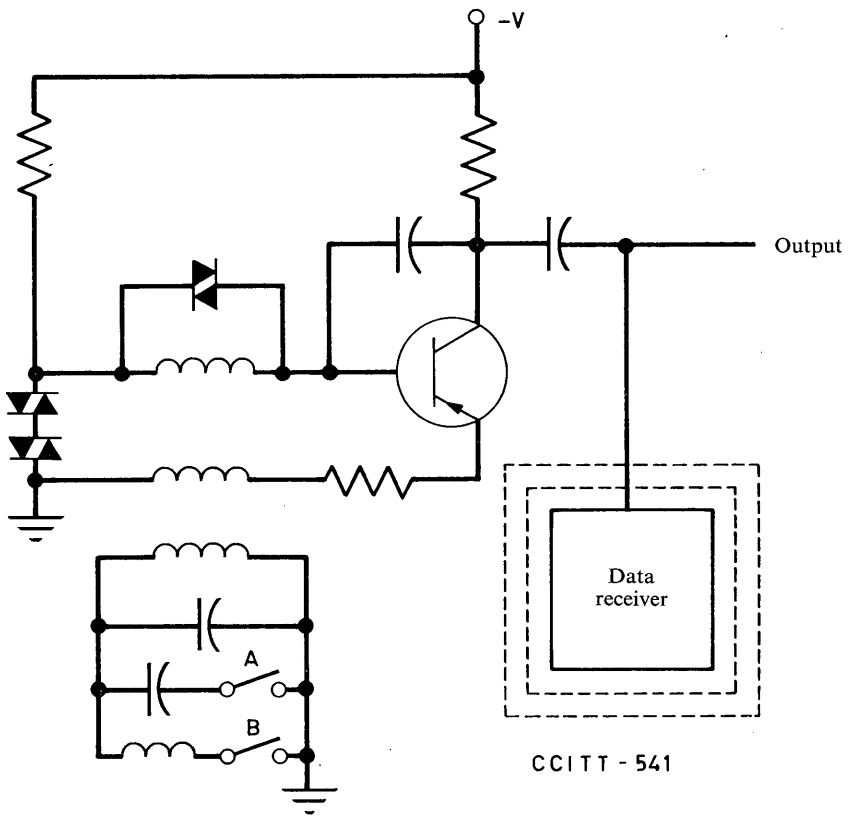
2.2.6 *Answer-back transmitter*

The answer-back transmitter (Fig. 5) is an oscillator which generates one of four frequencies under control of two relays. The answer-back signal is transmitted at −6 dbm. The transmitter can be transferred to the telephone line under control of the customer. When the answer-back transmitter is connected to the telephone line, the data receiver is disconnected. The customer controls the oscillator by grounding lead A or B (both may also be used). At the transmitting terminal the answer-back receiver will close the corresponding contact.

2.3 *Transmission characteristics*

The received level of the data signals can vary over a wide range and the automatic volume control amplifier is capable of maintaining an output, to the timing and data channels, constant within 3 db for an input signal varying from −6 db to −37 db. Furthermore, the AVC circuit can respond to a sudden change of 10 db without causing an error. Use of the sets should be limited to applications in which the over-all 1000-c/s loss, transmitter to receiver, is not expected to exceed 36 db.

Attenuation-frequency slope correction is built into the data set 402B. With the correction, the set will tolerate a 2200-c/s loss up to 7 db greater than the 1000-c/s loss, and a 700-c/s loss up to 3 db greater than the 1000-c/s loss. Envelope delay distortion is not expected to be a serious consideration in the use of the set, as it can tolerate 1500 microseconds or more within the 750-2200-c/s frequency band.



Switch closed	Output frequency
None	1152
A	1017
B	2025
Both	1785

FIGURE 5. — **Answer-back transmitter**

3. *Miscellanea*

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3.2.4 *Data channels*

The data channels are independent of one another at the transmitter so long as each delivers its intercharacter space-mark or mark-space transitions in accordance with the following:

1) The transitions at the beginning of a character must occur no later than 2 ms after the timing transitions for that character (see Fig. 8).
2) The transitions at the end of a character must occur no sooner than 12.3 ms after the timing transition signifying the beginning of that character.

Note that 2) is, in effect, the limiting factor on data transitions applied before the timing transition.

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3.4 *All-space*

When signals are absent at the input of the data receiver, noise may cause operation of the data set. Whenever the business machines change the mode of the data sets from data to answer-back and vice versa, transients are likely to be created in the timing and data channels. The business machine should guard against these noise conditions through the use of a start-of-transmission signal. After a change from answer-back mode to data transmission mode the receiving business machine should be blinded to all signals until it receives a valid start-of-transmission code. The start-of-transmission code would, for this case, consist of a spacing signal in all data channels persisting for several character intervals. The length of the all-space signal that must be transmitted is determined by the presence or absence of active echo suppressors. If active echo suppressors are present, the business machine must send all-space for a minimum of 250 ms. If no echo suppressors are present, the business machine must send all-space for a minimum of 25 ms. This signal will activate the all-space detector in the data set receiver. The operation of the space detector circuit is independent of the timing channel. Where the presence of echo suppressors is unknown the 250-ms figure should be used.

4. *The data set*

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4.2.4 *Timing*

The timing transitions are used by the receiving data set to determine when the data channels are to be sampled. Business machines should be designed to overlook spurious timing transitions that occur on changes from one mode to another by using the all-space indications. In rare cases spurious timing closures may occur as late as 5 ms (staying closed for 5 ms) after the all-space indication has been given to the receiving business machine. Business machine designers may guard against this condition by delaying their sample of the timing channel until at least 5 ms after all-space has been received.

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5.2 *Telephone lines*

In order to maintain high-quality service and to minimize interference from related devices, it is preferable to use data sets on individual lines with no extension telephones connected to them. Likewise, the sets should not be used on key telephone systems.

6. *Operation*

6.1 *Establishing a call*

A call is established using the telephone associated with the data set in the normal manner and the data key down. When both transmit and receive parties are ready for data transmission, the data keys of both stations are then pulled up which connect the data sets to the line in place of the telephone sets. The data keys shall remain up for the duration of data communication.

6.2 *Data transmission*

Operation of the mode control contacts in the business machines will condition the sets for data transmission. At the start of each data transmission the all-space code must be transmitted by opening all of the data contacts. The all-space signal must be transmitted for a minimum of 250 ms if echo suppressors are present and for 25 ms if echo suppressors are not present.

6.3 *Answer-back*

The business machines can operate the mode control contacts to condition the data sets for answer-back transmission. The first answer-back character should be transmitted for sufficient time to turn around echo suppressors, if present (250 ms) and to operate the answer-back receiver (a minimum of 25 ms).

6.4 *Disconnecting*

Upon completion of data communication, voice communication may be re-established by restoring the data keys to the voice position (down). If voice communication is not desired the handsets can be replaced on the hook switches thereby restoring the data keys automatically and disconnecting the line.

Supplement No. 59

CHILE TELEPHONE COMPANY — *(Extract from contribution COM Sp. A/No. 90 — August 1963)*

PARALLEL MODE DATA TRANSMISSION

1. *General*

A proposed system for the low-speed transmission of data over telephone connections is described.

The system is adaptable to punched-tape and punched-card readers and to a keyboard, this adaptation being effected by insertion of the proper coding matrix. Data input leads are d.c. operated, ensuring safe contact functioning.

2. *Modulation method*

The frequency coding is of the type "two out of eight", "three out of twelve" or "four out of sixteen" depending on the number of frequency groups in use, the modulator transmitting two, three or four frequencies simultaneously—i.e., one frequency from each group of four frequencies.

The modem interface comprises three data leads for each group, corresponding to the three lower frequencies (01, 10, 11). The highest frequency, the rest frequency (00), is generated if no lead is operated. When operating from a card reader one, or no, lead in each group is marked to represent a character.

A tape reader operates two leads in each group, enabling an eight-track tape to be transmitted. Frequency selection is controlled by keying circuits, so arranged that simultaneous marking of these data leads, which individually correspond to the two mid-frequencies (01, 10), gives the lowest frequency (11). At the receive side this is decoded into the original two markings.

Timing information is achieved in two ways. The simpler, which is to return to the rest frequency in all groups between characters, allows a maximum modulation rate of about 25 characters/second. Synchronous operation, with no spacing between characters, allows modulation rates up to about 50 characters/second ; this requires a frequency-shift timing channel using the shift frequencies 1470 c/s and 1530 c/s.

A half-duplex or full-duplex mode of operation may be used, the backward channel being either a frequency shift ($\pm$ 30 c/s) channel and mean carrier frequency of 420 c/s or, if an audible signal is required at the sending station, a 450-c/s tone. The backward channel is used to acknowledge or request repeats of complete blocks of data, e.g. a punched card. Full-duplex operation permits the use of continuous tone in the forward direction permitting very efficient error detection (see 4 below).

3. *Modulation frequencies*

The limited modulation rate allows all 16 frequencies to be contained within the range 900-2100 c/s, and the three groups used in the punched-card application within the range 900-1800 c/s. Numeric punched-card information can be transmitted by two groups of frequencies only in the range 900-1440 c/s. The following frequency groups are proposed :

Group A	Group B	Group C	Group D	Optional timing channel
930 c/s	1230 c/s	1590 c/s	1890 c/s	
990 c/s	1290 c/s	1650 c/s	1950 c/s	
1050 c/s	1350 c/s	1710 c/s	2010 c/s	1470 c/s
1110 c/s	1410 c/s	1770 c/s	2070 c/s	1530 c/s

Tone is present in all groups as long as the out-station is in the transmitting mode.

4. *Error-detection facility*

The receive equipment employs eight frequency-shift demodulators and individual modulation detectors. A pair of demodulators serves a frequency group, tone being present only in one or the other. The modulation detectors are used as AM detectors for the purpose of signal detection and can be designed also to give an alarm upon the detection of a signal disturbance. If suitable threshold levels are chosen, error detection can be performed with a very high efficiency. However, this calls for continuous tone operation because a slow-acting AGC amplifier must be used to compensate for varying receive levels.

5. *Interface*

Data signals are accepted at the out-station in the form of contact closures between respective data leads and a common lead. At the in-station signals are delivered in a similar manner. Functions affecting the telephone circuit are preferably incorporated in the modem equipments or in separate control units.

6. *Code conditions for transmission of punched-card information (Figures 1 and 2)*

6.1 *Conversion table*

Frequency c/s			
Condition	Group A	Group B	Group C
00	1110	1410	1770
01	1050	1350	1710
10	990	1290	1650
11	930	1230	1590

6.2 *Signalling conditions, alpha-numeric system*

Group A	Group B	Group C	Application
00	00	00	Rest condition
01, 10, 11	00	00	Numerals 1-3
00, 01, 10, 11	01	00	Numerals 4-7
00, 01	10	00	Numerals 8-9
00	00	11	Numeral 0
01, 10, 11	00	01, 10, 11	27 letters
00, 01, 10, 11	01	01, 10, 11	
00, 01	10	01, 10, 11	
11	10	00	5 functional characters
00, 01, 10, 11	11	00	
00	00	01, 10	21 conditions not used
10	10	00, 01, 10, 11	
11	10	01, 10, 11	
00, 01, 10, 11	11	01, 10, 11	

7. *Code conditions for transmission of punched-tape information (Figure 3)*

When adapted for punched-tape operation it is necessary, if the customer wishes to send complementary information from the keyboard, that the keyboard be coded in accordance with the code used on the actual tape system. For example, the coding would be changed when switching from, say, an eight-row tape to a six-row tape.

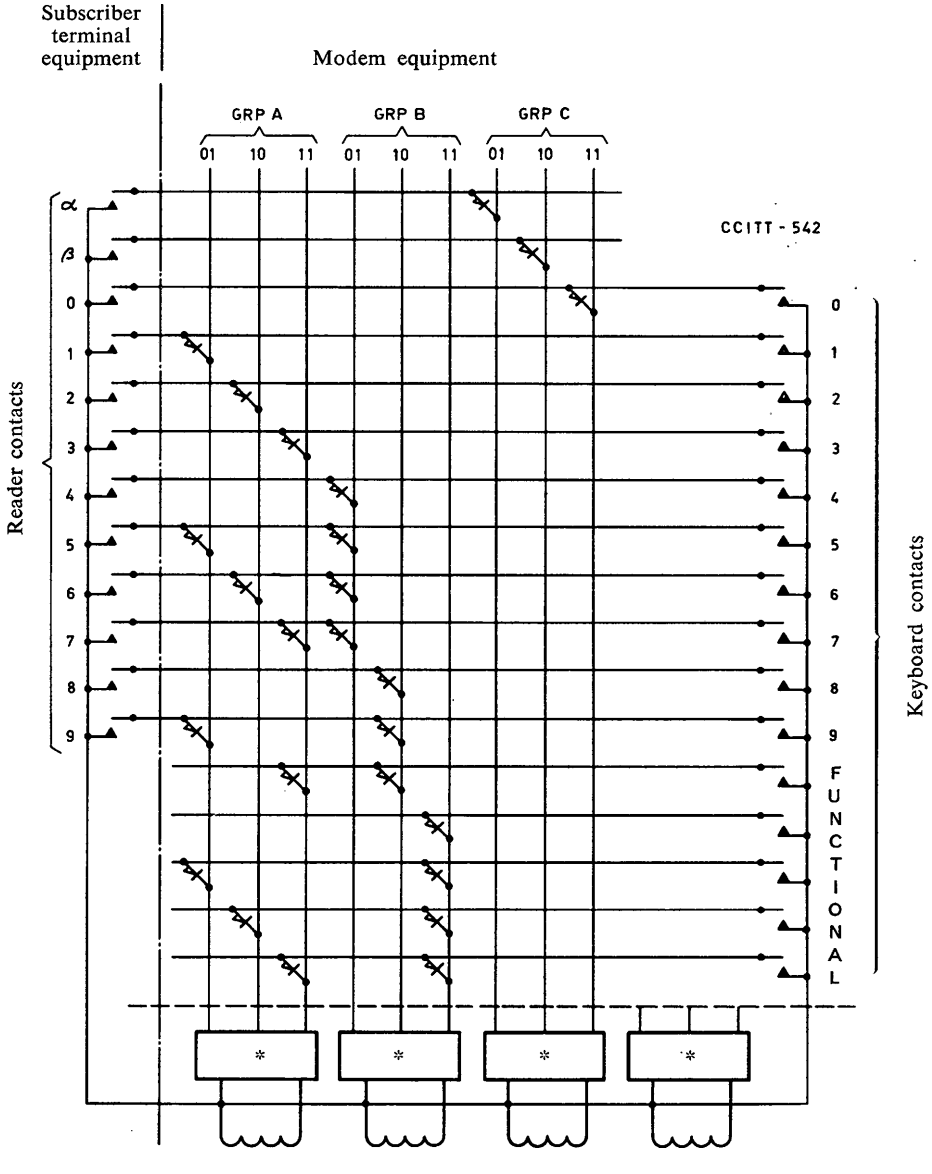
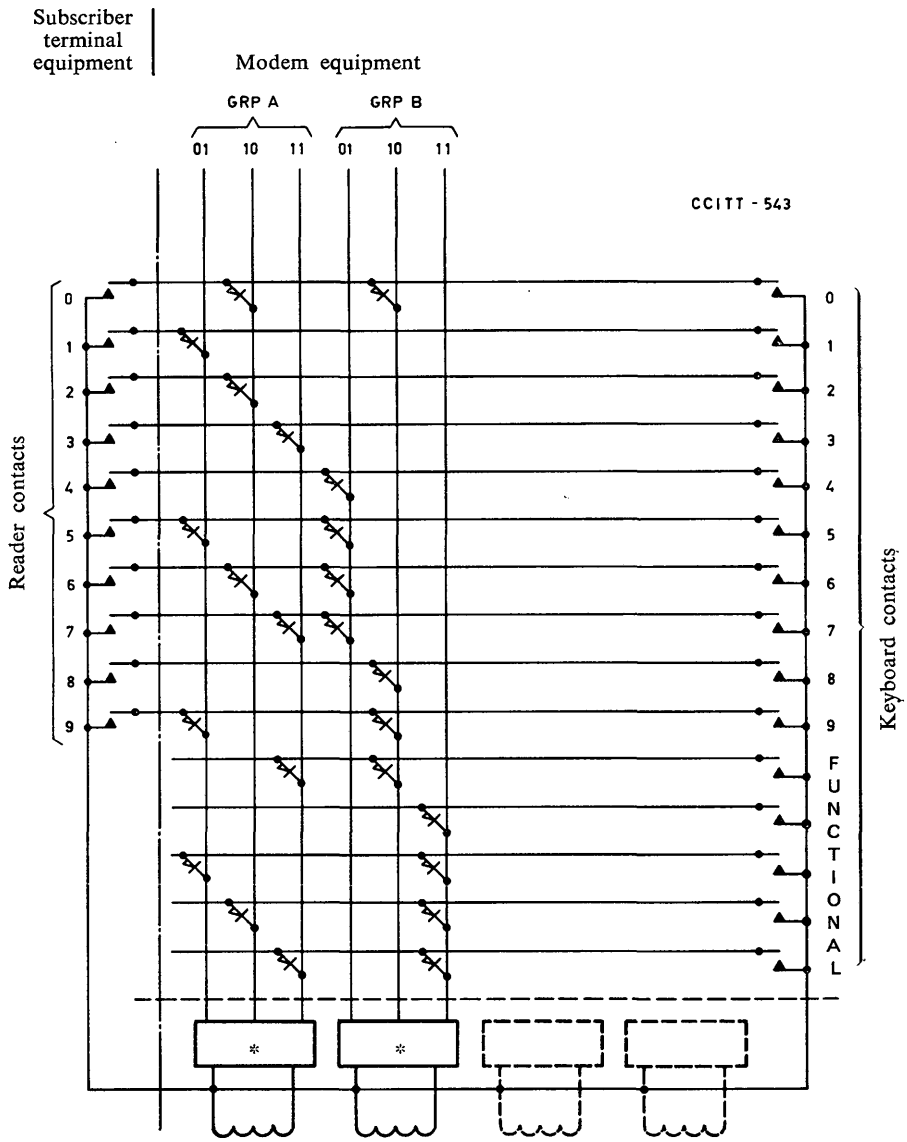
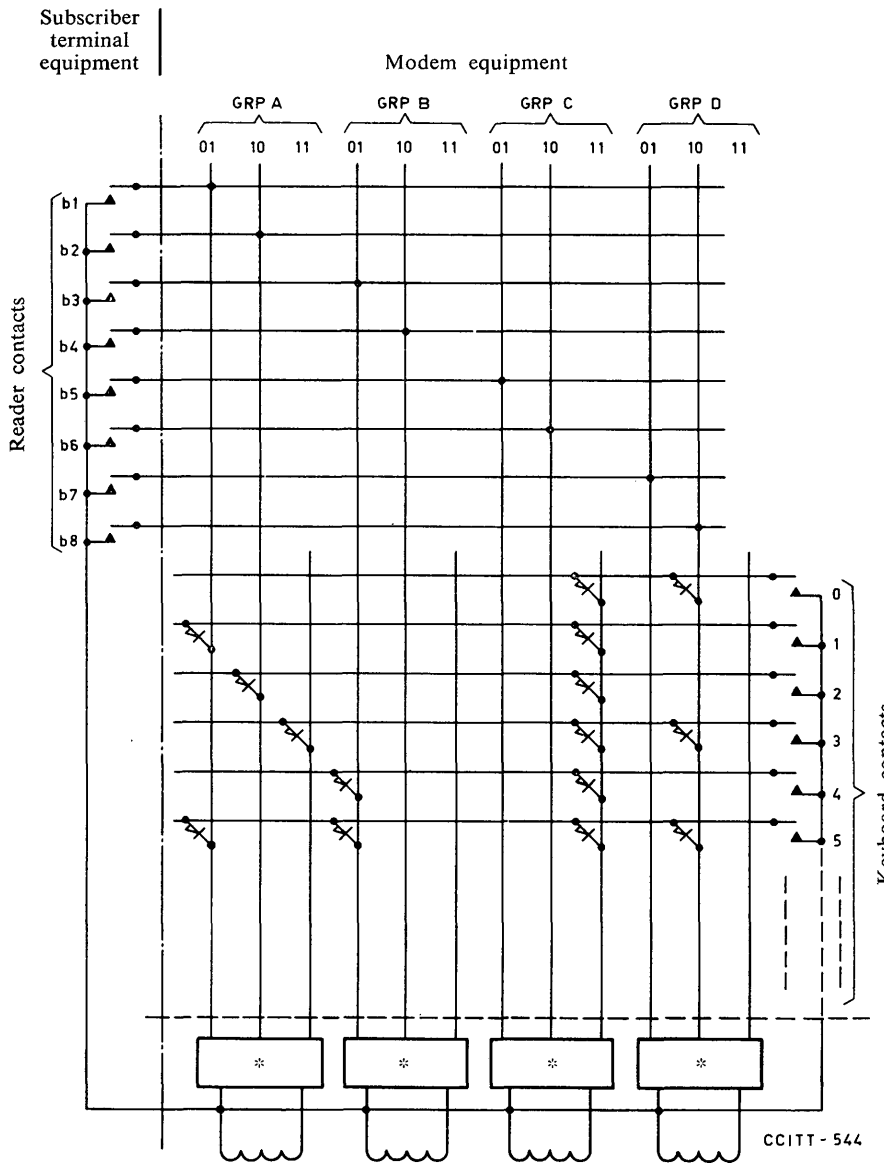


FIGURE 1. — **Out-station equipment for transmission of alpha-numeric punched-card information**



* Keying circuit.

FIGURE 2. — **Out-station equipment for transmission of numeric punched card information**



* Keying circuit.

FIGURE 3. — **Out-station equipment for transmission of punched-tape information**

Supplement No. 60

SPECIAL STUDY GROUP A — *(Extract from report COM Sp. A/No. 92 — October 1963)*

REPORT ON PARALLEL TRANSMISSION OF DATA

1. Special Study Group A tried to reduce parallel systems to two separate types.

These systems are primarily intended for use on national switched telephone networks, but their use may be extended for international transmission. It was not envisaged that these systems should be used for intercontinental operation, and therefore it does not seem necessary to provide means for disabling echo suppressors.

a) *Parallel data transmission system*

A system of data transmission for use at a maximum of 75 characters per second using typically up to four groups of four frequencies giving a maximum of 256 code combinations. This system can be used with a wide range of data-input machines and permits the transmission of up to eight unit codes.

b) *Parallel data collection system*

A very cheap system for data collection for use at a maximum of 20 characters per second using up to three groups of four frequencies giving a maximum of 64 code combinations. This system is limited to simple data-input machines, for example, a mechanically operated card reader or keyboard. It will permit the transmission of up to six unit codes.

Note. — Certain delegates expressed the view that it might be possible to meet both requirements with basically the same system.

c) The following table was prepared to compare the difference between these two basic types of system.

Both of these systems should have provision for timing or rest frequencies between or above or below the groups of data frequencies and for return channels if required.

System b) is primarily a low-cost system as it is intended for use for data collection from a number of out-stations into a central receiving station, so the cost of the out-station should be kept as low as possible. System a), however, is more sophisticated and its cost can be higher, being matched to its performance.

During the discussions another system using parallel/series approach was proposed whereby only a limited number of frequencies were used. If only limited code combinations were required these frequencies could be used directly. If, however, more code combinations were required, the characters could be sent in two parts. The Study Group thought that this system was of sufficient interest to be brought up for further study and a description of the system is given at the end of this Supplement.

2. *Frequency allocation (grouping)*

The Study Group felt that this should be held over for further study. The Administrations present (eight) were asked what frequency band was available for the systems under consideration. All the Administrations present, with the exception of the United Kingdom, said the band from about 400 c/s to about 2100 c/s was available. The United Kingdom said that at the present time

		System a)	System b)
Character set		$\leqslant 256$ or $\leqslant$ 8-unit code	= 64 (48)
Typical speed in characters/second		75	20
Power supply	(Transmitter)	Central battery if possible	Central or local battery
	(Receiver)	Mains supply	Mains supply
Timing channel		Optional (may not be necessary for lower speeds)	No, but may be necessary for higher speeds
Backward signals		Audible tones (but voice may be used) 75 bauds 2-condition Multi-state supervisory	Audible Acknowledgement
Typical data sources		Power-driven devices	Keyboard card reader and similar mechanically driven devices
Automatic answering	(Transmitter)	Optional	None
	(Receiver)	Optional	Optional

the band was restricted to 900 c/s to 2100 c/s with the additional restriction on frequency separation in the regions of 600 c/s and 750 c/s. The limiter in certain signalling receivers generates intermodulation products and this causes false operation. Italy said that the band from 400 c/s to 3000 c/s was also available. The Study Group wishes to ask Administrations their views on what bandwidths are available for parallel transmission.

3. *Interface*

The Study Group thought that consideration should be given to the interface between the modulator and the data-input machine.

4. *Line-feed current*

It is proposed that the modulators should be powered by the central battery in the telephone switching centre if possible. The Study Group wishes to ask Administrations and operating companies what minimum power is available to operate a simple modulator for parallel transmission.

5. *Speeds and timing*

A question that the Study Group would like to put up for study is that of the relationship of timing channels and the speed at which a system can send data.

6. *Output levels*

For further study, in the context of the revision of Recommendation V.2, possibly a different power level might be transmitted for each group of frequencies to assist equalization.

7. *Review of contributions received*

a) *United Kingdom* (Supplement No. 54)

The contribution from the United Kingdom is of a general nature. The first part compares parallel transmission with serial to decide which is economically more efficient in the effective utilization of bandwidth. It is concluded that, owing to parallel/serial conversion, the serial arrangement is more expensive at the lower speeds.

The second part discusses two methods of coding. One type which sends a fixed number of tones, say N out of a set of M tones, and the other which transmits up to a maximum of M tones out of a set of M tones. The first system restricts the number of tones sent to line and allows a higher level than if all tones were sent and also provides an inherent error-checking facility. The second system offers no error checking and the level of each tone is low. For the transmission of binary codes no code conversion is necessary.

b) *American Telephone and Telegraph Company* (Supplements Nos. 57 and 58)

Two systems are described : the "three out of fourteen" system and the 8-channel system for operation at rates up to 75 characters per second.

i) The first system has evolved from the "two out of eight" and "three out of twelve" parallel data transmission systems (*Red Book*, Volume VII, page 372). This system is commonly used for the transmission of information read from punched cards and keyboard.

The system uses 14 frequencies within the voice band. They are divided into three groups, two groups of five frequencies each and one group of four frequencies. The lowest frequency of each group is designated a "rest" (or "null") frequency. The higher frequencies are designated "data" frequencies. The lowest frequency is 600 c/s and the highest 2250. Some of the frequencies are those used in touch-tone calling, a system of multifrequency push-button "dialling" from telephone stations.

The transmitter has three oscillators, one for each group of frequencies. There is an interface lead for each data frequency. An additional pair of leads energizes the three oscillators, resulting in the simultaneous transmission of three frequencies.

Two coding systems are used ; both use one tone from each group. In the first the "all rest" combination is used as an inter-character separator, and the data frequencies used for sending data. This gives an alphabet of $4 \times 4 \times 3$ or 48 characters. Errors are detected by checking for three tones on the line. The second method also uses the "all rest" combination as an intercharacter separator but makes use of the "rest" frequencies as well as the "data" frequencies to send data. Thus the use of all 14 tones gives $5 \times 5 \times 4 = 100$ combinations, less one for the inter-character separator. Transmission errors may still be detected.

The maximum speed of this system is 20 characters per second. The receivers are equipped for automatic answering thus allowing unattended operation at the receiving station. The transmitter is powered entirely from the telephone line. Each frequency is transmitted at a power of

approximately −9 dbm, a combined power of approximately −4 dbm, slightly less than ½ mW. An audible answer-back acknowledgement signal is used.

Experimental work has been done with the use of a "spoken" reply to inquiries presented to a computer. It is planned that in the future it will be possible to use a touch-tone telephone set to transmit data using the same push-button "dial" that is used to set up the call.

ii) The second system has been developed to provide a binary, parallel data transmission system for use in service over the switched telephone network or for leased lines. The system will transmit any number of data levels up to 8 at any speed up to 75 characters per second. Timing signals are also transmitted. A non-simultaneous answer-back feature is provided at a maximum speed of 20 two-bit characters per second. The system offers advantages that parallel-serial conversion is not required and that a timing output is supplied to the receiving business machine.

The system uses nine frequency shift channels, the middle frequency being used for timing. The lowest mark frequency is 730 c/s and the highest 2090; to send a space the frequency is shifted up by 70 c/s.

The transmitter data set contains an answer-back receiver to receive a signal chosen on a "one-out-of-four" basis. One of the four frequencies will be a rest frequency not actuating a relay, two of the remaining three frequencies will operate one relay each and the fourth frequency will operate both relays. The lowest frequency is 1017 c/s and the highest 1152 c/s.

This arrangement does not provide a self-checking facility. It is powered from the a.c. mains supply.

c) *Federal Republic of Germany* (Supplements Nos. 55 and 56)

Two systems are described, one by Siemens & Halske and one by Standard Elektrik Lorenz AG.

i) The Siemens system is intended for the transmission of data originating from a punched-card reader, punched-tape reader or keyboard. The most general application envisaged is the non-simultaneous transmission of data from a great number of out-stations to a central station. The out-station equipment should be as simple as possible.

A three-times-one-out-of-four code is used, 12 V.F.T. frequencies, the lowest being 420 c/s and the highest 1860 c/s with a spacing of 120 c/s in three groups of four frequencies. An F.M. return channel is provided using frequencies 1620 and 1740 c/s.

The maximum transmission speed is 20 characters per second. The coding is 3×1 out of $4 = 64$ combinations.

ii) The Lorenz system is intended to be a universal parallel data transmission system sending data from punched tape, punched cards or a keyboard. The transmission frequencies are arranged in four groups with one-out-of-four frequencies. A character is represented by the transmission of one frequency from each group. The lowest frequency is 930 c/s and the highest 2070 c/s, within the groups the frequency spacing is 60 c/s. An optional F.M. timing channel is provided using 1470 c/s and 1530 c/s. Data signals are accepted by the modulator in the form of contact closures between respective data-leads and a common lead for all groups.

An optional frequency shift return channel is proposed for signalling at a maximum speed of 75 bauds. A frequency shift of ± 30 c/s is proposed with a mid-frequency of 420 c/s [1].

[1] This is to conform to the specification for the return channel in the medium-speed standard modem.

The speed of operation of this system is 25 characters per second if the rest frequencies of all groups are sent between characters. If an optional timing channel between two groups of frequencies is used, the maximum modulation rate is 50 characters per second because there is no pause between characters.

d) *Chile Telephone Company* (Supplement No. 59)

This system is adaptable to punched-tape, punched-card readers and keyboard, this adaptation being effected by insertion of the proper coding matrix. Data-input leads are d.c. operated, ensuring safe contact functioning.

The coding is "two-out-of-eight", "three-out-of twelve" or "four-out-of sixteen", depending upon the number of frequency groups in use, the modulation transmitting one frequency from each group of four frequencies.

The frequencies, optional timing channels and return channels are the same as above.

8. *Detailed summary of systems*

a) *Coding*

United Kingdom	General
A.T. & T.	i) $4\times4\times3 = 48$ combinations $5\times5\times4 = 100$,, ii) 8 out of $8 = 256$,,
Germany	i) 3×1 out of $4 = 64$ combinations ii) 4×1 out of $4 = 256$,, or 3×1 out of $4 = 64$,,
Chile	4×1 out of $4 = 256$,, 3×1 out of $4 = 64$,, 2×1 out of $4 = 16$,,

b) *Frequency allocations*

A.T. & T.	i) 14 frequencies 3 groups, the first frequency "rest" ii) 9 frequencies centre frequency timing channel
Germany	i) 12 frequencies (V.F.T. frequencies) 3 groups of 4 frequencies 120-c/s spacing ii) 16 frequencies 4 groups of 4 frequencies spacing 60 c/s in groups
Chile	i) 16 frequencies 4 groups of 4 frequencies spacing 60 c/s in groups

c) *Timing channel*

A.T. & T.	i) No timing ii) Single channel in centre

Germany	i) No timing
	ii) Optional F.M. 1470 c/s and 1530 c/s
Chile	i) Optional F.M. 1470 c/s and 1530 c/s

d) *Backward signals*

A.T. & T.	i) Audible Yes—No for repetition Future use to transmit voice signals from computer
	ii) One-out-of-four basis 20 characters per second
Germany	i) F.M. using frequencies 1620 c/s and 1740 c/s
	ii) Return channel same as 600/1200 bauds modem, 420 c/s ± 30 c/s, 75 bauds
Chile	As Germany ii) or audible 450-c/s tone

e) *Applications and speeds*

A.T. & T.	i) Punched cards I.B.M. tabulator cards Speeds up to 20 characters/second
	ii) Wide range Speeds up to 75 characters/second
Germany	i) Universal applications using code converter Handle paper tape up to 6-unit codes Speed up to 20 characters/second
	ii) Punched cards Punched tape Keyboard Speeds up to 25 or 50 characters/second
Chile	i) Punched cards Punched tape Keyboard Speeds up to 25 or 50 characters/second

9. *Contribution by Automatic Telephone & Electric Co.*

a) *General*

It is proposed that the parallel-series data transmission mode of operation for transmitting alpha-numeric information should be included in the list of systems for further study under parallel modulation systems for data collection.

The basic features of such a system are described as a guide for further study.

b) *Basic features*

b.1 The one basis 2 × 1-out-of-4 parallel modem with an additional timing channel can be used by simple parallel operation to transmit numeric information and by parallel-series operation can be used to transmit alpha-numeric information up to an eight-unit code.

b.2 Providing the two speeds could be made compatible (for example, 20 characters per second for alpha-numeric and 20 or 40 characters per second for numeric) the two system requirements could possibly be grouped into one, with the possible exception of the requirements for power-driven devices and non-power-driven devices as the data source.

b.3 Provided a separate timing channel is used, a basic numeric character transmission rate of 40 characters per second appears practicable. This means that 20 alpha-numeric characters per second can be achieved with parallel-series transmission. For numeric data, there would be a choice between operating at 40 characters per second if this was compatible with the data source and data sink, or operating at 20 characters per second with improved performance.

b.4 The character set would be 16 for the simple parallel mode of operation and 256 for the parallel-series mode of operation.

b.5 The maximum number of characteristic frequencies for either numeric or alpha-numeric operation would be 10, including an F.M. timing channel. This eases the problems of allocating frequencies within the restricted bandwidth available over switched networks, and avoiding interference with existing V.F. signalling receivers in the switched connection.

b.6 The desirable feature of the modem at the transmitting end being powered from the telephone line could be met for both numeric and alpha-numeric operation, even up to an eight-unit code. It is understood that, under such conditions, the return signalling would be restricted to a simple audible signal.

Supplement No. 61

SPECIAL STUDY GROUP A — *(Extract from report COM Sp. A/No. 92 — October 1963)*

REPORT ON METHODS OF ERROR CONTROL

1. Special Study Group A examined contributions received on error control of data transmission over telephone-type circuits with a view to considering the possibility of recommending a standard method or error control for serial bit stream input and output systems.

a) for use over the 200-baud duplex transmission system ;

b) for the 600/1200-baud transmission system.

2. It was agreed that the field of error control could be broken down for study purposes into the following :

2.1 *Error detection*

2.1.1 *Error-detection code*

2.1.1.1 No return path (receiver detects the errors only).

2.1.1.2 Decision feedback.

2.1.1.3 Redundancy feedback (i.e. redundancy check bits developed from received signals are sent back to the transmitter for comparison and decision).

2.1.2 *Information feedback*

2.2 *Error correction*

2.2.1 Error-correction code (no return path).

2.2.2 By retransmission of a part of the information in cases 2.1.1.2, 2.1.1.3 and 2.1.2.

2.3 *Mixed operation*

2.3.1 = 2.2.1 + 2.1.1.1 (i.e. partial correction and remainder detected only);

2.3.2 = 2.2.1 + 2.2.2 (i.e. partial correction by code and remainder corrected by retransmission).

3. *600/1200-baud error-control system*

As the 600/1200-baud system is the more general case—i.e., solutions for this may be applicable to the 200-baud case whereas the converse is less likely—the 600/1200-baud error-control system should be dealt with first.

3.1 No proposals were received for 2.1.2 or 2.2.1. However, 2.1.2 had been proposed for the 200-baud system. It was generally thought that the high redundancy required for high-accuracy error-correcting codes, coupled with the burst nature of errors on telephone lines, would reduce the over-all information-transfer rate of this system. It was therefore decided to concentrate on error correction by means of decision-feedback systems with retransmission. It is proposed, however, that further study of correction by error-correcting codes (in particular single-bit correction) should be pursued as the subject of new questions, as well as the possibility of adaptive control, i.e. variable amounts of redundancy to suit the channel conditions.

3.2 In principle, any of the techniques can be carried out within a data-processing system if this is economically justified; these are indicated in the following table:

Error-control techniques for 600/1200 bauds and higher rates
(Subjects for future questions)

Technique within communication data transmission equipment	Technique within data-processing system	Is adaptive control possible?
Correction by decision-feedback and retransmission	As in column 1	Yes
Partial error correction using error-correction coding. Residue corrected by decision-feedback and retransmission	As in column 1	Yes
Complete correction using error-correcting coding. No feedback	As in column 1	No

3.3 *Error correcting using decision-feedback with retransmission*

Two cases were considered :

a) block by block ;

b) character by character.

3.3.1 Taking the block system first, this in turn can be subdivided into systems with a simultaneous low-speed return channel (75 bauds) and those without a simultaneous return channel which use the data channel in the reverse direction, i.e. half-duplex systems.

3.3.2 *Size of error-control block*

A distinction must be made between the block on which the error control system is based and the size of the block of data that is required to be transmitted by the user over the system.

The factors affecting the choice of size of error-control block when using a simultaneous return channel for decision feedback are :

a) error-distribution characteristics of circuit ;

b) maximizing efficiency factor ;

c) improvement factor of error-control system ;

d) number of check bits ;

e) composition and length of feedback-decision signal ;

f) maximum loop propagation time of circuit plus propagation delays in equipment ;

g) cost of the stores for such systems.

Note. — The minimum size of the blocks in bits in this case cannot be less than f) plus the length in time of the decision signal, divided by the modulation rate, assuming that two blocks are stored for retransmission purposes.

The size in the half-duplex case, however, is mainly affected by the speed of turn-round at the terminals and, if they exist in the circuit, the time taken to reverse the direction of the echo suppressors.

It was suggested that the recommended block size should be selected from the following fixed sizes: 210 - 240 - 480 or 560 bits. (These figures do not include service or error-check bits.)

In the half-duplex case where echo suppressors are involved, the block size may have to be considerably larger.

It was alternatively suggested that the error-control block should be of variable length and vary with the data block size, as this would enable very small blocks of data to be transmitted more efficiently. This, however, requires an end-of-block signal to be recognized or indicated to the error-control equipment at both ends of the circuit.

3.3.3 *Size of data block*

The I.S.O. representatives and other data-processing industry members stated that the data block size or format should not be limited by the error-control block size and should not be conditioned on any particular code or range of character sizes.

It was suggested that buffers in the data source and data sink clocked from the error-control equipment might be a solution to this.

4. *Error control for the 200-baud system*

A short discussion on the methods applicable to this system showed the importance of single and economical methods based on the principle of character-by-character procedure or using blocks of short length. It is pointed out that the use of the first procedure with a single parity

bit and a modulation quality detector might provide a sufficient degree of security in many cases because of the small phase distortions observed in the channels foreseen for use at 200 bauds ; but this point needs testing.

5. In conclusion, the Study Group considers that, for error correction, three principal methods should be taken into consideration in future studies, viz. :

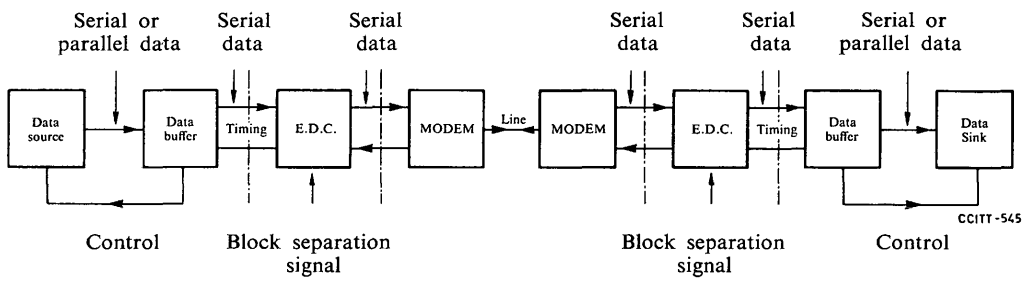
— information feedback with retransmission,

— character-by-character with decision feedback and retransmission,

— block system with decision feedback and retransmission.

6. *Serial bit stream error control : general concept*

Figure 1 shows the general concept of a system allowing for a serial stream input and output of data which is applicable to the error-control systems based either on block or on character-by-character error correction by decision feedback and retransmission. Characters can be regarded as very short blocks in this context.



This diagram is conceptual only and the interfaces shown between data source and data buffer and E.D.C. may or may not physically exist. Where they do exist they should be of the form shown.

In the case of the data source, data buffer and E.D.C. being combined in one equipment, they will generally not exist in this form.

FIGURE 1. — **Diagram of concept of error-control system**

7. *Fixed and variable block length*

Two systems of data transmission by blocks have been proposed in the documents of Special Study Group A and have received considerable attention ; they are :

a) transmission by error-control blocks of fixed length ;

b) transmission by error-control blocks of variable length.

In system a), the separation signal between the information bits and the check bits is obtained by decoding the state of a bit counter in either the data buffer or the E.D.C. In system b), the separation signal between the information bits and the check bits is obtained by decoding a character contained in the data signal.

The error-control blocks will consist of at least two sections, one containing the information bits and the other containing the check bits. The difference between systems a) and b) lies in the means of determining the separation point between the information bits and the check bits within each error-control block.

In system a), each error-control block will contain a fixed number of information bits followed by a fixed number of check bits. Considering the input bits stream at the receiving terminal, the point of separation can readily be determined by counting, at a synchronous rate, the number of bits received from the beginning of each block.

In system b), the separation between the information bits and the check bits within each block is defined by a particular character. This character is usually referred to as "end-of-block" code ; it must be decoded at both terminals, to define the separation instant. The particular character chosen for the end-of-block code can often conveniently be contained within, and be part of, the data received from the data source.

8. *Proposed characteristics to guide the study of future block-error correction systems using decision feedback and retransmission*

8.1 The case considered was limited to systems using 600/1200 bauds on the telephone network operating in a synchronous mode and using a low-speed return channel for the decision feedback signals.

8.2 *Error control block size*

8.2.1 *Fixed block length*

The information content of the block should be chosen from the range :

210 bits, 240 bits, 480 bits, 560 bits.

It should be noted that in a 200-baud system a block length of 64 bits or 128 bits may be preferred.

8.2.2 *Variable block length*

It was decided that the maximum or minimum length of the block should be studied.

8.3 *Checking code*

Three types of code were considered : cyclic code (generated by shift register) ; two co-ordinate parity ; spiral parity.

Generally it was felt that, for the type of error distribution encountered on telephone lines, the cyclic code was most efficient, but the other approaches mentioned above should not be ignored.

8.3.1 *Cyclic code*

If a cyclic code is employed, it is proposed that the choice should be made from 12, 16, or 18 check bits, but this will depend on :

a) error-control-block size ;

b) line-error distribution ;

c) the residual error rate.

It may be that 16 check bits will provide sufficient protection where the block size does not exceed 560 bits.

The polynomial defined by the shift register should be of such a form as to provide adequate protection against both uncorrelated bit errors and the burst type of a bit-error distribution.

8.4 *Block sequence number*

A numeration cycle of 3 or 4 would be preferred.
It should be noted that a numeration cycle of 2 is adequate for a half-duplex system.

8.5 *Decision feedback signal*

Two possibilities were considered.

8.5.1 *Compelled system*

Where the receiver calls for each specific block by number.

8.5.2 *Repetition demand system*

Where the transmitter sends a stream of blocks to line until a signal from the receiver demands a repetition.

8.5.3 It was considered that some kind of redundancy is required to protect the decision feedback signals.

8.6 *Transmission loop delay*

A parameter, upon which both the block length and the form of the decision signal depend, is the loop delay of the transmission circuit.
It was proposed that systems should be studied on a basis that the maximum loop delay of the line was 150 milliseconds. To this delay must be added any delay introduced by the error-control equipment or the modulator.
It was appreciated that longer delays would be encountered on intercontinental and satellite circuits; however, it was recommended that the problem involved in the use of circuits with a loop delay greater than 150 milliseconds should be studied as an additional item.

8.7 The start-stop method of block error control was discussed, but shortage of time prevented any conclusion being reached.

9. *Block-error correction using decision feedback and retransmission over a half-duplex data transmission connection*

9.1 *General*

The following general features of transmission are proposed for highly protected data transmission on telephone circuits:

9.1.1 The data are preferably transmitted at rates of 1200 or 600 bauds.

9.1.2 They are divided into blocks.

9.1.3 Sending and receiving terminals operate in synchronism. Bit synchronization is, for instance, effected and maintained by means of the incoming transitions. Provision of the block synchronism requires special arrangements.

9.1.4 The check bits for each block are generated in the shift registers in the sending and receiving terminals.

9.1.5 The check bits of each block generated at the sending and the receiving end are compared at the receiving terminal. For each received block, decision signals, e.g. "proceed" or "repeat", are transmitted from the receiving station to the sending station (decision feedback).

9.1.6 Correction of erroneous blocks is effected by block repetitions.

9.1.7 Block numbering allows the correct continuation of transmission upon a disturbed decision signal.

9.1.8 The period of time between two undetected errors will be considerably increased.

9.2 *Description of the half-duplex procedure*

9.2.1 The data blocks are transmitted one by one in the forward direction ; the associated decision signal is fed back upon receipt of the respective block (half-duplex operation).

9.2.2 All blocks are transmitted in a fixed element pattern ; the space between the blocks is not specified, but is determined by the respective delay of the line. It amounts, however, to an integral multiple of the bit length.

9.2.3 Block synchronism is effected by an individual sequence of bits, preceding each data block ; bit synchronism is also maintained during the transmission intervals between two blocks.

9.2.4 *Composition of blocks*

Each transmitted block comprises :

1. the heading signal ;
2. the start signal ;
3. the service bits ;
4. the data bits ;
5. the check bits.

9.2.4.1 *Heading signal*

The heading signal consists of reversals 1 : 1 and gets the receiving station ready for the following start signal. The heading signal is so long that the receiving station will still be able to recognize it after the dying out of the echo of the decision signal or, if there is no echo, after the operation of the level supervising unit.

9.2.4.2 *Start signal*

The start signal marks the beginning of a block. It consists of one bit having the same polarity as the last bit of the preceding heading signal. The polarity of this start signal bit should be standardized, and should preferably correspond to that of the digit 0.

9.2.4.3 *Service bits*

There should be three service bits containing the information essential to the transmission process. One of these bits represents the two sequence numbers A and B ; the other two may be used to indicate special blocks.

The service bits are protected by the error-control procedure.

9.2.4.4 *Data bits*

The information part of the block may consist of a number of bits, belonging to characters of any code. It will be possible to adapt the number of data bits per block to the terminal equipment used in the respective case. 210, 240, 480 and 560 data bits per block are preferred ; in special cases it may be possible to use larger blocks.

9.2.4.5 *Check bits*

Check bits are generated in a shift register for low transmission speeds. The number of stages of the shift register must be specified according to the properties of the telephone network.

9.2.5 *Decision signals*

There are two decision signals, "proceed" and "repeat". These signals must be so long that they can still be recognized upon the operation of the level supervising unit at the sending end. They should be composed in such a way that it is practically impossible for the two conditions "proceed" and "repeat" to be changed into each other by errors.

9.2.6 *Synchronization at the beginning of transmission*

At the beginning of transmission, the bit synchronism will be established by the first block transmitted by the sending station. If this block is received without error, the receiving station will send the decision signal "proceed" and the next block will be transmitted. The bit synchronism is maintained, for instance, by the transitions within the following incoming blocks. In this case, it should be borne in mind that bit synchronism may be lost by blocks containing almost exclusively bits of one polarity only. For each block, block synchronism is established by the start bit.

9.2.7 *Transmission process*

The transmission process is shown on the diagram opposite.

Line a : Block 1 with the sequence number A is received undisturbed. The decision signal "proceed" is fed back. Next, block 2 is transmitted with sequence number B, block 3 obtains sequence number A, block 4 sequence number B, etc.

Line b : Block 2 is received incorrectly (disturbance of the service bits, the data bits or the check bits). The decision signal "repeat" is fed back.

Line c : The sending station then repeats block 2. When this is received correctly (decision signal "proceed"), transmission is continued with block 3.

Line d : If block 2 is also erroneous when transmitted the second time, it is repeated once again.

Lines e and f : If a decision signal is disturbed and is consequently not recognized at the sending station within a certain period of time after the transmission of the last block, the sending station repeats the block sent last after this interval. (The number of these repetitions can be limited.) If this repeated block is not disturbed during transmission, it will in any case be acknowledged by "proceed". The repeated block will be passed on to the terminal equipment, provided that this block has the correct sequence number (line *e*). In case of an incorrect sequence number, it will be erased at the receiving station, in spite of an undisturbed transmission (line *f*).

9.2.8 If a receiving block contains the expected sequence number and if there is no transmission error, the data bits are passed on to the terminal equipment ; the additional signals required for transmission are normally suppressed (clean tape).

10. *Study of error control on character-by-character systems*
(200 and 600/1200 bauds synchronous operation)

10.1 The characteristics which should be standardized for compatible inter-working are as follows :

10.1.1 Character (or block) size.

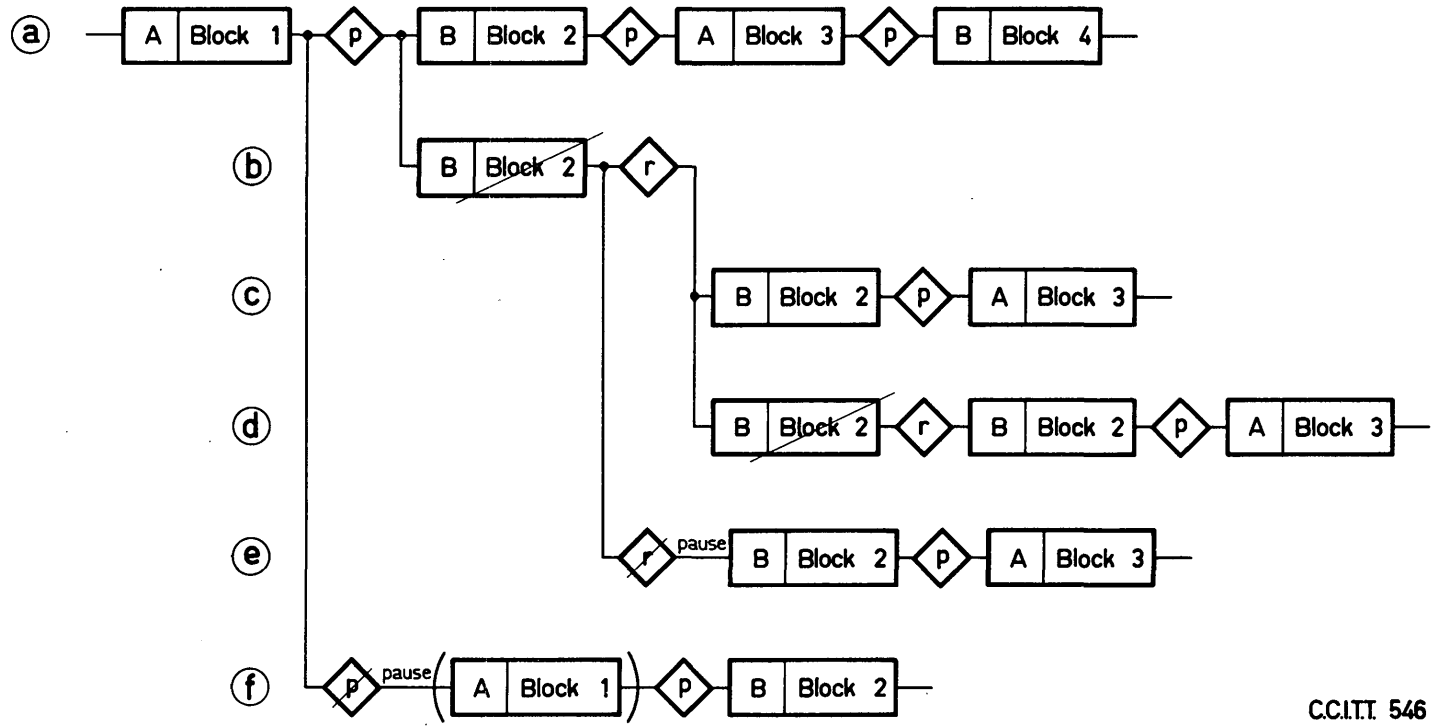
10.1.2 Detection code.

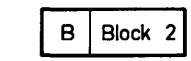
10.1.3 Line-signalling conditions, viz. synchronization, starting, terminating and repetition procedures.

10.1.4 Sending terminal store capacity and permissible loop propagation time.

10.2 The following values have been proposed for these characteristics :

Half-duplex medium-speed data transmission system on telephone network

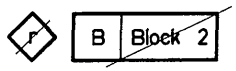




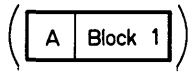
Sequence number B followed by the second block, transmitted forward



Acknowledgement signal p = proceed or r = repeat respectively



Transmission disturbed



Undisturbed block transmitted but erased

10.2.1 *Character (or block) size*

Although different character lengths could be arranged by mutual agreement between the terminal stations it would be desirable, for the general service, to standardize one character length only. Seven information bits per character are preferred. An additional parity bit may or may not be present in the source data and may or may not be delivered to the data sink. If not present, this parity bit is added by the error-control equipment.

The seven information bits per character, or block, may or may not correspond to the code combination of a particular alphabet and code; in other words, the system is not code restrictive.

10.2.2 *Detection code*

The single parity bit, occupying the eighth position in the character, provides an odd (or even) parity check.

An alternative possibility for the 600/1200-baud system is to provide a second parity bit, giving a nine-bit character or block. This ninth bit provides an odd parity check for bits 1, 3, 5 and 7.

10.2.3 *Line-signalling conditions*

The following have been proposed:

Data channel (forward)

Idle periods (also used for element synchronization):	reversals
Start of message:	A sequence of three characters duration consisting of: 1. an "x" character in which elements 3 and 4 consist of no signal to the line. The polarities of the other elements are immaterial. 2 and 3. the combination "1 0 1 0 1 0 0" (with normal parity).
End of message:	As for start of message but using the following code combination in positions 2 and 3: "0 1 0 0 1 0 0" (with inversed parity).
Repetition cycle:	One "x" character as defined above, followed by the characters retained in the sending terminal store.

Supervisory channel (backward)

Idle period:	Space (a change from "constant mark" to "constant space" constitutes an "end of message" acknowledgement).
Receiving data correctly:	Mark (a change from "constant space" to "constant mark" constitutes a "start of message" acknowledgement).
Request for repetition:	A "space" condition corresponding to the transmission time of four characters on the data channel.

10.2.4 *Sending terminal store capacity*

10.2.4.1 *200-baud operation*

Five characters storage:	Provides for approximately 150 ms loop delay and 50 ms for equipment delays.

10.2.4.2 *600/1200-baud operation*

Sixteen characters storage : Provides for approximately 150 ms loop delay at 600 bauds and approximately 80 ms at 1200 bauds.

For the special cases of frequent operation over very long intercontinental connections, and where a sacrifice of speed is not acceptable, a sending terminal store of 64 characters is suggested. Such a store would provide for loop delays of up to approximately 440 ms at 1200 bauds and approximately double this delay at 600 bauds.

Supplement No. 62

I.B.M. (UNITED STATES OF AMERICA) — *(Extract from contribution COM Sp. A/No. 13 — August 1961)*

STUDY OF ERROR CONTROL SYSTEMS

A. *Introduction*

This paper is concerned with, specifically, error control methods which are designed to safeguard the high reliability of computer data throughout their transmission over relatively noisy channels.

The simplest and most commonly used case will be considered : binary serial transmission over a single channel. The original message is assumed to contain no redundancy [1] ; redundant bits are introduced in the encoding process for the purpose of error detection and error correction. The final message is arranged in blocks, each block consisting of several six-to-eight-bit characters.

A conventional error control system might operate as follows : redundancy bits are included in the message (typically in the form of a parity check code) for the purpose of error detection. At the receiving end, the message is decoded in groups of binary digits ; if an error is detected, transmission is interrupted with a demand for repetition of the last, erroneously received, sequence. This system may be contrasted with one in which error control is achieved without a return channel by incorporating in the original message additional redundancy bits which enable the receiver not only to detect errors, but also to correct them. It is the purpose of this paper to compare these two methods, to indicate their domains of applicability, and finally, to suggest their simultaneous use in a system particularly well adapted to the stringent requirements of data transmission.

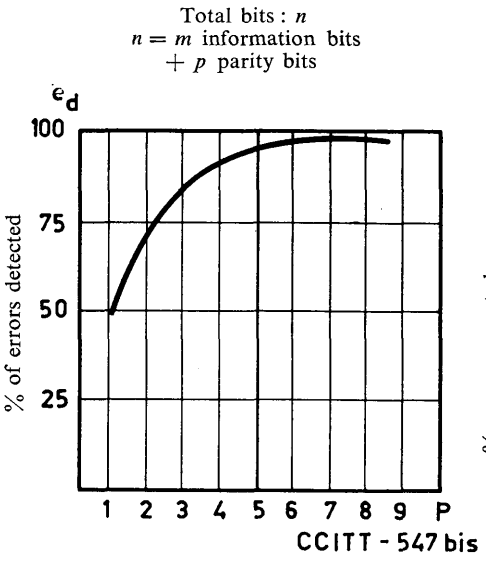
[1] These considerations are, of course, equally applicable to a message whose original redundancy has been eliminated by preprocessing, such as Huffman coding.

The gist of the argument is as follows: error-detection coding (EDC) and retransmission make an excellent system for reliable communication if the transmission channel introduces few errors, and if a high-quality return channel is available. The system deteriorates rapidly with increasing error rate. Error correction coding (E.C.C.) is particularly effective if the characteristics of the channel are such that a relatively small percentage of the possible error patterns account for virtually all observed errors. One may thus use E.C.C. to convert the actual channel into an effective channel having a relatively lower error rate; retransmission may then be used to its best advantage. There is one additional point which should be kept in mind: although the principle of E.C.C. has been known for years, it is only with the recent discovery of easily implemented codes that E.C.C. has become attractive for practical application.

B. *Error detection by coding (EDC)*

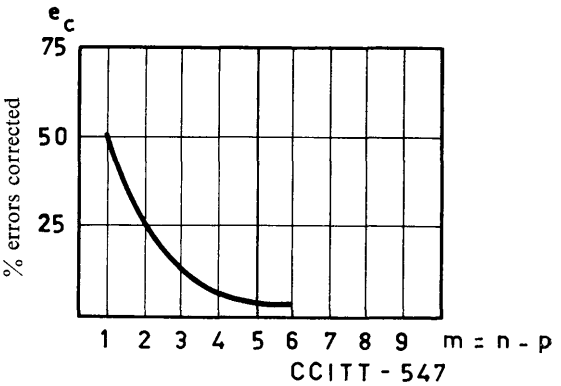
The simplest binary error detection code is the parity check, an additional digit chosen equal to the sum modulo 2 of all the bits in the message. The n-bit message including the parity check can be received in 2^n different ways, one of which corresponds to the original message. If an odd number of errors is present, the parity check will indicate an error. Thus, one-half of all possible errors can be detected with just one additional bit. The fraction of errors actually detected in transmission will depend on the characteristics of channel noise which assigns a given probability of occurrence to each of the 2^n possible versions of the received message. Additions of a greater number of parity bits will increase the protection afforded by the code. In a message of arbitrary length including p check bits, the number of undetected errors will amount to a fraction $\frac{1}{2p}$ of all possible errors if the error-detecting code is efficient (Figure 1). The undetected error rate can be made arbitrarily small by increasing p. The implementation of such codes requires only one storage element (flip-flop) per parity bit with very little added logic.

The addition of a given number p of EDC bits detects a fraction $1 - \frac{1}{2}p$ of all possible errors regardless of the total length of the code word. As the word length increases, the frequently occurring error patterns form an ever smaller fraction of the total number of possible errors. Thus, the efficiency of EDC increases as the word length is increased, and one should choose as the code word length a reasonably large structural subdivision of the message. Consider, for example, a block of 80 6-bit characters, required to transmit the information contained in an 80-column punched card. A single parity check per character will not detect any double errors in a character, and in particular will fail to recognize a double adjacent error, whose frequency of occurrence in many channels is second only to single error occurrence. Double adjacent errors can be detected by the addition of a second check bit, equal to the sum modulo 2 of every other bit in the character. This checking scheme, however, will fail to recognize other common error patterns, such as a burst of four consecutive errors, and will add a total of 160 redundant bits, or 33% of the information bits. In contrast, in a block detection scheme, 9 parity bits are sufficient to detect any double error in the message, or any burst up to 8 bits in length. The redundancy required is less than 2%. Another means of error detection by character is afforded by fixed weight codes, where the information is encoded to include a fixed number of "ones" per character. An error is detected whenever that number is altered by noise. Any odd number of bits in error is detected by this scheme as well as many even errors. An undetected error occurs when an even number of bits is corrupted in such a manner as to leave the total number of "ones" unchanged.



$$e_d \approx \left(1 - \frac{1}{2p}\right) \times 100$$

FIGURE 1. — **Error-detection efficiency**



$$r = \frac{2p-1}{2n-1} \times 100 \approx \frac{1}{2m} \times 100$$

FIGURE 2. — **Error-correction efficiency**

Fixed weight codes are attractive in unsymmetrical channels, where the probability of gaining "ones" by error is larger than that of gaining "zeros", or vice versa. A combination of fixed weight coding by character and parity checking by block has been successfully employed in some data transmission systems.

C. *Error correction by coding (ECC)*

ECC, just like EDC, involves additional check bits, which are the sum modulo 2 of certain information bits. The bit or bits in error can be pinpointed at the receiver by determining which parity bits fail to be equal to the sum of the corresponding information bits. In an n-bit message including p parity bits, $2p - 1$ erroneous version of that message can be corrected, since there must be a one-to-one correspondence between every erroneous message to be corrected and every possible failure of the p sums. Of $2n - 1$ possible erroneous versions of the transmitted message, the fraction $r = \dfrac{2p-1}{2n-1}$ can be successfully corrected with p check bits. But

$$r = \frac{2p-1}{2n-1} \approx \frac{1}{2n-p} = \frac{1}{2m}$$

where m is the number of information bits. The fraction of correctable errors is seen to depend only on the number of information digits, and to be independent of the number of parity digits (Figure 2). The maximum fraction of correctable errors occurs for $m = 1$ or for p parity bits

and 1 information bit, where $r = 1/2$. It is therefore impossible with an error-correction code consisting of separate information and parity bit groups to correct more than one-half of all possible errors in a message, even though it consists of just one information bit, and any number of check bits [1].

A system where each binary digit is repeated n times can be thought of as embodying the simplest example of an optimum error-correction code, consisting of one information bit and $n - 1$ parity bits. The decision at the decoder is based on a majority count of received bits. In such a scheme, it is possible to correct the received message only if half or less of the bits have been corrupted.

The effectiveness of ECC is due to the fact that not all errors in a message are equally probable. In some channels, the probability of certain small classes of errors has been shown to be orders of magnitude larger than the probability of other classes. It is apparent, then, that an error-correcting code can offer maximum protection only if it is designed specifically for the types of errors caused by the interference in the channel, which are generally also a function of the type of modulation system employed. Of particular interest is the distribution of the number of bits between successive errors, which will determine the optimum message lengths, and show whether errors are independent, or occur in bursts. Error statistics of this type for the U.S. switched telephone network were obtained by Alexander, Gryb, and Nast [1]. They clearly point out that errors are not independent but occur in bursts. The number of bits corrupted by a noise impulse is a function of the speed of transmission as well as the channel characteristics, and this factor must be taken into account in determining a suitable error-correction code.

D. *ECC implementation*

Until recently, it was unpractical to incorporate error correction in many systems because of the complexity of the decoding and correcting operations. This situation changed, however, with the advent of cyclic block codes, first developed by Abramson [2]. The implementation of the burst-error-correcting codes developed by Peterson is not much more complex than that of cyclic error-detecting codes [3, 4].

The encoding operations for ECC and EDC are identical and require only a p element feedback shift register, where p is the number of parity bits (Figure 3). The decoding operation is similar to the encoding operation (Figure 4). The feedback shift register is examined after information and parity bits have been passed through it and, if the storage elements are not all in the zero state, an error is present in the data. The error-detection process ends there; for error correction, a further cycle is necessary. In the case of burst-error correction, it is necessary only to add the contents of the feedback register to the data being shifted bit by bit out of storage, when a given bit configuration is tested in the feedback register. The storage device need only be a serial buffer to delay the data, rather than a core buffer providing random access. This addition actually performs the correction of corrupted bits.

Cyclic codes have been developed both for the correction of burst errors and for the correction of independent errors [5, 6, 7]. The same codes can be used for error detection only, or for a combination of error detection and correction. Since these codes all use the same implementation for error detection, they impart a large amount of flexibility to a system, and they can easily be modified to meet various reliability requirements.

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[1] By comparison, error detection of half of all possible errors can be accomplished with only one parity bit, regardless of the number of information bits.

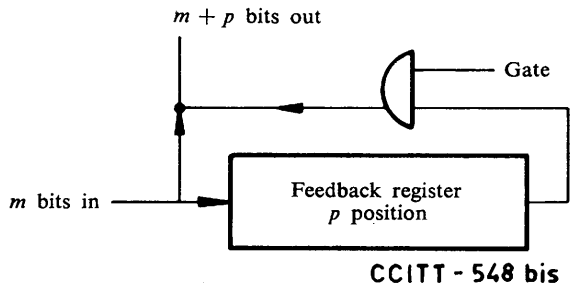


FIGURE 3. — **Encoding**

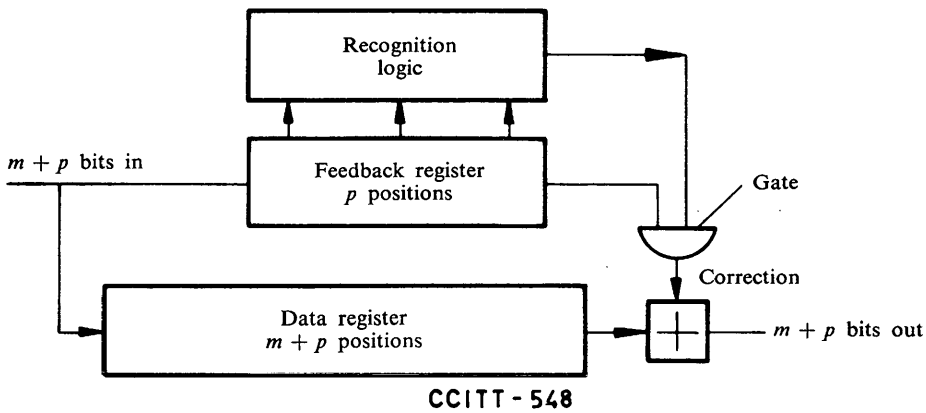


FIGURE 4. — **Decoding and correction**

The redundancy required for correction is not excessive. A code which corrects bursts up to 10 bits in length in the 480-bit block mentioned earlier requires 20 parity bits, or about 4% redundancy. For correction of any three errors occurring at random in the same block, 27 parity bits must be used for a redundancy of 5%.

An error-correction code designed to correct a specific class of errors will interpret any error as belonging to that class. If the error is of a type that cannot be corrected, the correction process takes place anyway, and some good bits are altered. Furthermore, the alternation is made in such a way that it cannot be detected if the same code is used again.

Any system with ECC must therefore have additional EDC performed to verify the correction cycle. An ECC system can thus be thought of as consisting of three cycles: an initial detection cycle where the parity relations are applied to the received data, a correction cycle performed if any of the parity relations fail, and a final detection cycle to ensure that no errors remain in the data.

E. *Error correction by automatic retransmission*

A most effective means of error control is the automatic retransmission of erroneous messages, where at the end of every message the receiver sends an acknowledgement to the transmitter; upon failure to receive such acknowledgement, the transmitter then retransmits the same message.

In order to have a reliable retransmission system, sometimes known as a feedback system, the following characteristics of the message are of importance:

1. The error detection applied to the transmitted message must be adequate since the whole system can be no more reliable than the error detection.

2. The acknowledgement signal must carry enough redundancy to be relatively immune to interference. Clearly, if the acknowledgement signal indicates a correct message, when actually the message has been detected as being incorrect, the transmission efficiency will be seriously impaired.

If the opposite happens, that is, if a feedback signal indicates that the message is incorrect, when the message was actually correctly received, it will be retransmitted needlessly and this of course will not impair the reliability of the transmission, but again result in decreased channel capacity.

An asymmetrical type of error correction could then be adapted for the acknowledgement signal. For example, let k "zero" bits represent acknowledgement of a correct message, and k "one" bits be a repeat signal; the transmitter might repeat the message when m of the k bits it receives are "ones" and $m > k/2$. Clearly, the correction will be more asymmetrical as m is decreased, and the probability of needless retransmission will increase. The possibility of this latter occurrence makes it necessary to identify new or retransmitted messages at the receiving end by adding a group of bits at the beginning of the message in the form of an address code. The address code should also have enough redundancy to be self-correcting, to protect against misinterpretation of a new message as a repeat. An excellent discussion of automatic retransmission is presented by Metzner and Morgan [8].

If redundancy is adequate in the address at the beginning of each message and in the acknowledgement signal, retransmission is a very powerful scheme of error control, since it can be made as reliable as the error detection. It is therefore possible to attain with this means an arbitrarily large degree of reliability, if channel capacity is not a factor. The main disadvantage of this method is that it requires two-way communication in the form of a feedback channel, not always available in transmission systems. Also it does not make very efficient use of the channel because it might require retransmission of a whole message regardless of the number of errors it contains. It requires more redundancy than one-way transmission with error detection, because of the high reliability imposed on the acknowledgement and the address codes.

The requirement of feedback transmission cannot always be met by the communication system, as shown in the following two examples:

1. When the transmission is in real time, as for radar data, or for guidance systems, automatic retransmission may not be possible since it would result in a variable time delay between the time the message is transmitted and the time it is correctly received, which cannot be compensated for easily.

2. When a master station is required to transmit data simultaneously to many different locations, automatic retransmission becomes extremely inefficient, since the message must be retransmitted to all receiving locations if only one of them sends a repeat request.

Automatic retransmission is particularly well suited to channels where noise conditions are either catastrophic, or practically non-existent. In this situation, retransmission would occur only when large parts of the message were deleted, and thus efficient use would be made of the channel.

When channel noise is such that only a small percentage of transmitted blocks are corrupted, the over-all efficiency obtained with retransmission is high and any redundancy and hardware created by addition of ECC would not be justified. This was found to be the case in many wire-line transmission tests where the fraction of blocks to be retransmitted was in the order of 2% or less.

Automatic retransmission *per se* is inefficient in types of channels where the noise is of a sporadic nature. In the worst case, when a noise pulse occurs in every consecutive record, automatic retransmission alone is totally inefficient. Error-correction coding, on the other hand, is useless in catastrophic conditions, but can be very effective for sporadic impulse noise. A combination of the two error control methods would thus result in a very powerful tool for error control and ECC can be achieved with little additional equipment since a buffer is available at the receiver for retransmission (Figure 5).

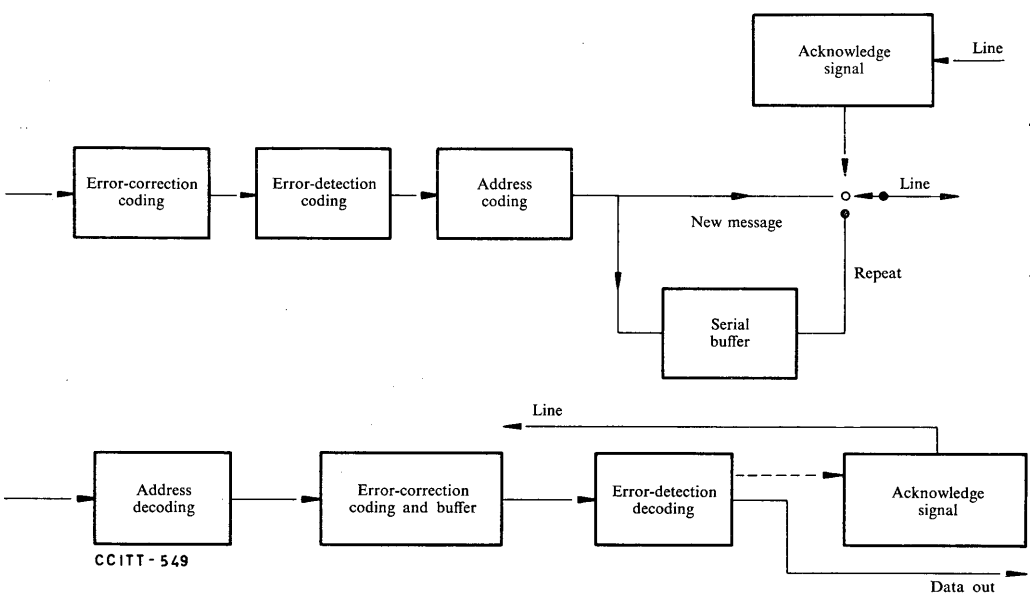


FIGURE 5. — **Block diagram of transmission terminal combining retransmission and ECC**

F. *Optimum message length*

An accurate determination of the optimum length of a transmitted message is perhaps the most important factor in ensuring best channel utilization, whether error correction is used by coding or by retransmission. The block length to be used can be determined after the error distribution of the channel is known, or calculated from an assumed noise distribution, as shown by Wood [9]. This message length might be very different from that dictated by other systems considerations, but can always be achieved for transmission if a buffer is present at the terminal, a necessary feature for any error correction system. Clearly, the noisier the channel the shorter the optimum block length, since this increases the probability of finding an error-free message, or one correctable by coding, and thus decreases retransmission time. Long blocks will result in less over-all redundancy, but in longer retransmission time.

The optimum block length for ECC need not be the same as that for automatic retransmission and will generally be shorter. This can be taken into account in a system where acknowledgement occurs only after several blocks have been received and where only the blocks in error are repeated.

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Supplement No. 63

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 14 — August 1961)*

BLOCK CHECKING BY MEANS OF A FEEDBACK SHIFT REGISTER

For building up the checking signal in blockwise data transmission the use of a cyclical code is recommended. This method was initially suggested by Huffman and has been substantially extended since (see reference [1] at the end of this Supplement).

A cyclical code with 12 checking bits allows to protect a basic block of $N_0 = 2^{12} - 1 = 4095$ bits (including the checking bits) with a Hamming distance of $d = 3$. This represents the maximum block length which can be protected with this Hamming distance by means of 12 checking bits. The same number of checking bits can also protect a basic block of $N_0 = 2^{11} - 1 = 2047$ bits (including the checking bits) with the Hamming distance of $d = 4$. This alternative is to be preferred because cutting down the length of the basic block appears to be justified for the benefit of error protection.

The probability of undetected errors in this case is approximately:

$p'_4 = 0.7 \times 10^6 p^4$ errored bits/bit $\qquad (N = 2035)$

(p = probability of an errored bit)

In the case where the block length is not used to its full extent, we get for a block length of N bits approximately:

$$p'_4 = \frac{(N)^3}{100} \times 82\, p^4 \qquad (N \leq 2035)$$

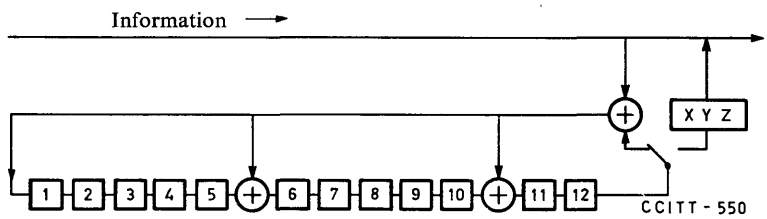
If, however, the block length exceeds the length of the basic block, the Hamming distance is reduced to $d = 2$ and the probability of undetected errors is approximately:

$$p'_2 = \left(\frac{N}{2047} - 1\right) p^2 \qquad (N \geq 2 \cdot 2047)$$

The protection against error bursts is likewise the maximum which can be achieved by means of 12 check bits: if all errors of a block are within a 12-bit sequence, the block is surely detected as faulty.

The shortest error burst which may remain undetected is 13 bits long. As regards longer error bursts, only 1 burst of a given length in every 4096 error bursts of this length goes undetected.

A set-up for realizing the proposed protection method can readily be established with the aid of a feedback shift register composed of 12 binary counters connected as shown in the following diagram:



Feedback shift register

After the end-of-block signal, the shift register in the block-checking device is read out and the 12 bits together with 3 more bits are transmitted in the form of 3 characters (named x y z in the diagram). Thereafter all binary counters of the shift register are restored to the basic position, which may consist of all ones.

The block-checking device at the receiving end has an identical function except that the 12 checking bits together with 3 more bits are fed into the comparator.

Note to data transmission with cross-checking

If one wants to transmit every single character protected in itself and if protection is to be increased further by block checking, cross-checking would be a common approach. This well-known procedure will not be described here anew. If z characters with $(n + 1)$ bits each (n information bits plus 1 parity bit) are put together to a block with N bits, the probability of undetected errors in the case of cross-checking will be approximately

$$p'_4 = N p^4$$

$$\text{or } p'_4 = \frac{n+1}{k} N p^4$$

if several characters with a total of k bits are used for checking purposes.

It is more efficient to use a feedback register with k stages instead of the counters for longitudinal parity. This results in approximately:

$$p'_4 = \frac{n\,(n+1)}{2^k} N p^4.$$

As compared with cross-checking the factor $2^k/n.k$ can be gained, that is, e.g., the factor 68 with $n = 5$ and $k = 12$.

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Supplement No. 64

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 14 — August 1961)*

THE DETERMINATION OF THE EFFICIENCY OF ERROR-DETECTION ARRANGEMENTS

1. *Introduction*

Many different error-detection codes have been proposed and the comparison of their relative merit is necessary for the design of data transmission systems.

The methods proposed below refer to the detection of errors in binary coded messages transmitted in blocks, each block being checked by adding redundancy signals which may be transmitted in the forward or backward direction.

The methods take account of the fact that errors encountered on actual telephone and telegraph connections may occur in bursts. Curves of the relevant parameters are appended. These have been obtained by actual measurements.

2. *Definitions*

In this section the quantities mentioned in the following sections are defined:

n = total number of elements in a block
m = number of message elements in a block
k = total number of redundancy elements associated with a block

Note 1. — $n = m + k$

e = number of erroneous elements in an erroneous block

p_E = element-error rate $= \dfrac{\text{erroneous elements}}{\text{total elements}}$

p_B = block-error rate $= \dfrac{\text{erroneous blocks}}{\text{total blocks}}$

p_{EB} = element-error rate for erroneous blocks

Note 2. — $p_E = p_B \cdot p_{EB}$

Note 3. — $e_{\text{average}} = n \cdot p_{EB}$

p_E^*, p_B^* = undetected-error rates (after application of error detection)

$r(e, n, k)$ = reduction factor for e-errors due to error-detection code (considering all possible pattern of e-errors to occur with equal probability)

$$= \frac{\text{undetectable patterns of } e\text{-errors in a block of } n \text{ elements}}{\text{all possible patterns of } e\text{-errors in a block of } n \text{ elements}}$$

$g(e, n, p_{EB})$ = distribution of erroneous elements in error patterns with e-errors

$$= \frac{\text{erroneous elements occurring in blocks containing } e\text{-errors}}{\text{all erroneous elements}}$$

Note 4. — $\displaystyle\sum_{e=1}^{n} g = 1$

$b(e, n, p_{EB})$ = error pattern distribution

$$= \frac{\text{erroneous patterns of } e\text{-errors}}{\text{all erroneous blocks}}$$

Note 5. — $\displaystyle\sum_{e=1}^{n} b = 1$

Note 6. — $g = \dfrac{e}{n \cdot p_{EB}} \cdot b$

Note 7. — r, b, g refer to blocks of n elements, r depends only on error-detection coding, while g and b depend only on transmission conditions.

$R_E(n, k)$ = total reduction in element-error rate (due to error detection)

Note 8. — $R_E = \frac{p_E^*}{p_E}$

$R_B(n, k)$ = total reduction in block-error rate (due to error detection)

Note 9. — $R_B = \frac{p_B^*}{p_B}$

3. *Conventional method*

The method described below is suitable for determining and comparing the efficiency of error-detecting arrangements taking account of the actual distribution of errors in real connections.

The method relies on data which must be acquired by tests on actual connections. The parameters are p_{EB}, p_B, and the distributions g and b.

This method is called a "conventional method" because it is assumed that all possible patterns of a given number of errors in a block are equally likely to occur. This may appear to limit the accuracy of the method. However, taking into consideration that the line conditions and consequently also the fine structure of the errors are subject to great variations from connection to connection and may differ also considerably between different networks, this assumption seems to be useful as a conventional condition.

The method consists of calculating the total reduction due to error detection of either the element-error rate or the block-error rate by applying the following formulae :

$$R_E = \sum_{e=1}^{n} r \cdot g$$

$$R_B = \sum_{e=1}^{n} r \cdot b$$

Thus the total reduction is obtained by multiplying the quantity r which depends only on error-detection coding with g or b, respectively, which depend only on transmission conditions. The total reduction for a given block size, calculated in this way, is a characteristic measure of the effectiveness of the error-detection code.

With the total reduction R_E or R_B, respectively, the undetected-error rate (after error detection) is given by

$$p_E^* = R_E \cdot p_E;$$

$$p_B^* = R_B \cdot p_B$$

4. *Relationship between the quantities considered*

In a data transmission system using the block repetition method there are two error rates which are of special interest to the designer of the error-detection procedure, i.e. the block-error rate p_B, which is a measure for the time lost due to repetition of erroneous blocks, and the element-error rate for erroneous blocks p_{EB}, which is important with regard to the error-detection capability of the coding arrangement. These two error rates may be measured on representative connections. They are related to the element-error rate p_E by the formula

$$p_E = p_B \cdot p_{EB}$$

For a given value of p_E the main effect of the concentration of line errors in bursts is to increase the value of p_{EB} and to decrease the value of p_B. For the purpose of explanation, this is demon-

strated in Figure 1 and Figure 2 where p_B or p_{EB}, respectively, are plotted against the number of elements in a block (n). Appendix A shows measured curves for different transmission conditions on telephone lines.

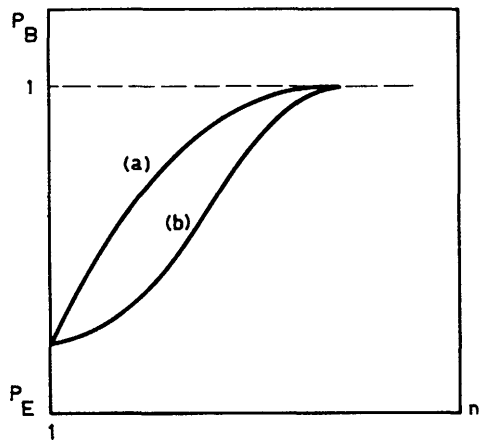


FIGURE 1. — **Block-error rate p_B as a function of block size n, with equiprobable or burst type error distribution**

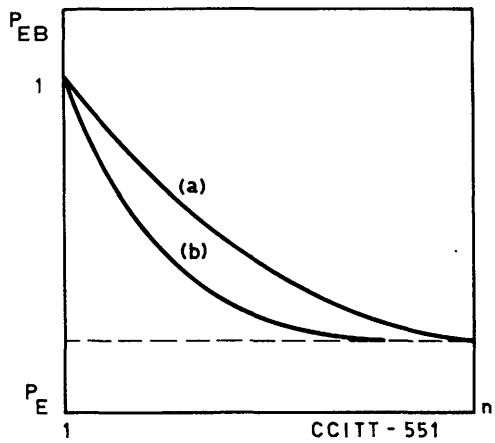


FIGURE 2. — **Element-error rate for erroneous blocks p_{EB} as a function of block size n, with equiprobable or burst-type error distribution**

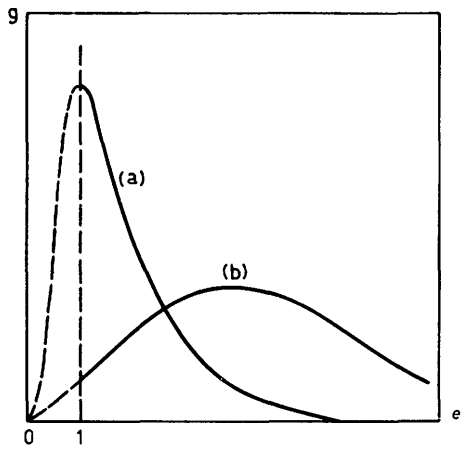
For an equiprobable distribution without error bursts, in which the errors would occur with equal probability p_E for any element, the following formulae would apply:

$$p_B = 1 - (1 - p_E)^n$$

$$p_{EB} = \frac{p_E}{1 - (1 - p_E)^n}$$

Figures 3 and 4 show typical curves for g and b as a function of e. The relatively high value of p_{EB} for the burst case leads to a relatively flat distribution g or b in which the occurrence of error patterns with a high number of e-errors cannot be neglected. This reflects the tendency of a burst-like noise to produce a high number of errors within a block. It might be noted that $n \cdot p_{EB}$ is the mean value of e for the distribution b.

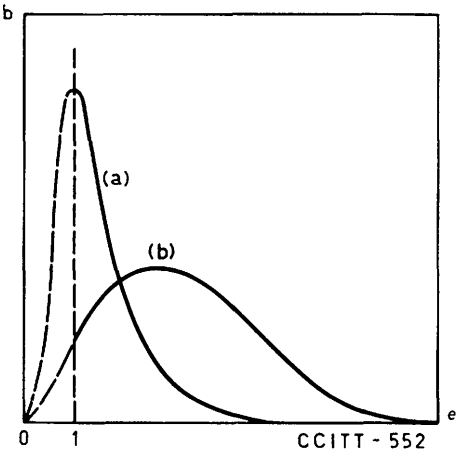
Appendix B shows measured curves g and b, and also for comparison a family of calculated curves assuming an equiprobable distribution of the errors within an erroneous block.



(a) = small p_{EB}
(b) = large p_{EB}

FIGURE 3. — **Distribution g of erroneous elements in error patterns with e-errors**

Parameters : p_{EB}, block size n



(a) = small p_{EB}
(b) = large p_{EB}

FIGURE 4. — **Error pattern distribution b for e-errors**

Parameters : p_{EB}, block size n

The reduction factor r as a function of e is obtained by calculating the number of undetected patterns of e-errors in a block of n elements and dividing it by the number of all possible error patterns of e-errors which is $\binom{n}{e}$. Figure 5 shows two typical curves for certain coding arrangements. Curve A relates to a code in which the minimum number d of errors which remain undetected [1] is $d = 2$. Curve B shows the same for $d = 4$. It might be noted that $r = 0$ for $e < d$. The particular codes illustrated in Figure 5 have $r = 0$ for all odd values of e.

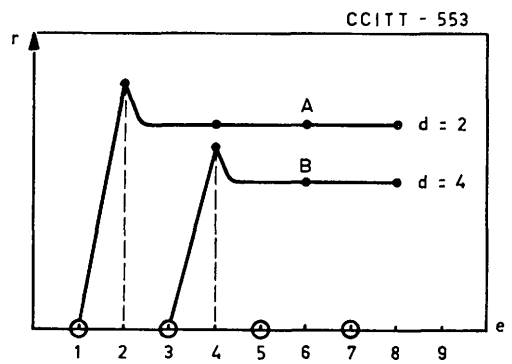


FIGURE 5. — **Reduction factor r for e-errors due to error-detection code (considering all possible patterns of e-errors to occur with equal probability) as a function of the number of e-errors**

Appendix C describes approximate calculations of curves of the reduction factor r as a function of e for coding arrangements using regularly spaced parity words. Appendix D shows calculated curves using binary-spaced parity words as coding arrangement.

[1] This corresponds to the so-called "minimum-Hamming-distance" d.

The curves for r and g (or b) can be used for calculating the nominal values of the total reduction R_E (or R_B) as already shown in section 3.

5. *Simplified method*

Approximate values for the distribution g or b can be calculated by assuming an equiprobable distribution of the errors within an erroneous block. The method described above can then be simplified. In this case only a range of values of p_{EB} need be known in order to calculate an approximate value of the total reduction.

Under this assumption, both g and b are given by a Bernoulli distribution, depending only on p_{EB}.

$$g = \binom{n-1}{e-1} \cdot p_{EB}{}^{e-1} (1 - p_{EB})^{n-1-(e-1)}$$

$$b = \binom{n}{e} \cdot p_{EB}{}^{e} (1 - p_{EB})^{n-e}$$

(Appendix B, Figure 4, shows the result of such calculations.)

Using these formulae the total reduction R_E (or R_B) can now easily be calculated and is only a function of p_{EB} for a given coding arrangement. The knowledge of the curve r as a function of e and the range of values of p_{EB} is therefore sufficient for calculating an approximate value of the total reduction. It might be noted that R_E or R_B, as a function of p_{EB} are often found to have a maximum at a certain value of p_{EB} reflecting the fact that for a particular coding arrangement a certain error density is least well detected. When no measured g (or b) distributions are available the above approach is still possible.

* * *

The methods described in the sections above take into account the burst character of line errors to a first approximation in so far as a distinction is made between periods with errors and periods without errors. The ratio of both is indicated by the block-error rate p_B. The element-error rate during periods with errors is given by the element-error rate for erroneous blocks $p_{EB} = p_E/p_B$. The knowledge of the interesting range of p_{EB} is sufficient for the application of the simplified method (section 5). If, furthermore, the distributions g (e) or b (e) are available from measurements on typical connections the more accurate method described in section 3 can be applied. In this way the efficiency of error-detection arrangements can be predicted in a comparatively easy and simple way (conventional evaluation).

6. *More complete methods*

The only assumption made in calculating the conventional reduction factor due to error-detection code r (e) was to consider that all possible error patterns of e-error within an erroneous block are equally probable. Thus the fine structure of the incidence of errors in actual connections has not been taken into account.

The following possibilities are being considered for more complete methods for comparing the efficiency of coding arrangements :

a) Sets of error separation curves which show the probability distribution of the time separation between errors may be used to programme a computer as an artificial error generator.

Alternatively the computer may also be used to imitate the error detection coding, whereby the R_E (or R_B) values are obtained directly.

By using this method a second set of curves for the reduction factor r (e) can be obtained which now take into account the corresponding fine structure of the error distribution. Using these r (e)-curves, the method described in the sections above can be applied without any change.

b) Another method of checking the efficiency of coding arrangements consists of using punched or magnetic tapes containing errors measured under actual line conditions. In order to reduce the time needed for checking, it is advisable to cut out noiseless periods compressing in this way the error tapes by a certain amount.

c) Still another method consists of using noise records for actual line conditions, for instance, recorded on magnetic tapes. The noise from these records (which may be compressed by cutting out noiseless periods) is injected at the input of the receiver. In this way, variations of the signal-to-noise ratio can be easily simulated by changing the signal or the noise level. This method does not take into account sudden changes in transmission loss or phase distortion.

d) For an actual system "on line" measurements are advisable as a final check. However, it must be mentioned that with this method no time compression is possible. Therefore, the measuring time may be extremely long to reach statistical stability.

APPENDIX A

This appendix shows a number of measured curves for both the block-error rate p_B and the element-error rate for erroneous blocks p_{EB}. The tests were carried out in the local network of Stuttgart on switched telephone connections. Figure 1 shows how these quantities depend on the block size n. For determining p_B the blocks containing errors were counted with an electronic counter. This number is divided by the number of blocks transmitted. For determining p_{EB} the mean number of errors within erroneous blocks is counted and divided by the number of elements in a block n. The dashed curves are calculated for an equiprobable distribution according to the formulae mentioned in section 4. The comparison between the measured and the calculated curves shows that the errors measured on telephone lines occur in "bursts".

Figure 2 shows the influence of the signal level on the block-error rate p_B and the element-error rate for erroneous blocks p_{EB}. Different block sizes have been measured. As it can be expected, the signal power has considerable influence.

Figure 2 was obtained by using a compressed noise tape recording (section 6c)) taken from actual connections. A check under actual line conditions was made at -3.6 N.

Figure 3 shows variations of block error rate p_B with time of day. Considerable variations can be observed. These are probably caused by traffic intensity. Fortunately, the block-error rate rarely exceeds a value of a few per cent.

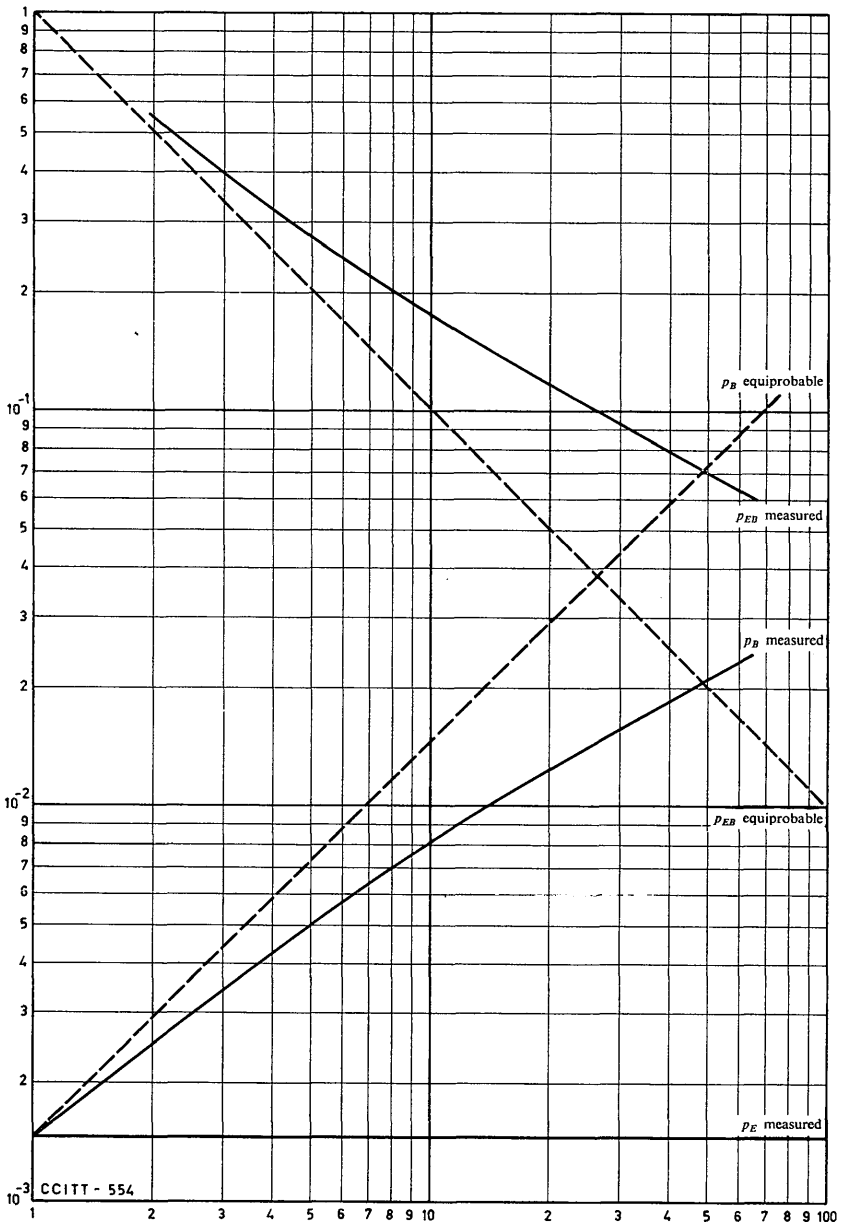


FIGURE 1. — **Block-error rate p_B and element-error rate for erroneous blocks p_{EB} as a function of block size n**

(Signal level at final selector : — 3.6 N) Measured Stuttgart (800 bauds)

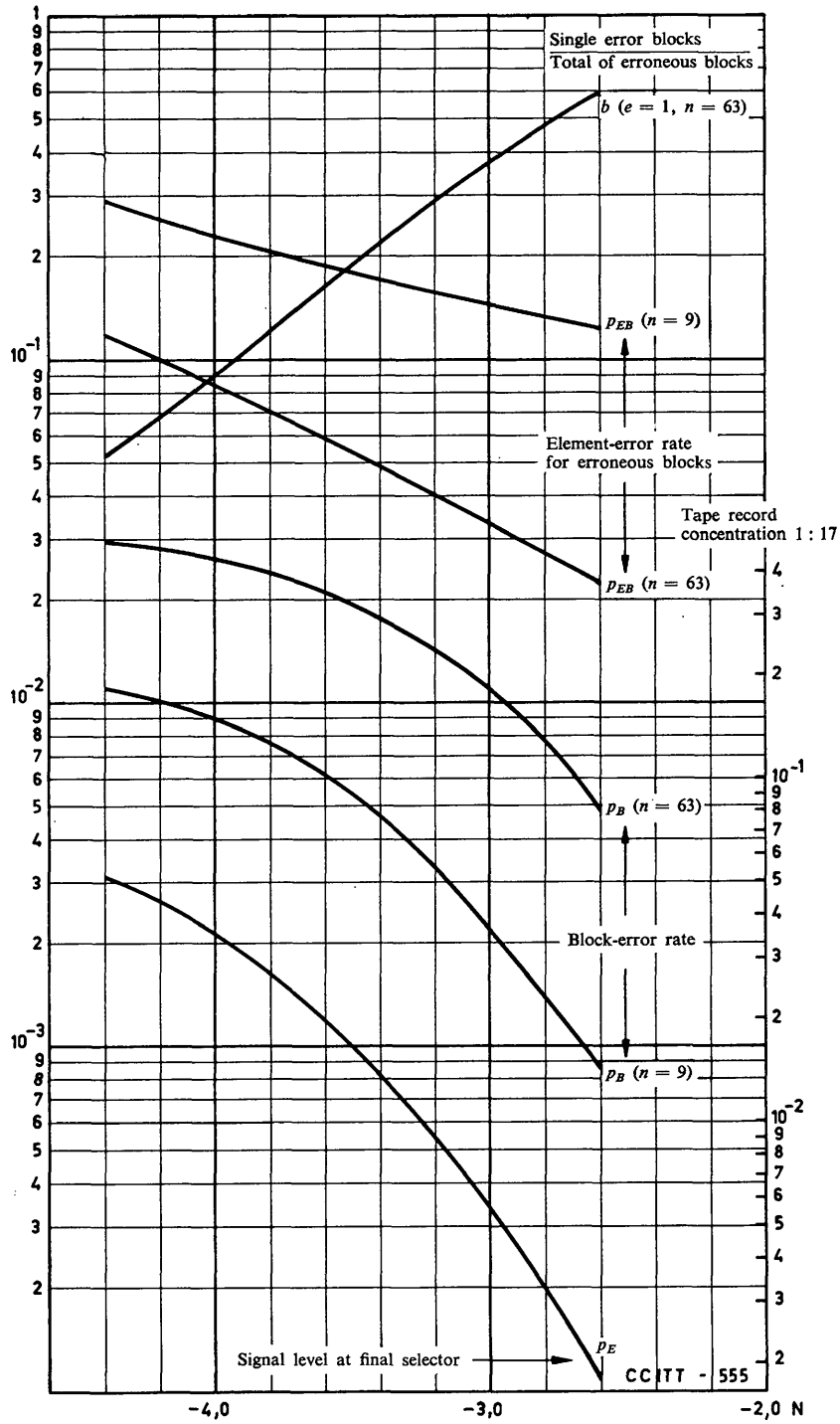

Single error blocks
Total of erroneous blocks
b (e = 1, n = 63)
pEB (n = 9)
Element-error rate
for erroneous blocks
Tape record
concentration 1 : 17
pEB (n = 63)
pB (n = 63)
Block-error rate
pB (n = 9)
pE
Signal level at final selector
CCITT - 555
-4,0
-3,0
-2,0 N

FIGURE 2

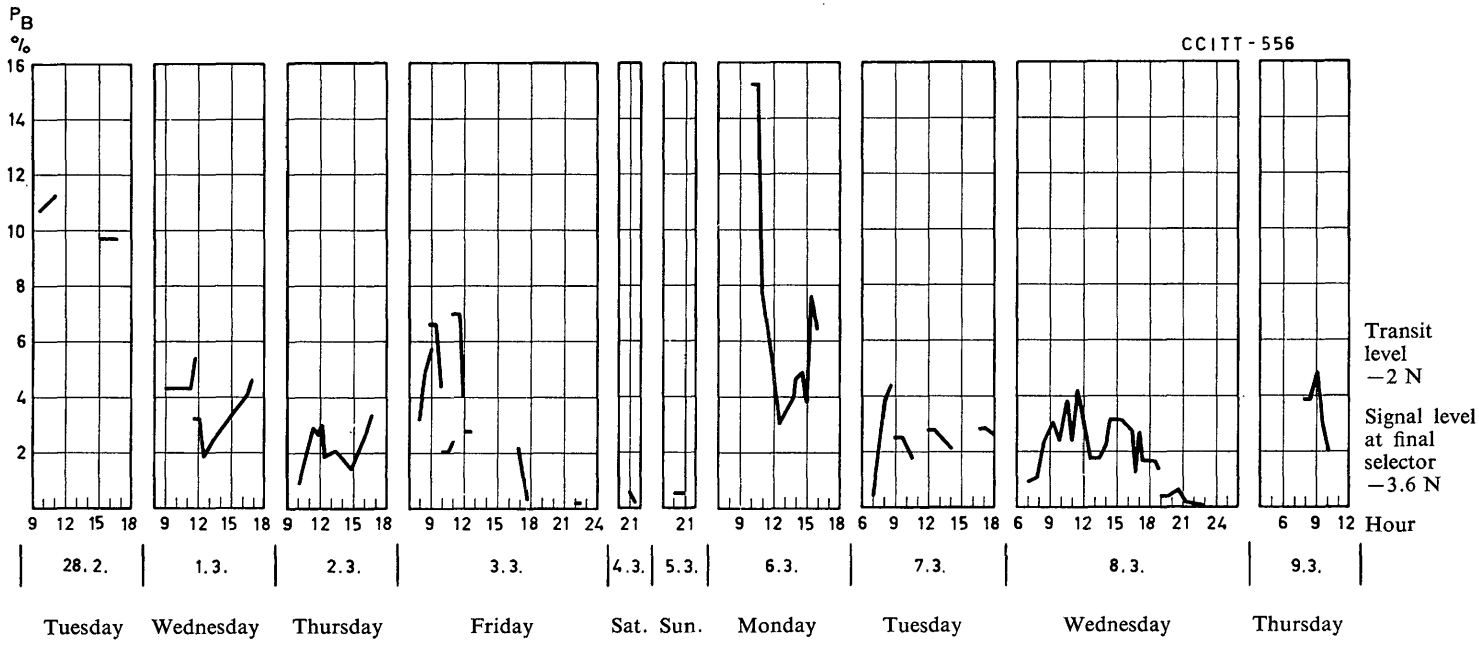


FIGURE 3. — **Block-error rate as a function of time and for different connections**

(Each curve segment refers to a particular connection in the local network of Stuttgart)

APPENDIX B

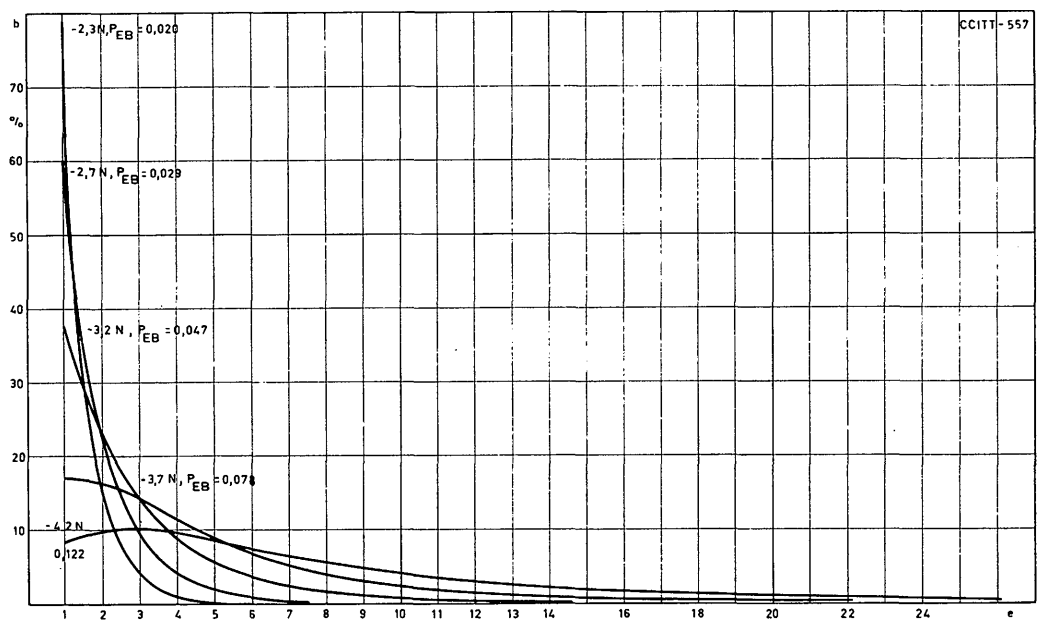
Appendix B shows curves of the distribution g and b measured on the telephone line network in Stuttgart.

Figure 1 shows b as a function of the number of errors within a block e. Each curve refers to a certain signal level and, therefore, to a corresponding value of p_{EB}. It can be observed that for small values of p_{EB} the curves concentrate at small values of e, while for larger values of p_{EB} the curves extend to greater numbers of errors. This is to be expected because $n \cdot p_{EB}$ is the mean value of the errors in erroneous blocks. Figure 2 shows the same for the distribution g.

Figure 3 shows the distribution g for different block sizes, the signal level being kept constant.

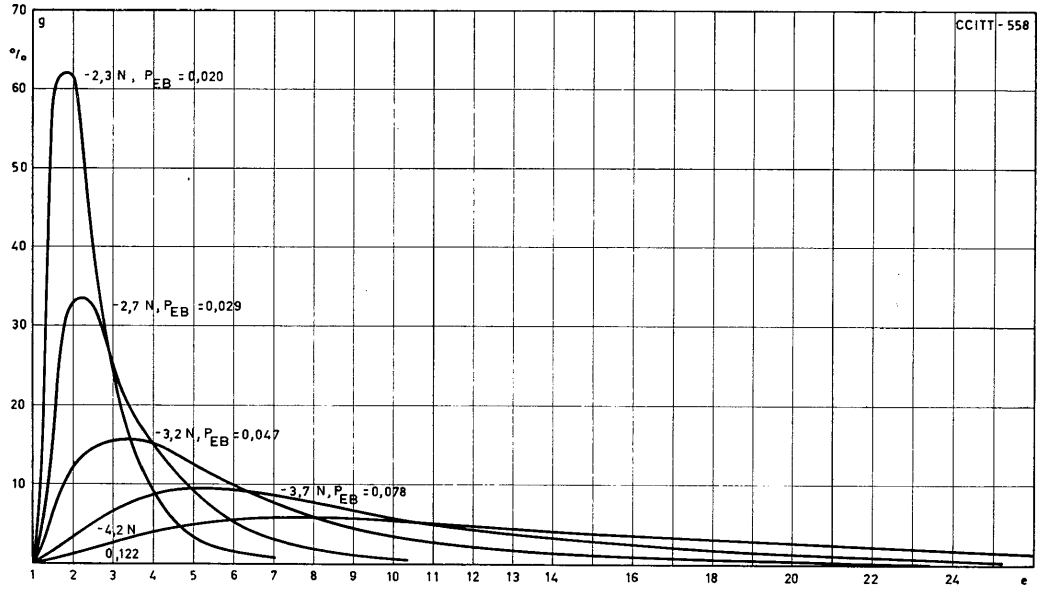
Figure 4 shows calculated distributions g for an equiprobable distribution (Bernoulli distribution) according to the formulae in section 5.

It can be observed that all of these curves for g and b show a similarity.



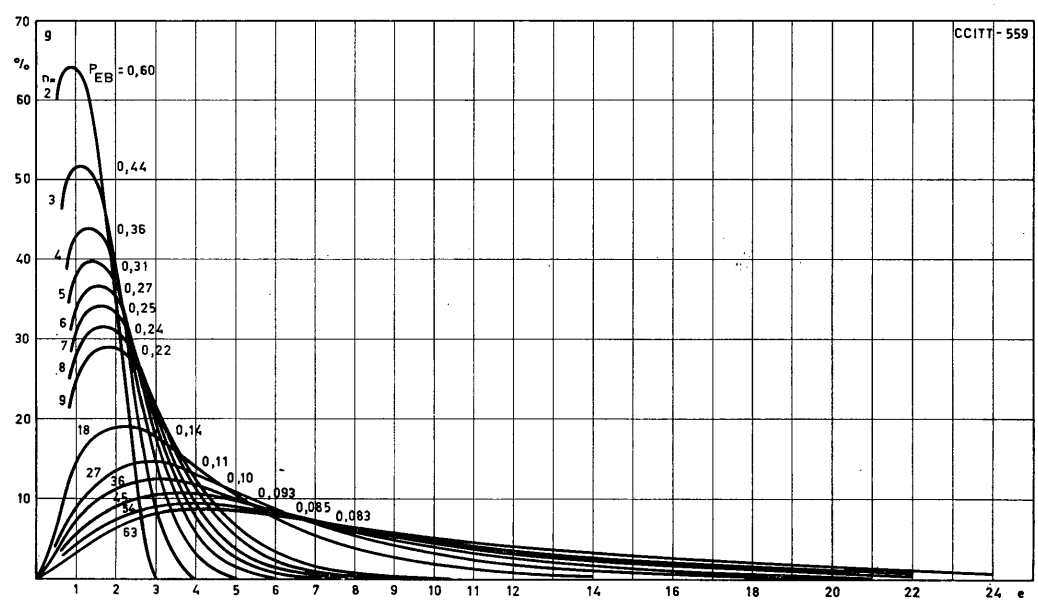
Measured Stuttgart (800 bauds)
$n = 63$
Parameter : p_{EB}

FIGURE 1. — **Error-pattern distribution**



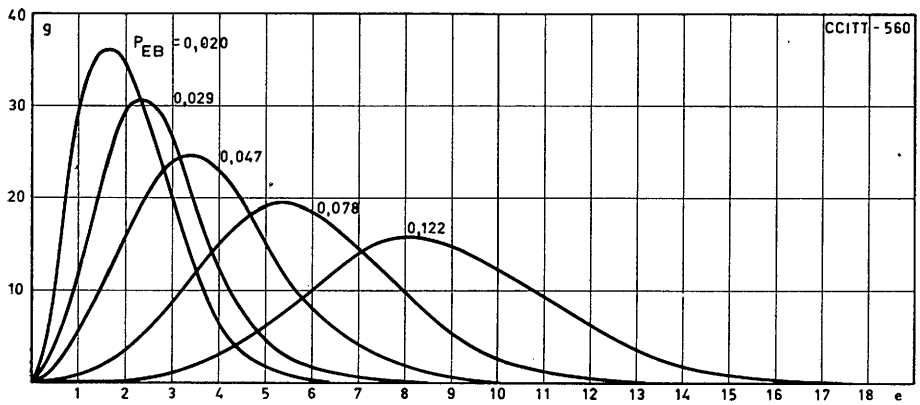
Measured Stuttgart (800 bauds)
$n = 63$
Parameter : p_{EB}

FIGURE 2. — **Distribution of erroneous elements in error patterns** g



Measured Stuttgart (800 bauds)
Parameter : block size n

FIGURE 3. — **Distribution of erroneous elements in error patterns** g



Equiprobable distribution
$n = 63$
Parameter : p_{EB}

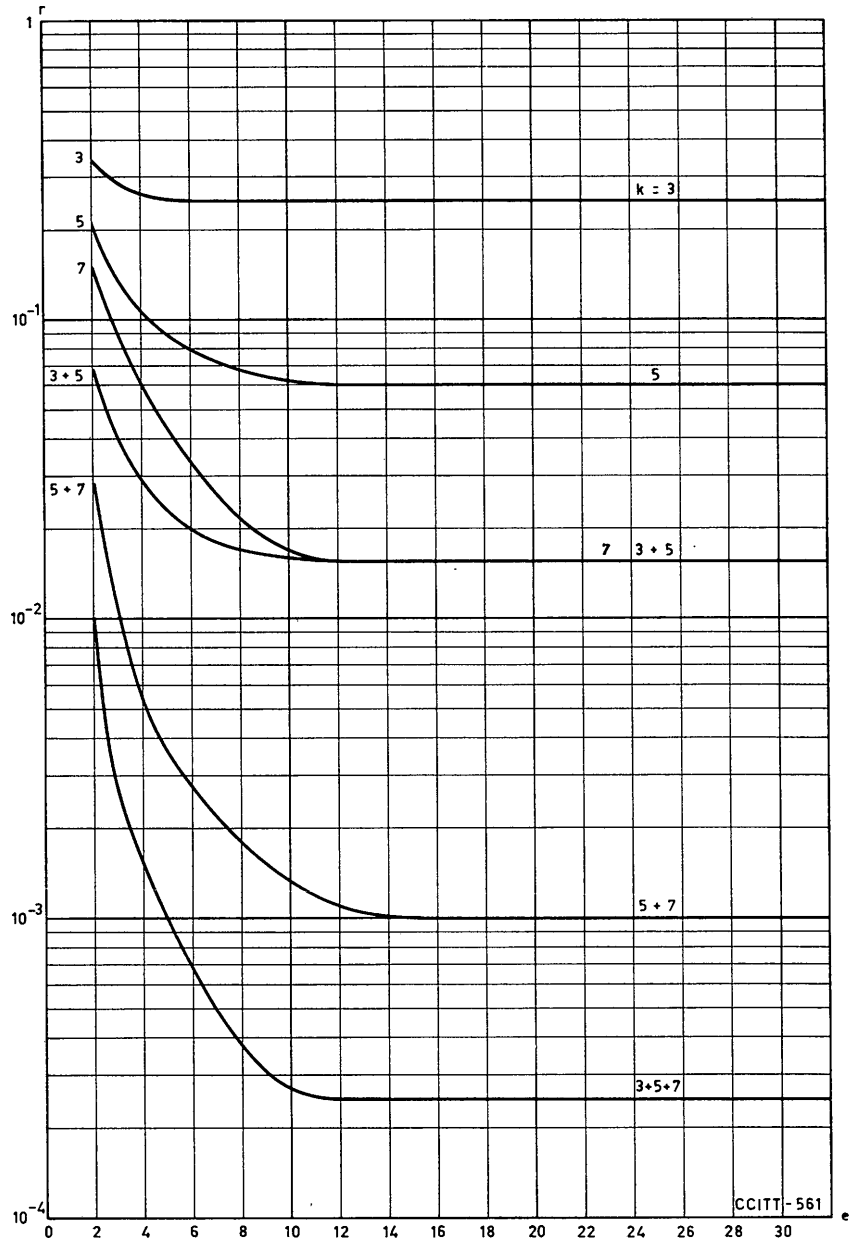
FIGURE 4. — **Distribution of erroneous elements in error patterns g (calculated)**

APPENDIX C

The diagram Figure 1 shows a number of curves for the reduction factor $r(e)$ for the regularly spaced coding arrangements. The curves were obtained by an approximate calculating assuming a large block size. The figures given as parameter refer to the number of parity words, i.e. 5 + 7 means that 5 parity words with a period of 5 bits and 7 parity words with a period of 7 bits are used, leading to a redundance of $k = 12$. For odd values of e the reduction factor r equals zero; this is not shown in the diagram.

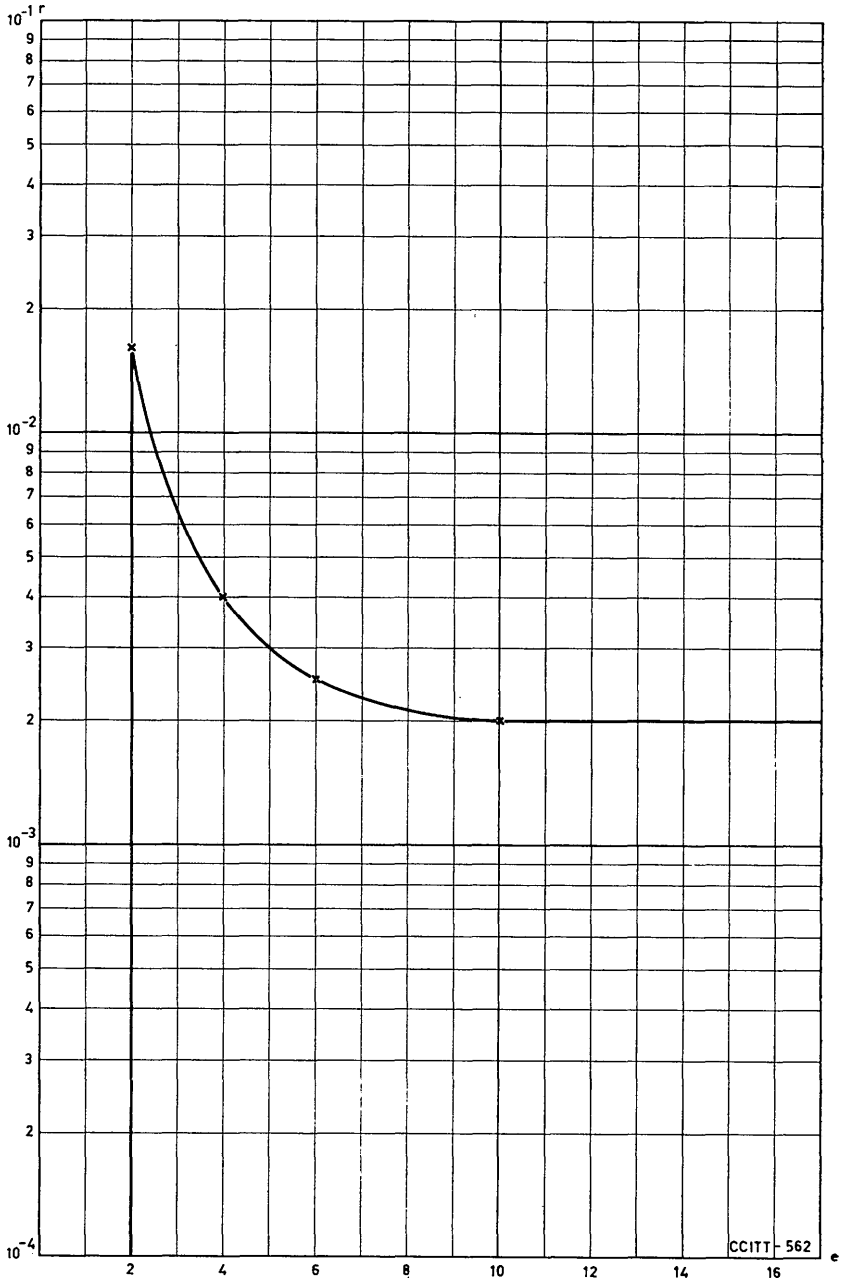
The diagram Figure 2 shows the result of an approximate calculation for the reduction factor $r(e)$ for a coding arrangement using the regularly spaced parity word arrangement 3 + 4 + 5.

For the block size $n = 360$ (corresponding to a teleprinter line) the minimum-Hamming-distance equals $d = 2$ and, therefore, the value of r for $e = 2$ is not zero. It should be mentioned that for odd values of e the reduction factor r equals zero ; this is not shown in the diagram.



Reduction factor r
Regularly spaced parity
(Large block size)

FIGURE 1



Reduction factor r
Regularly spaced parity (3 + 4 + 5)
$k = 12$

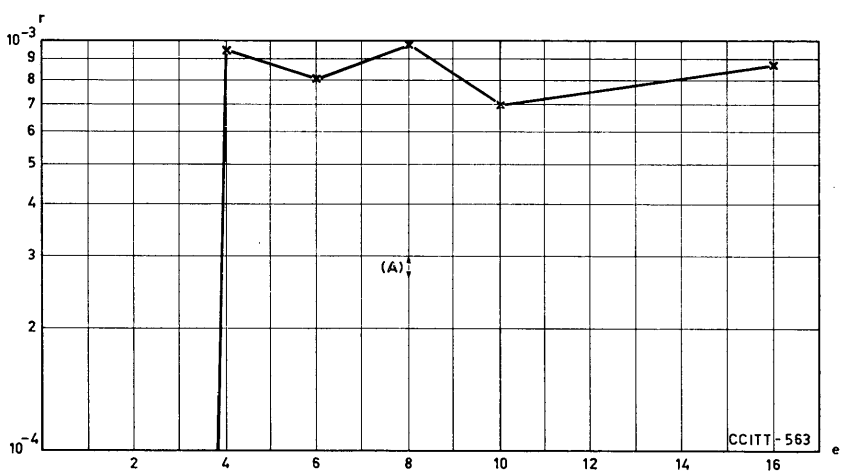
FIGURE 2

APPENDIX D

This diagram shows the reduction factor $r(e)$ for the binary-spaced coding arrangement with 11 parity checks. A 12th equally spaced parity check was added comprising the information elements Nos. 1, 5, 9, 13 ... and so on. The curve was obtained by programming an electronic computer with the coding method and by producing random errors by means of random numbers. For each value of (e) statistical stability was reached after about 20 undetected error patterns, which corresponds to about 20 000 blocks with e-errors being considered.

As the minimum-Hamming-distance for this coding arrangement is $d = 4$, the first non-zero value of r occurs at $e = 4$.

Furthermore, r equals zero for all odd number of errors e ; this is not shown in the diagram.



Reduction factor r
Binary-spaced parity (method Ma 4)
$n = 372$
$m = 360$
$k = 12$
(A) = approx. range of uncertainty

Supplement No. 65

CHILE TELEPHONE COMPANY — *(Contribution COM Sp. A/No. 20 — October 1963)*

ERROR-CORRECTION ARRANGEMENTS FOR DATA TRANSMISSION SYSTEMS

1. *Introduction*

Systems using error-correction codes have certain inherent disadvantages. The amount of redundancy included with the forward going message information is usually a very considerable percentage of the total information so that the effective speed at which information can be transferred is likely to be considerably reduced. Codes are available which can correct bursts of errors of a given length, but these require periods of error-free transmission to ensure satisfactory operation of the correction process. Extensive error-rate and error-distribution measurements on a wide variety of lines have shown that very frequently the essential "guard" periods contain errors, the effect of which is to make the production of an error-free copy difficult and expensive to obtain.

The preferred arrangement would appear to be error detection initiating the retransmission of data that have been mutilated. This implies an occasional very short delay which will be acceptable in most cases. The arrangement requires :

a) that information is sent forward in blocks, and

b) that a backward channel is available for the transmission of redundancy and/or control signals.

The forward going block of n bits might consist of m bits of data information, k bits of redundancy and s bits for a block sequence number providing block identification. On the backward channel would be transmitted q bits per block received, these comprising a decision signal to request retransmission when an error is detected. A preferred method of operation for some applications would be to transmit the redundancy for the received block over the backward channel and for the sending station to compare this with redundancy derived from the same block while it is being emitted ; the decision in respect to retransmission is then passed to the receiving terminal by means of the sequence number. In this case the forward going block consists of $n = m + s$ bits and the backward channel returns the k redundancy bits, this redundancy being transmitted simultaneously with the following block of data.

2. *Optimum block length*

A study has been undertaken to establish the optimum length of the data block in a system employing retransmission. Basically there appear to be four main parameters :

1. Percentage redundancy used ;
2. Percentage of retransmission ;
3. Loop propagation times ;
4. The expense of any stores needed to permit retransmission.

2.1 *Percentage redundancy*

The efficacy of a detection code—i.e. the error reduction factor for blocks containing any given number of errors, particularly those numbers capable of forming undetectable combinations—is closely related to the number (k) of redundant elements per block. Very approximately this reduction factor has a value 2^{-k}. For a predetermined number of redundancy elements the reduction factor changes very little with increase of block length; the evidence of many measurements suggests that in fact the reduction factor improves slightly as the block is increased above 100 bits to, say, 500 (see Appendix II).

2.2 *Percentage of retransmission*

It is clear that the larger the block, the higher the percentage of retransmitted blocks. The information in a previous contribution from the Chile Telephone Company (Supplement No. 22) gives details of retransmission rates derived from measurements on various types of connection for a variety of block lengths. The blocks retransmitted always include the block immediately following that in which the error has been detected. Because this second block may also have contained errors the ratio of the number of blocks retransmitted to the number of faulty blocks is always considerably less than two.

2.3 *Loop propagation time*

In medium-speed systems the decision whether to retransmit a block will not be available at the transmitter until some significant time after the transmission of the block has been completed. This period of time T may be represented by the expression:

$$T = D_d + D_b + \frac{k}{B_b} + C$$

where D_d is the propagation time in seconds of the forward data channel;

D_b is the propagation time in seconds of the backward supervisory channel;

k is the number of elements in the redundancy or decision signal returned over the backward channel;

B_b is the speed in bauds of the backward channel;

and C is the computation time in seconds to take the decision.

In order to simplify the logic circuits, it is desirable that the decision be made available at the sending station during the transmission of the subsequent block.

If n be the number of elements in the forward going block,

and B_d be the speed in bauds of the data channel,

then $$T = D_d + D_b + \frac{k}{B_b} + C < \frac{n}{B_d} \text{ seconds.}$$

The storage capacity necessary would then be for two blocks of minimum length n as determined above.

This is the most important factor in the determination of block length for medium- and high-speed systems. However, no definite recommendation is possible until the speeds of the forward and backward channels, the amount of redundancy (i.e. error-detection code arrangement) and the distance range over which the system is to operate are determined.

2.4 *Cost of store*

When a store is needed to allow a retransmission to be made, it is necessary to consider whether a larger block would increase the cost unduly.

2.5 *Message length*

A further consideration is the average length of message in elements. If this is considerably less than n, the block length, elements will have to be supplied to fill up the block ; this is a wasteful procedure if the traffic consists mainly of individual short messages. Of course, "fill-up" requirement is always necessary to some degree on the block containing the end of a message.

2.6 *Recommended block length*

The block length to be recommended should be a compromise based on the foregoing rather conflicting considerations. For the case of medium-speed transmission, we suggest that it should not be greater than 250 elements and that there is a lower limit set by the maximum tolerable redundancy (paragraph 2.1) and by loop propagation time (paragraph 2.3) considerations.

For low-speed applications we consider that a somewhat larger block could be justified.

3. *Error-detection codes*

Using block retransmission it is only necessary to detect the presence of an error or errors in a block; this is a far less complex process than locating each individual error for correction.

The error reduction factor of a code to detect errors on a block is a function of 2^{-k} approximately, k being the number of redundancy elements associated with the block. This approximation assumes a random distribution of errors ; in practice, owing to the "burst" nature of line interference, errors are less evenly distributed. Studies have been made of a number of codes to compare their theoretical performance with that obtained in the presence of line noise, momentary interruptions and breaks. The codes investigated range from simple row parity to more complex arrangements of two- or three-co-ordinate parity. They are :

1. Row parity
2. Constant ratio
3. Interleaved parity
   a) Column parity
   b) Shared column parity
   c) Two-co-ordinate parity
   d) Three-co-ordinate parity
4. Binary parity
   a) Straight binary (Hamming type codes)
   b) Scrambled binary
5. Mixed co-ordinate parity

3.1 *Row parity code*

Row parity is equivalent to character parity, each parity check element combining with a group of consecutive elements sent serially to form a parity word. This code cannot detect even numbers of errors in a parity word, and so it is quite inadequate for detection purposes, particularly where certain types of modulation system are used. Measurements indicate that only some 80% of the errors are detected.

3.2 *Constant ratio codes*

These codes are well known but are generally inadequate for medium-speed data transmission.

3.3 *Interleaved parity codes*

The simplest form of these codes, column parity, has theoretically the same power as a row parity code of equal magnitude. However, due to error burst effects, the column parity is several times more effective than row parity (see Appendix II).

A two-co-ordinate parity code increases security against errors by combining two sets of parity words, e.g. row and constant ratio or row and column parity are applied to the same block of data. With such an arrangement all groupings of errors will be detected other than certain patterns involving even-numbered groups of errors from 4 upwards.

Three-co-ordinate parity is an arrangement combining three sets of interleaved parity words and provides even further security. Only certain patterns of even numbers of four or more errors can remain undetected. One arrangement investigated known as 3-5-7 provides 15 parity elements arrived at by arranging every third element of data in the block to form a parity word (three words), then every fifth element and then every seventh element, making a total of 15 words or a word of 15 elements for the block parity. This gives an arrangement which, in general, provides an adequate degree of security using logic circuits of simple design.

3.4 *Binary parity*

Binary codes, generally developments of the Hamming code, have been investigated in recent years for error-correction purposes. As detection codes they are very valuable, but they suffer the disadvantage that odd-numbered groups of errors can remain undetectable as well as even-numbered groups.

Straight binary, in which the numbered position of every element in the block is directly related to numbers in the binary code, has shown disadvantages in the face of error bursts. This can be improved by scrambling the binary order. In one application the block of data is fed to a series of parity counters, through a series of gates which are opened and closed in accordance with the sequence cycle generated by a feedback shift register.

3.5 *Mixed co-ordinate parity*

With interleaved parity, say 3-5-7, it is obvious that undetected two-error patterns can occur when the block size exceeds ($3 \times 5 \times 7$) or 105 bits, i.e. element No. 106 has the same parity as element No. 1. By using a suitable binary code in conjunction with two interleaved parity co-ordinates, a code is obtained which obviates this difficulty for the larger blocks and detects all odd-numbered groups of errors. The use of a three-co-ordinate mixed parity scheme of this nature is very flexible as one or other of the co-ordinates may be disregarded when working to subscribers who are satisfied with a lower degree of security. Thus a subscriber fully equipped to work with, say, interleaved 3 and 5 and binary 4 can work with three co-ordinates to a subscriber similarly equipped or with simple interleaved 3 to a subscriber who does not demand more than the lowest degree of security offered.

3.6 *Coding arrangements using feedback shift registers to generate parity words*

The use of feedback shift registers for coding has been investigated in recent years for error-correction purposes. As detection devices, such arrangements are about as powerful as three

co-ordinate interleaved parity schemes with the same number of elements as stages in the registers. They are, however, vulnerable, as are all binary codes, to groups containing odd numbers of errors.

4. *Code assessment*

Analysis of error measurements made on typical connections has enabled a computer to be programmed to generate errors with similar error distributions. Using the computer and such artificially generated errors the capabilities of a number of codes can be compared fairly rapidly for different block lengths.

Appendix I presents a typical sample of the results of such measurements for a block size of 100 bits with 2, 3, 4, 6 and 8 errors per block for various single and combined interleaved and binary parity arrangements. It will be seen that the measured figures are generally more favourable than the theoretical, this being due to the simulation of line error distribution by the artificial error generator.

These results assume that the parity element in each code is determined according to an "excess modulo 2" law. Some improvement can be brought about in both the measured and theoretical E.R.F. values for certain multi-coordinate arrangements selected parity elements are determined instead according to an addition law including carry from word to word. If the "excess modulo 2" law operates in the case of a first set of parity words of uniform spacing, then it is advisable, in the case of a second (and third) set using uniform spacing, to determine the parity elements according to the addition law with carry from word to word. Such an arrangement improves the measured and theoretical values for codes 3.5 ; 3.7 ; 5.7 ; and 3.5 (3) by a factor of two. It improves the values for 3.5.7 by a factor of four.

5. *Summary of coding arrangements*

Row parity	Very inadequate
Constant ratio	Inadequate
Column parity	Inadequate
Interleaved parity	
Two co-ordinates	Marginal
Three co-ordinates	Acceptable in certain cases
Binary	
Straight binary	Marginal
Scrambled binary	Good
Mixed co-ordinates	Adequate and flexible

6. *Conclusions*

For medium-speed data transmission it is desirable that the block length should not exceed 250 elements. The acceptable minimum limit depends on the tolerable percentage redundancy and considerations involving propagation time. For low-speed systems a somewhat larger block can be justified.

When considering the choice of a suitable code, the incidence of errors and their distribution in the block size chosen should be borne in mind. The use of two co-ordinates of interleaved parity in conjunction with one of binary enables all blocks containing two or odd numbers of errors to be detected. One great advantage in this arrangement is the flexibility it provides in that varying degrees of security may be obtained.

APPENDIX I

Error-reduction factors for 100-bit blocks

Errors per block	2		3		4		6		8	
Detection arrangement	Measured E.R.F.	Theoretical E.R.F.	Measured E.R.F.	Theoretical E.R.F.	Measured E.R.F.	Theoretical E.R.F.	Measured E.R.F.	Theoretical E.R.F.	Measured E.R.F.	Theoretical E.R.F.
	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$	$\times 10^{-2}$
3	31.8	33.3	—	0	22.8	26	25.6	25.1	24.8	25.0
(3)	—	—	14.6		13.4	—				
5	19.2	20	—	0	7.3	10	5.23	7.7	5.99	5.7
(5)	—	—	3.79		4.96					
7	9.15	14.3	—	0	2.26	5.5	1.66	3.21	1.38	2.1
(7)	—	—	1.15		3.8					
3, 5	3.18	6.67	—	0	2.01	2.6	1.69	1.93	1.51	1.65
(3, 5)	—	—	0.85		1.17					
3, 7	1.4	4.76	—	0	0.72	1.4	0.27	0.81	0.446	0.54
(3, 7)	—	—	0.72		1.11					
5, 7	0.8	2.86	—	0	0.167	0.55	0.052	0.26	0.106	0.165
(5, 7)	—	—	0.93	2.3	2.3					
3, 5, 7	0	0	—	0	0.027**	0.14	0.0167	0.068	0.064	0.036
(3, 5, 7)	—	—	0.61		0.77					
3, (3)					3.24					
5, (3)					1.05					
3, 5 (3)					0.29					
(7) S.B.					0.60					
(12) S.B.			0		0*		0		0	

Note 1. — Detection arrangement "3" denotes an interleaved arrangement with three parity elements.
Bracketed figures refer to binary arrangements.
S. B. denotes scrambled binary, using a feedback shift register to derive parity.

Note 2. — Measured E. R. factors are based on error sample size averaging 30 000 errors, with the exception of the figures with asterisks.
Single asterisk based on 5000 errors, double asterisk 146 000.

Note 3. — Theoretical E. R. factor for binary codes codes may be taken as 2^{-k}, where k is the number of elements in the parity word.

APPENDIX II

Results of computer analysis of errors occurring on data transmission tests over leased circuits

All tests used frequency modulation equipment (frequency shift $\pm$ 400 c/s, mean carrier frequency 1500 c/s) transmitting at 800 bauds. Mean transmitting signal level was -12 dbm. Emphasis networks giving 6 db advantage to the higher frequency were fitted at both transmitting and receiving terminals.

Test code	Undetected errors							
	Row parity (10 % redundancy)				Single-column parity 20 %	10 %	5 %	2 %
	50	100	200	500	50	100	200	500
EH1	0	0	0	0	0	0	0	0
EH2	0	0	0	0	0	0	0	0
EH3	0	0	0	0	0	0	0	0
FH1	68	34	34	0	0	0	0	0
FH2	0	0	0	0	0	0	0	0
FH3	0	0	0	0	0	0	0	0
FH4	0	0	0	0	0	0	0	0
GH1	0	0	0	0	0	0	0	0
GH2	0	0	0	0	0	0	0	0
GH3	0	0	0	0	0	0	0	0
GH4	138	34	26	4	12	12	14	6
IH1	0	0	0	0	0	0	0	0
IH2	8	8	2	8	0	0	0	0
IH3	14	14	10	10	0	0	0	0
IH4	14	12	12	8	0	2	0	0
JH1	2	2	2	2	0	0	0	0
JH2	2	2	2	2	0	0	0	0
JH3	0	0	0	0	0	0	0	0
JH4	0	0	0	0	0	0	0	0
KH1	0	0	0	0	0	0	0	0
KH2								
KH3	0	0	0	0	0	0	0	0
KH4	4	2	2	2	0	0	0	0
LH1	8	8	2	0	0	0	0	0
LH2	12	10	10	10	0	0	0	0
LH3	0	0	0	0	0	0	0	0
MH1	18	18	4	2	8	8	0	0
MH2	260	144	62	18	2	62	2	0
MH3	2	2	2	2	0	0	0	0
TH1	40	34	34	22	0	0	0	0
TH2	6	6	6	6	0	0	0	0
TH3	0	0	0	0	0	0	0	0
UH1	0	0	0	0	0	0	0	0
UH2	0	0	0	0	0	0	0	0
UH3	2	2	2	2	0	0	0	0
UH4	2	2	2	2	0	0	0	0
VH1	0	0	0	0	0	0	0	0
VH2	0	0	0	0	0	0	0	0
Totals	600	334	214	100	22	84	16	6

Supplement No. 66

SWEDEN — *(Contribution COM Sp. A/No. 59 — June 1963)*

CONTROL OF ERRORS FOR DATA TRANSMISSION ON SWITCHED TELEPHONE CONNECTIONS

1. *Introduction*

During recent years investigations have been carried out in Sweden on a method for error correction by means of retransmission employing novel means for error detection. The receive side redundancy control has been arranged to be supported by modulation detector control. It has been shown that by employing such a method it is possible to achieve a considerable cost reduction as compared with a pure redundancy control method, without sacrificing the degree of error reduction.

The aim has been to develop a data transmission system capable of correcting line errors. An error-free output is regarded as essential. Such a system, including suitable peripheral equipment, is considered to be a suitable device to lease to data subscribers.

2. *System design*

The system is based on the C.C.I.T.T. type F.M. modem for 600 bauds (mid-frequency 1500 c/s) or 1200 bauds (mid-frequency 1700 c/s) with a supervisory channel for up to 75 bauds (mid-frequency 420 c/s). Transmission at a character rate of 100 ch/s 900 bauds (mid-frequency 1500 c/s) has also been considered. At the receive side the modems are equipped with modulation detectors which check that the incoming signal level is within certain limits, e.g. between 50% and 180% of the mean level. An AGC amplifier allows the mean level to vary within a 40-db range. The modulation detectors are used to initiate requests for repetition in the same manner as the retransmission requests initiated by the redundancy control. The use of the modulation detector has resulted in the number of control elements required per block being reduced to two. It has therefore been possible to reduce the length of a block to such an extent that each block corresponds to only one character without the percentage lost time, due to the redundancy, becoming excessive.

The resultant individual error control of each character has several operational advantages. The characters are released one by one from the data source and arrive at the receive equipment in the same way. The reduction in the length of the individual blocks also means that the required memory capacity has been substantially reduced. In the send side it is only necessary to store the number of characters which corresponds to the propagation time from the send equipment to the receive equipment and back again. At the receive side it is only necessary to store one character in order to ensure an error-free output. In an error-correcting system which depends solely on redundancy control, four or five times this storage capacity is required.

A very high degree of error-reduction capability has been achieved, which can be explained as follows.

One of the two control elements, which is preferably generated within the data source, is used for normal parity check, while the other is chosen so that it will detect errors which affect two adjacent character elements. Hence any disturbance must be spread over at least three signal elements before the redundancy control is likely to show signs of unreliability. In such

cases, however, the disturbance is almost bound to be accompanied by a change of signal amplitude which the modulation detector can detect. During the period that measurements have been proceeding over the Swedish telephone network no error has occurred, up to the time of writing, which has not been detected and corrected.

3. *Measuring results*

A model of the system has been tested on the switched telephone network of Sweden. By suitable level adjustments, a very low signal-to-noise ratio was used in order to obtain a large number of errors within a short period. Tests were carried out at 1200 bauds using a test message comprised of all possible 8-bit characters with odd parity. Character speed = 133 ch/s.

The following results were obtained :

Number of transmitted characters	8 827 000
Number of alarms from parity check	45 800
Number of alarms from modulation detector	108 000
Number of retransmissions	150 600
Number of erroneous characters (estimated)	50 000
Number of undetected erroneous characters	0

The number of retransmissions is three times the number of erroneous characters. The time lost due to these excessive retransmissions would be negligible under normal level conditions when the error rate would not be so high.

Supplement No. 67

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — July 1963)*

CONTROL OF ERRORS FOR DATA TRANSMISSION ON SWITCHED TELEPHONE CONNECTIONS

Source of error and responsibility for correction

The sources of error in an integrated data-processing system are those occurring during data preparation, data transmission (by telephone lines, etc.) and data processing. Any errors in the original data preparation stage cannot in general be detected in the subsequent stages of transmission and processing ; such errors must therefore be detected and corrected at the data source.

Similarly, any errors generated during transmission should be detected and corrected before the data are processed. It is of course possible during the data processing to introduce feasibility checks on the original data or on the subsequent results of the processing. Such a method of detecting errors produced by data transmission would vary with the feasibility limits for each

set of data or job. Consequently, it would not be economical to employ a general purpose computer on this procedure as, apart from the fact that it would be a trivial task in relation to the work the machine could be doing, there would be the inevitable system delays or restrictions which would accrue in interrogating the transmitting end for correction.

Use of an error-correcting code would avoid some of these difficulties but would result in the loss of the advantages of an error-detecting/automatic repeat (ARQ) system, which is particularly suited to the error distributions on telecommunications networks.

This suggests that errors occurring on the transmission path should be dealt with within that link ; in a similar way that errors in the data preparation and data processing parts of the over-all system should be detected and corrected at their source. The causes and effects of error in each of the three parts of the system are different and unrelated.

Modulation and coding as methods of error control

Errors in the transmission path are due to deficiencies and limitations of the telecommunications network. The means of combating them are conditioned by :

a) the fact that, to date at least, the network has been designed primarily for speech transmission ;

b) the method of modulation, which must be compatible with the network and also give the minimum error rate having regard to

c) the availability of error control methods of encoding.

The most economical solution for a given information and error rate involves an interaction between these three. Under a) it may be that the technical needs of data transmission will be an increasing factor in the design requirements of new telephony transmission systems. Both b) and c) may be related and are component techniques in the over-all assessment of the economics of the method of error control.

An example of this is an amplitude-modulated (A.M.) system where errors due to noise can be counteracted by raising the transmitted power or alternatively encoding the digital signals to increase the redundancy and the possibility of error detection. The relative merits of these alternatives have to be weighed both technically and economically. Similarly a frequency-modulated (F.M.) system achieves a signal-to-noise error advantage over an A.M. system and as before the merits of increasing the redundancy of the digital signals against the more complicated F.M. transmission system has to be considered. Finally, we give an example of a phase-modulated system. This demands a phase reference. One way of achieving this is by differential coding of the binary signals, which may in turn modify the error pattern by turning single-element into double-element errors.

In considering the form of the error-control system and encoding, it is not wise to dissociate this from the method of modulation. In fact the modem is one stage of error control and one point where redundancy may be inserted or removed depending upon the most economic solution.

Standardization and flexibility

For maximum flexibility in the design of data-processing and data transmission equipment the minimum of constraints on one or the other should be exercised. Most of these for the medium-speed range have been defined.

There will be advantages purely from the data-processing point of view in having a standardized data alphabet designed to be optimum for use with data-processing machinery. However, the bit structure of this should not be conditioned by the requirements of error control or modems associated with the line, as this would produce inflexibility and make difficult any subsequent revision if the two standards were locked together. Furthermore, any redundant coding added to the data within the error-control equipment necessary to combat errors on the line transmission path should not limit the form, content and method of presentation of the original data in size of data block, alphabet, etc.

Ideally the data from the data-processing source should only include sufficient redundancy to deal with detection or correction of errors that may arise within the data-processing part of the system. Generally, the signal error rate at this point is very much lower than that experienced in the data path over the telecommunications network and in principle a small number of check digits will suffice. The redundancy required over the telecommunications network is much higher and will involve coding capable of detecting a larger number of errors and particularly groups of adjacent element errors (error bursts). The data should not have to bear this (with consequent inefficiency in the data-processing machinery) except over the line transmission path.

Such an approach is consistent with the earlier discussion based on the philosophy that errors should be dealt with at their source.

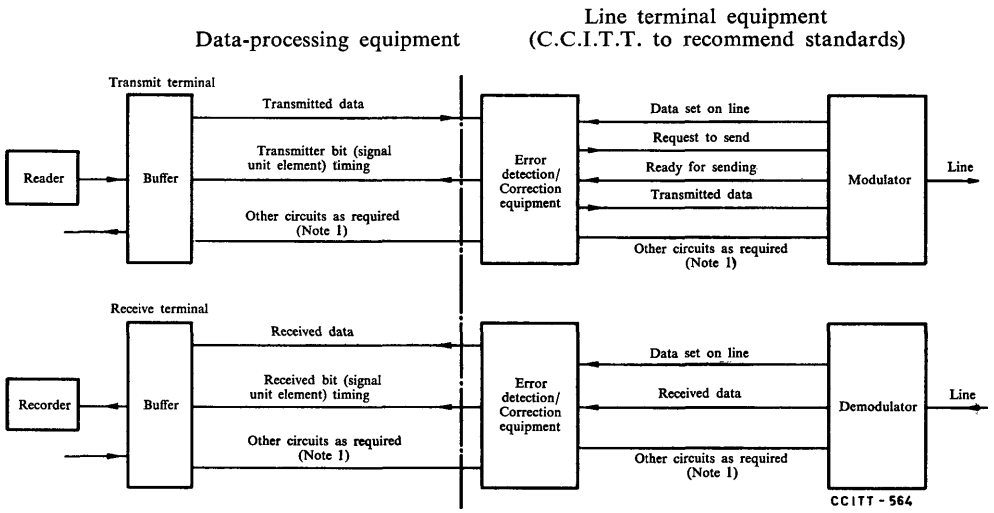
The United Kingdom Administration considers that the C.C.I.T.T. should lay down standards for error control for medium-speed systems 600 and 1200 bauds based on the foregoing considerations.

Basic form of proposed 600/1200-baud error control system

Figure 1 shows the block schematic arrangement of a proposed error-control system working at modulation rates of 600 and 1200 bauds. The interface circuits between the data-processing, error-detection/correction equipment and modem are in accordance with the C.C.I.T.T. specifications. The line terminal equipment has been shown as including error-control equipment and the modem, and these are regarded as those parts for which C.C.I.T.T. should make standards.

The arrangement allows the device forming the source of data signals within the data-processing equipment to be of any type, e.g. punched tape, cards, magnetic tape etc. Similarly the error-detection/correction equipment could be of any type and independent of the data source. Compatibility of speed is not essential between the data source and the error control. This is achieved by the error-detecting/correcting equipment accepting a stream of signal elements (a serial form for the moment is assumed) over the "transmitted data" wire from a buffer within the data-processing equipment.

The buffer store may be of any size to suit the particular data-signal source, e.g. for a seven-track punched tape a seven-element buffer store would be required. A signal for clocking the signal elements from the buffer would be supplied over the "transmitter bit timing" wire from the error-detection/correction equipment. The signals timed (clocked) out of the buffer store would be assembled into a standard block and stored within the error-detection/correction equipment. Simultaneously, the appropriate redundancy would be calculated and added at the end of the block of signals and transmitted to line serially via the modem.



Note 1. — These circuits may include circuits such as speed selector.

FIGURE 1. — **Basic system for error control**

With this arrangement, so long as no errors occur on the transmission path, a continuous succession of blocks of signals would be sent. The buffer store in the data-processing equipment, under the control of the clock signal, would feed the data signals continuously to the line terminal equipment. The buffer store would have to be continuously refilled by the data signal source, e.g. tape reader. Whatever signal was in the buffer store would thus be transmitted. In the event of errors on the transmission line, an error-correction cycle is started, and during the cycle no new data can be transmitted. The signal element clock signal over the "transmitter bit timing" wire must therefore be interrupted and no further data signals taken from the buffer store. As soon as more data signals were required, e.g. after correction had been completed, the contents of the buffer store would continue to be clocked out from the point where it had been left off.

At the receiving end the error-detecting/correcting equipment would deliver the original stream of data signals (having checked the accuracy and removed the redundancy) to the receiving buffer store in the data-processing equipment. These would be clocked by signals over the "received bit timing" wire. The buffer store would be any size suitable to the recording device. As explained for the transmitting end, during correction cycles the clock signal would be interrupted and filling of the buffer store would be temporarily held up.

Advantages of proposed basic arrangement

1. A clear demarcation is made between line terminal equipment (error-detection/correction plus modems) for which the C.C.I.T.T. would be responsible for standards and the data-processing equipment (data-reading/recording device and associated buffer stores), for which the C.C.I.T.T. would not be primarily responsible. Such an arrangement would allow a high degree of independence in standardizing the two sets of equipment.

2. The data-processing code structure and alphabet would be independent of the line terminal equipment.

3. It allows the fusion of error-detecting/correcting techniques, signal regeneration and modulation methods to minimize costs and undetected error rates.

4. The received data signals sent to the customer would be regenerated and the excursions of element-error rates controlled. Any division of responsibility for maintenance between the line equipment and the data-processing equipment would be eased by having unambiguous signals at the demarcation point.

5. The long-term error rate achieved by the error control of the basic standard system should be so chosen as to be adequate for the vast majority of users.

There may be, however, a customer with a specialized application, or so situated that he requires extra protection. Such extra code protection would be his responsibility and be included in the data-processing equipment. With the basic system as proposed in Figure 1, additional error-detection/correction equipment could be interposed between the reader and recorder and their associated buffer stores in the data-processing equipment. This equipment could work in cascade with and independently of the error control of the line terminal equipment.

6. Although in the basic system Figure 1, the error-detection/correction equipment is shown as physically separate from the data-processing equipment, and connected to it by a standard interface, this does not preclude the following proposed standard error-control system being achieved by stored programme techniques within the data-processing equipment.

Points to be standardized in error-detection/correcting equipment

1. *An ARQ control to be used*

It is proposed that the system should use an error-detection code in a standard block transmitted at 600 or 1200 bauds. Correction would be by detection of errors at the receiving end and signalling the fact to the transmitting end over a low-speed channel of 75 bauds. With the error distributions likely to occur an ARQ form of error control is proposed, as this allows a lower redundancy and consequently a greater information transfer rate for a given error rate than would be the case if an error-correcting code is considered.

2. *Error-detection code*

The requirements for the code are :

a) It should be independent of data coding both as regards the number of elements per data character and pattern of 0's and 1's.

b) It should have low redundancy to ensure maximum efficiency in the transmission of information over the line.

c) It should be capable of detecting all small numbers of element errors occurring within a block say up to 3 and all single bursts of errors up to 12 elements in length, as well as any number and arrangement of odd element errors within a block. A very high proportion of all other possible element-error combinations should also be detected by the code.

d) Equipment required to generate the code should be simple.

A code which satisfies the requirements listed is of the cyclic error detecting type. Such a code containing N binary digits per block of which K digits are information, carrying digits would thus contain $(N-K)$ check digits. If element errors on the telecommunication network followed a distribution such that each error was discrete and unrelated, then the probability of an undetected block error would be 2^{-N+K}. However, as a large number of errors per block may consist of single elements as well as a number of bursts of consecutive multiple errors it is possible by consideration of error distributions to arrange the check digits so as to give an improvement better than 2^{-N+K}.

The particular requirements detailed under c) above could be met by appropriate design of a shift register for generating the cyclic code. The number of check digits required to deal with the suggested requirements under c) would be 12.

3. *Block size*

The block repetition rate on a circuit is related to the error rate. If the block is made large the percentage retransmission time will increase. Conversely for a fixed number of checking elements the smaller the block the greater the redundancy and the lower the information transfer rate.

Consideration of the error distribution on the switched telephone network suggests that, with the number of redundant elements proposed, a block size of 256 (a binary number of elements in the block is likely to be more convenient) will give the maximum transfer rate of information. One other consideration is that in an ARQ form of feedback error control there must be a minimum of two blocks at the transmitter for repetition purposes. The transmission time round the circuit loop in this case must be less than—including delays in each terminal—the total number of signal elements in a block divided by the modulation rate. At 1200 and 600 bauds this would correspond to 213 and 426 milliseconds respectively.

For normal international connections this should be sufficient, but it is possible that on very long connections, particularly those via space satellites, this will not be enough. In this event the number of blocks stored would have to be increased, pauses between blocks introduced, modulation rate reduced, or the block size enlarged. If the latter alternative were followed, then in order to keep the undetected-error rate down it may be necessary to increase the redundancy beyond 12 elements.

4. *Synchronization*

It is necessary to perform element and block synchronization. As the system is required to accept any data signal pattern, then the worst condition for element synchronization would be no signal transitions for a whole block, i.e. approximately 250 elements. With a short-term oscillator stability of 1 in 10^{-4}, a maximum correction of 5% of a signal element per block time would be required. Signal transitions for correction would be provided by the checking elements and a block start signal.

A block start signal consisting of, say, four elements at the commencement of every block would also provide block synchronization and block identification. Block identification is necessary and, incidentally, can provide a small amount of additional error detection for the block.

It would be necessary to lay down a synchronizing procedure following loss of synchronism. This could be achieved by filling a block with 0's and 1's and commencing it with a special block identification for the block start signal and replacing the checking elements with an easily recognized pattern. It has also been proposed that a synchronizing block be inserted in every repetition cycle.

5. *RQ command signal*

The RQ signal will be sent back to the transmitter over the low-speed channel to the receiver. It is proposed that the condition of this channel when no repetition is called for should be a space or binary 0 and that a change to mark or binary 1 shall be a request for a repetition, i.e. the RQ signal.

Summary of proposals for standardization

1. To standardize on the basic system concept as in Figure 1 using an ARQ mode of operation.

2. If it is agreed that sufficient experience of error distributions at 600 and 1200 bauds are now available, then the foregoing suggestions for :

a) type of code and number of check elements,

b) block size,

c) method of synchronization,

d) RQ signal,

should form the basis for proposals for standardization of the error-detection/correction method.

Supplement No. 68

CHILE TELEPHONE COMPANY — *(Extract from contribution COM Sp. A/No. 74 — July 1963)*

PROPOSAL FOR A MEDIUM-SPEED DATA TRANSMISSION SYSTEM ON THE TELEPHONE NETWORK

1. *General*

This proposal is for the method and procedure to be employed for the detection and automatic correction of errors introduced during the transmission of data over private wire or switched telephone circuits at the C.C.I.T.T. proposed speeds of 600 and 1200 bauds.

The system proposed employs transmission blocks of fixed length, with forward redundancy and decision feedback. Automatic correction is accomplished by block retransmission from the sending terminal upon the receipt of a request (or decision signal) from the receiving terminal via a backward channel.

2. *Technical summary*

2.1 *Data source*

The data to be transmitted are stored, for example, on a punched paper tape. The tape is read by a tape reader and the electrical signals are passed to the EDC (error detection and correction) equipment, the advancement of the tape being under the control of a signal from the EDC equipment to the tape reader.

2.2 *Transmitted information*

The incoming data are subdivided into discrete data blocks of 240 bits each. Each block of data has added to it 16 additional bits for control and protective purposes, four service bits and 12 redundancy (or check) bits. The blocks, as transmitted, are thus constituted of 256 bits.

2.2.1 *Service bits*

The four service bits control the flow of data through the EDC link and provide block sequence numbers and other control information.

2.2.2 *Data bits*

The data can either be transmitted directly from the input medium, or they can be stored within the EDC equipment before being transmitted. Alternatively the 240 data bits in a transmission block may be replaced by a particular pattern of signals and used for automatic synchronization of the transmitting and receiving terminals.

2.2.3 *Redundancy bits*

The last 12 bits of each transmission block are the checking information used for the detection of errors occurring during the transmission of the data. The redundancy is generated in a feedback shift register.

All 244 service and data bits are equally protected against errors by the redundancy bits.

2.3 *Transmission on error protection*

The cyclic code used for generating the redundancy will give protection against all faulty blocks containing 1, 2 or 3 errors, and larger numbers of errors contained within 12 bits. The block-error rate reduction factor, assuming randomly distributed errors, is 1 : 4096.

In practice, the system offers a block-error improvement factor of approximately 1 : 15 000 over typical telephone circuits, i.e. only one block containing errors out of every 15 000 blocks in error will get by undetected and be uncorrected.

2.4 *Synchronization*

2.4.1 *Synchronizing blocks*

Before accurate error detection can commence it is necessary that the transmit and receive EDC equipments shall be in complete synchronism ; not only in element (or bit) synchronism, to give the greatest margin against errors, but also block synchronism.

To achieve this initial synchronism quickly at the commencement of transmission, and to ensure that synchronism has not been lost upon the occasion of a large burst of errors a special synchronizing block is used.

A large number of the 244 service and data bits in a synchronizing block make up a pattern of 1 : 1 reversals, the 244th bit being "0" polarity. A part of the block could be devoted to purposes other than synchronization, for example, echo disabling. The starting pattern in the shift register is such that the redundancy transmitted after a synchronizing block will be eleven bits of "1" polarity followed by a single bit of "0" polarity.

2.4.2 *Initial synchronization*

Upon first being switched into circuit, the EDC equipment at the receive terminal transmits over the return path a continuous request for repetition (RQ). This request is maintained until complete block synchronism is achieved when it is replaced by an acknowledgement signal (OK).

The sending terminal upon receipt of the returning RQ signal proceeds to send synchronizing blocks and continues to do so until the OK signal is received. Following the complete transmission

of the current synchronizing block, the first data block is sent, prefixed by sequence number A. From then on the data blocks are sent consecutively, their sequence numbers being respectively A, B, C, D, A, B, etc., and the redundancy for each block being specific to the data content of each block.

2.5 *Error detection*

At the receiving terminal, the incoming blocks are checked :

i) for the expected sequence number, and

ii) that the received redundancy agrees with the received data content of the block.

Should the redundancy check show an error present, an RQ signal is sent back over the return path requesting a repetition of the block in error.

Should the redundancy check correctly but the pattern of the four service bits does not agree with that of the expected sequence number, the receiver requests a repetition of the block.

Should a synchronizing block sequence number be received and should the redundancy check correctly, then the block is accepted as the start of repetition cycle even though a repetition may not have been requested.

2.6 *Repetition cycle*

2.6.1 *Repetition request*

This signal is sent back over the return path from the receive EDC upon the detection of an error. Normally the return path is held in a steady acknowledgement state, i.e. steady ON condition. When an RQ signal has to be sent, the condition on the return path is reversed to the OFF condition.

2.6.2 *Repetition cycle*

Upon receipt of the RQ signal, at the sending terminal, the next block consists of a synchronizing block and this is followed by a repetition of the message commencing with the block indicated to be in error.

Assuming these repeated blocks are received without error at the receiving terminal, normal transmission of further data blocks will continue.

If the first of the repeated data blocks contains errors, then another request for repetition will be sent back and a further repetition cycle instigated, starting with a synchronizing block.

If the first data block of the repetition is accepted as being correct but the second data block contains errors, an RQ signal is again returned. This time the RQ signal will not arrive back at the sending terminal until a further data block has started to be transmitted.

2.7 *Block sequence numbers*

The block sequence numbers, each made up of the four service bits at the beginning of each block, have the following composition :

Block A	1 1 0 0
B	0 0 1 1
C	0 1 0 1
D	0 1 1 0
Synchronous block	1 0 1 0

3. *Other data storage media*

The system described in this proposal has been based on punched paper tape as the storage medium for the data. The system can equally well be applied, in principle, to other forms of storage media, such as punched cards, magnetic tape, or even on-line data processors.

CHARACTER-BY-CHARACTER ERROR CORRECTION

1. *Introduction*

Extensive error measurements on telephone connections have shown that error correction codes are not suited to the error bursts characteristic of those connections. Error correction is effected much more efficiently by retransmission of those parts of messages affected by line disturbances.

The efficiency of an error-detection code—i.e. the probability with which it detects a block containing errors—depends little on the size of the block. A code of, say, 12 bits provides a very adequate error-rate improvement and if used in conjunction with a block of several hundred bits, only a small percentage of transmission time is utilized in transmitting check bits. When a very small block is used, of the order of one character for example, a block retransmission system depending only on a detection code requires, for the same error-rate improvement, a code of very high percentage redundancy and which is consequently much more wasteful of line transmission time.

Nevertheless, a recommended system of error correction character by character is desirable for a number of reasons :

i) economy of transmission time when the majority of messages are short in relation to the block of a retransmission system using only a detection code ;

ii) adaptability to various kinds of peripheral apparatus ;

iii) modest storage requirements make for an economical error control equipment.

2. *Proposed means of error detection*

Powerful error control on a character-by-character basis can be achieved by the combined use of two means of error detection :

i) a simple detection code ;

ii) an analogue type disturbance detector, which would cause the rejection of characters when the received signals are abnormal.

The repetition of a character would therefore be initiated from the receiving terminal when an error, or the probability of an error, is indicated, either by the detection code or by the analogue detection. In view of the small size of the block a switched simplex (reversible) mode of operation is not suitable and only a full duplex mode, permitting backward signals to be transmitted simultaneously with those in the forward direction, although not necessarily at the same speed, should be contemplated.

In a prototype system the characteristic of the data channel which is continuously monitored to provide the analogue detection is the amplitude of the received carrier wave, any excessive variation of the incoming carrier initiating a repetition. Alternatively, one of several other characteristics could be supervised. Standardization of the analogue detector is not, in fact, necessary for satisfactory interworking but is of course desirable if a system providing a uniform error-rate improvement is required.

3. *Characteristics of a character-by-character error control system for synchronous data transmission*

Each automatic block retransmission system requires, of course, standardization of the block size, in this case the number of elements per character. A C.C.I.T.T. system would clearly be built around the alphabet and telegraph code which may be standardized by the C.C.I.T.T. for data.

Detection code : In the prototype system two parity bits are used for a seven-bit character. The eighth bit provides an odd parity check for bits Nos. 1-7. The ninth bit provides an odd parity check for bits Nos. 1, 3, 5 and 7.

The eighth bit may or may not be present in the source data and may or may not be delivered at the receiving point. The ninth bit is added in the error-control system and is abstracted before delivery of the data.

Signalling : The signalling conditions on the line which would need to be standardized, with, by way of example, those conditions which have been applied in a prototype system are as follows :

Data channel (forward)

Idle periods	Reversals.
Start of message	A period of no-tone, of duration corresponding to one character, followed by two "start of message" code combinations [1].
End of message	As for "start of message" but using "end of message" code combinations [1]. The characters in store are then repeated until the end of message is acknowledged.
Repetition cycle (ARQ)	A period of no-tone, of duration corresponding to one character, followed by the characters retained in the store at the transmitting terminal.

Backward channel (supervisory)

Idle period	Space (a change from constant mark to constant space constitutes an "end of message" acknowledgement).
Receiving data correctly	Mark (a change from constant space to constant mark constitutes a "start of message" acknowledgement).
Request for repetition (RRQ)	A space condition corresponding to the transmission time of four characters in the data channel.

Character storage and repetition cycle

At the data-receiving terminal, storage for one character only is necessary.

At the transmitting terminal, storage is needed for a number of characters sufficient to allow for the sum of the propagation times in each direction, including signal recognition times, and the transmission time of the RRQ signal. In the prototype system 16 characters are stored.

When the transmitting terminal recognizes an RRQ on the backward channel a period of no-tone replaces the next character due to be transmitted. This is followed by the characters retained in the store. Then, in the absence of any further RRQ signal, normal transmission is resumed.

At the receiving terminal, the reception of data is suspended on the reception of a faulty character, indicated by the failure of the parity check or by the analogue detector, for a period corresponding to 1 + 16 characters, i.e. the no-tone character plus the number of characters in the sending-terminal store. One RRQ signal is returned to the transmitting terminal.

In this way automatic phasing is achieved between the transmitting and receiving terminals, regardless of the actual propagation times, so long as the character storage is adequate. This phasing is not affected by false RRQ signals caused by disturbances on the backward channel.

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[1] These code combinations are used within the communications equipment only for the purpose of controlling the start and end of a message. There are no restrictions imposed on the code combinations used for the data to be transmitted.

Supplement No. 69

FEDERAL REPUBLIC OF GERMANY *(Extract from contribution COM Sp. A/No. 78 — August 1963)*

CONTROL OF ERRORS

1. The following general features of transmission are proposed for highly protected data transmission on switched telephone circuits at speeds greater than 200 bauds :

1.1 The data are preferably transmitted at speeds of 1200 or 600 bauds.

1.2 They are divided into blocks.

1.3 Sending and receiving terminals are operating in synchronism. Bit synchronization is, for instance, effected and maintained by means of the incoming transitions. Provision of the block synchronism requires special arrangements.

1.4 The check bits for each block are generated in the feedback shift registers in the sending and receiving terminals.

1.5 The check bits of each block generated at the sending and the receiving end are compared at the receiving terminal. For each received block decision signals, e.g. "proceed" or "repeat", are transmitted from the receiving station to the sending station (decision feedback).

1.6 Correction of erroneous blocks is effected by block repetitions.

1.7 Block numbering allows the correct continuation of transmission upon a disturbed decision signal.

1.8 The period of time between two undetected errors shall be considerably increased by these systems.

These general features result in two types of systems :

— one system with continuous transmission of blocks and low-speed return channel for decision feedback ;

— one system without return channel, i.e. working alternately, where decision signals are fed back in the intervals between two blocks.

2. *Proposals for a system with return of decision signals in a low-speed return channel obtained by dividing up the frequency band of the telephone channel*

2.1 *Basic properties*

The error detection and correction (EDC) equipment subdivides the information offered into blocks of fixed length and adds for transmission service bits and redundancy bits. The redundancy bits will be derived in a feedback shift register, e.g. comprising 12 stages as discussed by Special Study Group A for use on telex lines.

The service bits are included in the coding. The information blocks transmitted are cyclically numbered by four sequence numbers. Other combinations of service bits can be used to designate other than information blocks.

The repetition of blocks which are found to be in error is controlled by a decision signal in the backward channel, whose modulation rate will be 150 bauds.

A repetition cycle comprises three blocks. One dummy block, the block having been in error, and the subsequent block. Dummy blocks contain a synchronizing pattern, which is used to check the block synchronization of the data-receiving station.

2.2 *Block format*

All blocks transmitted to the line comprise four service bits, a fixed number of information bits (e.g. 240) and redundancy bits (e.g. 12) in the same positions.

2.2.1 *Service bits*

The four service bits are used in various combinations for control purposes :

a) Four combinations are used to number the information blocks, i.e. to provide the block sequence number.

b) Other combinations can be used with dummy blocks, e.g. to indicate a dummy block used during the repetition cycle, to indicate a dummy block used as start-of-transmission signal or to indicate a dummy block used as end-of-transmission signal.

2.2.2 *Information bits*

These bits carry the transmitted data in the information blocks.
All dummy blocks contain a special combination for block synchronizing purposes.

2.2.3 *Redundancy bits*

The last positions (e.g. the last 12) of a block are reserved for redundancy bits in both dummy blocks and information blocks. These check bits protect all service and information bits.

2.3 *Block sequence on line*

2.3.1 *Operation without errors*

After the connection has been established and the EDC terminals are connected to line, dummy blocks are transmitted continuously. The receiving terminal will automatically gain bit synchronism and also block synchronism by testing for the synchronizing pattern carried by the dummy blocks. The combination of service bits indicating "start-of-transmission" will be detected by the receiver and, if the redundancy check is correct, the receiver will indicate "ready to receive" by means of the "proceed" decision signal.

The start of transmission of data is signalled by the appropriate combination of service bits (sequence number) and this may be initiated after the return of the "proceed" signal from the receiving station either automatically or under manual control of the transmitting operator.

The service bit combination and the redundancy bits are checked by the receiving station and a "proceed" signal sent to the transmit station, if the redundancy check is correct.

The "end-of-transmission" condition is signalled by sending dummy blocks with a service bit combination indicating "end-of-transmission". This criterion can also be used for automatic switch-off, for releasing the connection from the data-receiving terminal, and for bringing the telephone set in normal condition.

2.3.2 *Operation in the presence of errors*

Errors occurring during the transmission are detected by the redundancy check. Failure of this check causes a "repeat" signal in the backward channel which must be available at the sending station before the first half of the subsequent block has been transmitted. On this signal the data transmitter interrupts the transmission of the current block at its mid-instant and fills the second half of the block with dummy information. Then a complete dummy block containing the synchronizing pattern is transmitted, followed by the block having been in error and the subsequent block. These last blocks can either be transmitted from a buffer store or from the input device, if the latter is capable of stepping back and repeating parts of the information. Unnecessary repetition due to an error in the backward channel can be rejected by the receiver on grounds of the sequency number test.

A disturbance of the "repeat" signal will cause the data transmitter to proceed, although a repetition is wanted. This, too, can be recognized by the number test. However, there is no means to retransmit the lost blocks without manual intervention. By suitable choice of the decision signals this case can be made very unlikely. If, for example, a simple two-state signal is used as decision signal it should be made as long as possible and the data transmitter (i.e. the backward channel receiver) should interpret the signal as "proceed" signal only if it is in the proper polarity for the whole of its duration.

3. *Proposals for a system without return channel, i.e. working alternately*

3.1 The data blocks are transmitted one by one in the forward direction ; the associated decision signal is fed back upon receipt of the respective block (half-duplex operation).

3.2 All blocks are transmitted in a fixed element pattern, the space between the blocks is not specified, but is determined by the respective delay of the line. It amounts, however, to an integral multiple of the bit length.

3.3 Block synchronism is effected by an individual sequence of bits, preceding each data block ; bit synchronism is also maintained during the transmission intervals between two blocks.

3.4 *Composition of blocks*

Each transmitted block comprises :

1. the heading signal ;
2. the start signal ;
3. the service bits ;
4. the data bits ;
5. the check bits.

3.4.1 *Heading signal*

The heading signal consists of reversals 1 : 1 and gets the receiving station ready for the following start signal. The heading signal is so long that the receiving station will still be able to recognize it after the dying out of the echo of the decision signal or, if there is no echo, after the operation of the level supervising unit.

3.4.2 *Start signal*

The start signal marks the beginning of a block. It consists of one bit having the same polarity as the last bit of the preceding heading signal. The polarity of this start signal bit should be standardized, and should preferably correspond to that of the digit 0.

3.4.3 *Service bits*

There should be three service bits containing the information being essential to the transmission process. One of these bits represents the two sequence numbers A and B, the other two may be used to indicate special blocks.

The service bits are protected by the error-control procedure.

3.4.4 *Data bits*

The information part of the block may consist of a number of bits, belonging to characters of any code. It shall be possible to adapt the number of data bits per block to the terminal equipment used in the respective case. 210, 240, 480 and 560 data bits per block are preferred; in special cases it may be adequate to use larger blocks.

3.4.5 *Check bits*

Check bits are generated in a feedback shift register, constructed similar to that described in Annex 2 of Supplement No. 12 for low transmission speeds. The number of stages of the shift register must be specified according to the properties of the telephone network.

3.5 *Decision signals*

There are two decision signals "proceed" and "repeat". These signals must be so long that they can still be recognized upon the operation of the level supervising unit at the sending end. They should be composed in such a way that it is practically impossible for the two conditions "proceed" and "repeat" to be changed into each other by errors.

3.6 *Synchronization at the beginning of transmission*

At the beginning of transmission the bit synchronism will be established by the first block transmitted by the sending station. If this block is received without error, the receiving station is going to send the decision signal "proceed" and the next block will be transmitted. The bit synchronism is maintained, for instance, by the bit transitions within the following incoming blocks. In this case it should be borne in mind that bit synchronism may be lost by blocks almost exclusively containing bits of one polarity only. For each, block synchronism is established by the start bit.

3.7 *Transmission process* (see Supplement No. 61 — point 9.2.7)

Supplement No. 70

OLIVETTI — *(Extract from contribution COM Sp. A/No. 88 — October 1963)*

BLOCK-ERROR DETECTION

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Among the many possible methods that might be used for error detection, cyclic codes appeared to be the most convenient for the following reasons:

a) the circuitry to implement a given scheme is very simple and economical;

b) the information block is left unaltered, which is often desirable ;

c) with few simple changes in feedback connections, the code can be varied, thus it can be made to approach the optimum in each particular case ;

d) mathematical studies in this field are particularly advanced and offer some powerful means of research for optimum codes both from an *a priori* and from an *a posteriori* viewpoint ;

e) in any case, cyclic codes seem to be particularly well suited for switched telephone channels where errors have a tendency to occur in bursts.

As mentioned previously, the code used in the present equipment makes use of 18 check bits, in other words, the cyclic code has an 18-degree generator polynomial. In a cyclic code generated by such a polynomial, the fraction of all possible error patterns that remains undetected is :

$$K = \frac{1}{2^{18}} - \frac{n}{2^{n+2}}$$

where n is the length of the code. This is the factor which should multiply the rate of occurrence of any error to obtain the probability of undetected errors, provided all possible error patterns of any length are equally probable.

In practice, of course, not all error patterns are equally probable and with a well-chosen generator polynomial, the factor k can be expected to reduce to 1/10 or 1/100 or even to smaller values, if n does not exceed the code length (equal to the smallest integer n such that $1 + x^n$ is divisible by the generator polynomial in the algebra modulo 2). It is then clear that with the bit-error rates usually encountered, say 10^{-3}, an undetected error rate of 10^{-8} to 10^{-9} is easily achievable with the 18 check bits used in the equipment. In fact, a number of generator polynomials were studied after a statistical examination of some switched telephone lines, then they were applied to data transmissions over several kinds of lines. In all cases the block lengths were shorter than the code lengths which ranged between 1397 and 8184 bits. All the transmitted samples, of about 10^8 bits each, revealed no undetected errors, even when some very noisy lines were used (error rates about 0.5×10^{-2}).

The error-detecting circuits are also used for signalling the occurrence of a read error or a perforation error. In the case of a read error, its occurrence must be signalled to the receive terminal. This is done very simply by forcing an error in the check computations in progress, as shown in Figure 1. This causes the rejection of the block and its retransmission.

The same thing is done when a perforation error occurs. An error is forced in the check computations in course (see Figure 4). The block is marked with the error code and its retransmission is asked for.

It may be noted here that, while a perforation error positively causes the rejection of the block and a request of retransmission, a read error is protected with the same probability as an error in the channel : in fact, the wrong check emitted might be affected by line errors and appear at the receiving end as correct.

Block numeration and signalling

Reliability of transmission requires not only an efficient detection of errors, but also that block transmission take place at the right moment starting with the right character ; it further requires that no block be omitted or repeated twice due to a wrong interpretation of the return signal. Among several different methods, the one that proved to be the very best was cyclic block enumeration and correspondingly the cyclic permutation of return signals with cycle of order

three. This method is well known, but it is described here to point out the criteria adopted to make it suitable to the present case in which the information and the control signals are simultaneously present on the line.

In brief, it works as follows :

During the transmission, a prefix character (which will not appear on the output tape) is added to each block : there are three different prefixes (A, B, C) that are cyclically shifted and assigned in turn to the blocks. A prefix is ideally bound to its block, in the sense that, if a block is repeated, it will be headed all the times by the same character.

There are also three different return signals, that shall be named A′, B′, C′, that are cyclically shifted ; which of the three is chosen depends on the preceding history of the blocks, but indicates the next block to be expected, at the receiving end. In other words, a signal B′ means that a block marked B is demanded, and that any other block will be rejected.

It is clear that this method holds also in the present case in which a signal is returned when the next block is already being transmitted, provided some conditions are satisfied. First of all, the case of a block headed by a wrong prefix is not rare, as it occurs at least every time an error is detected in a block. In fact, assume that block A contains an error : at the end of the check tail, the receiver is able to detect it, hence it returns the signal A′ ; at the same time it sets itself in such a way as to accept only block A. But block B begins to come in immediately after block A, and must be rejected. During the time block B comes in, the control signal A′ is acknowledged and becomes effective, the transmitter goes back to the beginning of block A and repeats it, thus after the rejected block B, block A follows and is accepted.

From the preceding description one sees how the method of block enumeration has been applied. Shifting from A′ to B′, from B′ to C′, from C′ to A′ occurs only if two conditions are satisfied : a) the last block was headed by the right prefix and b) the check bits proved that no errors affected it.

In this way, the method automatically solves both the problems of block-error control and block-sequence control. It may be noted that, while a block affected by errors is perforated as it is and is closed at the end by a special error code, a block headed by the wrong prefix can be prevented from being perforated. This helps in reducing the number of bad blocks on the output tape.

It remains to be seen how A, B, C, A′, B′, C′ are constructed and protected. The heading characters A, B, C, are any three characters of the alphabet used (remember that they are not perforated) and, as such, they are protected as any other character in the block, in the sense that the check bits are computed also on them.

As to the control signals A′, B′, C′, the following criterion has been adopted. The three signals are chosen in such a way as to render negligible the probability that one be transformed into another by transmission errors.

If one of these signals is not recognized because altered by transmission errors, transmission stops, after having completed the block in progress and awaits another signal. Another signal is received due to the last completed block. If it is recognized correctly, the transmission is resumed and proceeds as before ; if again it contains some erroneous bits, the equipment still makes no decision and awaits another signal. In this situation, at the receive terminal a special circuit is provided to repeat the control signal that was last sent. This circuit is triggered whenever information has not been received for a certain period of time t, and continues to be triggered every t, until some information is received. Every time it is triggered, it causes the retransmission of the control signal that was sent last. It may be noted that, when a return signal is lost, the interpretation of the first signal received after it must be used in a somewhat different way, as compared with the normal case of uninterrupted transmission.

In fact, assume that during transmission of block B, the return signal, which was a consequence of block A, is not recognized. The transmission then stops until a recognizable signal is received, but the latter is caused by block B and must be interpreted as such (for example, if the signal is B′, that means that block B has not been received correctly and must be repeated). This and other details have to be taken into account, but their exhaustive description is outside the scope of the present contribution. Instead, it may be interesting to point out the general criteria that led to the adoption of this method, excluding any error-correcting redundancy in the return signal.

Statistical tests that have been conducted at Olivetti laboratories on many types of telephone lines have proved that no practical redundancy is feasible that is able to correct the types of errors that may affect the control signal.

The present system has the advantage that, as soon as the line is re-established after an interruption, the transmission is automatically resumed from the first mutilated block and no information is lost or mixed incorrectly, no matter how long the interruption may have been. All this happens without the intervention of the operator.

The question remains now, whether it is convenient to add redundancy to the signal to enable the correction of certain more frequent error patterns and to rely on logic such as that described, whenever an error pattern occurs that is not corrected. To achieve this, however, the signals should contain a considerable number of check bits. For example, to correct a single error and simultaneously detect a double error in a three-bit code, four check bits must be added. The number of check bits grows rapidly as correction and detection are required for more complicated (but nevertheless realistic) cases. Along this line, a situation is soon reached in which the control signal takes more time to be transmitted than the information block. In fact, it must be borne in mind that the control signal is transmitted at a much lower speed than the information bits, and that the present system deals with blocks of variable length, that in some cases may be formed by as few as 20 or 15 characters. Below this limit, the transmission would stop at the end of each block, waiting for return signal to be completed. All this results in a low transmission efficiency that overrides the advantage of a smaller number of stops due to errors.

Extensive experiments carried out at Olivetti laboratories have shown that three nine-bit signals, transmitted over the supervisory channel at a speed of 50 to 75 bauds, if properly chosen, have a negligible probability of being transformed into one another. In fact, three such signals can be made to differ from one another by six bits; then transformation of one into another takes place only when the sequence of nine bits is affected by six errors having two fixed patterns. If the probability of any six-error pattern is denoted by p, assuming any pattern equally probable, the probability of transformation is:

$$p_t = p \frac{2}{\binom{9}{6}} = p \frac{1}{4\,2}$$

Even if such an improbable event should happen, block prefixes would indicate that a misinterpretation has occurred; thus no irregularity would escape control. Only when such patterns of errors affect two consecutive signals a block might be lost, which is extremely improbable. This event is expected to have the same order of probability as an error burst, affecting the information blocks, of escaping detection.

7. *Description of a data transmission set with block memories*

The only difference between this equipment and the one previously described is that the input and the output tape units are replaced by two memories. This permits some modifications in

the operating methods of the equipment, that improve the over-all efficiency and make it more versatile.

The capacity of each buffer is two blocks, the reason being clear from the preceding section. In fact, at the transmit terminal, each block must be held for a possible repetition while the block next to it is being transmitted. At the receive terminal, utilization of a block does not start until that block has been checked for correctness, thus, while one part of the buffer delivers one block to the user, another part is receiving the next block.

The use of the buffers allows omission of the start and stop bits on each character, thus allowing a better over-all efficiency. In fact, the transfer of each character to the serializer can be timed by the clock of the latter, and no interval is left between a character and the next.

As compared with the performance of the first type, no time is lost in going back two blocks at each repetition, which also means better efficiency.

One last advantage of the buffered system is that the speed of character transmission over the line is completely independent of the speeds of input and output units (the speeds that, of course, must be the same as the block speeds) which makes it possible to adapt the equipment to different kinds of data media, as mentioned in the introduction. It is only necessary to standardize some interface signals between the data source and the data transmission set, and between the data transmission set and the data processor.

. . . . . .

Choice of cyclic code for block-error detection

It is believed that a data transmission by blocks, with error control obtained by means of a redundancy based on the properties of cyclic codes, is most effective and convenient.

Some basic protection characteristics are common to all cyclic codes, however a proper choice of a particular code may considerably improve the performance of a system.

1. *Choice of cyclic code for data transmission on telephone lines at 600/1200 bauds*

In some cases (particularly in point-to-point connections) the structure of the code could be optimized according to the characteristics of the link involved.

Nevertheless, in many applications such optimization may not be possible; in such cases the choice of a suitable cyclic code can be based on the knowledge of the most frequent patterns of errors observed on the general switched telephone network.

An efficient protection is ensured by a code having the following properties:

a) a sufficient Hamming distance;

b) the ability to detect single-error bursts of sufficient length in the block;

c) the ability to detect double-error bursts of sufficient over-all length in the block.

The number of the check bits must be dimensioned according to the block size. For the most usual and convenient block sizes it is believed that a reliability satisfactory for all practical demands can be obtained by employing 16 to 18 check bits.

For block size up to 614 protected bits the generator polynomial

$$1 + X^5 + X^6 + X^{10} + X^{15} + X^{16}$$

provides the following properties :

a) Hamming distance : 4 ;

b) detection of all single-error bursts of length up to 16 bits ;

c) detection of all double-error bursts of over-all length up to 11 bits ;

d) the residual probability that error bursts of length greater than 16 remain undetected is :

$$p_r = p \frac{1}{2^{16}}$$

where p is the probability of error bursts longer than 16 bits. This polynomial requires the use of 16 check bits.

For block length up to 1379 protected bits, the generator polynomial :

$$1 + X^3 + X^7 + X^{11} + X^{14} + X^{18}$$

provides the following properties :

a) Hamming distance : 4 ;

b) detection of all signal-error bursts of length up to 18 bits ;

c) detection of all double-error bursts of over-all length up to 12 bits ;

d) the residual probability that error bursts of length greater than 18 remain undetected is :

$$p_r = p \frac{1}{2^{18}}$$

This polynomial requires the use of 18 check bits.

The implementation of the two codes described is most simply obtained by the two shift registers with feedback connection shown in Figure 6.

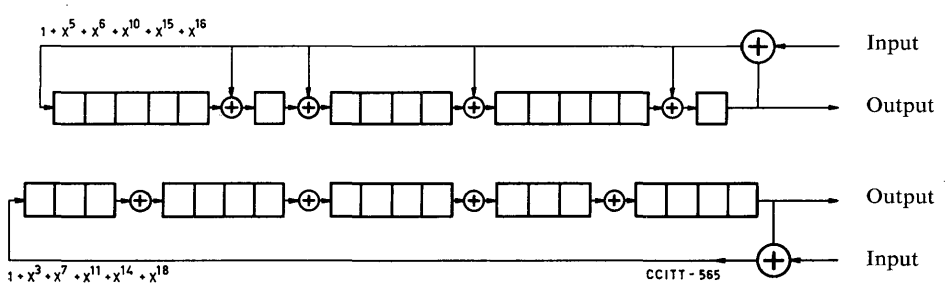


FIGURE 6

2. *Choice of cyclic code for data transmission on telex*

In the case of telex network, for block size up to 502 protected bits the generator polynomial

$$1 + X^3 + X^6 + X^{10}$$

provides the following properties :

a) Hamming distance : 4 ;

b) detection of all single-error bursts of length up to 10 bits ;

c) the residual probability that error bursts of length greater than 10 bits remain undetected is :

$$p_r = p\,\frac{1}{2^{10}}$$

This polynomial requires the use of 10 check bits.

For block size up to 623 protected bits, the generator polynomial

$$1 + X^3 + X^5 + X^7 + X^8 + X^{12}$$

provides the following properties :

a) Hamming distance : 4 ;

b) detection of all single-error bursts of length up to 12 bits ;

c) detection of all double-error bursts of over-all length up to 6 bits ;

d) the residual probability that error bursts of length greater than 12 bits remain undetected is :

$$p_r = p\,\frac{1}{2^{12}}$$

This polynomial requires the use of 12 check bits.

In Figure 7 are shown the two shift registers with feedback connections corresponding to the polynomials examined.

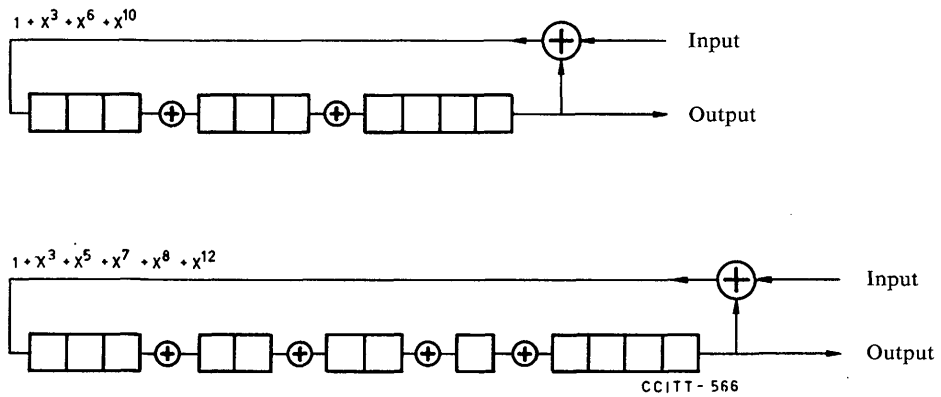


FIGURE 7

Supplement No. 71

N.V. PHILIPS T.I. — *(Extract from contribution COM Sp. A/No. 16 — September 1961)*

PHASE DISTORTION

The distortion was measured as group delay in ms. The group delay measurements were made in two manners:

a) A hybrid transformer and a frequency counter were used to determine the frequencies at which the output signal of the circuit showed a 180° phase shift with respect to its input signal.

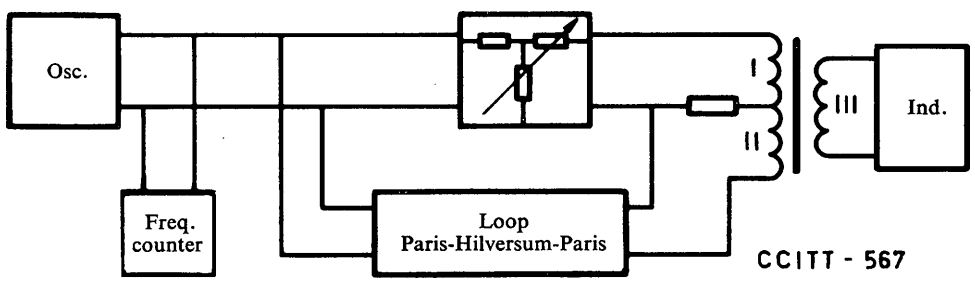


FIGURE 5

The attenuator was used to make the amplitude of the signal at winding I equal to the amplitude of the output signal at winding II. If the signal at winding II has a phase shift of 180° with respect to the signal at winding I, there will be a minimum output signal at winding III. By sorting out the frequencies at which the 180° phase shift occurs (minimum deviation of the indicator), the frequency interval was measured at which the signal shows a 360° phase shift.

The signal frequency was read from a frequency counter with great precision.

This method can be used only if there is no frequency departure on the circuit.

The group delay is $t_g = \frac{d\varphi}{dw}$ (φ in radians, w in c/s).

Expressed in degrees $t_g = \frac{2\pi\varphi}{360}$.

Since only 360° phase shifts were measured,

$$t_g = \frac{2\,\pi\,360}{360\,\Delta w} = \frac{1}{\Delta f}.$$

b) A group delay meter was used for the second method. The meter applies a modulated signal to the line. The envelope has a fixed frequency of 85 c/s. The carrier can be varied between 500 and 3000 c/s. At the receiver a 85-c/s reference signal is compared with the detected signal which is subject to phase variations as a result of the phase distortion on the circuit.
The phase difference between the detection signal and the reference signal is a measure for the group delay.

Supplement No. 72

STUDY GROUP III OF THE C.C.I.R. — *(Contribution COM Sp. A/No. 37 — May 1962)*

MEASUREMENTS OF PATH TIME DELAY DIFFERENCES AND THEIR INCIDENCE ON TYPICAL RADIO LINKS

1. *Measurements on high-frequency radio circuits of the fixed service*

1.1 *Summary*

Some 4000 facsimile pictures received over a number of important radiotelegraph circuits terminating in London during the period from sunspot minimum to sunspot maximum (1953 to 1957) have been examined to ascertain the incidence of multi-path propagation and to measure the path time delay spreads. Multi-path conditions were found to obtain for a considerable proportion of the time throughout the whole period and path time differences were observed up to 2.5 ms. The measurements were made on pictures received from New York (1420 pictures), Melbourne (1600 pictures) and Moscow (350 pictures), together with a few less frequently used circuits.

1.2 *Method of measurement*

The facsimile transmissions use frequency-modulation so that, in general, the received picture will be derived from the predominating path. If, due to fading, signals from different unequal length paths predominate at different times, a sharp, straight line in the transmitted picture at right angles to the line of scan will appear as a jagged line in the received picture. By measuring the width of the ripple of the received line, it is thus possible to determine the difference in propagation time of the shortest and longest paths that predominate from time to time during the transmission of the picture.

The spread in path time delay, as seen on the facsimile pictures, was obtained by measuring the ripple on a line, at right angles to the direction of scan, which could be safely assumed to be sharp and straight when transmitted. A low-power microscope having a graticule divided into squares was used to measure the ripple. The magnification was adjusted so that one square represented a time difference of 2 ms for the machines most generally used. Displacements were estimated to the nearest 0.5 ms and where a number of different delays were observed only the maximum delay difference was recorded. In some cases, this maximum delay was not typical of the distortion throughout the picture as a whole.

A check was made under controlled conditions using a fading machine which was adjusted to produce random fading with known path time delay difference and with various levels of median signal level on the two paths. The incidence of multi-path observed in these tests was almost identical with that obtained by mathematical analysis.

1.3 *Measurement on facsimile pictures*

Each picture was examined to determine the maximum path delay difference. Since the values of delay were approximated to the nearest 0.5 ms, the probable distribution of path time delay differences between 0 and 2.5 ms was obtained by the usual method of apportioning the number of pictures at each value of delay other than zero equally between the adjacent delay ranges.

The incidence of multi-path distortion on each picture was assessed according to four categories, viz : none, rare, frequent, continuous.

For the pictures received in each month, the percentages falling in each range of delay difference and in each category of incidence were obtained. For simplicity, these monthly percentages have been averaged over a complete year, and the results are shown in Figure 1.

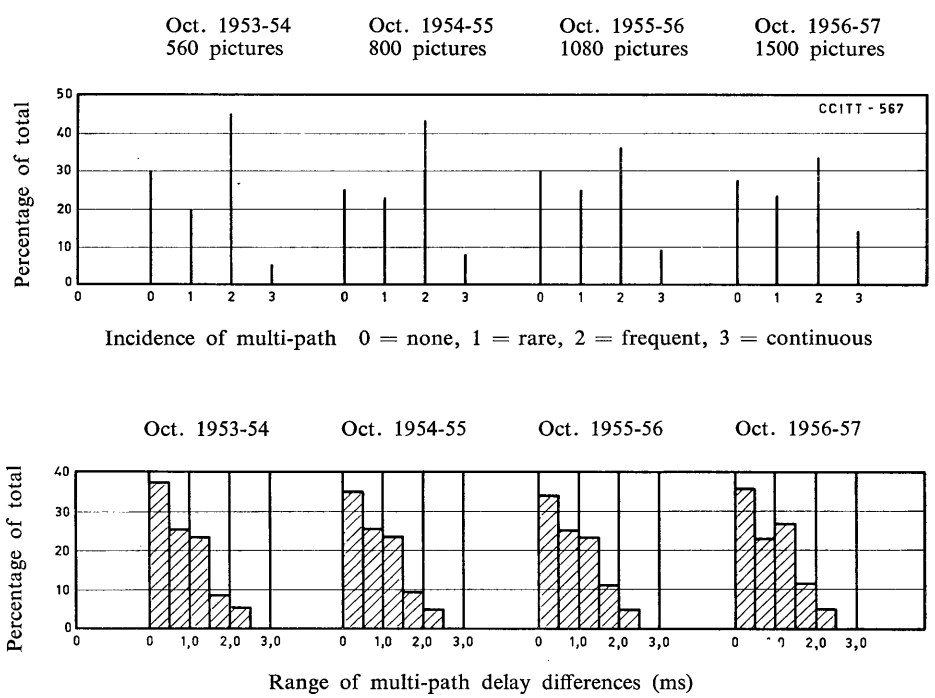


FIGURE 1. — **Multi-path propagation on H.F. fixed-service radio circuits**

1.4 *Discussion of results*

The results show that, for each of the five years 1953 to 1957, frequent or continuous multi-path was in evidence on between 40 and 50% of the pictures analysed. Approximately half the

pictures affected by multi-path showed path time delay difference of 1 ms or more, and nearly 30% had delay differences of 1.5 ms or more.

Multi-path effects such as these are not particularly troublesome in facsimile transmission. Facsimile pictures have been used merely as a convenient means of obtaining data on incidence and delay difference over a long period on typical high-frequency fixed-service radio links. The effect of such multi-path propagation could be more serious on telegraphy and data circuits, particularly where the path time delay difference is appreciable in relation to the duration of the telegraph element. For example, a path delay difference of 2 ms would have an adverse effect on the performance of a telegraph circuit working at 200 bauds, since the delay difference is equal to 40% of the duration of the signalling element. Circuits working at lower modulation rates would be less affected by multi-path propagation since the path delay difference would be smaller in relation to the duration of the telegraph element. For example, a 2-ms delay difference would equal only 20% of the element duration at 100 bauds.

2. *Measurement on meteorological broadcast services*

Section 1 of this Supplement shows the incidence of multi-path propagation and path time delay differences observed on typical point-to-point high-frequency radio circuits. More severe multi-path effects may, however, be experienced when frequencies below optimum have to be used. Such circumstances often arise, for example, in high-frequency meteorological broadcast services, and an analysis has been made of meteorological charts received in the United Kingdom by facsimile transmissions from Washington (D.C.) and from Japan.

Some 1600 charts received during the period June to September 1961 have been analysed. Of these, 1000 were received from Washington and 600 from Japan. The method of measuring the path-time delay differences was similar to that described in section 1 of this Supplement. The results are tabulated below and are shown graphically in Figure 2.

Multi-path time delay difference (ms)	Percentage of charts for each circuit	
	Washington (D.C.) to U.K. (6000 km)	Japan to U.K. (9600 km)
0 - ½	10	0
½ - 1	20	5
1 - 1½	28	9
1½ - 2	21	10
2 - 2½	10	30
2½ - 3	6	26
3 - 3½	2	11
3½ - 4	2	6
4 - 4½	1	2

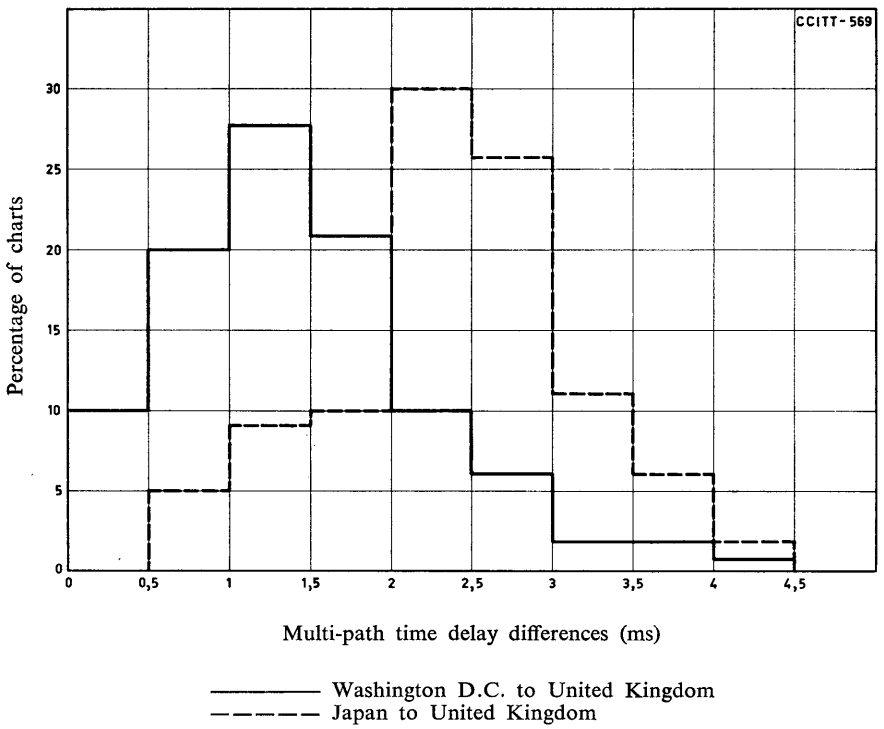


FIGURE 2. — **Multi-path propagation on the meteorological broadcast service**

Supplement No. 73

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — July 1963)*

MEASUREMENTS OF PHASE DISTORTION BETWEEN SUBSCRIBERS AND LOSSES IN SUBSCRIBER-TO-SUBSCRIBER CONNECTIONS

Measurements have been made of the insertion loss/frequency and insertion group delay/frequency characteristics of a few sample connections between unfavourably-situated subscribers within the London multi-exchange area and the results are given in Figures 1 to 4. Some of the connections involved long, heavily-loaded junction cables ; the incidence of this type of old plant is being gradually reduced as part of an improvement programme but imposes a definite limit at the present on the exploitation of the switched telephone network for data transmission at modulation rates in excess of 600 bauds.

After a study of the characteristics of plant and typical routings, nominal group delay/frequency and loss/frequency characteristics have been established for the guidance of the designers of data-transmission equipment to be used on switched connections. These characteristics are given in Figure 5 and, in the United Kingdom telephone system, the performance of data transmission modems will be studied with reference to these nominal, extreme characteristics. Provision would be made for an equalizer compensating for one-half of the characteristics to be incorporated in data modulator/demodulator equipment where desired.

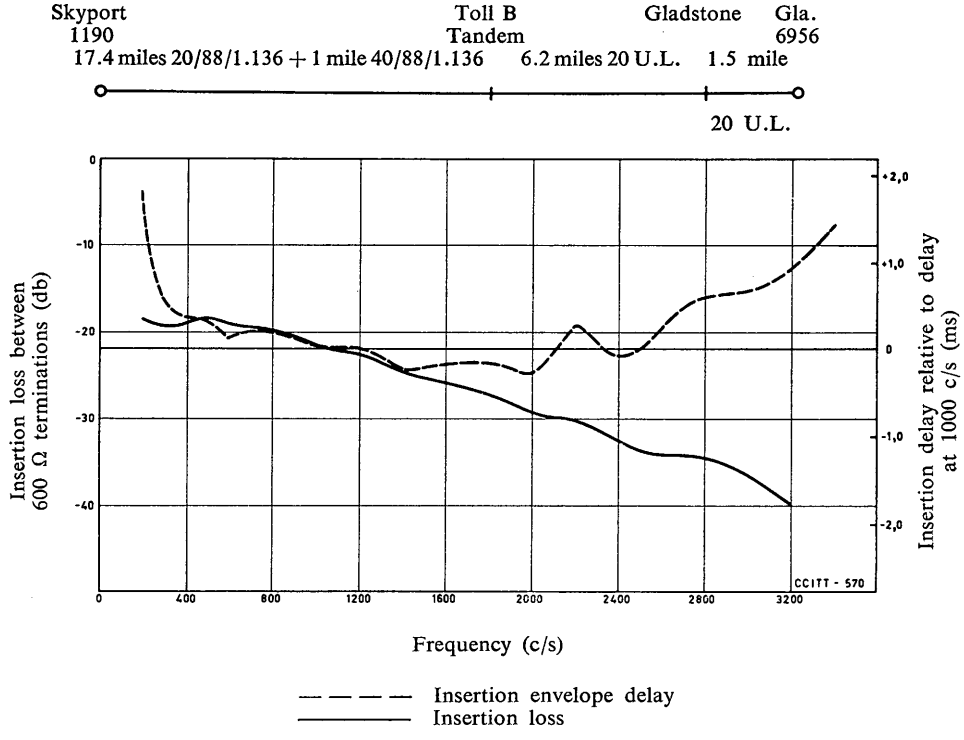


FIGURE 1. — **Insertion loss and delay**
Skyport 1190 — Gladstone 6956

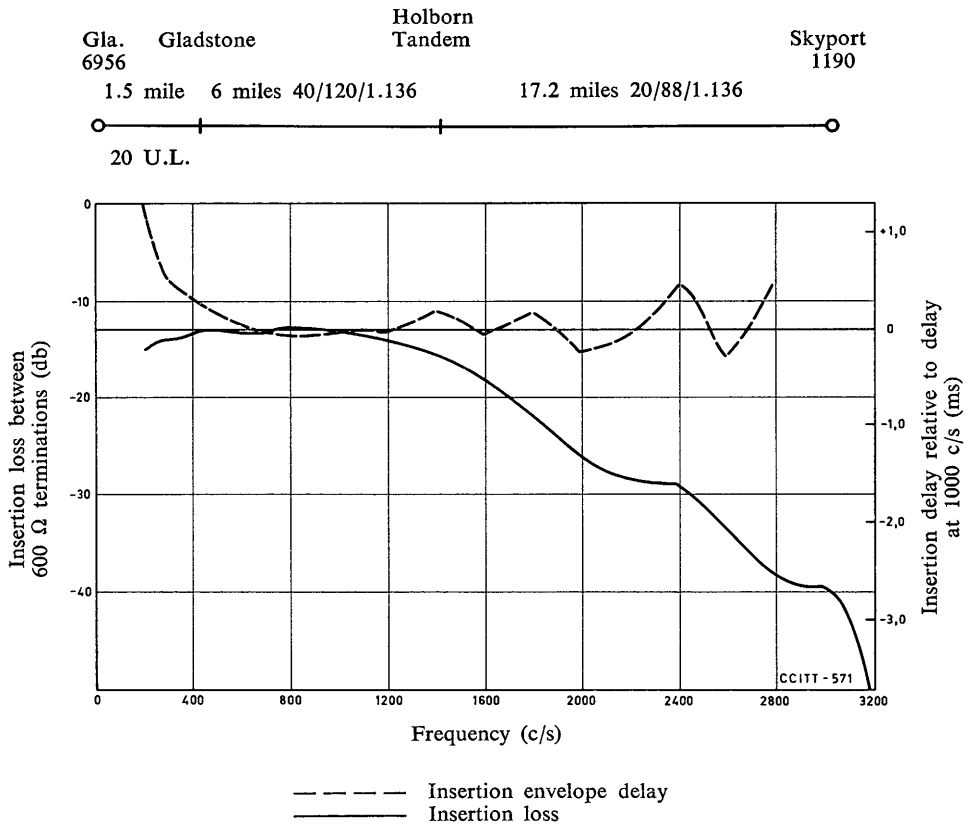


FIGURE 2. — **Insertion loss and delay**
Gladstone 6956 — Skyport 1190

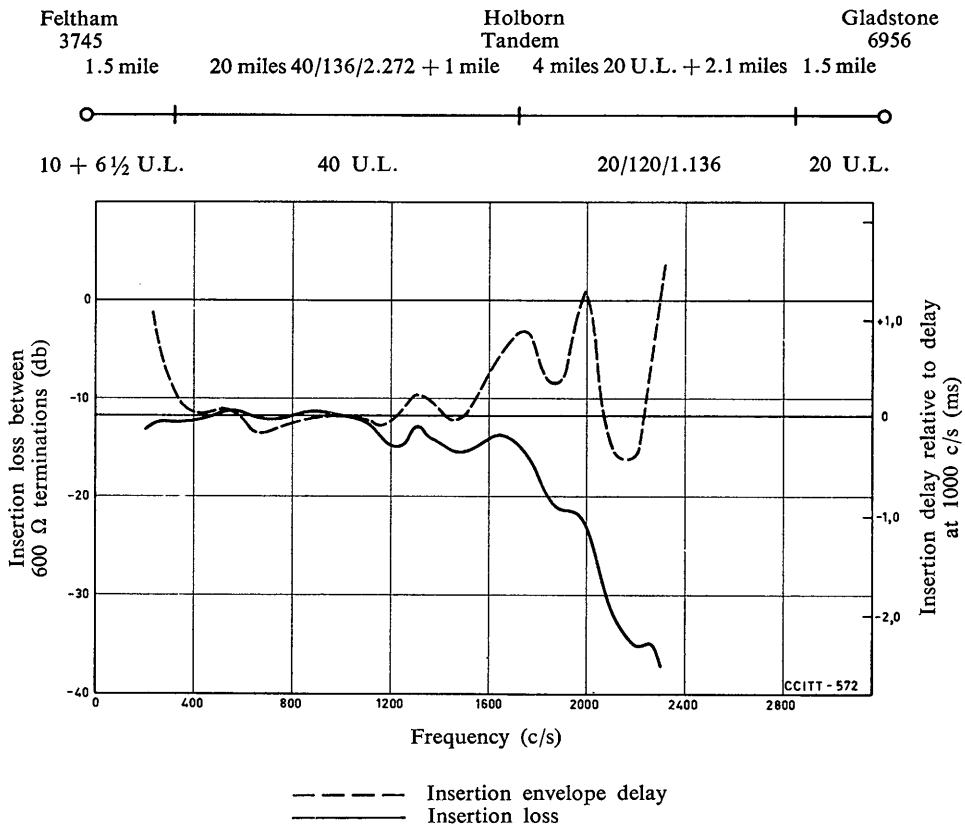


FIGURE 3. — **Insertion loss and delay**
Feltham 3745 — Gladstone 6956

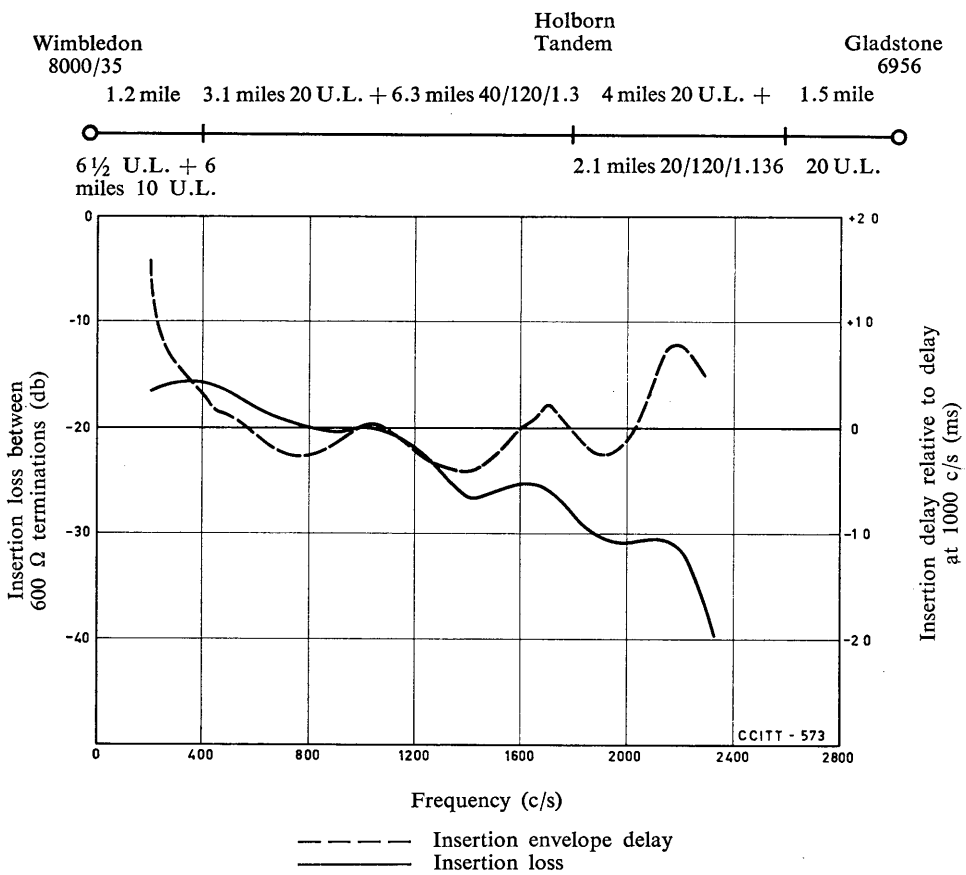


FIGURE 4. — **Insertion loss and delay**
Wimbledon 8000 — Gladstone 6956

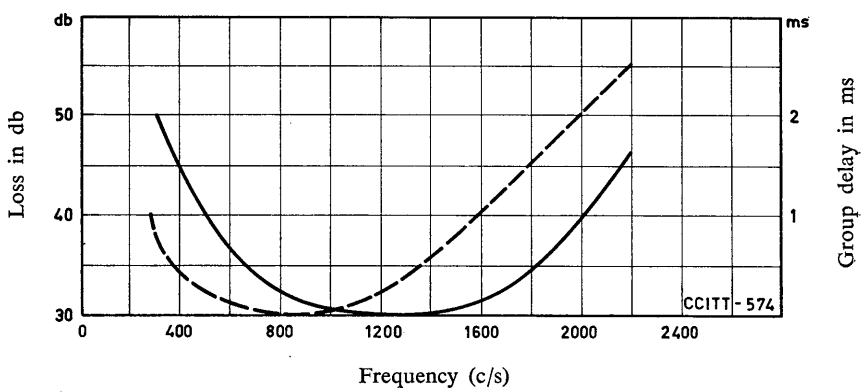


FIGURE 5. — **Loss/frequency and group delay distortion/frequency characteristics of a nominal extreme connection involving long junction and trunk circuits in the United Kingdom**

Supplement No. 74

AMERICAN TELEPHONE AND TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A/No. 75 — August 1963)*

SPECIFICATIONS FOR PRIVATE LINE VOICE-GRADE CHANNELS USED FOR DATA TRANSMISSION

Considerable need has developed and continues to develop for general purpose voice-grade data channels, which need certain specifications. Three of these findings of the greatest application in the American Telephone and Telegraph Company are shown in the following table, together with their principal characteristics and field of application.

As will be noted, the Type 4 channel is not specially conditioned, but utilizes voice channels as they are. Types 4A and 4B are conditioned for increasingly severe requirements.

Specially treated voice bandwidth circuits for data use

	Schedule 4 data	Schedule 4A data	Schedule 4B data
I. *Circuit designation*			
Use	Alternate voice/data or data only	Alternate voice/data or data only	Alternate voice/data or data only
II. *General characteristics*			
Type of service	2 point of multi-point	2 point or multi-point	2 point or multi-point
Mode of operation	Half or full duplex	Half or full duplex	Half or full duplex
Method of termination	2-wire or 4-wire	2-wire or 4-wire	2-wire or 4-wire
Imped. (source and load)	600-ohm-resistive-balance	600-ohm-resistive-balance	600-ohm-resistive-balance
Maximum signal power	−8 dbm for composite data signal, −8 vu for voice	−8 dbm for composite data signal, −8 vu for voice	−8 dbm for composite data signal, −8 vu for voice
III. *Attenuation characteristics*			
Meas. between 600-ohm impedances at line-up	8 db ± 1 @ 1000 c/s	8 db ±1 @ 1000 c/s	8 db ±1 db @ 1000 c/s
Expected max. var. of (L) [a]	Short term ± 3 db Long term ± 4 db[b]	Short term ±3 db Long term ±4 db[b]	Short term ±3 db Long term ±4 db[b]
Frequency response	*Freq. range Var.-db* 350-2000 c/s −2 to +6 2000-2500 −3 to +12	*Freq. range Var.-db* 300-1000 c/s −2 to +6 1000-2400 −1 to +3 2400-2700 −2 to +6	*Freq. range Var.-db* 300-500 c/s −2 to +6 500-2800 −1 to +3 2800-3000 −2 to +6
IV. *Delay characteristics*			
Circuit delay	Not specified	Not specified	Not specified
Envelope delay	Less than 1000/μs over band from 1000 to 2400 c/s	Less than 1000/μs over band from 1000 to 2400 c/s	Less than 500/μs 1000-2600 c/s; less than 1500 μs 600-2600 c/s; less than 3000/μs 500-2800 c/s
V. *Noise characteristics*			
Circuit noise (s/n ratio) [c]	48 dba F.I.A. (26 db)	48 dba F.I.A. (26 db)	48 dba F.I.A. (26 db)
Impulse noise	Not specified	Not more than 70 noise peaks/hr will exceed −18 dbm (long-term average)	Not more than 70 noise peaks/hr will exceed −18 dbm (long-term average)

a (L) is net loss as measured at 1000 c/s.

b Short term is a few seconds or less; long term is for longer periods, including daily and seasonal variations.

c This is noise limit for data, facsimile or control tones; limit for voice operation (if involved) may be tighter.

Supplement No. 75

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — July 1963)*

LINE CHARACTERISTICS FOR DATA TRANSMISSION

a) *Leased circuits*

a 1) Normal practice is to provide renters with telephone circuits which are suitable for speech when used point-to-point. Such a circuit would be provided as a two-wire circuit or as a four-wire circuit terminated and extended on two-wire tails into the renter's premises. The over-all loss between the renter's terminals should not exceed 13 db at 800 c/s and the variation of loss with frequency should be within the "standard quality" limits shown in Figure 1. (The 800-c/s loss of leased circuits forming part of a private switched network is limited so that the most unfavourable cascade combination of circuits will not have a loss at 800 c/s which exceeds 21 db.)

An examination of the measured characteristics of a large number of existing leased circuits showed that the loss at the frequency of 800 c/s and the loss/frequency response relative to the 800-c/s value was as shown in Figure 2. These characteristics have been given to designers of general purpose data transmission equipment as a guide in the design of modems for use over normal speech-type leased circuits.

a 2) The United Kingdom Administration finds that there is already a demand for leased telephone type circuits for applications where data transmission at a relatively high modulation rate (1200 bauds and above) is a requirement. Provision of such a circuit requires consideration being given to the following factors :

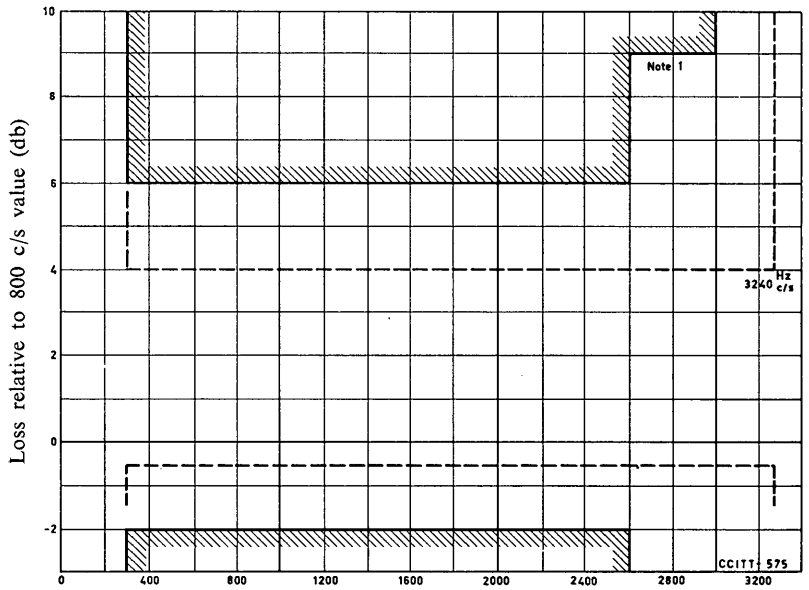
a) Selection of modern, high-quality plant (e.g. avoiding long or heavily loaded audio cables), with a good performance in respect of impulsive noise and interruptions.

b) Uniformity of circuit, e.g. correct termination of loaded cables and correct spacing of loading coils.

c) Equalization of loss and group delay, including provision of additional amplifiers where necessary to offset losses in equalizers.

d) Provision of four-wire local ends to the renters' premises.

Subject to the availability of suitable plant the target performance which is being aimed at for such a circuit is that which would be desirable for a 24-channel voice-frequency telegraphy circuit with an additional requirement that the group delay distortion should not be worse than 0.5 ms over the band 600 to 2600 c/s. Figures 1 and 2 show the proposed characteristics for high-quality leased circuits ; these characteristics are thought to represent the highest quality which it is practicable to offer and maintain as a regular service. The group delay frequency

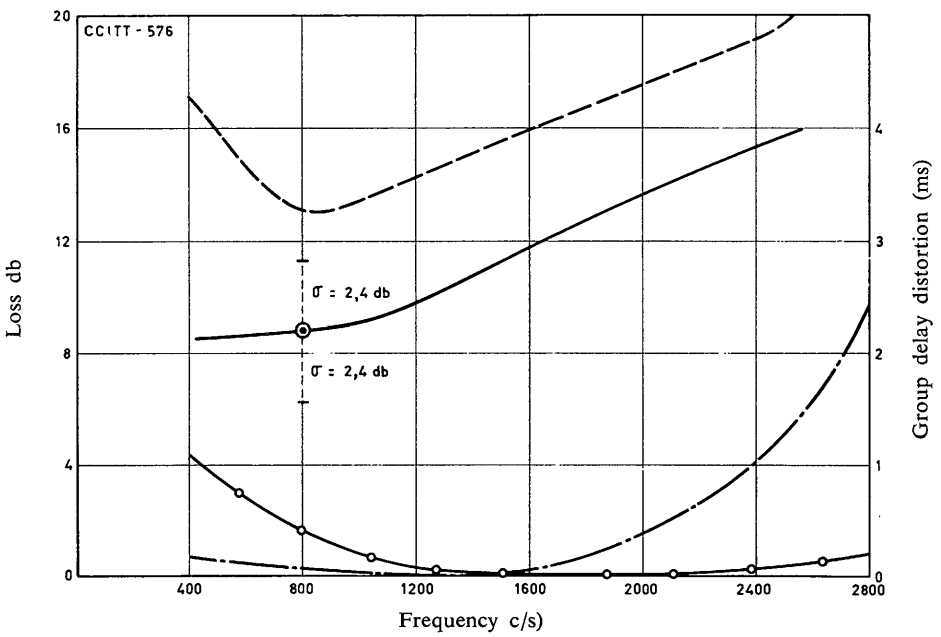


Standard quality

Note 1. — Response 2600-3000 c/s conditional on available plant

Special quality for 24-channel V.F. telegraphy or special data transmission systems

FIGURE 1. — **Loss/frequency characteristics proposed for private circuits in the United Kingdom**



— — — — — Suggested limit of loss
———— Mean loss
— o — o — Delay distortion per single carrier link
— - — - — Delay distortion per 100 miles loaded audio cable (88 mH 1.136 mile)
⊙ Possible specification : loss at 800 c/s : 3 to 13 db
Loss/frequency characteristics : +1 to −6 db (relative to 800 c/s over the band 300 to 2600 c/s)
Variation with time of 800 c/s loss : $\sigma < 2$ db

FIGURE 2. — **Typical loss and delay characteristics of private circuits in the United Kingdom (two-wire into renters' premises at each end)**

characteristic is approximately met by three channel modems of standard type or 25 miles (40 km) of standard loading [88 mH 1.136 mile (1.830 km)].

As far as possible the group delay requirement would be met by the selection of suitable plant but provision is being made of a range of standard types of group delay equalizer which can be fitted on a prescriptive basis on a complex circuit according to the items of equipment or length of line. The table below shows the proposed types of delay equalizer and their nominal characteristics.

Table of proposed group-delay equalizers

Type Equalizing:	Nominal characteristics of plant relative to frequency of minimum delay							
	600 c/s	800 c/s	1200 c/s	1600 c/s	2000 c/s	2400 c/s	2600 c/s	2800 c/s
A1 40 km loaded cable [a]	0	0	0.015	0.085	0.18	0.34	0.50	0.65
A2 80 km loaded cable [a]	0	0	0.06	0.17	0.35	0.68	1.0	1.3
A3 160 km loaded cable [a]	0	0	0.12	0.33	0.70	1.35	2.0	2.6
B3 Three-channel modems (without out-band signalling) .	1.23	0.69	0.24	0	0	0.27	0.48	0.87
C2 Two-channel modems (with out-band signalling) .	1.4	0.8	0.16	0	0	0.14	0.36	0.76

[a] 88 mH/1.83 km, 0.053 microfarad/km cable.

b) *Switched networks*

The United Kingdom Administration has on order F.M. modems (designated data transmission set No. 1A) to the recommendations of Special Study Group A. These will be capable of transmitting and receiving serially up to 600 or 1200 bauds, using characteristic frequencies $F_z = 1300$ c/s ; $F_a = 1700$ c/s, or $F_z = 1300$ c/s ; $F_a = 2100$ c/s, respectively. In addition a supervisory channel will be available capable of modulation rates up to 75 bauds using $F_z = 390$ c/s ; $F_a = 450$ c/s.

At the present time various commercially available modems are used by subscribers for transmitting data signals over the United Kingdom switched telephone network. With the availability of data transmission set No. 1A it will be obligatory for this equipment to be used. The equipment

which will be on rental terms from the Administration will be available in various configurations to suit subscribers' requirements, e.g. a data channel A—B with or without supervisory channel for control purposes working simultaneously B—A.

The data transmission set No. 1A will accommodate standard equalizers for correcting the group delay/frequency and amplitude/frequency distortion of the speech path connection. One of each of these will be supplied in the receiver of the data channel when it is considered necessary. They will have a characteristic giving half correction of the curves shown in Figure 5 of Supplement No. 73, which are typical of a connection on long junction and trunk circuits on the United Kingdom switched network.

Supplement No. 76

UNITED KINGDOM — *(Contribution COM Sp. A/No. 39 — July 1962)*

THE EFFECTS OF TRANSMISSION TIME DELAY AND SWITCHING DISCONTINUITIES

ACTIVE EARTH SATELLITE COMMUNICATION SYSTEMS FOR TELEGRAPHY

The effects of transmission time delay and switching discontinuities

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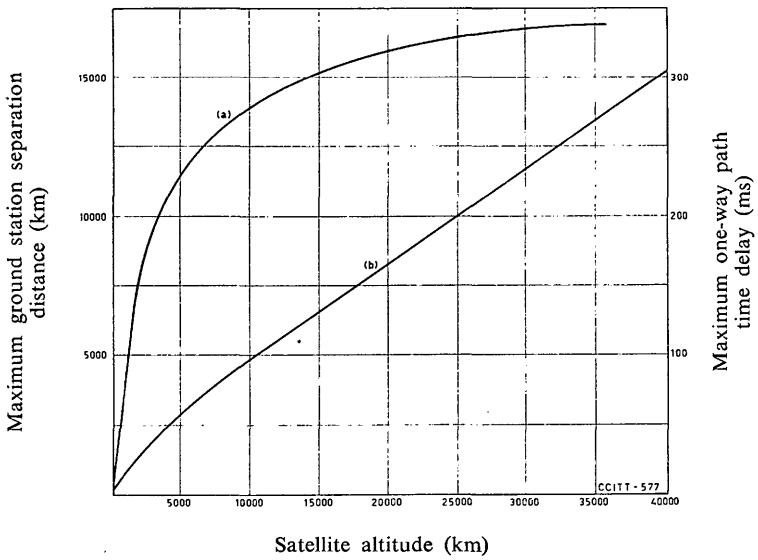
2. *Time-delay factors to be considered*

The time-delay factors affecting the use of telegraphy in a communication satellite system fall into two groups: those occurring normally (i.e. during the normal course of transmission) and those occurring only when switching from one satellite to the next. The following may be considered to fall in the first group:

2.1 *Transmission time delay (D_1)* in the propagation path between satellite ground terminals

The figure shows the maximum path-time delays and maximum separation distances between ground stations as a function of satellite altitude for a single-hop link with 5° minimum angle of elevation of the ground station antennae. Typical values are given below.

Altitude of satellite (km)	Maximum separation distance between ground stations (km)	Maximum time delay (ms)
2 000	7 900	33
5 000	11 400	59
10 000	13 800	97
15 000	15 700	132
35 900 (synchronous satellite)	17 000	275



(a) Ground separation distance
(b) Time delay

Signal-hop radio link via satellite (minimum angle of elevation = 5°)

With two-hop satellite links the delay times and separation distances would be rather less than twice the above values, according to the amount of overlap of the coverage areas of the two satellites. For example, a circuit relayed via two hops with satellites in a circular orbit at 10 000 km would suffer a delay of the order of 170 ms, while for circuits relayed via two satellites in the synchronous orbit the delay would be of the order of 530 ms.

2.2 *Time delay (D_2)* between the satellite ground station terminal equipment and the Central Telegraph Office including time taken to translate individual channels up to, or break down from, the wide-band system. Taking the two ends together, these delays may be of the order of 25 to 50 ms for each direction of transmission.

2.3 *Time delay (D_3)* accounted for by an H.F. or scatter radio tail which might be necessary at one end. Since this would require the use of a channelling system suitable for use over such radio links, substantial additional delays would be introduced. In each direction of transmission, between 25 and 50 ms may be absorbed in this way.

The following may be considered to fall in the second group, and of course apply only to non-synchronous satellite systems.

2.4 *Time difference (D_4)* due to path length changes when switching from one satellite to another. (This might be a positive or negative amount. Differences of the order of 30 ms between maximum and minimum delays can arise depending on circuit length and orbital altitude, but with a suitably arranged system the time differences when switching from one satellite to another need not exceed about 20 ms.)

With a random satellite system, time may also be lost due to non-availability of a satellite at switchover. This will be a function of the number of satellites and their orbits. Periods of non-availability will be predictable and will need to be met by special traffic arrangements, which might include storage, delayed transmission or re-routing.

3. *Effects of normal time delays D_1, D_2 and D_3 (i.e. while no switching is taking place)*

On a telegraph system without error-correcting equipment (and this presupposes no H.F. radio tail) none of these delays need be considered. On the other hand, with the most common error-correcting system in use and an H.F. radio tail linked to the satellite terminal the total "transmission loop" delay ($2D_1 + 2D_2 + 2D_3$) would be important. From section 2, these delays might be expected to be :

— for a 11 000-km ground distance circuit relayed via a single hop from an orbiting satellite at 5000 km, of the order of 260 ms ;

— for a 14 000-km circuit relayed via a satellite at 10 000 km, of the order of 340 ms ;

— for the synchronous satellite, with a ground distance of 17 000 km, of the order of 700 ms.

These figures are not beyond the capabilities of present error-correcting systems provided an eight-character repetition cycle is employed. The maximum allowable loop delays for an ARQ system with four, five and eight character repetition cycles and using equipment now available or shortly to be available are approximately 250, 400 and 400 to 850 ms respectively.

For a two-hop circuit using satellites in synchronous orbit the loop transmission delay would be about 1200 ms and this would require the use of an ARQ system with greater storage capacity than those indicated above.

4. *Effects of the transmission delay difference at switching (D_4)*

The differences in time due to path-length changes (D_4), would affect any telegraph circuit using synchronous printing or error-correcting systems in which terminal synchronization was required. At 100-baud transmission speed for example, each element of a seven-unit code would occupy 10 ms and, with a two-channel system, it would be possible to "slip" a fraction less than half an element, say 4 ms without loss of phase. The values of D_4 suggested in section 2 are considerably in excess of this, so loss of phase might be expected at each switchover. Modern ARQ equipment would automatically re-phase, and there would be an interruption of only about 4 s provided that a suitable out-of-phase detection device is employed. The loss of phase at switching would not occur with synchronous (stationary) orbit satellite systems. It may be possible to overcome the difficulty in non-synchronous orbit satellite systems by arranging the transfer of traffic from one satellite to the other when the time delay for the new satellite is the same as the delay for the old one. This would introduce a serious restriction on the availability of each satellite and is not considered to be a feasible solution for a multi-station system. However, the loss of 4 s while re-phasing takes place at each switchover will not be a serious disadvantage unless the switchovers are very frequent.

Switching delay differences of the order of 20 ms would be likely to cause one or possibly more character errors in an unprotected 50-baud start-stop telegraph circuit. Since switching from one satellite to another would occur only infrequently, such errors may be considered to have only a slight effect on over-all circuit performance.

Various techniques might be adopted to overcome the effects of delay differences at switching. For example, one arrangement would be to introduce into each ground-satellite-ground path

a variable time delay such that when combined with the propagation delay (which is itself variable) a constant over-all delay would be obtained. The magnitude of the constant delay would be fixed and equal to the maximum propagation delay anticipated.

Supplement No. 77

SPECIAL STUDY GROUP A — *(Extract from report COM Sp. A/No. 92 — October 1963)*

REPLY BY SPECIAL STUDY GROUP A TO QUESTION FROM SPECIAL STUDY GROUP C ON NOISE LIMITS FOR DATA TRANSMISSION

1. Published information on the performance of current types of data modems using two-condition modulation shows that, in the absence of other serious impairments, the limiting signal-to-white-noise ratio (for an arbitrarily chosen continuous error rate of 10^{-6}) is :

$$10 \log_{10} \cdot \frac{E}{N_0} > 12 \text{ to } 15 \text{ db (according to Sunde, } B.S.T.J., \text{ November 1959)}$$

where E = signal energy per signal element = $\dfrac{\text{signal power (in watts)}}{\text{modulation rate (in bauds)}}$

N_0 = noise power per unit bandwidth.

Thus, a 1200-baud system, operating, for example, at a power level of −10 dbm0 would reach the 10^{-6} error rate in the presence of white noise in the range −20 to −23 dbm0p. A 600-baud system could theoretically accept a signal/noise ratio 3 db worse.

2. Very considerable margin, certainly not less than 10 db in the most favourable case, must be allowed on these figures to give satisfactory service on practical circuits and connections, noting that the error rate would worsen by one or two orders for only a few decibels worsening in signal/noise ratio. The factors to be considered are :

a) reduction in signal level from the assumed value of −10 dbm0, due to statistical variation in loss between the transmitters and the zero relative level point of the international circuit ;

b) delay distortion in carrier terminal equipment ;

c) delay distortion in loaded cables ;

d) impulsive noise ;

e) low level of received signal due to long, local cable connections.

Some of these factors can be made unimportant on leased circuits, but all may become dominant on switched connections.

3. A reasonable objective for uniform spectrum random noise for a data-transmission *leased* circuit, assuming that plant liable to impulsive noise interference is avoided and that as high

a modulation rate as possible is to be used without significant error rate, would appear to be −40 dbm0p.

4. For switched connections, short-duration transmissions may be assumed, with an appreciable proportion of unusable data calls on long, complex connections. A design objective, for intercontinental circuits, if compandors must be foreseen, of say, −36 dbm0p (uncompandored) would probably not seriously increase the proportion of unusable connections.

5. The value of −30 dbm0p, envisaged by Special Study Group C, would not be acceptable as a design objective for circuits used for data transmission.

6. Future demands for increased data-signalling rates over telephone circuits will undoubtedly force consideration of other modulation techniques; additional signal-to-noise margins would then become necessary.

Supplement No. 78

L. M. ERICSSON (SWEDEN) — *(Contribution COM Sp. A/No. 21 — October 1961)*

THE INFLUENCE OF SOME TRANSMISSION DEFECTS ON BINARY DATA SYSTEMS: A THEORETICAL EVALUATION

Summary

Transmission quality can always be assessed by direct measurements of error rates. It is, however, often preferable to judge the impairments by evaluation of known transmission quantities.

As high-speed data transmission becomes more common, realistic evaluation methods of transmission defects will be much needed, especially with regard to delay distortion.

Previous evaluations have mainly followed two different lines of approach. Theoreticians have made very thorough investigations for some types of equipment and special forms of delay distortion. The complicated computations, however, make general application difficult. The practical engineer has gone to the other extreme and resorted to rule-of-thumb methods, as for instance an absolute limit on permissible delay deviation. This results in too broad generalizations and inaccuracies.

This Supplement attempts in avoiding these extremes to give a numerical defect evaluation for an arbitrary transmission medium. The procedure is based on a statistical determination of the error rate for a generalized data system, which uses the transmission medium in question.

Many defects are shown to be transformable to equivalent noise signals, which add on a power basis to give a composite noise signal. The power ratio between this and the normal information-bearing signal determines the fault ratio.

Certain general noise signals and synchronization faults are examined briefly, but the main emphasis is placed on attenuation and phase distortion. It is shown that the *sum of the mean square deviations of attenuation and non-linear phase is a direct measure of signal-to-noise ratio.* This leads to simple rules for optimum delay equalization.

The derivations and calculations are carried out strictly for a binary phase shift data system only. However, some reasons are also given to justify the use of the rules for a binary frequency shift system as well.

Introduction

This investigation deals only with *high-speed* binary data transmission. The following general conditions apply.

Type of information

The information to be transmitted is a continuous flow of binary data with equal probability for mark and space.

Special start or stop signals, block synchronization signals, etc., are not used for the data transmission system. Timing or clock signals are not transmitted. (The clock frequency is, however, very exactly controlled and has a high stability.)

Transmission medium

A characteristic feature of the transmission medium is that it has very small bandwidth compared to the information which is to be transmitted. (One expects, for instance, a double sideband A.M. system to transmit B bits/s in a bandwidth of substantially B c/s.) The transmission medium may contain physical lines with pupin coils, carrier systems, radio-link systems, all subject to delay and attenuation distortion, frequency faults and noise.

The data transmission system

The evaluations are based on a generalized model of a data transmission system according to Figure 1.

The transmitter according to Figure 1 receives binary data symbols, which are transformed to a time-varying signal. The signal passes through the transmission medium, where it is subjected to degradations, and comes to the receiver.

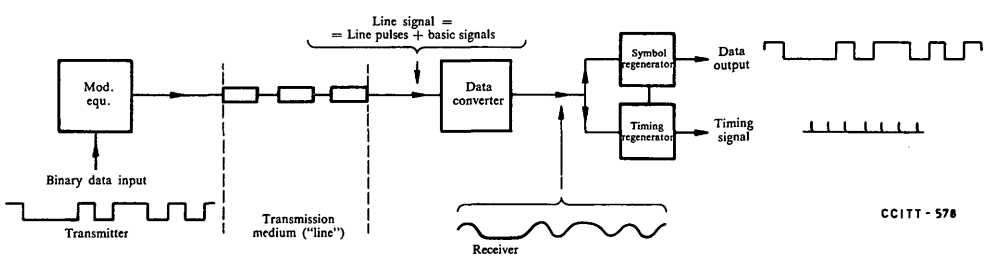


FIGURE 1. — **Generalized data transmission system**

All practical data transmission receivers can be considered to have some form of data converter output, where mark and space signals are presented as respectively positive or negative signals. Thus a long row of mark or space symbols corresponds to a constant positive or negative signal.

Mark and space symbols are transmitted with equal time intervals. At the receiver the symbol detection is achieved by sampling the converter output signal with the same time intervals, but displaced some necessary amount of time with regard to the transmitter. The sampling is done in the "symbol regenerator", which regenerates the information to square pulses.

Transmitter and receiver are mutually synchronized by means of the so-called "timing regenerator". This fulfils two necessary functions, i.e. to give out timing signals and to achieve optimum synchronization. The regenerator can be designed in many ways. A discussion of this is, however, outside the scope of this Supplement. The way in which the signal at the converter output varies with time depends of course on the distribution of mark and space symbols. Each symbol gives its separate signal contribution, which can be considered as being linearly added to the others. The mark or space signal element can thus be considered to be a separate pulse. We will use the designation *symbol pulse* for the signal element at the converter output. The symbol mark or space is thus indicated by one or the other polarity of the symbol pulse. The frequency range at the converter output goes of course down to direct current, and is called *base band.*

The corresponding pulse elements can be found at the input of the converter, i.e. on the line, where they are transposed in frequency range. The designation *line pulse* will be used as the signal element on the line.

The converter is assumed to perform the detection in an optimum way, i.e. it automatically corrects for any frequency errors between transmitter and receiver, as well as level variations.

In the general case in addition to the information carrying line signals there may be a signal containing no information, which we shall call *basic signal.*

For a phase shift data system (amplitude modulation $\pm$ 100%) the basic signal is of course zero.

For amplitude modulation with carrier rest the basic signal is equal to the signal mean between mark and space signals on the line.

A frequency shift modulation can also *very* roughly be described as a phase shift system, with the basic signal consisting of two sinusoidal signals. The line pulse contains the same two sinusoidal signals, but one of them is of opposite polarity with regard to the basic signal. By changing the polarity of the line pulse one or the other of the sinusoidal signals becomes blocked.

With a knowledge of the transmission line phase and loss it is possible to calculate how the line and symbol pulses are changed for different cases. We shall come back to this later on.

Waveform and frequency spectrum for the symbol pulse

As already mentioned the smallest possible bandwidth is to be used for the binary transmission. If the bits are spaced T_0 apart, the symbol pulse (i.e. the converted data signal) needs at least a bandwidth of $f_0 = \frac{1}{2} T_0$ if symbol interference is to be avoided. In practice a real data system needs a somewhat larger bandwidth. Therefore we shall reckon with an *effective bit length* τ_2, somewhat shorter than T_0. As a practical rule:

$$\tau_2 = 0.9\ T_0 \text{ or } 0.8\ T_0 \qquad (1)$$

Because of the minimum bandwidth requirement, the ideal symbol pulse can only assume the following well-known form :

$$\left.\begin{aligned} S(t) &= \frac{1}{\omega_2}\int_0^{\omega_2} \cos \omega t \, d\omega = \frac{\sin \omega_2 t}{\omega_2 t} \\ &\text{where } \omega_2 = \pi/\tau_2 \end{aligned}\right\} \quad (2)$$

See Figure 2.

This may represent the symbol "mark", which is to be sampled at $t = 0$ and with the amplitude value normalized to 1. It can be seen that for this *normalized pulse* we get :

$$\left.\begin{aligned} &S(0) = 1 \\ &S(n\,\tau_2) = 0 \text{ for } n = 0 \end{aligned}\right\} \quad (3)$$

Thus these pulses can be placed at whole multiples of τ_2 without symbol interference.

$$S(t) = \int_0^{\infty} a(\omega) \cos \omega t = \frac{\sin \omega_2 t}{\omega_2 t}$$

$$\omega_2 \tau_2 = \pi$$

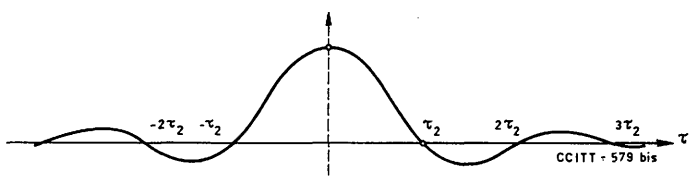


FIGURE 2. — **Normalized symbol pulse**

The frequency spectrum for the normalized pulse is rectangular up to $\omega = \omega_2$ and contains only cosine components. See Figure 3.

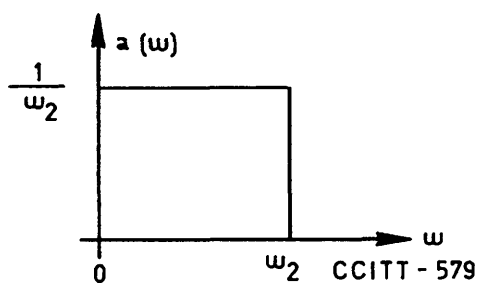


FIGURE 3. — **Frequency spectrum for the normalized symbol pulse of Figure 2 (contains only cosine components)**

One can see that the so-called "pulse power", i.e. the integrated square divided by the pulse interval τ_2, is equal to 1.

If the normalized pulse is subjected to a frequency dependent attenuation $A(\omega)$ and phase shift $B(\omega)$ the distorted pulse form will be given of the integral:

$$S_d(t) = \frac{1}{\omega_2} \int_0^{\omega_2} e^{-A(\omega)} \cos\left[\omega t - B(\omega)\right] d\omega \tag{4}$$

According to a known formula for Fourier integrals, the "pulse power" will now be:

$$E_p = \frac{1}{\tau_2} \int_{-\infty}^{\infty} S_d^2(t) dt = \frac{1}{\omega_2} \int_0^{\omega_2} e^{-2A(\omega)} d\omega \tag{5}$$

which is independent of $B(\omega)$.

As the distorted symbol pulse contains frequency components only up to $\omega = \omega_2$ it will be completely characterized by its values at equidistant sampling points spaced $\tau_2 = \pi/\omega_2$ apart. It can be shown that the "pulse power" can be written also in the following form:

$$E_p = \sum_{\eta=-\infty}^{\infty} S_d^2(t+n\tau_2) = \frac{1}{\omega_2} \int_0^{\omega_2} e^{-2A(\omega)} d\omega \tag{6}$$

independent of the value of t.

These formulae will be used later on in connection with loss and phase shift evaluations.

It turns out that the normalized pulse is very convenient and leads to simple expressions. A comparison is made in the Appendix between the normalized pulse and pulses in practical data systems. The errors are shown to be negligible in typical cases.

Frequency spectrum of symbol pulse and line pulse

We are now going to discuss how the frequency spectra for symbol pulse and line pulse are interconnected for different modulation systems, and how attenuation and phase shift are transformed between the line and base band frequency ranges.

Phase shift modulation is of course the same as *double sideband suppressed carrier amplitude modulation*. In the general, ideal, case, the line will have the following Fourier integral expression :

$$\left.\begin{aligned} S_1(t) &= \int_0^{\infty} a(W)\cos Wt\,dW + \int_0^{\infty} b(W)\sin Wt\,dW = \\ &= \frac{1}{\omega_2}\cos(\omega_m t + \varphi)\int_0^{\omega_2}\cos\omega t\,d\omega = \\ &= \frac{1}{2\omega_2}\int_{\omega_m-\omega_2}^{\omega_m+\omega_2}\cos(Wt+\varphi)\,dW \end{aligned}\right\} \quad (7)$$

where ω_m is the angular modulation frequency.

For the sake of clarity,

ω = means angular frequencies in the base band ;
W = means angular frequencies in the line band.

It can be seen from equation (7) that the frequency components have equal amplitude in the band $\omega_m \pm \omega_2$. Figure 4 shows the frequency spectrum $a(W)$ for the special case $\varphi = 0$, when $b(W) = 0$.

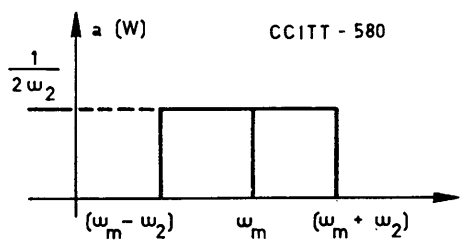


FIGURE 4. — **Line pulse frequency spectrum for phase shift modulation**

At the data receiver an ideal demodulation can be supposed to take phase with the carrier $\cos(\omega_m t + \varphi)$. The base band signal is built up of equally large contributions coming from the demodulated upper and lower sidebands.

Suppose the line pulse is subjected to a frequency dependent attenuation $A_1(W)$ and phase shift $B_1(W)$ according to Figure 5.

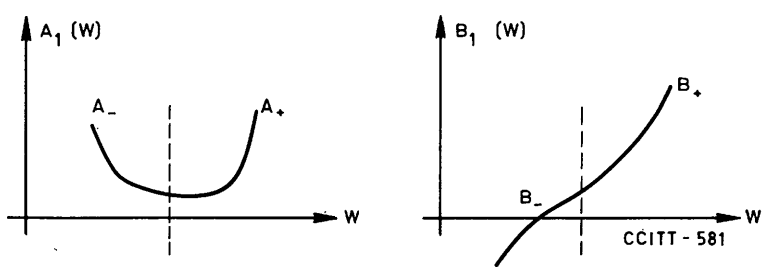


FIGURE 5. — **Example of line attenuation $A_1(W)$ and phase $B_1(W)$**

In the demodulation process the two separately attenuated and phase-distorted sidebands are added together linearly. The equivalent attenuation $A(\omega)$ and phase shift $B(\omega)$ for the base band can now be calculated from the expression :

$$e^{-(A\omega)-jB(\omega)} = \frac{1}{2} e^{(-A_+ - jB_+)} + \frac{1}{2} e^{(-A_- + jB_-)} \tag{8}$$

here,

$$\begin{cases} A_+ = A_1(\omega_m + \omega) \\ A_- = A_1(\omega_m - \omega) \end{cases} \qquad \begin{cases} B_+ = B_1(\omega_m + \omega) \\ B_- = B_1(\omega_m - \omega) \end{cases} \tag{9}$$

(Compare with the designations in Figure 5.)

The signs B_+ and B_- in equation (8) should be noticed. In the interesting special case $A_+ = A_- = 0$, we get :

$$\left.\begin{aligned} A(\omega) &= \ln \left| \cos \frac{1}{2}(B_+ + B_-) \right| \\ B(\omega) &= \frac{1}{2}(B_+ - B_-) + m\pi \end{aligned}\right\} \tag{10}$$

$$\left.\begin{aligned} m &= 0 \quad \text{for} \quad \cos \frac{1}{2}(B_+ + B_-) > 0 \\ m &= 1 \quad \text{for} \quad \cos \frac{1}{2}(B_+ + B_-) < 0 \end{aligned}\right\} \tag{11}$$

If the demodulation is made with a phase error of B_m in the modulation voltage, this will correspond to $B_1(W)$ being increased by a constant amount B_m over the whole frequency band.

Of course the formulae can also be used for *single sideband modulation* by letting $A_+ \longrightarrow \infty$ or $A_- \longrightarrow \infty$.

Vestigial sideband modulation corresponds to the demodulated signal receiving a contribution from both sidebands in a narrow range around the modulating frequency, and outside this from only one sideband. Equation (8) also applies here if the attenuation and phase shift of the vestigial filters are included in $A_1(W)$ and $B_1(W)$.

Frequency shift modulation for binary data transmission has a small modulation index, i.e. it occupies about the same bandwidth as the double sideband system. One can therefore argue with some justification that the line distortion affects the converted data signal, i.e. the symbol pulse, roughly in the same way for frequency shift modulation as for double sideband modulation.

Error evaluation for symbol signal plus noise. General case

We are now going to study the general error probability when there is some form of noise present beside the symbol signals at the output of the converter.

Suppose the received pulse at its proper sampling moment has the amplitude value of $S_0 > 0$ in undisturbed conditions. The symbol regenerator is also subjected to a number of *noise signals* which at the same sampling moment have instantaneous values D_1, D_2,... D_n. The symbol regenerator gives the right answer as long as the sum of the signals exceeds 0, but an error is registered if

$$\left.\begin{aligned} &S_0 - D < 0 \\ &\text{where } D = \sum_{\eta} D_n \end{aligned}\right\} \tag{12}$$

The noise signals are not constant but vary from one sampling moment to the other, quite independently. The error probability is governed by the statistical distribution of the added noise signals. To gain some knowledge of this distribution we are going to seek its *mean value* and the *second order moment around the mean* respectively.

As is well known, these can be *exactly* calculated as a *linear sum* of the corresponding values for the separate statistical variables.

$$\left.\begin{aligned} \overline{D} &= \sum_{\eta} \overline{D}_n \\ \sigma_D^2 &= \sum_{\eta} \sigma_n^2 \end{aligned}\right\} \tag{13}$$

Expressed in another way σ_n is quite simply the *root-mean-square* (r.m.s.) *value* at the sampling instants for the noise signal n, and σ_D is the corresponding value for the combined noise signals.

It would be too difficult and confusing to try to calculate the exact statistical distribution for different noise disturbances. However, for our purposes, the distribution of errors can be quite accurately represented by a *normal distribution* as it contains many independent stochastic variables, of which no one can be said to dominate over any other. Equation (13) will then characterize this normal distribution completely.

For the actual noise disturbances it turns out that their mean value is

$$\overline{D}_n = 0 \tag{14}$$

(equal probability that the disturbance will amplify or weaken the signal, equal probability that mark or space is transmitted).

According to known formulae the error probability will then be:

$$P_f = P(D > S_0) = \Phi\left(\frac{S_0}{\sigma_D}\right) \tag{15}$$

where

$$\Phi\left(\frac{S_0}{\sigma_D}\right) = \frac{1}{\sqrt{2\pi}} \int_{\frac{S_0}{\sigma_D}}^{\infty} e^{-u^2/2} du \tag{16}$$

We shall define the symbol *signal-to-noise ratio*

$$A_D = \ln \frac{S_0}{\sigma_D} \tag{17}$$

The error probability P_f as a function of A_D is shown in Figure 6.

For a number of simultaneous noise signals with the corresponding signal-to-noise ratios A_{D1}, A_{D2} ... nepers, the resulting signal-to-noise ratio becomes

$$A_D = \frac{1}{2} \ln \left\{ e^{-2A_{D1}} + e^{-2A_{D2}} + \dots \right\}^{-1} \tag{18}$$

Other pulse combinations can also be used as a criterion for mark and space. In the case of so-called "differential phase shift systems", we have:

space = the same pulse polarity is transmitted as the preceding one

mark = opposite pulse polarity is transmitted as the preceding one.

For such a system the error probability will be

$$P_f^1(A_D) = 2\,P_f(A_D) - P_f^2(A_D) \tag{19}$$

(This is the probability that any two adjacent pulses will be in error minus the probability that both of them are in error.)

The error probability P_f^1 is also shown in Figure 6. (These curves are of course well known, but are included here for the sake of completeness.)

The r.m.s. value is quite easy to identify for some important types of noise signals.

It can be shown that the *r.m.s. value* of the noise at the sampling instants is equal to the *effective value* for the following types of disturbances :

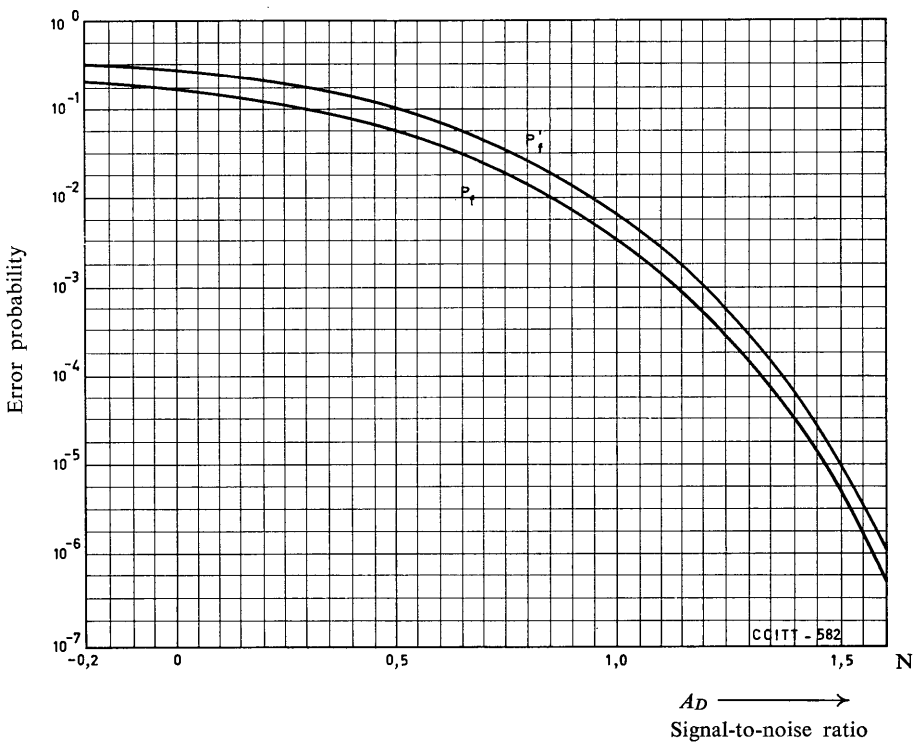


FIGURE 6. — **Error probability as a function of signal-to-noise ratio**

$A_D = \ln \dfrac{S_0}{\sigma_D}$

The curve P_f applies to ordinary phase-shift systems

The curve P'_f applies to ordinary differential systems

S_0 = sampled pulse amplitude for data symbol

σ_D = r.m.s. of the noise

White noise

A *limited number of sinusoidal signals*, where none of the frequency components or its second order intermodulation products are synchronous with the sampling frequency.

Non-periodic signals with the *same* bandwidth as the symbol pulse

If the disturbance is *synchronous* with the sampling frequency the same amplitude value is always measured at the sampling instants. This will thus be the r.m.s. value.

Symbol interference due to attenuation and phase distortion

Although it is not so obvious, *symbol interference* can also be expressed as noise with a definite r.m.s. value.

Exact expressions

Suppose a symbol pulse according to equation (2) is sent out at the instant $t = 0$. It is subjected to the complex insertion loss $A(\omega) + jB(\omega)$ and is sampled at the moment $t = \tau_0$ at the converter output. The amplitude at the sampling instant is found from equation (4):

$$S_{d0} = S_d(\tau_0) = \frac{1}{\omega_2} \int_0^{\omega_2} e^{-A(\omega)} \cos \left\{ \omega\tau_0 - B(\omega) \right\} d\omega \tag{20}$$

See Figure 7.

An arbitrary symbol pulse which has been sent out earlier at the instant $t = n\tau_2$ will at the sampling moment τ_0 give the following noise contribution:

$$D_n = \pm S_d(\tau_0 + n\tau_2) \tag{21}$$

As there is equal probability of plus or minus sign, the mean square value is:

$$\sigma_n^2 = \frac{1}{2} \left\{ - S_d(\tau_0 + n\tau_2) \right\}^2 + \frac{1}{2} \left\{ + S_d(\tau_0 + n\tau_2) \right\}^2 = S_d^2(\tau_0 + n\tau_2) \tag{22}$$

The data flow consists of an infinite binary pulse train, and the total symbol interference results in the following noise mean square value:

$$\sigma_D^2 = \sum_{\eta = \infty}^{\infty} S_d^2(\tau_0 + n\tau_2); \; n \neq 0 \tag{23}$$

If the term S_{d0}^2 is added, we shall get the expression for pulse power according to equation (6).

$$\sigma_D^2 + S_{d0}^2 = E_p = \frac{1}{\omega_2} \int_0^{\omega_2} e^{-2A(\omega)} d\omega \tag{24}$$

Thus:

$$\sigma_D = \sqrt{E_p - S_{d0}^2} \tag{25}$$

Signal-to-noise ratio

$$A_D = \ln \frac{S_{d0}}{\sigma_D} = \frac{1}{2} \ln \frac{S_{d0}^2}{E_p - S_{d0}^2} \tag{26}$$

In the important special case $A(\omega) = 0$ we shall get $E_p = 1$.

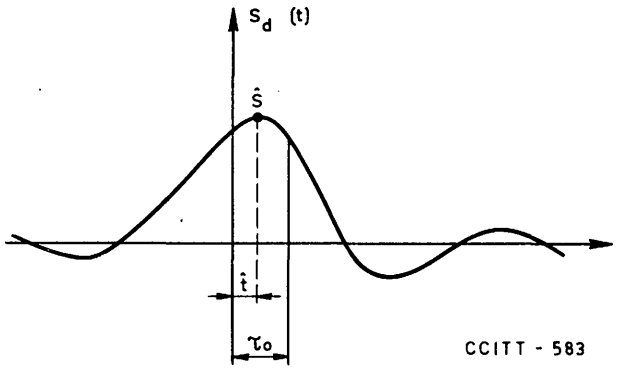


FIGURE 7. — **Symbol pulse, degraded by phase and loss distortion**

A_D will clearly be a *maximum* when $S_{d0} = \hat{S}$ is a *maximum*—i.e. the *optimum sampling instant* is the *peak value* of the distorted symbol pulse. See Figure 7.

This *maximum signal-to-noise ratio* at the optimum sampling instant is a convenient measure of how the *loss and non-linear phase distortion* affects the data pulses. It may be appropriate to give it a special name : "*pulse fidelity index*".

$$A_p = \frac{1}{2} \ln \frac{\hat{S}^2}{E_p - \hat{S}^2} \tag{27}$$

The optimum sampling instant $\tau_0 = \hat{\tau}$ is found from the expression :

$$\frac{\delta S_{d_0}}{\delta_{\tau_0}} = 0 \tag{28}$$

i.e.
$$\int_0^{\omega_2} \omega \ e^{-A(\omega)} \sin \left\{ \omega\hat{\tau} - B(\omega) \right\} d\omega = 0 \tag{29}$$

(In the special case $B(\omega) = 0$ we shall get $\hat{\tau} = 0$.)

With the help of (20), (24) and (29) the pulse fidelity index A_p can be calculated numerically from (27).

Approximate formulae

Numerical comparisons have shown that the exact but complicated equations (20) to (29) can be replaced by the far simpler but for most cases sufficiently accurate formulae which follow.

For those amounts of attenuation and phase distortions which are encountered in practice, the exponential and cosine functions can be replaced by series expansions, and *only the very first terms need be considered.*

Likewise, higher-order terms can be discarded in the multiplying of the exponential and cosine expansions with little loss of accuracy. With these hints, the calculations are quite straightforward. The end result is :

The pulse fidelity index :

$$A_p \approx \frac{1}{2} \ln \left\{ A_e^2 + B_e^2 \right\}^{-1} \tag{30}$$

Here A_e and B_e are the *minimum r.m.s. values of loss and non-linear phase variation* over the base band :

$$\left.\begin{aligned} A_e^2 &= \frac{1}{\omega_2} \int_0^{\omega_2} \left\{ A(\omega) - A_0 \right\}^2 \, \mathrm{d}\omega \\ B_e^2 &= \frac{1}{\omega_2} \int_0^{\omega_2} \left\{ B(\omega) - \omega\hat{\tau} \right\}^2 \, \mathrm{d}\omega \end{aligned}\right\} \tag{31}$$

(A_e in nepers, B_e in radians).

"Minimum" means here that the constant A_0 and $\hat{\tau}$ are to be chosen optimally. They are given by the formulae :

$$\left.\begin{aligned} A_0 &= \frac{1}{\omega_2} \int_0^{\omega_2} A(\omega) \, \mathrm{d}\omega \\ \hat{\tau} &= \frac{3}{\omega_2^3} \int_0^{\omega_2} \omega B(\omega) \, \mathrm{d}\omega \end{aligned}\right\} \tag{32}$$

Figure 8 gives equation (30) as a diagram. For ease of computation, B_e is given in degrees. The insert figures indicate how the variations are calculated from the reference lines $A = A_0$ and $B = \omega\hat{\tau}$ respectively. Equation (32) means quite simply that they are drawn according to the *least square method.*

If the transmission line consists of a number of links with mutually different phase and distortion curves, the resulting r.m.s. variations can at least approximately be added on a root-mean-square basis :

$$\left.\begin{aligned} A_e^2 &= \sum_{\eta} A_{en}^2 \\ B_e^2 &= \sum_{\eta} B_{en}^2 \end{aligned}\right\} \tag{33}$$

Faults in synchronization

The designer of a data system is interested to know how accurately the sampling instants are to be synchronized by the receiver and the transmitter. A general idea can be had by evaluation of the normal pulses according to equation (2). Suppose :

$$\begin{aligned} &\text{optimum sampling instant} &&= \hat{\tau} \\ &\text{the real sampling instant} &&= \hat{\tau} + \Delta t \end{aligned}$$

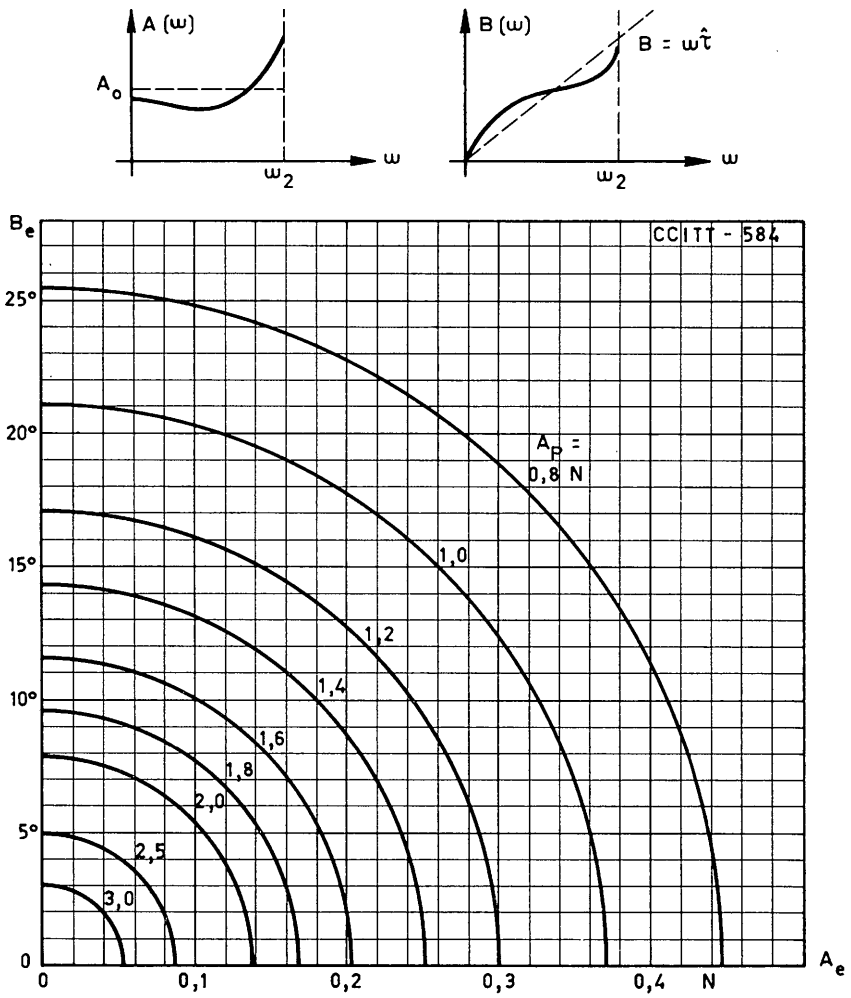


FIGURE 8. — **Pulse fidelity index A_p as a function of r.m.s. values of loss (A_e) and non-linear phase (B_e):**

$$A_e = \sqrt{\frac{1}{\omega_2}\int_0^{\omega_2}\left[A(\omega) - A_0\right]^2 d\omega} \; ; \; A_0 = \frac{1}{\omega_2}\int_0^{\omega_2} A(\omega)d\omega$$

$$B_e = \sqrt{\frac{1}{\omega_2}\int_0^{\omega_2}\left[B(\omega) - \omega\hat{\tau}\right]^2 d\omega} \; ; \; \hat{\tau} = \frac{3}{\omega_2^3}\int_0^{\omega_2} \omega B(\omega)d\omega$$

We shall only consider the case :

$$\Delta t \ll \tau_2 \,;\; A_e \ll 1 \,;\; B_e \ll 1$$

We can now start from equations (28) and (29). If $(\hat{\tau} + \Delta t)$ is substituted for $\hat{\tau}$ the signal-to-noise ratio becomes

$$A_D = \frac{1}{2} \ln \left\{ A_e^2 + B_e^2 + \left(\frac{\Delta t}{\tau_2}\right)^2 \cdot \frac{\pi^2}{3} \right\}^{-1} \tag{34}$$

i.e. even synchronization faults have the same effect as a noise signal which can be added on an effective value basis to the other types of disturbances.

If the time displacement Δt varies with time one must use the mean value of $(\Delta t)^2$, i.e. the square of the effective value

$$(\Delta t_e)^2 = \Delta t^2 \tag{35}$$

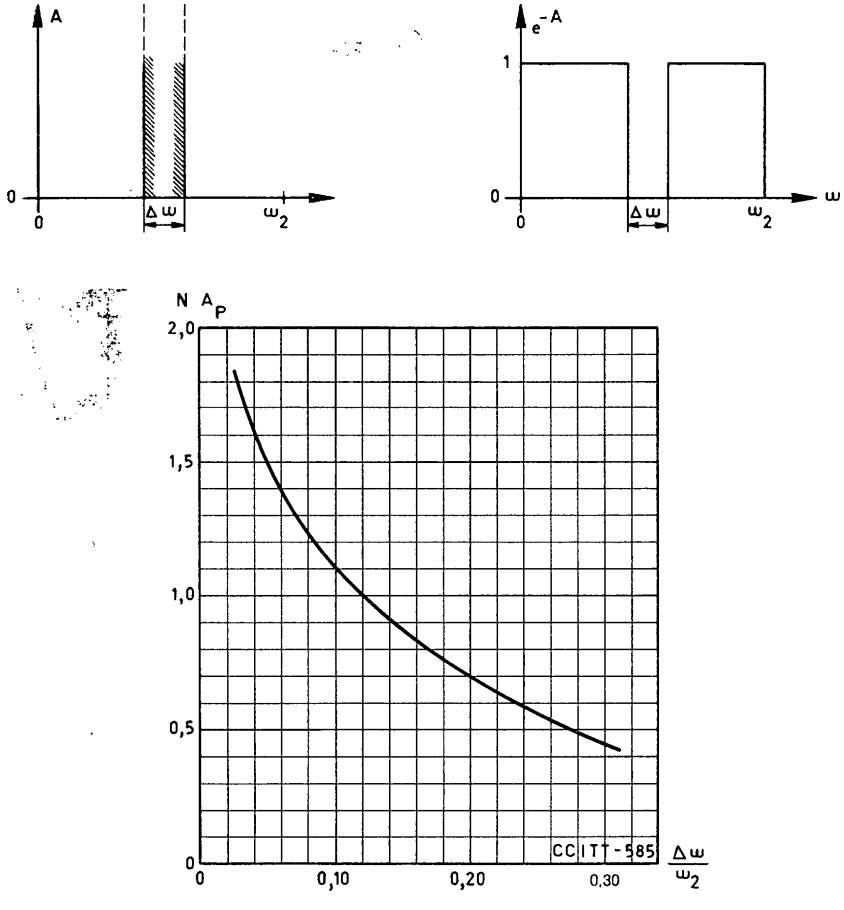
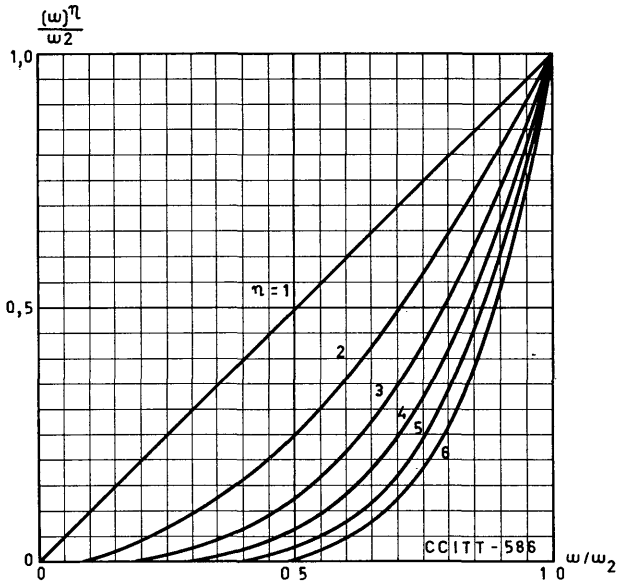


FIGURE 9. — **Pulse fidelity index A_P when the base band has a "gap"**

(A_P depends only upon the width of the gap, not on its location.)



Type-curves for variations in the baseband of :

Attenuation	*Phase shift*	*Delay*
$A(\omega) = \Delta A \left(\frac{\omega}{\omega_2}\right)^n$	$B(\omega) = \Delta B \left(\frac{\omega}{\omega_2}\right)^n$	$\tau(\omega) = \frac{\delta B}{\delta \omega} = \Delta\tau \left(\frac{\omega}{\omega_2}\right)^n$
$A_e = k_A \cdot \Delta A$	$B_e = k_B \cdot \Delta B$	$B_e = \frac{\Delta\tau}{\tau_2} \cdot k_\tau$

n	k_A	k_B	k_τ
1	0.289	0	10.°1
2	0.298	0.112	9.°1
3	0.283	0.151	7.°5
4	0.267	0.167	6.°2
5	0.251	0.172	5.°0
6	0.238	0.173	4.°4

FIGURE 10. — **Constants for computation of the r.m.s. values A_e and B_e for certain typical variations in A, B or τ**

The synchronization fault signal-to-noise ratio will thus be :

$$A_{DS} = \ln \frac{\tau_2}{\Delta t_e} \cdot \frac{\sqrt{3}}{\pi} \tag{36}$$

(τ_2 = effective bit length).

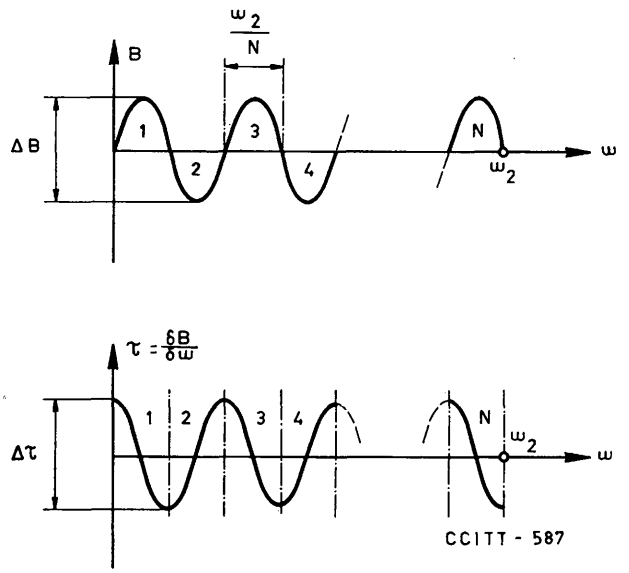
Evaluation of typical phase and attenuation curves

The approximate formulae (30) to (32) and, when necessary, the exact expressions (20) to (29) have been applied to evaluate the following cases :

— A "gap" in the baseband transmission (Figure 9)

— Variation in the baseband of A, B, τ of the type $\Delta \cdot (\omega/\omega_2)^\eta$ (Figure 10)

— Ripple in the baseband of B and τ (Figure 11)



N complete half-cycles

$$B_e = \frac{\Delta B}{2\sqrt{2}} \sqrt{1 - \frac{6}{(\Pi N)^2}} = \Delta B \cdot k_B$$

respectively

$$B_e = \frac{\Delta \tau}{\tau_2} \frac{1}{N \cdot 2\sqrt{2}} \sqrt{1 - \frac{6}{(\Pi N)^2}} \cdot \frac{180°}{\Pi} = \frac{\Delta \tau}{\tau_2} \cdot k_\tau$$

N	k_B	K_τ
1	0.177	12.°7
2	0.261	10.°1
3	0.353	6.°8
4	0.353	5.°1
5	0.353	4.°1
6	0.353	3.°4

FIGURE 11. — **Constants for computation of the r.m.s. value B_e for sinusoidal ripple in B and τ**

— Superposition of linear and parabolic delay variation with D.S.B.S.C.-modulation (Figure 12) and with S.S.B.S.C.-modulation (Figures 13 and 14)

The diagrams speak largely for themselves. A few facts may be worth some additional comments, however:

— In the case of attenuation curves of the type $A = \Delta A \left(\frac{\omega}{\omega_2}\right)^n$, A_e depends to a first approximation *only* on ΔA, and not very much on n.

— In the case of delay curves of the type $\tau = \Delta\tau\left(\frac{\omega}{\omega_2}\right)^n$, B_e depends *strongly* on n in addition to $\Delta\tau$. (For instance, to give the same signal-to-noise ratio $\Delta\tau$ can be twice as large for $n = 6$ as for $n = 3$.)

— For the same signal-to-noise ratio, the allowable amplitude of delay ripple increases directly with the number of ripple periods in the band. It is quite *meaningless* to try to define the quality of a delay equalization in terms of the delay *spread* in the band without considering the *form* of the delay curve.

From Figures 12, 13 and 14, it can be seen how simultaneous slope (linear) and sag (parabolic) delay distortion affects the pulse fidelity index for double and single sideband systems respectively.

(For simplicity, single sideband systems are considered instead of vestigial systems.)

The S.S.B. system can of course transmit data twice as fast as the D.S.B. value for the same available line frequency band. (To permit a quick comparison between D.S.B. and S.S.B. for the same amounts of slope τ' and sag τ'', the parameters in Figures 12, 13 and 14 are given as τ'/τ_2; τ''/τ_2; $\tau'/2\tau_2$; $\tau''/2\tau_2$ respectively). It can be seen that the price to pay for the increased speed with S.S.B. is of course an increased sensitivity to delay distortion, especially when the phase of the demodulating carrier is not optimum.

Noise impairment

In the preceding paragraphs a data transmission disturbance has been characterized by its (baseband) *signal-to-noise ratio*, A_D nepers. It can also be specified in terms of "*noise impairment*" A_I nepers. A_I is, however, not a unique property but depends on what error rate is tolerated.

Suppose that for a certain specified error rate, the data system can stand white noise A_N nepers below the signal level. When some transmission disturbance with signal-to-noise ratio A_D is added, the noise level must be decreased by an amount A_I nepers to give the same error rate. We get the following relation between A_D, A_I, A_N:

$$e^{-2A_N} = e^{-2(A_N + A_I)} + e^{-2A_D} \tag{37}$$

i.e.,

$$A_I = \frac{1}{2} \ln \left\{ 1 - e^{-2(A_D - A_N)} \right\}^{-1} \tag{38}$$

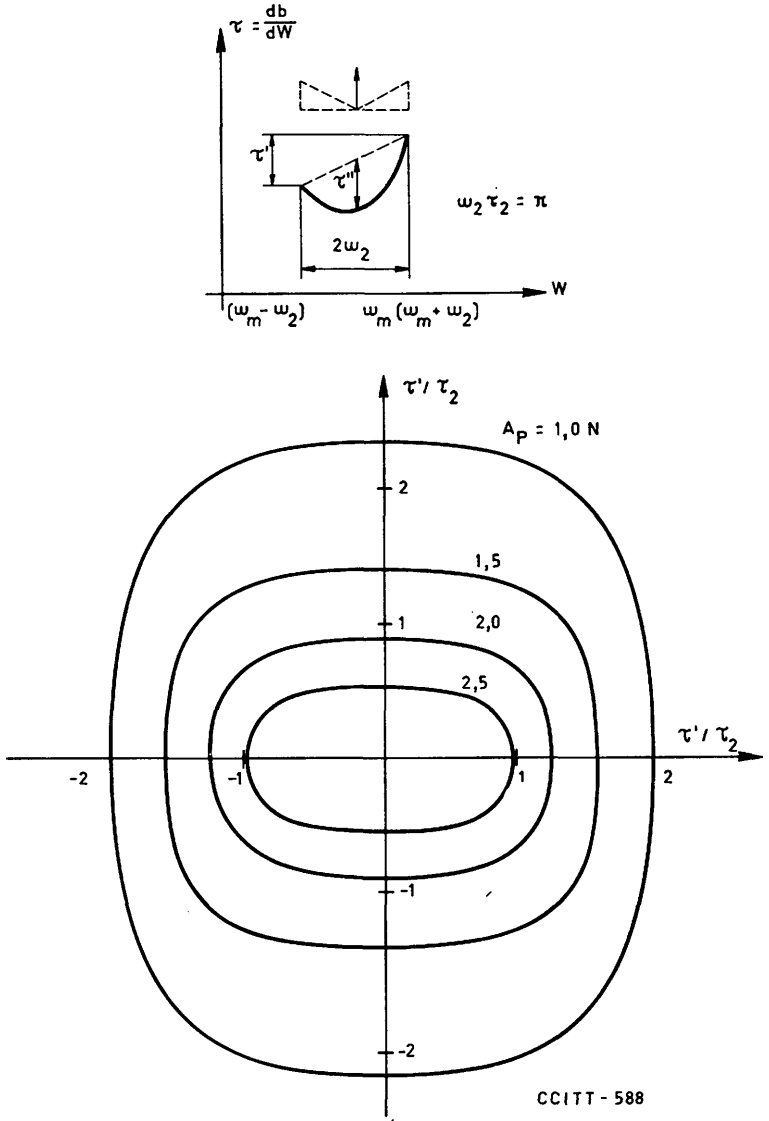


FIGURE 12. — **Pulse fidelity index A_D for the case of slope (linear) and sag (parabolic) delay distortion**

Double sideband suppressed carrier modulation (phase shift modulation)

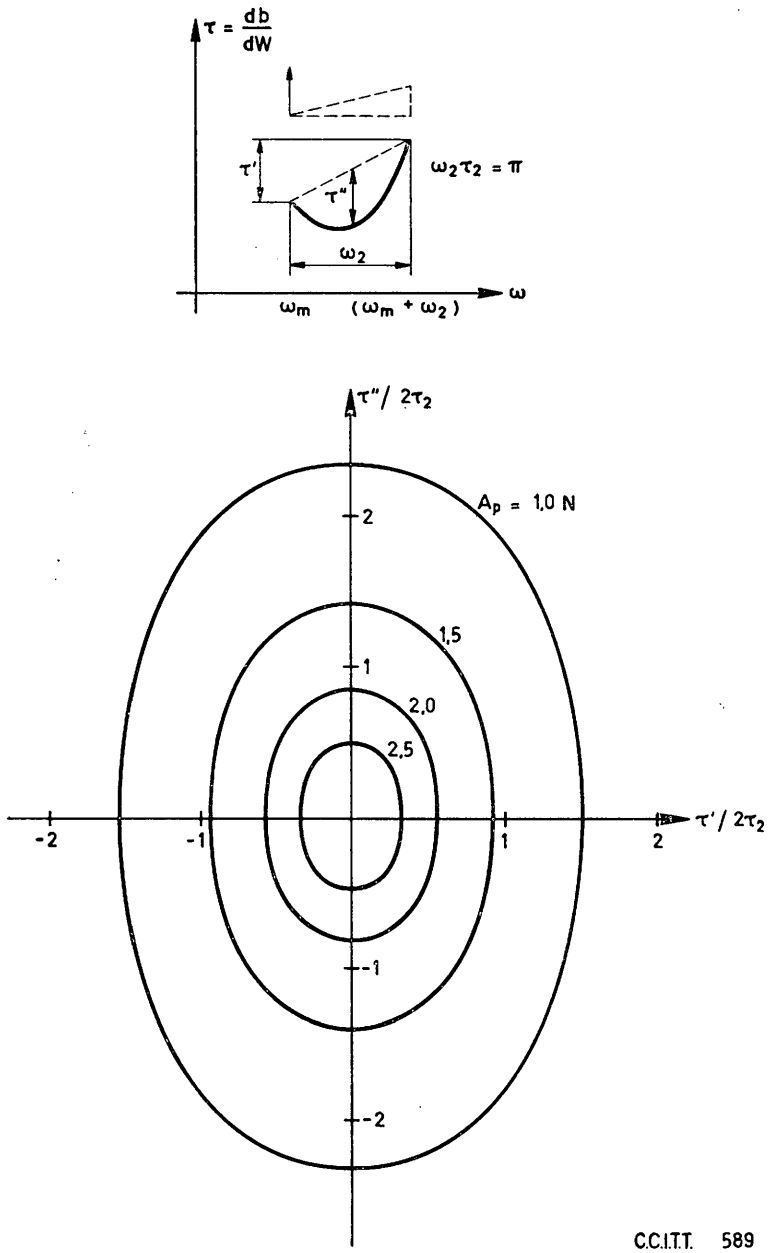


FIGURE 13. — **Pulse fidelity index A_p for the case of slope (linear) and sag (parabolic) delay distortion**

Single sideband, suppressed carrier modulation. The demodulating carrier at the receiver is adjusted for *optimum phase.*

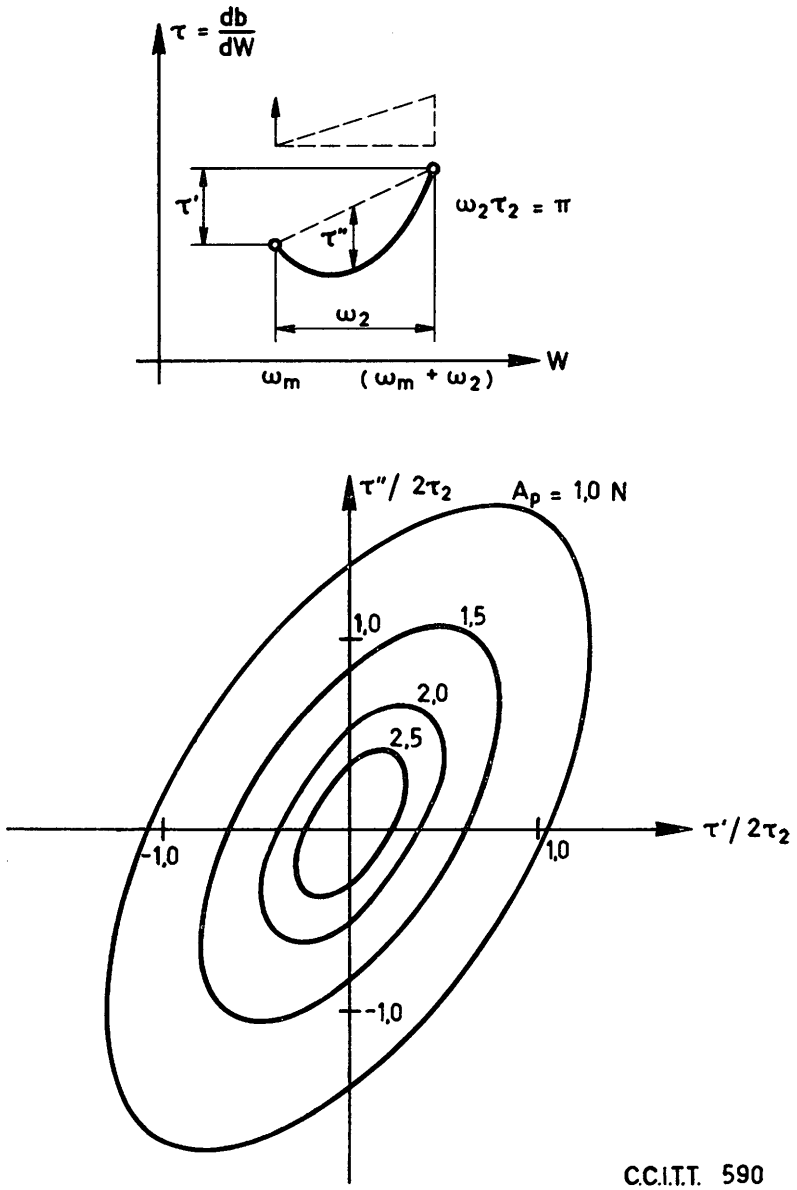


FIGURE 14. — **Pulse fidelity index A_P in the case of slope (linear) and sag (parabolic) delay distortion**

Single sideband, suppressed carrier modulation. The demodulating carrier at the receiver has the *same phase* as a virtual carrier transmitted at the frequency $W = \omega_m$.

Conversely,

$$A_D = A_N + \frac{1}{2} \ln \left\{ 1 - e^{-2A_I} \right\}^{-1} \tag{39}$$

For A_I to have a meaning, the disturbance signal-to-noise ratio must of course be greater than the reference value:

$$A_D > A_N \tag{40}$$

General recommendations for equalization

A permissible error rate of 10^{-3} means a baseband signal-to-noise ratio A_D of at least 1.1 or 1.2 nepers, and 10^{-6} means about 1.6 nepers. This corresponds to a non-linear phase r.m.s.-value B_e of about 19°-17° or an attenuation r.m.s. deviation A_e of about 0.33-0.30 neper respectively, if those were the *only* transmission defects. In practice, some margins are needed. It might be appropriate to set the limit on permissible attenuation and delay distortion as a *pulse fidelity index* $A_p \geqq 2.3$ *nepers (20 db)*. This corresponds (in the baseband) to

$B_e \leqq 5.^\circ 7$ (delay distortion only)

$A_e \leqq 0.10$ neper (attenuation distortion only)

In the case of a combination of delay and attenuation distortion, compare with Figure 8.

If the attenuation follows a uniformly ascending or descending curve, the maximum deviation limit becomes approximately:

$$\Delta A \leqq 0.35 \text{ neper for } A_p \geqq 2.3 \text{ nepers}$$

(Compare with Figure 10.)

If the type of data system is known, the delay and attenuation of the transmission medium can always be transformed to the baseband frequency range, and the equalization readily evaluated. Sometimes it is necessary to equalize the *delay distortion* of a transmission medium "in the best way" within a certain frequency range for an unspecified data system. In that case, the following procedure is proposed for assessing the equalization quality in a useful and general way:

Suppose the delay $\tau(W)$ is given in accordance with Figure 15a. Integrate this curve to give the phase-shift $B(W)$ (Figure 15b).

Draw a straight line $B_s(W) = W\hat{\tau} + B_0$ based on the method of least squares, and integrate the square of the difference $B(W) - B_s(W)$. This gives the appropriate mean square deviation of non-linear phase:

$$B_e^2 = \frac{1}{\omega_2} \int_{\omega_m}^{\omega_m+\omega_2} (B(W) - \omega\,\hat{\tau} - B_0)^2 \, d\omega \tag{41}$$

The constants $\hat{\tau}$ and B_0 are found to be:

$$\left.\begin{aligned} \hat{\tau} &= \frac{12}{\omega_2^3} \int_{\omega_m}^{\omega_m+\omega_2} WB(W)\,dW - \frac{\sigma}{\omega_2^2} \int_{\omega_m}^{\omega_m+\omega_2} B(W)\,dW \\ B_0 &= \frac{4}{\omega_2} \int_{\omega_m}^{\omega_m+\omega_2} B(W)\,dW - \frac{\sigma}{\omega_2^2} \int_{\omega_m}^{\omega_m+\omega_2} WB(W)\,dW \end{aligned}\right\} \tag{42}$$

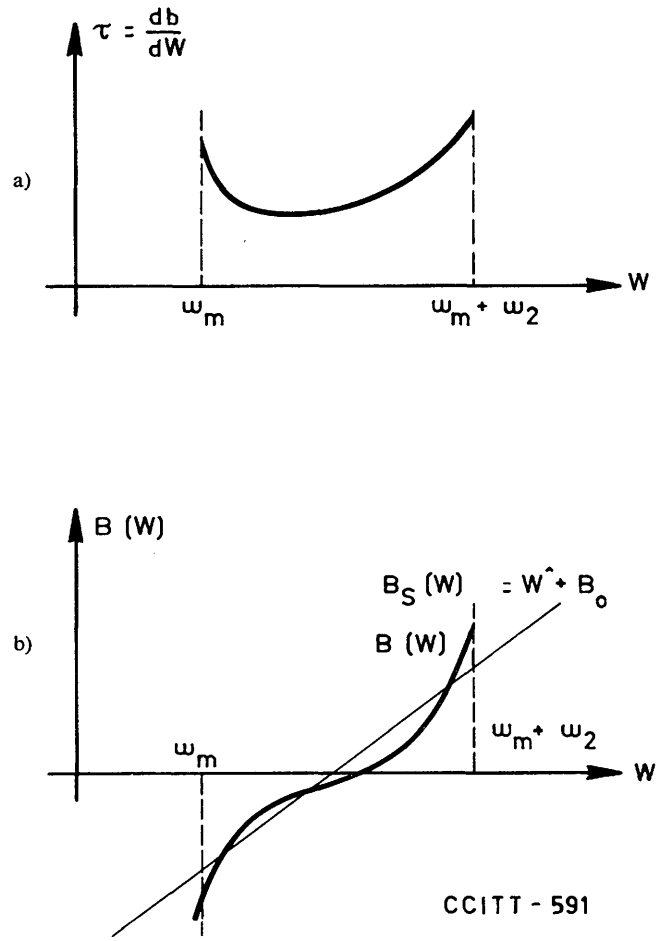


FIGURE 15. — **Delay and phase-shift. Evaluation of *Be* in a general case**

APPENDIX

Idealized versus practical symbol pulses

Practical data systems do not have the idealized symbol pulse with a perfectly rectangular frequency spectrum which has been assumed in the previous calculations. The practical symbol pulse is still symmetrical around $t = 0$ as the idealized one, but its frequency spectrum is more rounded-off at the upper edge.

We shall now estimate the error to be expected because an idealized pulse was used instead of the real one in the computation of inter-symbol interference.

The rounded-off spectrum is presumed to be *symmetrical* around $\omega = \omega_2$ in the range $\omega_2 \pm \Delta\omega$ in accordance with Figure 1A.

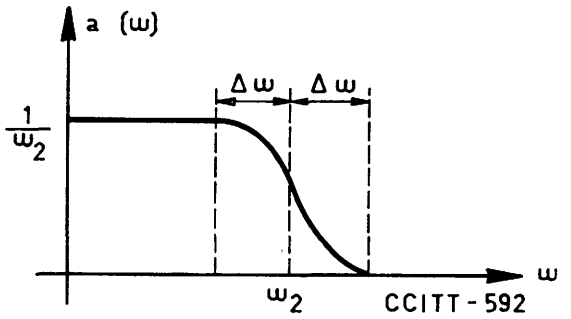


FIGURE 1A. — **A typical frequency spectrum for a practical symbol pulse**

I.e., in this range is :

$$\left.\begin{aligned} a(\omega) &= \frac{1}{2\omega_2} + \Delta a(W) \text{ for } \omega < \omega_2 \\ a(\omega) &= \frac{1}{2\omega_2} - \Delta a(W) \text{ for } \omega > \omega_2 \\ W &= \omega - \omega_2 \end{aligned}\right\} \quad (1A)$$

As the pulse was assumed to be symmetrical around $t = 0$, its Fourier integral contains only cosine components. After some calculations we get the following expression for the practical symbol pulse :

$$S_p(t) = \int_0^\infty a(\omega) \cos \omega t \mathrm{d}\omega = \frac{\sin \omega_2 t}{\omega_2 t} = 2 \sin \omega_2 t \int_0^{\Delta\omega} \Delta a(\omega) \sin W t \mathrm{d}W \quad (2A)$$

The first term is the idealized, normal symbol pulse. The last term thus represents the difference $\Delta S(t)$ between idealized and actual pulse. It can be seen from equation (2A), that

$$\left.\begin{aligned} S_p(n\tau_2) &= 0 \text{ for } n \neq 0 \\ (\tau_2 &= \Pi/\omega_2) \end{aligned}\right\} \quad (3A)$$

as was the case with the idealized, normal pulse. Thus, if there is *no* non-linear phase and attenuation distortion, there will be no symbol interference either.

With non-linear phase distortion present we need merely consider its effect on the "difference pulse" $\Delta S(t)$. This contains major frequency components only in a narrow band around $\omega = \omega_2$, where we assume that the phase can be written approximately as :

$$B(\omega) \approx B' + (\omega - \omega_2) \cdot \tau' \quad (4A)$$

reckoned from the reference line $B_0 = \omega\hat{\tau}$ ($\hat{\tau}$ = optimum sampling instant).

At the nth sampling instant $t_n = \hat{\tau} + n\tau_2$ the amplitude of the difference pulse becomes

$$\Delta S(t_n) = \pm 2 \sin B' \int_0^{\Delta\omega} \Delta a(\omega) \sin W(\tau' + t_n) \mathrm{d}W \quad (5A)$$

The mean-square value of this disturbance over all sampling instants is thus

$$\sigma_\Delta^2 = \sum_{\eta=\infty}^{\infty} \Delta\, S^2\,(t_n) \tag{6A}$$

which also can be written as :

$$\sigma_\Delta^2 = 4\, \sin^2 B' \cdot \omega_2 \int_0^{\Delta\omega} \Delta a^2\,(W)\, \mathrm{d}W \tag{7A}$$

If, for instance, $\Delta a(W)$ is a parabola in accordance with Figure 1A, after some calculations we get

$$\sigma_\Delta^2 = 0.20 \cdot \frac{\Delta\omega}{\omega_2} \cdot \sin^2 B' \tag{8A}$$

A numerical example :

With sinusoidal ripple having an r.m.s.-value B_e, we get :

$$\sin^2 B' \leqslant \sin^2 \sqrt{2} \cdot B_e \approx 2 \cdot B_e^2$$

Assume $\Delta\omega/\omega_2 = 0.25$, and we get :

$$\sigma_\Delta^2 \leqslant 0.10 \cdot B_e^2$$

Instead of the value B_e one should accordingly use

$$B'_e = \sqrt{B_e^2 + \sigma_\Delta^2} \approx 1.05 \cdot B_e$$

Thus using the rectangular spectrum as an approximation for the rounded-off frequency spectrum gives an insignificant error.

Supplement No. 79

SWEDEN — *(Extract from contribution COM Sp. A/No. 1 — June 1961)*

EFFECT OF WHITE NOISE ON DATA TRANSMISSION

White noise performance with ideal line

The optimum error probability for ideal systems has been calculated by several theoreticians as a function of the signal-to-noise ratio. The transfer function of the line has generally been assumed to be ideal, which implies a_L being a constant and $b_L\,(\omega)$ being a straight line. In the ideal case, with phase reference not being perturbed by noise, the characteristic of PSM systems can be written as

$$p_e = \frac{1}{\sqrt{2\Pi}} \int_{-\infty}^{-\sqrt{2R_{mr}}} \exp\,(-\frac{1}{2}\, t^2)\, \mathrm{d}t \tag{1}$$

where R_{mr} is the normalized signal-to-noise ratio at input of receiver in terms of average signal power and average noise power in flat band of width equal to the modulation rate. An equivalent definition of R_{mr} sometimes used is

$$R_{mr} = E/N_0$$

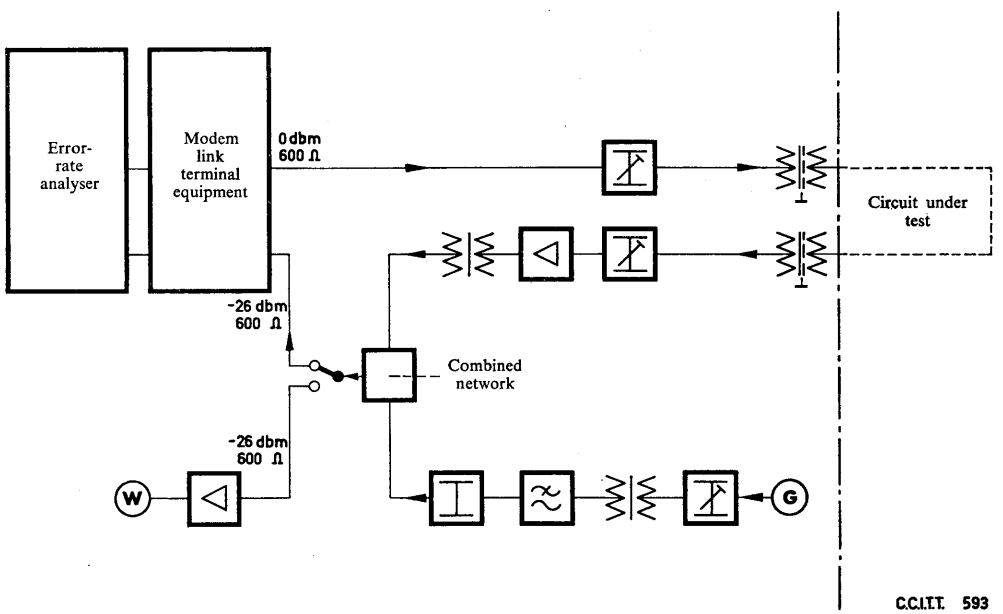
where E is the energy per received element and N_0 is the noise power density per unit bandwidth. The relationship p_e (R_{mr}) according to equation (1) has been represented in Figure 5. The curve obtained in this manner is of special interest since there is no question of its validity. Thus, this curve may be used as a reference for the comparison of other theoretical and measured curves.

The optimum characteristic as obtained from equation (1) cannot be achieved in practice owing to unavoidable disturbances of the phase reference. The error probability of differential coherent PSM systems, also represented in Figure 5, can be written as :

$$p_e = \frac{1}{2} \exp (- R_{mr}) \qquad (2)$$

In this case, the phase of the received VF signals during each unit interval is compared with the phase during the preceding unit interval.

As can be seen from Figure 5, the error probability in this latter case is less advantageous, the characteristic curve being moved 1 to 3 db towards higher values of R_{mr}.



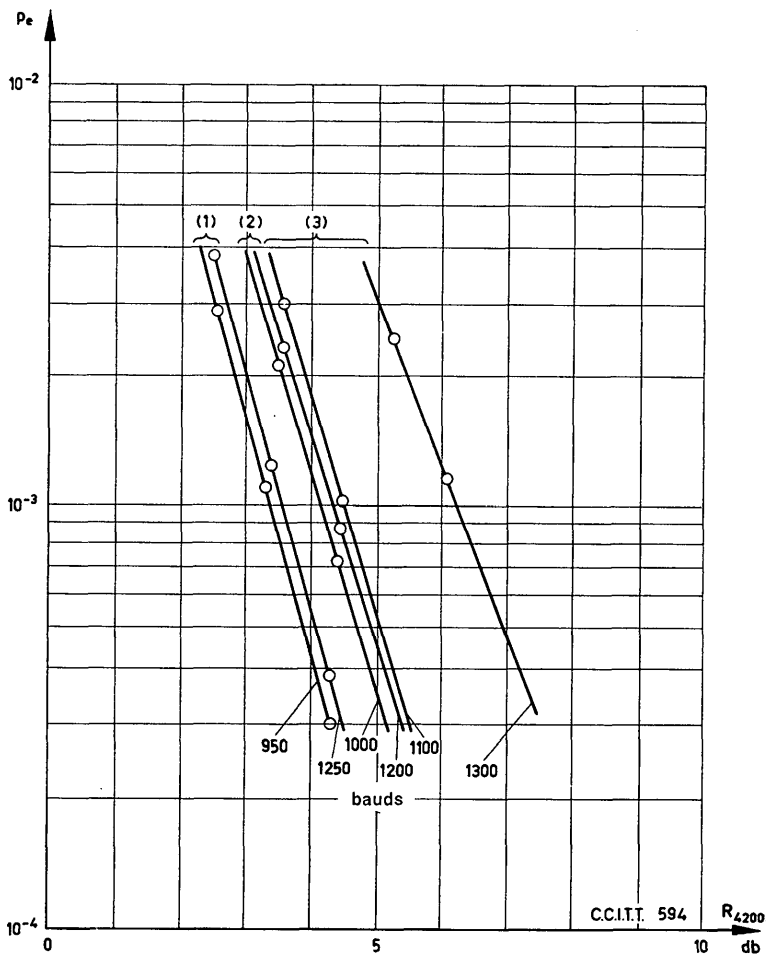
G = White noise generator 0.03 kc/s-20 kc/s
W = Thermocouple instrument for measuring a power of 1 mW

FIGURE 1. — **Data transmission measuring set-up with white noise injection**

Due to mathematical impediments, it is very difficult to calculate the optimum error probability curve for FSM as FSM is a non-linear modulation method. There are not many theoreticians who have tried an extensive analysis of this problem. However, many have investigated the two-tone case under the heading of frequency shift keying, which is a modulation method based on two on-off amplitude modulated channels, the tone being sent alternately on the two. This modulation method exhibits certain advantages only when the transmission medium is subject to selective fading and should not be mixed up with ordinary FSM, applying limiter, discriminator and post-detection low-pass filter.

. . . . . .

4. *Measurements of FSM modem link terminals connected in a local loop*



Frequency shift : (1) 880 c/s (2) 960 c/s (3) 600 c/s

FIGURE 2. — **Element error probability p_e versus signal-to-noise ratio R_{4200} with various variables as parameters**

4.1 *Optimum relation between shift and channel bandwidth*

At an early stage, a bread-board model of the modem link terminal was constructed and equipped with exchangeable sets of filters of different bandwidths. In order to explore what shifts should be chosen, a series of measurements were undertaken. In Figure 2, some results of the basic measurements have been plotted in order to illustrate the principles. Using one particular set of filters, the element error probability has been measured at two or three different signal-to-noise ratios for each combination of the two parameters-shift and modulation rate. Through each set of the two of three measured points a curve has been drawn and at $P_e = 10^{-3}$ the signal-to-noise ratio R_{4200} has been read from the curve. After normalization to 3000-c/s noise bandwidth, the slightly modified ratio R_{3000} has been inserted on a graph as a function

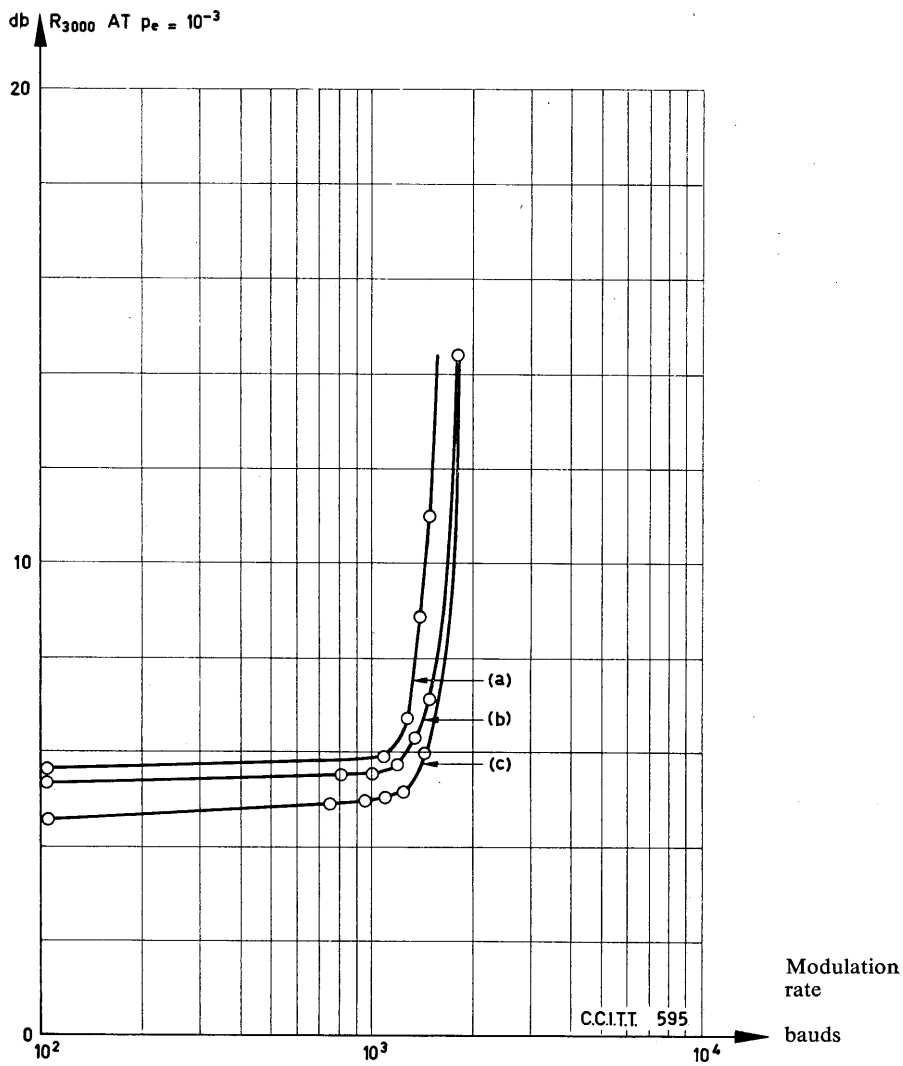


FIGURE 3. — **White noise performance versus modulation rate at various shifts**

of the modulation rate, as appears on Figure 3. The required signal-to-noise ratio (at $P_e = 10^{-3}$) is found to be almost independent of the modulation rate below 1100 bauds, the optimum shift being 880 c/s ($\pm$ 440 c/s).

The filters have been designed according to the same principles as those applied in multi-channel VF telegraph systems, the bandwidths being increased by certain factors. In the case of standard 120-c/s spaced channels, the result reported above would correspond to a modulation rate of 75 bauds and an optimum shift of 56.5 c/s. It is interesting to note that the shift of 56.5 c/s, found to be an optimum, is only a few per cent below the shift of 60 c/s ($\pm$ 30 c/s) that has been recommended by the C.C.I.T.T. for FSM VF telegraph channels of this particular bandwidth. Hence, these standard channels provide a very efficient medium for synchronous transmission.

4.2 *Optimum relation between modulation rate and channel bandwidth*

As has been seen from Figure 3, any modulation rate below a certain figure requires about the same signal-to-noise ratio in order to keep the error probability below a specified level. However, at low modulation rates, the bandwidth used does not give the best possible performance, since by choosing a narrower receive filter a reduction in noise power passed to the detector is possible. The following question then arises : which is the optimum channel bandwidth for a particular modulation rate ? The answer has been found by normalization of the signal-to-noise ratios obtained, as shown in the previous paragraph to a noise bandwidth equal to the modulation rate. By means of a graphic representation, it has been found that the system exhibits a pronounced minimum signal-to-noise ratio, R_{mr}, with a certain relation between modulation rate and channel bandwidth. The curves in Figure 4 illustrate how this ratio varies, the modulation rate over

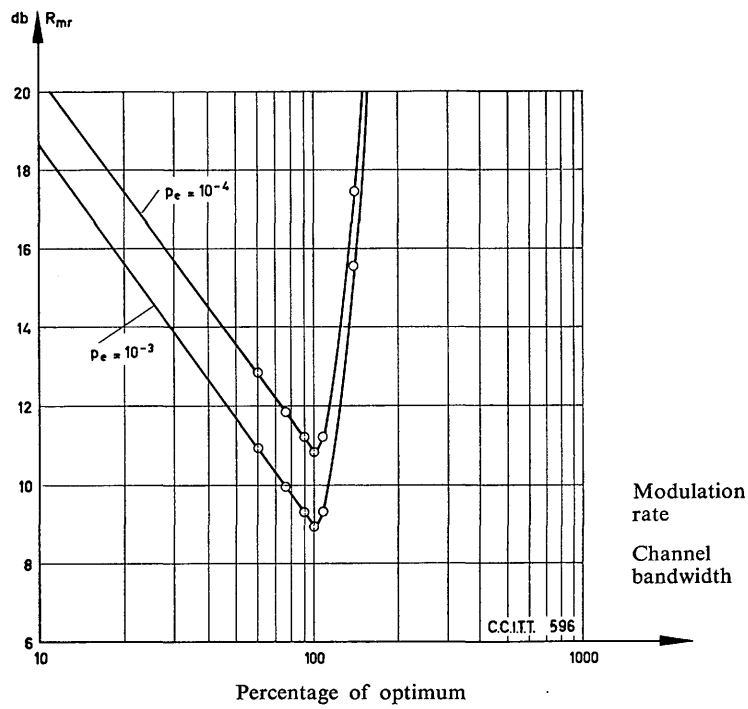
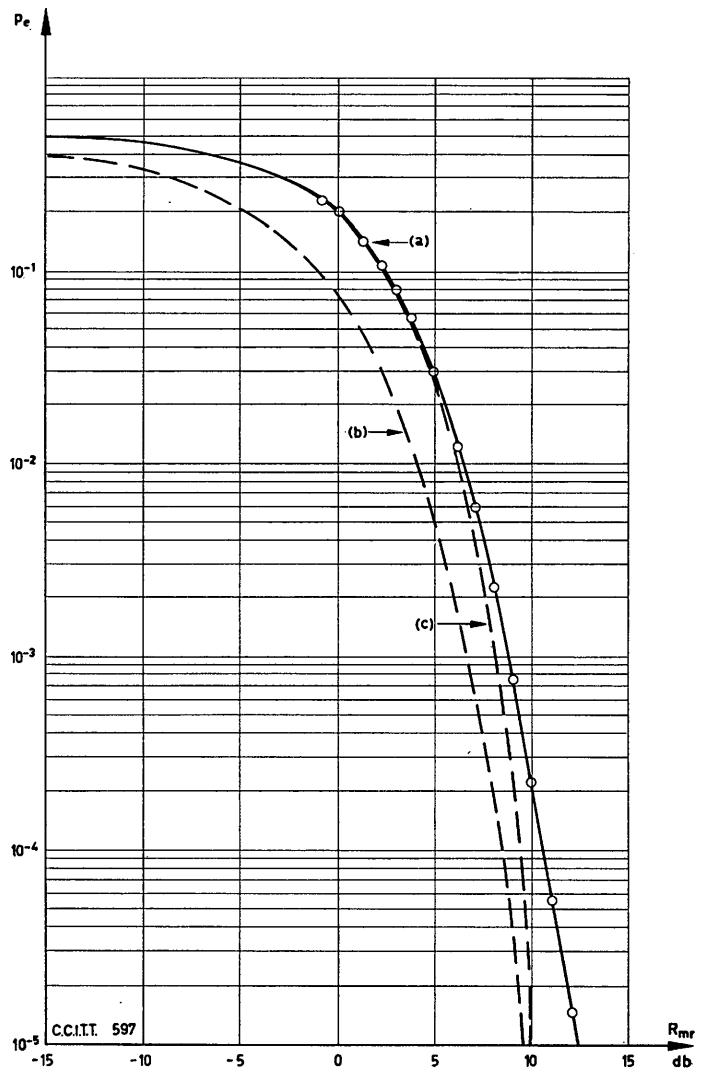


FIGURE 4. — **Normalized white noise performance versus percentage of optimum relation between modulation rate and channel bandwidth**



(a) = FSM, measured
(b) = Ideal phase reference PSM theoretical optimum
(c) = Differential coherent PSM, theoretical optimum

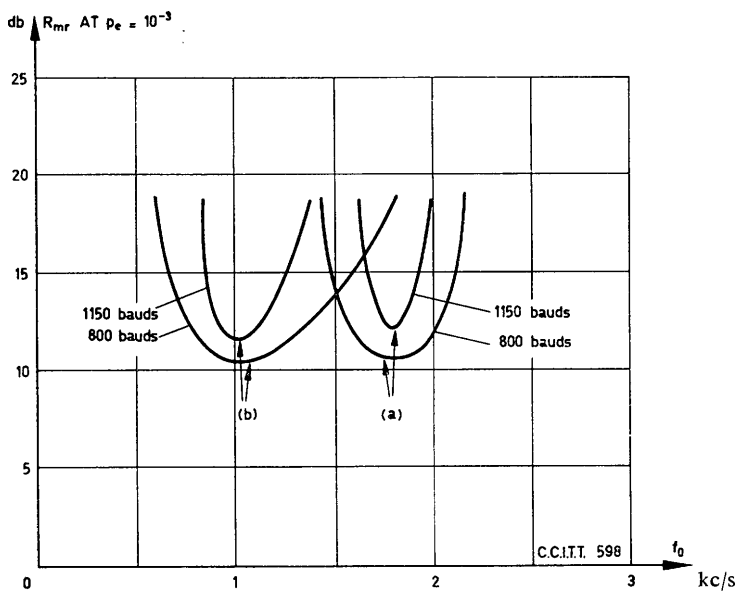
FIGURE 5. — **Element error probability p_e versus normalized signal-to-noise ratio R_{mr}**

channel bandwidth being used as a variable on the abscissa and expressed as a percentage of the optimum. For the particular bandwidth related to above, this optimum has been obtained at 1260 bauds, which in the case of standard 120-c/s spaced channels corresponds to 81 bauds.

The results reported above have been obtained from a system designed without a send filter. However, in multi-channel systems, send filters have to be used and in such cases the optimum signal-to-noise ratio may be obtained at a slightly lower modulation rate.

4.3 *Optimum white noise performance*

The optimum value of the signal-to-noise ratio R_{mr} obtained in Figure 4 is of special interest. Maintaining the shift and modulation rate thus found to give optimum performance, the error probability has been measured against the ratio R_{mr}. For comparison, the result has been shown in Figure 5 with two curves representing theoretical optimum characteristics for phase shift modulation. The curve to the left relates to PSM in the hypothetical case when an ideal phase reference is provided, and the curve in the middle relates to PSM when differential coherent detection is used. A comparison shows that the measured FSM characteristic happens to follow the theoretical differential coherent PSM curve fairly well at high values of error probability, and that it is only 2 to 3 db inferior to the theoretical ideal phase reference PSM curve. The measured FSM characteristic has been obtained by measurements in a system where the filters have been designed in a straightforward manner, but employing a special type of limiter. By insertion of a pre-emphasizing type of send filter and by modification of the post-detection low-pass filtering, it will no doubt be possible to move the FSM curve a few db to the left.



(a) = Measured on four cascade connected carrier circuits
(b) = Measured on the phantom circuit of 93 km loaded multiple twin cable ($f_1 = 3500$ c/s)

FIGURE 6. — **Normalized white noise performance versus channel mid-frequency f_0**

5. *Measurements on FSM modem links including a variety of loop-connected lines*

5.1 *White noise performance at various modulation rates and channel mid-frequencies*

The measuring set-up has been used for measurements on a variety of connections. White noise performance at various modulation rates and channel mid-frequencies has been measured, with and without equalization. The result will be the subject of a separate report.

However, some typical results of measurements of this kind will be given here in order to illustrate the difference in characteristics obtained on loaded cable circuits and carrier circuits. From Figure 6, it can be seen that the optimum channel mid-frequency of a particular loaded circuit is about 1000 c/s while a typical carrier circuit gives best performance at 1800 c/s. It should be noticed that there is no common channel mid-frequency usable for both these circuits except at the lower speed.

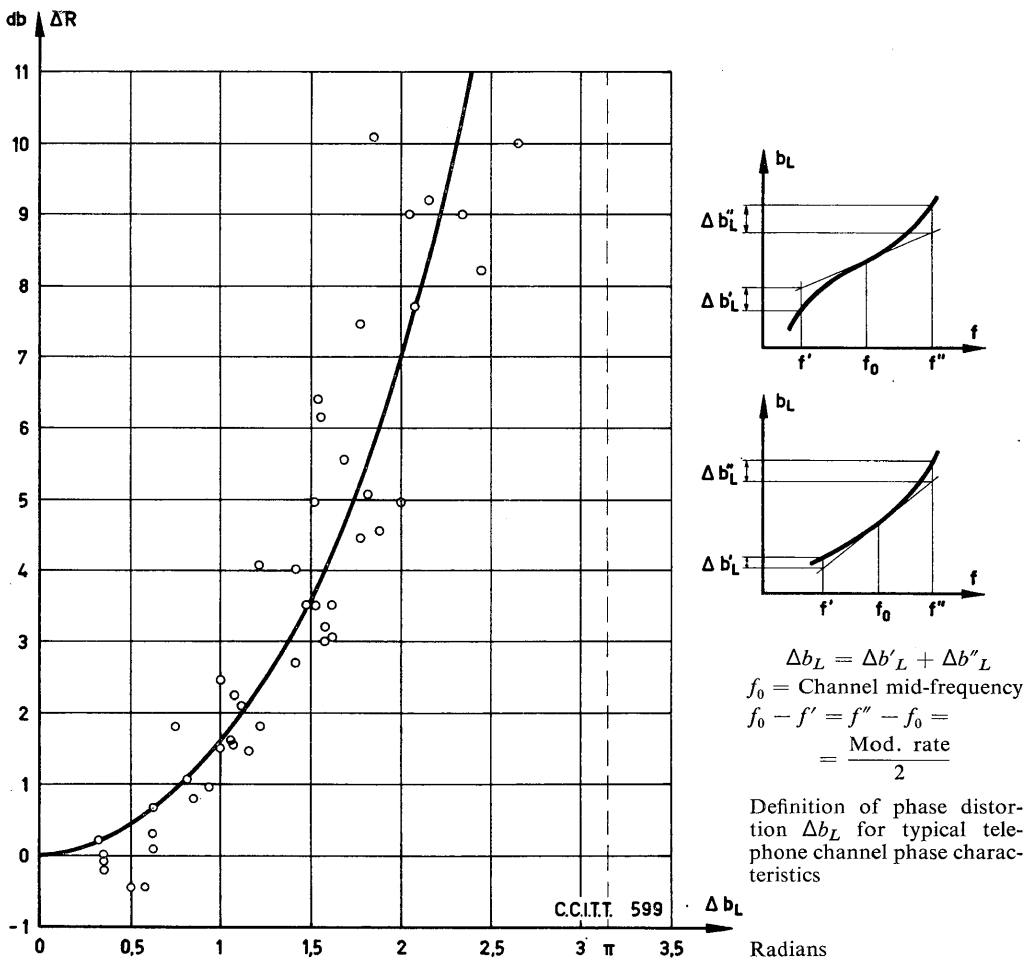


FIGURE 7. — **Impairment of white noise performance ΔR versus phase distortion Δb_L**

5.2 *Impairment of white noise performance due to non-ideal phase characteristic*

A large number of measured characteristics have been treated in order to find a relationship between the white noise performance and the amount of phase non-linearities of the transmission medium. Graphic illustrations in Figure 7 define two angles, $\Delta b'_L$ and $\Delta b''_L$, whose sum has been taken as an approximate measure of the phase distortion.

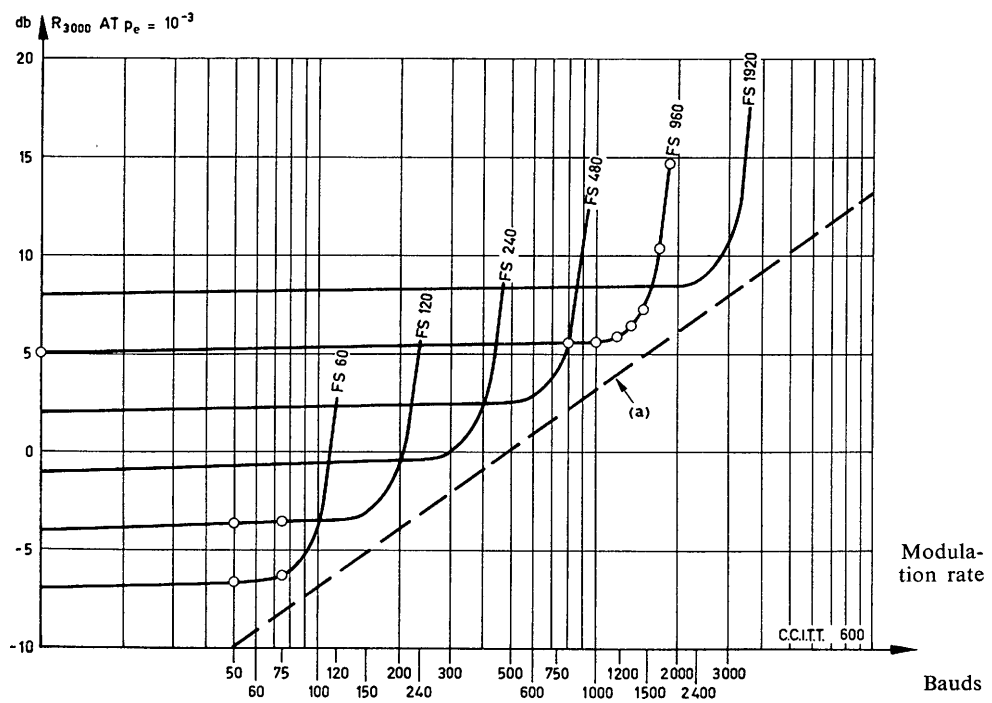
The main graph gives the relationship between the impairment of white noise performance and the so-defined phase distortion. More than 40 measured points have been worked out and a mean curve drawn. The tendency of the curve indicates that the impairment will go to infinity as the phase distortion approaches π radiants.

The same basic measurements have also been treated in a different manner in an attempt to obtain a relationship between noise performance and group delay. In this case, the spread appeared to be greater and it was not possible to find any significant mean curve. It is believed that group delay does not correspond to a physical reality in this case where differences in delay between various parts of the same spectrum, comprising one single channel, are considered.

It seems very difficult to find a theoretical relationship between white noise performance and phase distortion in the generalized case. However, some preliminary considerations have led to the following suggestion, which has been found to be in considerable agreement with the experimental result.

$$\Delta R = -20^{10} \log \left[\frac{\cos \Delta b'_L + \cos \Delta b''_L}{2} \right] \text{db}$$

. . . . . .



(a) = Differential coherent PSM, theoretical optimum

FIGURE 9. — **White noise performance versus modulation rate for various modem channel types**

Supplement No. 80

AUSTRALIA — *(Extract from contribution COM Sp. A/No. 99 — November 1963)*

NOISE MEASUREMENT

1. *Types of noise*

From our experience of the interference caused to carrier channels under operational conditions, the following types of noise are considered important when assessing the over-all noise performance of a telephone circuit :

a) Noise of very low frequencies, for instance due to ripple on the d.c. power supply. Such low frequency noise may prevent the proper operation of the transmission equipment used to provide the trunk channels, notably equipment associated with pilot channels and out-of-band signalling channels.

b) High frequency noise, say about 5 kc/s, which by modulation and demodulation processes in the transmission equipment may be translated into the voice-frequency band and finally appear as audible noise.

c) Noise at voice frequencies, which may interfere with any type of communication over the channel.

d) Impulse noise. This type of noise interferes with the transmission of high-speed signals.

2. *Measurement*

Individual measurement of each type of noise could be undertaken and the over-all noise performance assessed from these measurements. This Administration has conducted such measurements on a large number of circuits over a period of years and is of the opinion that, by using a suitable instrument, the over-all noise performance can be assessed in a single measurement. The type of instrument proposed for consideration by the Study Group has been arrived at from a study of these measurements. The proposed measuring instrument would be a peak-reading voltmeter similar to the United Kingdom peak programme meter (C.C.I.T.T. *Red Book*, Vol. III, Annex 49, p. 429) having a weighting time constant of 0.25 ms instead of 2.5 ms. A time constant of 0.25 ms has been chosen as half the duration of one bit of a 2000-baud data signal. The voltmeter would be used in conjunction with the following weighting curve :

0 — 150 c/s — 29.0 db,

150 — 4700 c/s, in accordance with the 1951 table of weights for the C.C.I.T.T. psophometer,

above 4700 c/s — 29.4 db.

Note. — The C.C.I.T.T. 1951 weighting figures are 29.0 db at 150 c/s and 29.4 db at 4700 c/s.

3. *Use*

The proposed method of measuring would be used for assessing the noise performance of circuits which may be extended or tandem switched to carrier derived channels. The proposal to use this type of instrument does not imply that the present C.C.I.T.T. psophometer should cease to be used for the measurement of noise performance apparent to subscribers ; indeed the weighting curve proposed for the new instruments has been chosen to agree as far as possible

with the C.C.I.T.T. weighting curve to allow the maximum amount of correlation between readings on the two instruments.

It may be of interest to note that the proposed instrument would appear to facilitate the use of recording instruments since the time constants are controlled in the electrical section of the instrument rather than in the mechanical meter movement as in the case of the psophometer. These recordings could be used for statistical analysis along the lines of C.C.I.T.T. Recommendation G.222.

4. *Noise objectives*

The proposed noise objectives when using the noise-measuring instrument described would be those currently recommended as objectives when using the C.C.I.T.T. psophometer.

Supplement No. 81

M.I.T. (UNITED STATES OF AMERICA) — *(Extract from contribution COM Sp. A/No. 13 — September 1961)*

STANDARD PERFORMANCE TESTS FOR DATA MODEMS

Summary

The great variability in field conditions under which data transmission modems must operate makes it highly desirable to have reproducible laboratory tests which, while not reproducing field conditions, will indicate the kind of performance that can be expected under field conditions. Laboratory tests with additive white Gaussian noise are easy to instrument and interpret, albeit such noise is not typical of field conditions. The laboratory simulation of impulse noise is for the present dependent on the availability of a library of recorded noise tapes ; but, given such a tape, it is possible to specify reproducible testing procedures. Other tests are suggested to check the sensitivity of a modem to phase crawl and other transmission distortions, to the transient effects of drop-out, and to certain gross dependence of modem performance on the transmitted data format.

Introduction

The past few years have seen an enormous growth in applications for the transmission of digital data at rates of the order of 1000 bits per second and up. Corresponding to these applications (both military and commercial) there have appeared an equally wide variety of data transmission systems, each of which is claimed to set a new milestone in reliability, speed, flexibility in operating under adverse conditions, or low cost.

It might be thought that at least with respect to performance there might exist data with which to resolve the conflicting claims ; on the contrary, there seem to exist performance data to support and disprove almost any claim. Part of these inconsistencies in experience with data transmission modems can be traced to the variability and statistical inhomogeneity of the circuits

over which they operate. But a large part of the difficulty in comparing data transmission schemes is that there exists no widely agreed-on standard set of performance tests for data transmission modems. The experimental conditions arbitrarily selected by one worker are incompletely described, inadequate in scope, or not reproducible by another worker; it is fortuitous that their results sometimes agree.

We propose here several possible standard tests for the performance of the modulation-demodulation terminal equipment used in "high-speed" data transmission systems. When one attempts to provide such standard tests, one faces two major kinds of problems. The first is an interface problem, in that there exists no universally accepted standards as to the kind of information that must be delivered *to* the data modems or of the kind of information which is delivered *by* these devices. The second problem relates to the simulation of conditions under which the devices must operate: either we must provide in the laboratory transmission conditions which are a reasonable cross section of those to be met in the field, or we must develop laboratory tests which, while themselves not necessarily typical of field conditions, will be indicative of the performance of the equipment in actual use.

With respect to the simulation of operating conditions, we choose the latter of the two possibilities. The actual conditions under which data transmission equipment must operate are so highly variable from time to time and circuit to circuit that it requires many hundreds of hours of testing over a wide variety of facilities to establish statistically meaningful conclusions about the performance of any one particular data transmission system. The Gaussian noise and impulse noise performance tests to be suggested below are not necessarily typical of conditions to be met in actual practice (indeed the additive noise in voice bandwidth circuits is rarely Gaussian in its amplitude distribution); but rather to provide easily instrumentable laboratory tests which experience has shown to be highly indicative of relative performance under actual field conditions.

With respect to the interface problem, it must be realized that the performance tests to be described below require the use of a word generator at the transmitting end and a corresponding word generator and match and error counting arrangement at the receiving end. To make such a system function, it is necessary that the transmitting data modem be able to accept from the word generator both data and some form of synchronizing information that indicates the start of a particular data sequence. At the receiver, the modem must be able to deliver to the match and error counting arrangement not only the data and the synchronization information but some indication of when the state of the output end indeed represents a given data or synchronizing symbol. In modem systems which have been designed to operate with word start, data, and timing information accepted and delivered at the two ends of the system, these requirements present no problem. Where a modem designer has undertaken to provide less than this, he must be able to provide at the receiver whatever arrangements are necessary to operate the match and error counter; and the performance of these arrangements must be considered as part of the performance of the data transmission modems themselves. When such intermediate arrangements are used, they should be sufficiently simple and clearly described so that the tests can be repeated by other persons with equipment readily (hopefully, commercially) available to them.

The standard of performance used in the tests to be enumerated is the average number of bits in error per bit of information transmitted. Because the disturbances to be introduced in the tests are statistically homogeneous, this standard of performance may be readily translated into terms of other statistical standards of performance. Other standards of performance, such as word error rate or error-free minutes per hour may be more meaningful in actual field use. One should not suppose that these field performance statistics can be deduced in any definite

fashion from the results of these laboratory measurements; for, as we remarked above, the object of the laboratory tests is not to simulate the field conditions, but rather to effect convenient quantitative comparison of those characteristics which contribute to good field performance.

The test procedures outlined below are in some respects arbitrary. They reflect, on the one hand, testing procedure that have been used in several independent laboratories for estimating the performance of data transmission systems; and, on the other hand, testing procedures that are used in related branches of the communication and radar art.

Basic test set-up

The basic set-up to be used in all the tests below is shown in Figure 1. The salient features of this arrangement are that the levels of the signal and of the noise to be added are controlled through calibrated attenuators which immediately precede a mixing pad. All necessary distortions and alterations to the transmitted signal are made ahead of this attenuator arrangement; nothing is to be inserted between the mixing pad and the data receiver input terminals. No adjustments are to be made in either the data transmitter or receiver during the course of any one series of tests. No tests are to be begun unless the set-up has a record of at least 24 hours of error-free operation with no inserted distortions and no added noise.

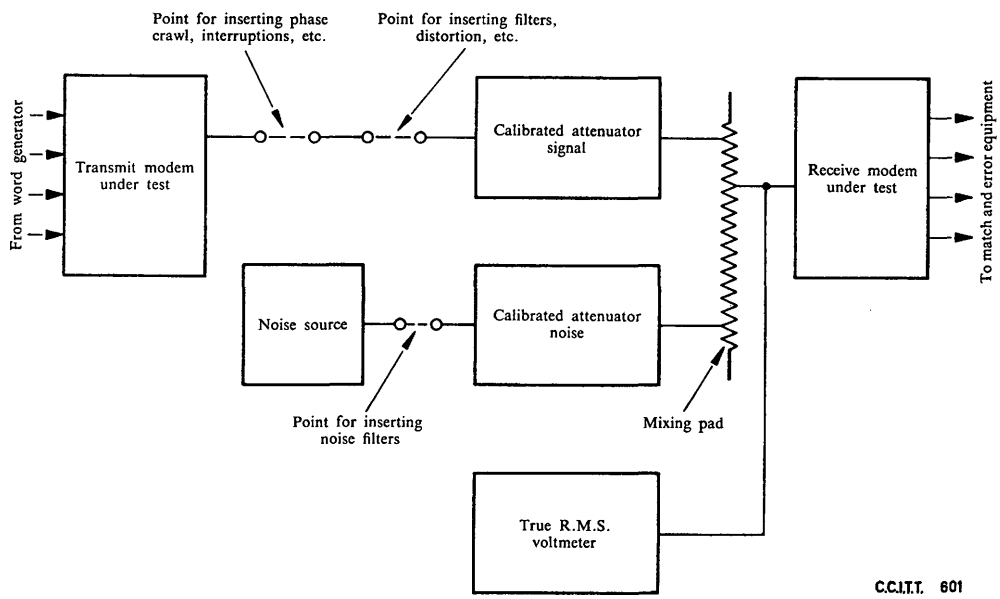


FIGURE 1. — **Data modem performance test set-up**

In testing many so-called "serial data" modems in which word-synchronizing data must be coded into the data stream, many workers have "looped-across" this word start information in a parallel channel, on the grounds of expediency, expense, etc. This is hardly a fair test of the

errors that may be incurred in actual operation, where no such parallel channel is provided [1]. In the tests outlined here, it is assumed that there are to be no interconnections between data transmitter and data receiver other than that provided by the artificial transmission system shown in the figure.

A basic test run consists of a measurement of bit error rate as a function of signal-to-noise ratio, generally with the signal level held fixed at some standard point. In general, three or four points measured one or two db apart around the region where the system makes one error a second will suffice to define a performance curve plotted on logarithmic co-ordinate or normal probability paper. For a data system which runs at rates of the order of a thousand or so bits per second, the minimum time interval during which a measurement is to be made is about ten minutes ; and in any event, enough measurement time should be allowed for any point that at least ten errors accumulate. This means that to demonstrate an error rate of one part in 10^6 in a 2000-bit-per-second system requires approximately an hour and a-half of measurement.

Gaussian noise tests

The additive noises which significantly disturb data transmission channels generally do not have either a Gaussian amplitude distribution or a flat power spectrum. Moreover, the disturbances which appear on many transmission facilities are capriciously variable in nature and have statistics which "settle down" (if they do so at all) only after hundreds of hours of data taking. Such noise sources, even if enough were known to simulate them in the laboratory, are clearly inappropriate for rapid bench-test evaluation. What is needed in the laboratory is a test noise source that is easily available, whose statistics are known, standardized, and homogeneous in time.

The laboratory instrument that best fulfils these last requirements is the white Gaussian noise generator : it is inexpensive, it is widely used in system tests, and its characteristics are well understood both experimentally and analytically ; moreover, it is usually possible to calculate how well a given kind of modulation scheme ought to perform when such noise is present [2]. Accordingly, there is a strong argument from convenience for comparing the operation of data modems in the presence of additive Gaussian white noise. Of course, the error rates obtained in such tests will be (for any given signal-to-noise ratio) only roughly related to those met in field operation, the field error rate generally being orders of magnitude larger. But it is a matter of experience that those systems which perform best in a Gaussian noise test tend to perform best in an impulse noise environment ; or, more precisely, if a system requires many more db of signal to reach a given standard of performance with Gaussian noise than theory says is necessary, then that system may make orders of magnitude more errors in the presence of impulse noise than a system which requires only a few more db than are theoretically necessary in the Gaussian environment. Figure 3 illustrates the situation. In Gaussian noise tests, white Gaussian noise (whose bandwidth is substantially greater than that of the transmission facility for which the data modem

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[1] According to J. MERCURIO, of the Mitre Corporation (private communication), there is about a 1-db improvement in the performance against Gaussian noise in the performance of modems which accept and deliver word start information when word start is "looped across" in a parallel, noiseless channel.

[2] See C. R. CAHN : "Performance of Digital Phase Modulation Communications Systems", *Trans. I.R.E. Comm. Syst.*, vol. 7, pp. 3-6, May 1959. Also E. BAGHDADY, Editor, *Lectures on Communication System Theory*, especially Chapter 11, "The Design of Signals".

is intended) is added to the signal. Signal-to-noise ratios can either be expressed in terms of average signal power to noise power in some standard bandwidth (often taken as 3 kc in voice channel measurements); or they can be normalized to a signal-energy-per-bit to noise-power-per-cycle ratio. (This latter ratio, designated E/N_0, is obtained by dividing the signal power by the bit rate and the noise power by the effective noise bandwidth.)

a) *Back-to-back noise test*

Here, no distorting filters are inserted between data transmitter and receiver. The signal attenuator is adjusted so that the signal level at the receiver is some standard level, say −15 dbm. The noise level is varied to obtain a performance curve. A typical curve of this sort is shown in Figure 2. For such a test, a theoretically optimum performance curve exists. The back-to-back Gaussian noise performance of the system may be specified either by giving the signal-to-noise ratio at some standard error rate such as 10^{-3} or 10^{-5}, or by comparing it with the theoretical signal-to-noise ratio for this standard error rate. For a 10^{-3} error rate, the theoretical E/N_0 of an FSK type system is 10 db; a phase-reversal type system should be 3 db better [1].

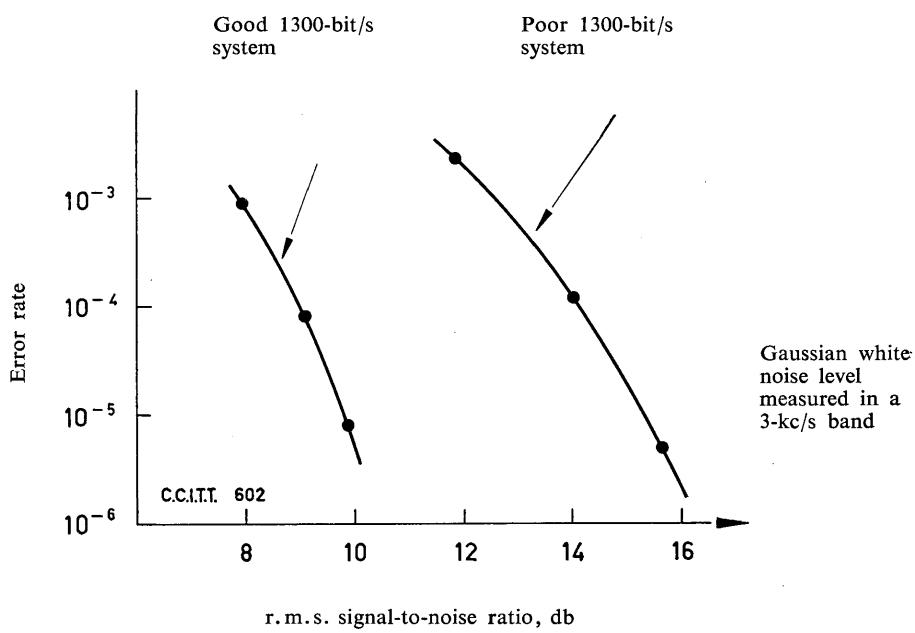


FIGURE 2. — **Typical back-to-back Gaussian noise performance curves**

[1] See footnote 2, page 653.

b) *Dynamic range test*

In this test, the back-to-back noise test above is repeated for each of a number of different signal level settings taken 5 or 10 db apart. Generally, the curves so obtained will lie more or less on top of each other over the useful dynamic range of the system and exhibit large increases in the signal-to-noise ratio required for a given error rate at the bounds of this dynamic range.

c) *Phase crawl tests*

Imperfectly synchronized local oscillators in single-sideband carrier systems introduce into the transmitted signal a continuously varying phase shift known as "phase crawl". Experimentally, phase crawl can be introduced into the signal at the insertion point "A" shown in Figure 1. The phase crawl insertion equipment should have, except for the amount of phase crawl, characteristics at least as good from the point of view of frequency passband and distortion as those which are available in K- or L-carrier type terminal equipment. The signal level is adjusted to be the standard −15 dbm (which should lie somewhere in the middle of the dynamic range as found in test b) above). A noise performance curve is then made for each of a number of inserted phase crawl rates. The degradation introduced by the phase crawl is usually expressed in terms of the number of db by which the signal-to-noise ratio must be raised in order to maintain a given error rate for that amount of crawl.

d) *Effects of amplitude and delay distortion in transmission system frequency response*

In this test, filters having the desired degree of passband amplitude distortion and passband delay distortion are inserted at point A and error rate curves taken as in a). Once again, the degradation is measured in db of equivalent signal-to-noise ratio. The difficulty with making such tests lies in the lack of any particular rational basis for picking the amplitude or delay distortion characteristics to be used.

e) *Drop-out test*

Only recently has it been realized that momentary drop-outs (ranging from a few milliseconds up in duration) are responsible for a substantial fraction of the errors incurred in transmission over certain facilities [1]. It is therefore desirable to assess the sensitivity of the modem to transient phenomena produced by such short drop-outs. The following procedure is suggested.

A minimum-phase filter having a passband from approximately 500 c/s to 3000 c/s is placed at point A to simulate the effect of the passband of the transmission system. Ahead of this filter is placed a switch, either mechanical or electronic, which can be operated to open the circuit for periods ranging from a few milliseconds up to substantial fractions of a second once every few seconds. The signal-to-noise ratio is then set to whatever value will give an arbitrary but moderate error rate (say, approximately 1 in 10^4) without any circuit interruption. We then introduce periodic circuit interruptions of a given duration. Let us define "drop-out efficiency" as the ratio of the actual increase in bit errors per second thus induced to one-half of the number

[1] J. KELLY and J. MERCURIO: "Two Measuring Techniques for the Investigation of Impulse Noise and Drop-outs on Phone Lines", Conference Paper at *A.I.E.E.* 1961 Fall Meeting.

of bits per second theoretically dropped during the interruption. The measurement of drop-out efficiency over a range of drop-out duration and duty cycle should reveal whatever deficiencies exist with respect to operation in a drop-out environment (especially difficulties with respect to proper synchronization of the transmitted and received word format).

Word format

Generally, even the back-to-back noise performance of the system as measured in a) above will vary slightly with the length and actual structure of the test word used in transmission. Ideally, one should make tests such as a) with a succession of words whose symbols are chosen at random, and average the results. So far as the "randomness" of a word of data is concerned, it is from the point of view of time and convenience much better to fix on a standard word having a balanced number of transitions of all sorts. Such a word is to be found in the maximal length shift register sequences having lengths of 7, 15, 21, etc. Equally as important as obtaining a good "random" test word is to try the system performance under a) and e) above with whatever system input signal represents no data. This "no data" condition will exist in the system a large fraction of the time, and it is essential that short drop-outs, for instance, do not affect its synchronization more seriously during these periods than might be expected in the presence of a random data word. Some data transmission schemes are particularly error-prone when a single "one" or zero occurs in the middle of a string of symbols of the opposite kind. In any extensive programme of comparative testing, one would want to check on this kind of tendency to error.

Even when a data transmission system is to be used only with words of a fixed length, it is interesting if not necessary to know how that performance (as measured in a) or e)) varies with gross changes in word length. Certainly, one should test the system at least with the word length that one plans to use. When the transmission system must cope with a variety of word lengths, it would seem reasonable to make spot checks of the performance over this range of expected word lengths, even if most of testing is done at a single anticipated field-use word length. In spite of the seeming obviousness of these remarks, an inordinate amount of testing of systems intended for use with words having lengths up to several hundred bits has been done with 16-bit words. The difficulty with this kind of test is that the amount of time devoted to word start or synchronization arrangements is an artificially large proportion of the total transmission time of any given word, and that the word starts come too often to show clearly the effect of synchronization failures due either to momentary drop-outs or to false indications of word start.

Impulse noise

The major type of additive disturbance in voice bandwidth communication channels is usually not noise with a Gaussian distribution, but noise in which the occurrence of high peaks is orders of magnitude more probable than would be the case with thermal noise. In Figure 3, we have plotted some results of performance tests made under equivalent conditions by Mercurio and Kelly [1]. The horizontal scale marks the signal-to-noise ratio for which the system error rate was 10^{-5} in a Gaussian noise test. The vertical scale indicates the error rate when impulse noise from a particular tape recording was added to the signal at a given standard level. Each point represents a data set of a particular manufacture.

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[1] J. Mercurio and J. Kelly: "Comparative Performance of Digital Data Modems", *Technical Memorandum TM 3037* (not generally available) and private communications.

The data rates and modulation schemes vary, but in all cases the signal-to-noise ratio for 10^{-5} error rate should have been 10 db or less ; thus, points far to the right on the diagram represent far departures from theoretical Gaussian noise performance. The performances of enough of the systems tested tend to lie on a straight line to warrant the statement : in general, the better the Gaussian noise performance, the better the impulse noise performance. But at least two systems (A and B in the figure), one an A.M. system, and the other a phase reversal system, are simply bad from an impulse noise point of view irrespective of their ranking in the presence of Gaussian noise.

In spite of the rough indication that systems which do well in a Gaussian noise test also tend to do well in the presence of impulse noise, one would like to have some quantitative measure of system performance in the presence of impulse noise itself. Experience with actual circuits has shown, however, that impulse noise waveforms vary in type from circuit to circuit and exhibit activity which is irregularly distributed in time. Only a minimum amount of information is yet available about the distribution of these noise occurrences in amplitude, duration, and in time. Lacking the time with which to make live tests in which the statistics of the impulse occurrences "settle down" (if, indeed, they ever do), and lacking a good statistical model from which to generate such noise in the laboratory, our only recourse is to assemble in a systematic manner recordings of such noise during intervals of high activity. Since recordings representing noise bursts from several thousands of hours of operation over a variety of types of facilities are available, a standard tape of this variety would not be difficult to make ; several such tapes have in fact

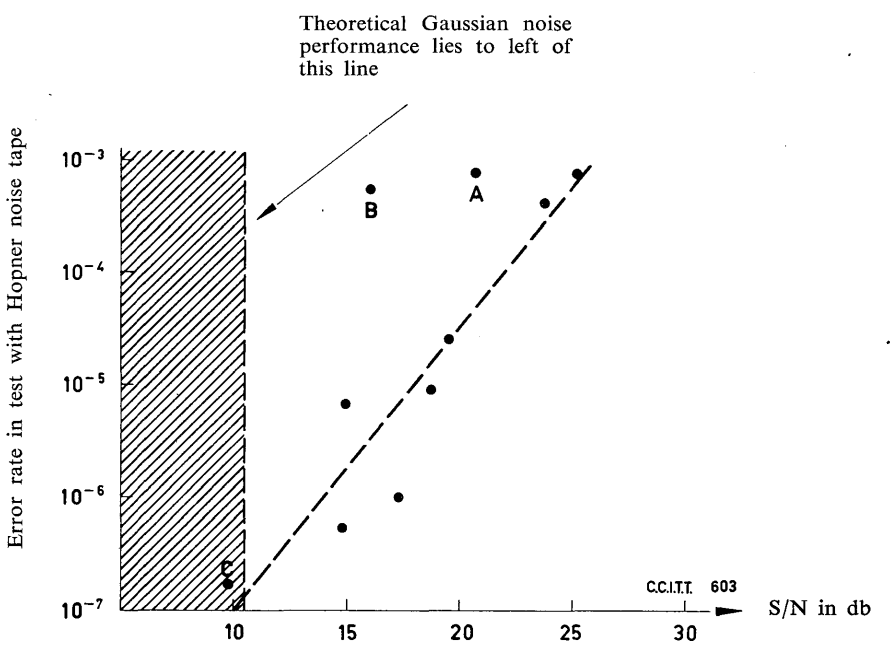


FIGURE 3. — **Comparative impulse and Gaussian noise performance**

been made [1]. In assembling several such tapes from several sources to form a single library of impulse noise, one should not seek so much to maintain the original circuit noise levels as to maintain some sort of statistical homogeneity from sample to sample; this can be done, for example, by recording the noises from different sources on the master library tape in such a way that, say, 10 per cent of the impulses from each source exceed some standard test tone level.

Lacking a theoretical indication of best system performance in the presence of such noise, one is driven to comparing a system under test with a "standard" transmission system. It does not matter a great deal what that "standard" system is so long as one can publish its performance curves; but to avoid problems of commercial favouritism and to permit a variety of people to assemble the data from which to construct similar tapes, the standard system should be very simple and easily constructable and tested with both timing and word synch information strapped across for the purposes of the test [2]. The noise bursts from a given facility should be recorded on the test tape together with a test tone at such a level that a fixed percentage of the noise bursts cause errors in the standard data modem when the single test-to-test ratio is, say, 0 db. Induced error rates can then be plotted in terms of signal-to-test-tone ratio rather than signal-to-noise ratio, which, in this case, would not be too meaningful.

Conclusions

It should not be thought that the author regards any of the test procedures suggested here as rigid and valueless if altered. It is clear that some of them (such as the question of test words and the construction of a useful and reproducible set of impulse noise test procedures) are somewhat speculative in nature, whereas others (such as the back-to-back Gaussian noise tests) are test procedures in wide use in related branches of the communications art. Nevertheless, the need for standardization in laboratory procedures for testing digital data modems is immediate and critical. The suggestions of one man cannot and ought not to be the standards for an industry; but the author hopes that in his discussion here there are at least the seeds of a set of generally accepted standard tests for data transmission modems.

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[1] If any reader has such recorded impulse noise data in either reduced or unreduced form, the author would appreciate being informed of the fact, with a view towards using such recordings in assembling a composite test tape.

[2] It does not matter that *(for the standard system only)* we here violate the "no other interconnections but the artificial transmission system" condition imposed above: this is not a system to be used for the transmission of information, but merely as an arbitrary standard of comparison; the primary consideration is that it be as simple as possible to construct.

Supplement No. 82

OLIVETTI (ITALY) — *(Extract from contribution COM Sp. A/No. 89 — October 1963)*

A SYSTEM FOR ANALYSING AND PROCESSING RANDOM PHENOMENA ON TELECOMMUNICATION LINES

1. *Introduction*

A comprehensive investigation of the conditions and capability of a telecommunication network to transmit data at high speed must include a detailed analysis of the impulse noise present in the connection and of the rapid variations in level during transmission. Such information is not generally available, because it is not usually of great interest in telephone communications.

Capability of a network to transmit data at any fixed speed depends to a large extent, not on the number of noise pulses, but rather on their distribution.

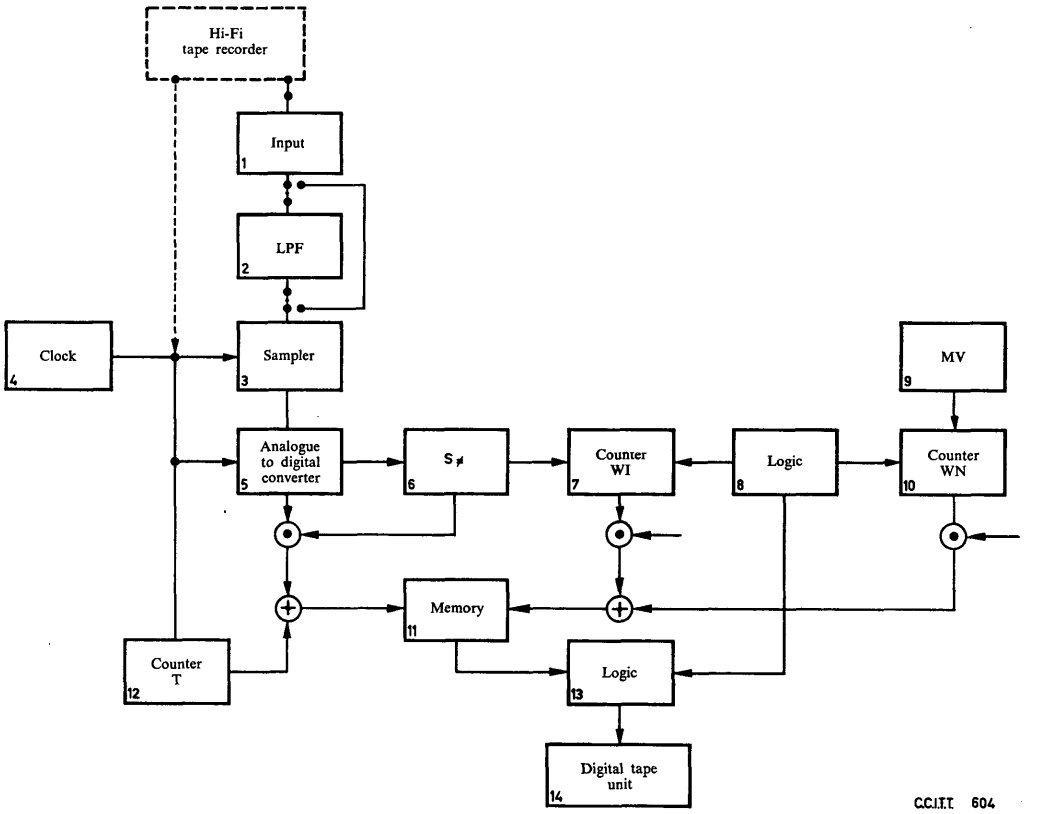


FIGURE 1. — **CDFC block schematic for noise measurements**

This is especially true when the purposes of the investigation are : the design of a data system with the most suitable protection against errors, the choice of the most effective redundancy, etc.

For these reasons Olivetti has developed, with the encouragement of the Istituto Superiore delle Poste e Telecomunicazioni, Rome, an electronic measuring apparatus which can convert any continuous or random event, present in a communication channel, up to 4 c/s, into digital information. This converter, called "Convertitore Digitale di Fenomeni Casuali" or CDFC, is described in the following pages.

It can, in brief :

1. Analyse impulse noise on a communication channel, due to the exchanges, to crosstalk from signalling or metering pulses, etc. ;
2. Analyse, when a steady tone is present at the origin of the circuit, variations in gain due to statistical disturbances and interruptions ;
3. Record and analyse the errors during data transmission.

2. *General description of CDFC*

As mentioned above, the purpose of CDFC is to :

a) measure noise ;
b) measure variation of insertion loss ;
c) register errors in a data link.

As far as points a) and b) are concerned, the basic principles of CDFC are the following, with reference to Figure 1.

In (1) the input signal passes through a low-pass filter with cut-off frequency of 4 kc/s, then it is amplified by a variable amplifier and rectifier.

In (3) it is scanned at a rate :

$$\tau = \frac{1}{2 f_{max}} = \frac{1}{2 \times 4 \text{ kc/s}} = 125 \ \mu s$$

At each sampling the instantaneous level is compared with eight fixed thresholds, numbered 0 to 7, 0 being the most sensitive threshold. The separation between them in db appears in Table 1.

TABLE 1

7	
	3 db
6	
	3 db
5	
	6 db
4	
	6 db
3	
	6 db
2	
	6 db
1	
	6 db
0	

The total dynamic range is 36 db and by means of the variable attenuator the gain of the input amplifier may be adjusted and a reference level may be made coincident with any of the thresholds.

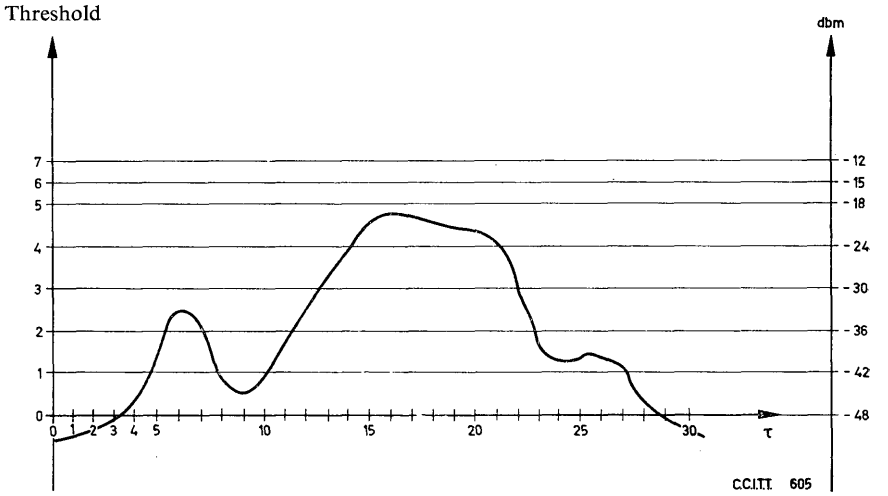
Facilities are provided for changing the intervals between the thresholds, if a different fineness of testing is required.

The maximum sensitivity of threshold 0 can be brought to −67 dbm/600 Ω.

The output of the analogue to digital converter (5) is the reference number of the highest threshold which has been crossed by the sampled signal. The result of each sampling could be written directly on the magnetic core memory (11), but this would result in general in an unnecessary high number of characters.

A compression of the number of characters to be stored in the memory can be obtained without loss of information by recording a threshold only when a change in level has occurred — that is, only when the threshold crossed is different from the previous one. Information consists of the threshold level followed by the time co-ordinate of the sampling. This time co-ordinate is derived by counter (12), which counts the number of elementary intervals of 125 μs, timed by clock (4).

When one-half of the memory is filled, circuit (8) enables address register WN to transfer its contents on magnetic tape (14) through the circuit (13), at a rate given by multivibrator (9) : in the meantime, the other half of the memory is connected to the input circuits in order to store any information that might come in. The functions of the two halves of the memory are then interchanged in time.



L	T
″0	″0000
″1	″0005
″2	″0006
″0	″0008
″1	″0011
″2	″0012
″3	″0013
″4	″0015
″3	″0022
″1	″0023
″0	″0028

FIGURE 2

In Figure 2 a typical burst of impulsive noise on a telephone line and its digital representation, obtained by conversion on CDFC as delivered on the magnetic tape (14), are shown.

a) *Noise measurements*

CDFC can be connected directly on a communication line to analyse the noise on an idle channel when the connection is established. Of particular interest are the measurements on switched connections, where a great contribution to the impulsive noise is given by the switching station, metering and dialling pulses, etc.

As mentioned in the introduction, these observations can be recorded on a Hi-Fi tape recorder and converted later on CDFC, as shown in dotted lines in Figure 1.

If the tape recorder is provided with a clock track, this track times the whole equipment, thus ensuring a better repeatability of the observed phenomena ; otherwise the internal clock can be used.

An advantage in the use of the tape recorder lies also in the possibility of feeding the same phenomena to CDFC more than once, each time with a different sensitivity of the input amplifier, thus extending or interlacing the effective number of thresholds.

b) *Measurements of variation of line insertion loss and interruptions*

In this case a constant sinusoidal tone is sent from one terminal of the line. The procedure for quantizing and recording the signals received at the other end of the line is identical with case a) except for the following points.

A low-pass filter (2) is inserted between the rectifier and the sampler, in order to filter out the mean level of the rectified signal.

If no disturbance or alteration in the line occurs, the output signal of (2) is a constant level which does not cause registration in the memory (no threshold crossed).

If a change of level or an interruption occurs, the mean rectified level changes, crossing thresholds and producing information that is stored in the memory with the procedure described before.

c) *Registration of errors in a data link*

CDFC can also be used for analysing the distribution of errors in the transmission of data, accepting characters up to 8 bits in length.

This registration assumes that the transmission of data takes place on a closed loop, or that the transmitted message is stored in a memory (not provided by CDFC).

The following description refers to the closed loop case. CDFC must be connected to both send and receive data terminals. Reference is made to the block schematic in Figure 3.

A repetitive message, of a maximum length of 2048 bits, is continuously transmitted on the loop.

Every character, at the time it is sent on the line, is also stored in a buffer (15) to be compared in (16) with the received character.

When the comparison finds an error, the incorrect character is recorded together with a co-ordinate that tells its position in the message.

The procedure for storing in the magnetic core memory and for transferring to the magnetic tape is identical to that used for noise measurements, and in fact the circuits are the same.

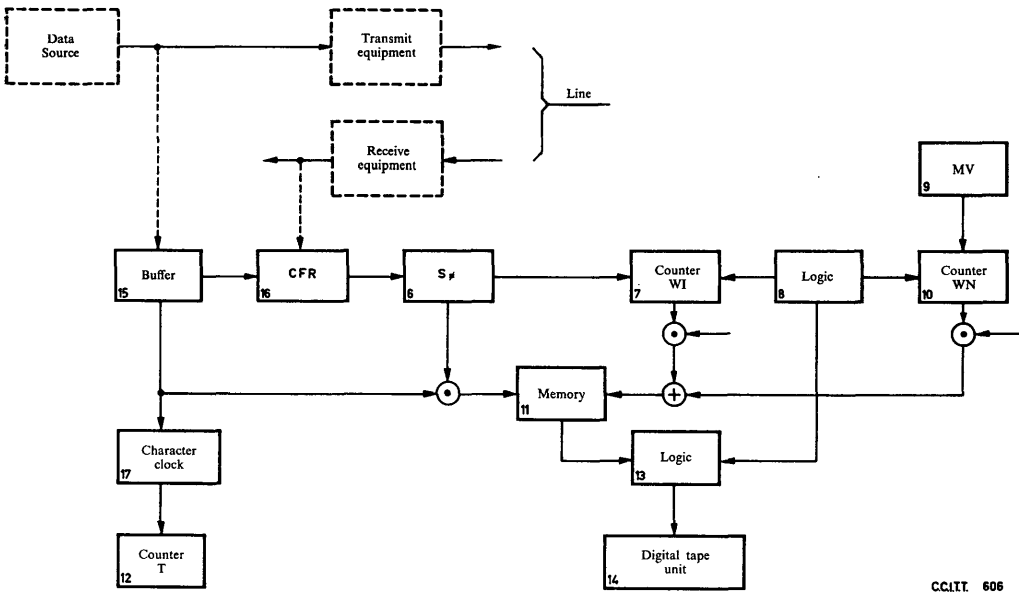


FIGURE 3. — **CDFC block schematic for registration of errors**

3. *Circuit description*

Input circuit

The input filter (1) has a cut-off frequency of 4 kc/s and an impedance of 600 ohms, with passband characteristic similar to a telephone channel filter.

The output of the amplifier is rectified, as the interest of the analysis lies mainly in the absolute value of the disturbances.

When variations in level are to be observed (case b) a second low-pass filter is connected at the output of the rectifier to smooth out the ripple of the sinusoidal tone sent on the circuit under measurements. This filter does not restrict the capability of the machine to detect changes of level up to the maximum frequency of 4 kc/s, when the test tone has a frequency greater than 2 kc/s.

In case of lower test frequencies, the suitable filter is provided.

The maximum speed of change of level that can be detected is changed accordingly.

The sampler is essentially a gate which is permanently open so that its output follows the phenomenon under examination. At regular intervals of 125 μs the gate is closed for a short period, so that the output is temporarily frozen on a short-term memory and the level existing at the instant of sampling is compared with eight fixed thresholds.

The coded level of the threshold immediately lower than the sampled signal is stored in a register. This coded level is compared with the coded level obtained in the preceding sampling. If the two levels are equal, no new information is transferred to the magnetic core memory. If the levels are different, the thresholds and their time co-ordinate, indicated by the state of counter

(12), are transferred to the core memory. Counter (12) counts the number of elementary time intervals (125 μs) up to 4095. When the counter has completed one full cycle, before returning to zero, the level of the threshold and its time co-ordinate are transferred to the memory, irrespective of whether a different threshold has been crossed or not.

This permits the maintenance of a complete record of the absolute time elapsed.

Magnetic core memory and central control

The magnetic core memory has a capacity of 1024 characters of 6 bits. Of these, one is used as a parity check, one for programming purpose, and four for information.

The read-write cycle of the memory is approximately 10 μs.

The memory is divided in two halves, each of which is addressed by two independent counters, so that, while one is connected to the output circuit (magnetic tape unit) the other can be connected to the input, in such a way that no information is lost during the transfer to the magnetic tape. The functions of the two halves of the core memory are interchanged in sequence.

When the frequency of the significant events at the input of the memory is such that counter WI might overrun WN, a logical circuit prevents it from happening, without loss of information, by reducing the number of characters to be stored in the memory.

This is done by changing the time co-ordinate coding and referring accordingly to the proper section in the computer programme.

Magnetic tape unit and controls

The magnetic tape unit speed is 50 000 characters per second. For processing purposes the information is divided in blocks.

The head and tail of each block are marked with special reference characters.

Format and code of the blocks have been chosen to be compatible with the Olivetti range of computers.

Additional circuit for registration of errors

A comparison is made on a closed loop basis of the characters transmitted on a circuit. Bits can be grouped in packets of 8 to form "characters". A local buffer stores groups of characters transmitted to allow for the loop delay.

After comparison the characters which have been affected by errors are transferred to the memory followed by co-ordinates indicating the position of the characters in the message. A key character marks the start of message.

The information transferred on the magnetic tape, through a routine similar to that one described in the preceding paragraph, is thus restricted to the incorrect characters and their positions in the message.

The magnetic tape is then fed to a computer, in the memory of which the original message is stored. The computer is programmed to supply statistics on bit errors, characters, etc.

The maximum length of the message is 256 characters or 2048 bits.

4. *Processing on the computer*

a) *Presentation of statistical results*

The tape obtained from the digital tape unit (14) is fed to a computer and a number of statistics can be obtained by programming.

b) *Use of results*

As mentioned in the previous paragraph, the computer presents, in a condensed form, the results of the statistical examination, as processed by one of the computer programmes available.

As an example, suppose that the number of disturbances having a level of −18 dbm (threshold 2) is of interest, the sensibility of the converter for threshold 3 being of −12 dbm.

During the interval of observation, the following number of disturbances of this level have occurred :

Number of disturbances	*Duration*
5583	125 µs
2275	250 µs
188	500 µs
39	1 ms
12	2 ms
2	4 ms
1	32 ms
14	256 ms

If disturbances of, for example, 1 ms duration are of interest, the computer supplies the following information :

Number of disturbances	*Level dbm*
146	−24
39	−18
27	−12
9	−6
2	0

A tendency to accumulate figures in specific time-level areas is a function of the type of switching station, its age, state of maintenance, etc.

For example, some isolated noise pulses of 250 ms duration up to a level of −12 dbm were observed. The remaining noise consists mainly of pulses of low level.

5. *Conclusion*

As has been described in the previous pages, CDFC can be used to examine random phenomena. The characteristics are the following :

useful band :	0-4 kc/s
sampling period :	125 µs
maximum sensitivity :	−67 dbm/600 ohms
number of thresholds :	8
total dynamic range :	adjustable by substitution of plug-in boards

In case of analysis of errors in data transmission, maximum length of a message is 2048 bits. CDFC can be used for :

a) full analysis of the noise in a communication channel ;

b) analysis of changes in level during transmission ;

c) analysis of errors in a transmission of data.

CDFC produces a digital tape for processing on a computer. With suitable programmes, a wide range of information can be obtained, such as, for example :

a) relative evaluation of the ability of different channels or switching stations to transmit data ;

b) choice of optimum length of blocks of data for maximum efficiency of transmission ;

c) choice of the optimum system of error protection. In case of block redundancy by means of cyclic codes, an effective research of the optimum generator polynomial can be made ;

d) research of a mathematical model of line noise and of transmission errors ;

e) an estimate of the conditions of a communication channel for maintenance purposes. A correlation seems to exist between an increasing quantity of impulse noise or short interruptions and an approaching faulty condition.

Supplement No. 83

BELL TELEPHONE LABORATORIES — *(Extract from contribution COM Sp. A/No. 13 — September 1961)*

DIGITAL DATA TRANSMISSION TEST SETS

The increasing interest in high-speed data communication facilities for business, industry, and military use has made available a number of terminal equipments. The installation and maintenance of these equipments will be expedited by testing facilities which can generate complex data trains to modulate the transmitting unit and others which can measure characteristic transmission distortion and detect and record errors in the recovered data at the receiver. Units embodying these functions will be described.

In order adequately to test any data transmission system, the driving generator must cause the output signal to embrace the entire frequency spectrum which is to be utilized. This requirement suggests a "digital-noise" generator ; however, the detection of transmission errors in this type of signal would require complex encoding and decoding equipment. A subterfuge to the provision of "digital-noise" is a quasi-random word generator which can generate a function which has many random properties but which is repetitive. This repetitive feature is all-important to the construction of a simple system. The quality of transmission may be ascertained by measuring the peak characteristic distortion and the error total of a received sample transmission of suitable duration.

In order to measure these two quantities, the equipment functions shown in Figure 1 are required.

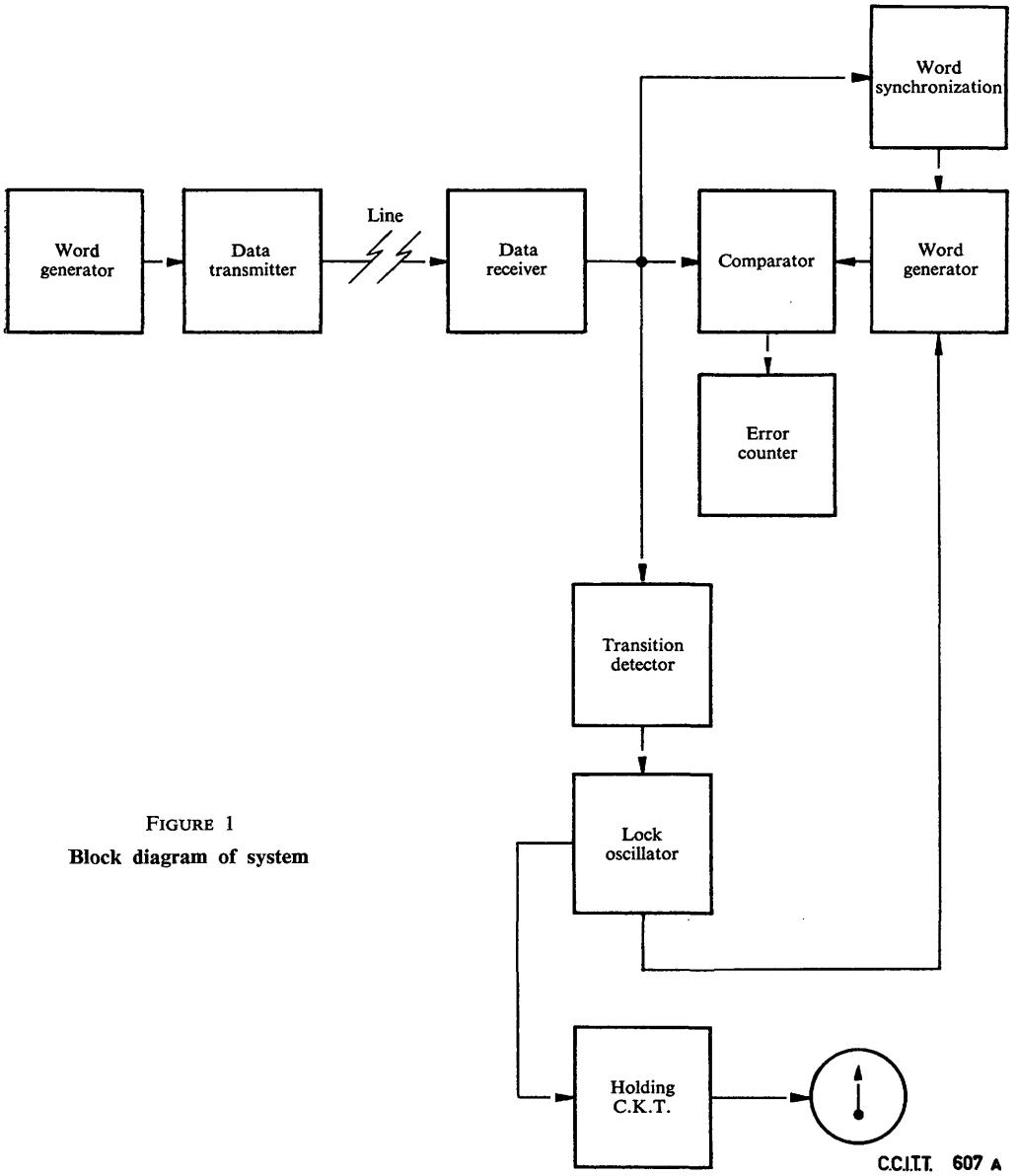


FIGURE 1
Block diagram of system

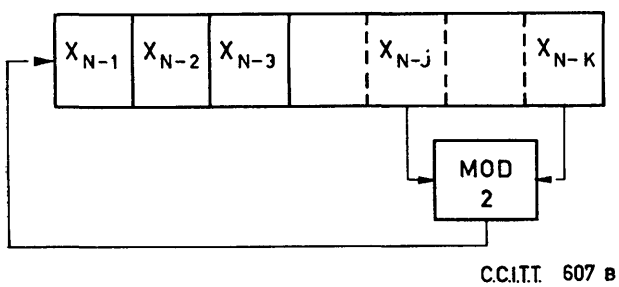


FIGURE 2
Sequence generator

As shown, the "random" word generator drives the digital data transmitter, presenting all possible combinations of sidebands to the transmission facility. The receiver converts the sub-carrier signal to the baseband. A transition detector and stabilized bit synchronization recovery complete the recovery of the transmitted information. In many receiving systems, these functions are a part of the receiver proper.

Characteristic distortion is found by measuring the temporal displacement of a transition from an average integral period. A holding circuit detects the peak displacement and a meter, or other means, indicates the percentage deviation.

A "random" word generator, of identical word structure to the unit at the transmitter, is driven at the recovered bit synchronization rate. Means must be provided to gain the proper phasing of the words, referred to in the figure as "word synchronization". In this condition, a bit-by-bit comparison may be made to determine when errors occur. A counter records such errors.

The quasi-random word generator, required at both terminals, may be constructed of a k stage shift register and a modulo 2 adder so combined that:

$$X_n = X_{n-j} + X_{n-k} \bmod 2.$$

$X_n = n$th binary digit

j, k are integers, $0 < j < k$.

This generator is shown schematically in Figure 2. From this figure it is seen that the shift register contains a word found by shifting the previous contents one bit to the right and filling the first register with the modulo 2 sum of the previous jth and kth bits. Any starting word is permissible except all zeros. If j is properly chosen (it varies with k), the register produces a word of length 2^k-1 and repeats it indefinitely. Incorrect choices of j cause the output word to be shorter.

The specific circuit to be described employs a six-stage register. The proper value for j in this instance is 5, which then yields a 63-bit word. The binary waveform of this word is shown in Figure 3. It is seen that this word contains strings of six ones, five zeros and the configurations 000010, 111101 and 1010101. These sequences are important because they are most susceptible to distortion.

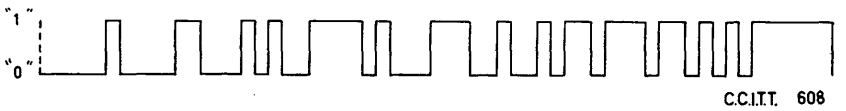


FIGURE 3. — **Output of generator**

In spite of the fact that X_n is determined by preceding bits, the sequence X resembles in some respects sequences which are produced when X_n is chosen at random with probabilities 1/2 for 0 and 1/2 for 1 and independent of preceding bits.

Without attempting rigorous mathematical exercises to show the extent of the randomness of X it may suffice to display the Fourier transform and the autocorrelation function. The Fourier transform takes the form

$$F(\omega) = T \frac{\sin \frac{\omega T}{2}}{\frac{\omega T}{2}} \sum_{n=1}^{2^k - 1} X_n e^{-j\omega nT}$$

T = one bit length in seconds.

Because the sequence is repetitive, frequency components appear only at harmonics of the word rate, i.e., bit rate/63.

The incorporation of the foregoing sequence generator into a 63-bit word generator is shown in Figure 6. The shift register is a magnetic core register manufactured by DI-AN Controls Inc. of Boston. These registers employ single cores which have separate output windings (making a total of four) for each stage. The "advance" pulse generator, a blocking oscillator, is an integral part of the final package furnished by DI-AN. This unit is in turn driven by a saw-tooth oscillator of the general negative resistance variety as shown in Figure 7. The negative resistance is obtained by means of complementary transistors in a feedback arrangement. Rotary switches provide ten possible clock rates as follows:

75, 180, 600, 800, 1000, 1200, 1600, 2000, 2400, 3000 c/s.

An additional switch position allows the clock to be driven externally by sine waves, square waves, positive or negative going pulses. Further, either the positive or negative transition may be utilized as the trigger.

The modulo 2 addition is accomplished by the circuit of Figure 8. The positive pulse outputs of stages 5 and 6 are applied to the "exclusive-or" gate which performs the addition in parallel form. Transistor $Q1$ is back-biased whenever the two inputs are the same and forward-biased into saturation when the inputs differ. Part of the charge upon $C1$ is transferred to $C2$ during this saturated interval, causing an oscillation to be set up in $LC2$. This oscillation drives $Q2$ into saturation after 2 μs which in turn records "one" in the first register. The 2-μs delay is required to allow the advance pulse to clear the first register prior to writing in the next bit. The all "one" word is detected with an AND gate and is utilized as a word synchronization output.

The pulse output of the modulo 2 adder and a delayed clock pulse drive the flip-flop of Figure 9. The inclusion of this circuit is to convert the pulse train to a baseband output. The relative lengths of the two pulses cause the flip-flop to be dominated by the modulo 2 adder output. In the absence of such an output, indicating "zero", the flip-flop changes to the "zero" state on command of the clock. An optional switch position converts the flip-flop into a binary scaler for alternate mark-space output. The output amplifier is capable of driving a 1000-ohm load with a ± 8-volt output. This is in accord with the EIA standards requirements. An internal power supply provides both positive and negative 12-volt supplies.

Detected errors are recorded in a seven-stage binary counter. No conversion to a decimal system is made. The total errors are indicated by lighted incandescent lamps in the counters.

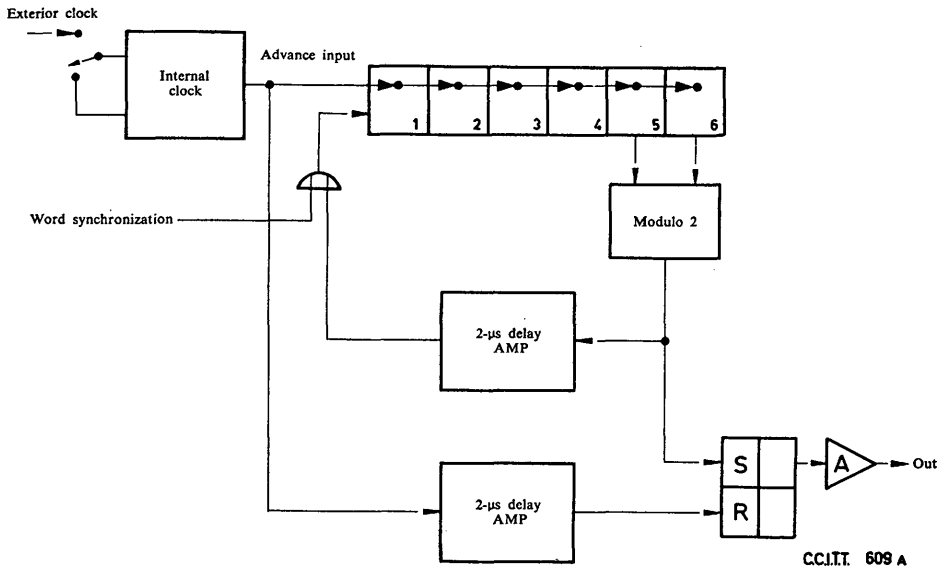


FIGURE 6. — **Word generator**

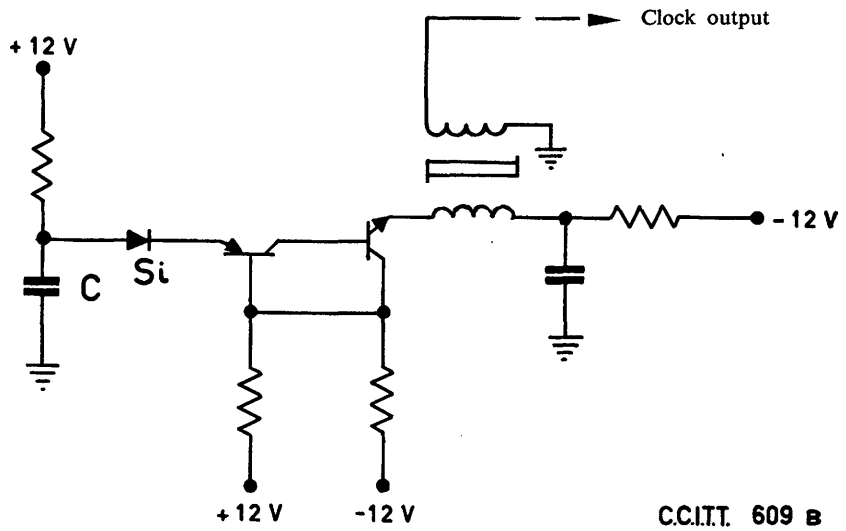


FIGURE 7. — **Saw-tooth wave generator**

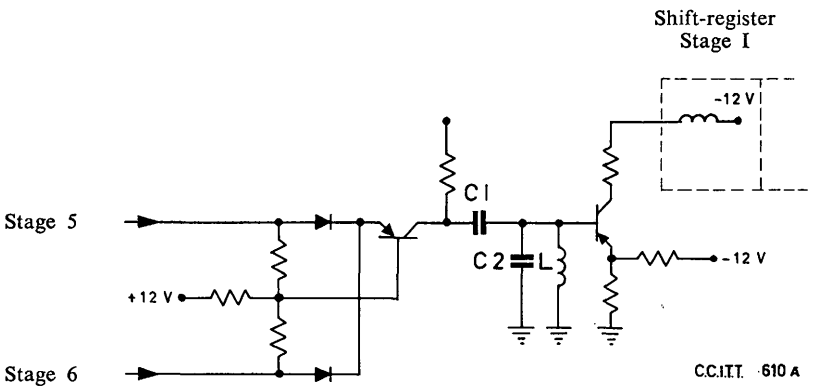


FIGURE 8. — **Modulo 2 adder and delay amplifier**

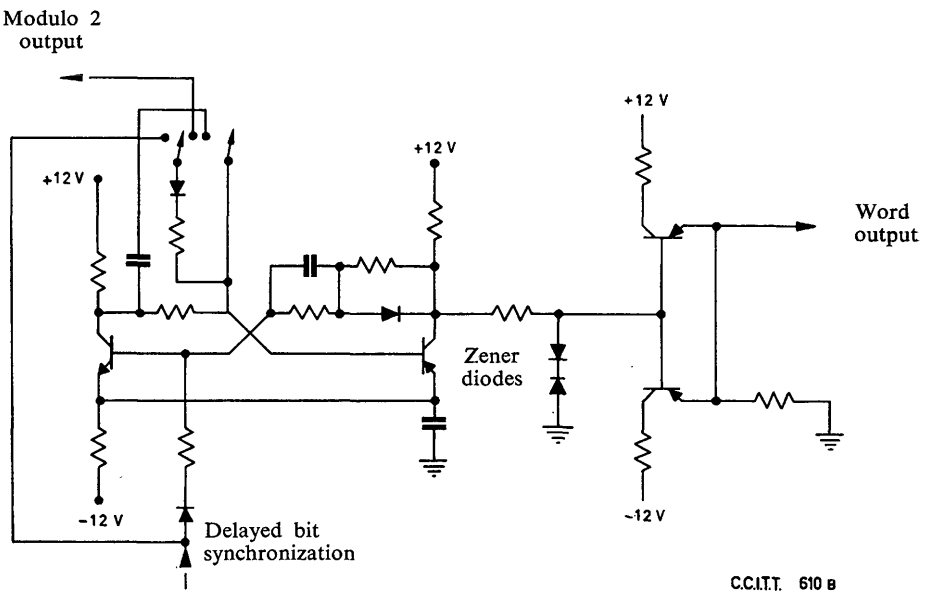


FIGURE 9. — **Baseband converter and output amplification**

Each lamp's decimal value is lettered by the lamp, decimal conversion being mental addition of the sums indicated.

Upon completion of an error total of 128, an eighth lamp lights, indicating that succeeding error totals are erroneous. The counter must then be reset manually.

A brief summary of the manual operations required for the use of this instrument may be of interest here.

The local word generator speed is set for external drive and the bit synchronization recovery speed is set for the incoming bit rate. A five-position switch, in its first position connects the data signal to a bias-measuring circuit, which has not been discussed here because of its simplicity. A bias adjustment is made, if indicated, and the switch is thrown to its second position. In this position the distortion-measuring circuit is zeroed by means of *P* 2 of Figure 14. The third switch position adjusts the quiescent output voltage of the direct-coupled amplifier by *P* 1 of Figure 13. Position 4 eliminates the phase error between the incoming data and the local phase-locked oscillator by means of *P* 2 of Figure 13. The final position gives the distortion reading. No further adjustments are made.

Supplement No. 84

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 7 — September 1963)*

BINARY-DIGITAL TESTING METHODS FOR DATA COMMUNICATION

Test procedure

Testing in the event of a fault on the modem equipment situated at the customer's premises is most conveniently carried out by connecting the modulator back to the demodulator locally. It is possible to do this only if the installation has facilities for sending and receiving data on the same carrier frequency. However, in many cases a customer's installation will be equipped for transmitting data only, or perhaps in addition receiving control signals on a separate carrier over a low-speed supervisory channel. In this event the United Kingdom Administration proposes to establish test centres in convenient telephone exchanges which can be connected to the customer's equipment over the switched telephone network and exchange digital test signals.

Digital data test set

For this purpose a digital data test set is required both at the test centre and at the customer's premises. In the latter case a portable form will be necessary.

The facilities required from such a test set could be as follows :

1. *Interconnection*

The data test set to be connected to the modem through an interface plug.

2. *Interface circuits*

The following circuits should be available with appropriate keys and lamps:

2.1 Common circuits
- 2.1.1 Connect data set to line
- 2.1.2 Data set on line
- 2.1.3 Speed selector

2.2 Transmit circuits
- 2.2.1 Request to send
- 2.2.2 Ready for sending
- 2.2.3 Transmitted data

2.3 Receive circuits
- 2.3.1 Received data
- 2.3.2 Data carrier detector

3. *Test transmitter signals*

The following test signals should be available for transmission over the transmitted data interface circuit :

3.1 Mark (binary 1)

3.2 Space (binary 0)

3.3 Reversals (1 : 1 ratio)

3.4 Pseudo-random signals repeating every 511 elements

3.5 The modulation speeds in bauds of the signals under 3.3 and 3.4 to be variable in steps as follows : 50, 75, 200, 600, 1200, 1800, 2000, 2400, 3000 and an external control.

4. *Test receiver*

The test receiver should be capable of accepting the signals generated under 3 above. The receiver should be capable of synchronizing to the incoming signals. Automatic initial synchronization is not required.

The following facilities should also be available:

4.1 Indication of mark or space on "Received data" circuit

4.2 Indication of condition on "Data carrier detector" circuit

4.3 Measurement and indication of state of signals on "Received data" circuit
- 4.3.1 Bias distortion
- 4.3.2 Peak distortion on any signal element

4.4 Detecting element errors on signal, counting and displaying the count. The counting circuit should be able to register up to 255 errors.

Supplement No. 85

AMERICAN TELEPHONE AND TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A/No. 75 — July 1963)*

DISABLEMENT OF ECHO SUPPRESSORS

A. Consideration in providing reverse channel control signalling simultaneous with data transmission

Many of the data transmission services now in use and being planned for use in the near future are of a nature that require end-to-end control of terminal equipments simultaneous with data transmission. Such controls may be for starting or stopping terminal machines for example, for circuit assurance, or for data-error control.

If control signals are to be transmitted in the reverse direction of data signals, the communication channel must of course possess full duplex characteristics. This is no problem when the channel is a private line extended four-wire to the data terminal equipment. The need, however, exists in those data systems which utilize either two-wire private lines or the switched telephone network.

The one factor which in the past has prevented simultaneous transmission of signals in both directions (on a nation-wide basis) is the use of echo suppressors. With the advent of dial TWX service echo suppressors were arranged to be disabled by transmission of specific control tones, thereafter allowing bi-directional transmission. This plant capability can be exploited for high-speed data transmission in one direction with low-speed control transmission in the other direction or by low-to-medium speed data transmission in both directions simultaneously.

In general, it has been found that there is a real economy and need in data transmission for high speed, making it necessary to use the maximum available bandwidth for the data signal spectrum. Now that it becomes necessary to transmit simultaneous control signals, the choice lies between retaining high-speed data and using very-low-speed control or buying control speed at the expense of data speed.

Until a point is reached where the control speed can be equal to the data speed, several factors affect the choice in trade-off. These are :

1. Cost to the business machine to generate very-low-speed control signal ("break" signal, on-off control, etc.). This type of control seems to be in high demand.

2. Cost to the business machine to generate low-speed control signals (20 to 200 bits per second as compared with 1000 to 2000 bits per second data speed).

3. Comparative cost of very-low and low-to-medium speed control signalling in the data set.

4. Time-saving by transmitting simultaneous control signals as opposed to stopping data transmission periodically and reversing direction for transmission of control.

First consider these factors in the order listed. The cost to a business machine to generate very-low-speed control signals such as would be required to start or stop a transmitting machine from the receiving location or to signal for an attendant-operator can be very low. A simple push-button or toggle switch can be used to generate the binary signal.

To generate low-speed (coded) control signals—for example, in the 20-to-200-bit-per-second range while data transmission is in the 1000-to-2000-bit-per-second range—presents complications in the business machine. Such controls could be quite flexible and very effective for digitally encoded replies. However, to use such an arrangement requires that the business machine incorporate complete circuitry for operation at two different speeds and to do so independently in the two directions of transmission. This may or may not include duplicate synchronization recovery circuits. Indications are that very few business machine manufacturers would be willing to complicate their machines to the degree required, possibly at the expense of pricing themselves out of competition.

A data set may handle very-low-speed control channels with relatively inexpensive modems. When the control speed is pushed upward, the complexity of the control channel modem becomes equal to that of the high-speed data modem. Every effort must be made to maintain the data speed as high as possible and this necessitates very good and very expensive band separation filters to keep data from interfering with control or vice versa.

The prime reason to have simultaneous control signal transmission is of course to avoid having to stop data transmission periodically in order to reserve the direction of transmission just to ascertain that everything is operating properly at the receiving end and that the data are being received without errors. Almost without exception, business machines which include error control have resorted to "block" or "record" transmission with reversal of direction and transmission of encoded control signals after each block. The control signals are transmitted at the same digital bit speed as the data. It is necessary that the control signal transmission be fully as dependable as the data transmission. Therefore, a control signal consists generally of two or more digital characters which are protected by error detection.

If the encoded control signals are simultaneously sent back to the transmitting business machine while the next block of data is being received, the entire control message must be shorter than the shortest data block if bookkeeping for association of data block and control are not to become difficult and require costly storage. This then dictates the minimum acceptable speed of control transmission, compatible with a nominal data-block length (the data-block length may occasionally be only a few characters and much shorter than the nominal block length used in the system design).

The nominal data-block length in general use is about 1000 characters. If these are eight-bit characters, the block length is 8000 bits. At 2000 bits per second a block takes four seconds to transmit. The nominal control message in general use is four characters, or 32 bits. The control message could then be transmitted easily at ten bits per second with time to spare. If, however, a data block is very few characters in length, this would mean wasting approximately three seconds getting a control message returned; not an acceptable situation when certainly the direction of transmission could have been reversed and the control message completed in much less time. The minimum acceptable speed for simultaneous control transmission, then, must be such that a control message can be transmitted in equal or less time than it takes to reverse direction and transmit the control message at high speed.

To determine the total time to reverse the direction of transmission over circuits which are equipped with echo suppressors which have not been modified to permit disabling, transmit a control message, and re-start transmission of data involves the following:

1. Reversing echo suppressors for control transmission . . .	(approx.)	120 ms
2. Control transmission of 32 bits at 2000 b/s	(approx.)	16 ms
3. Reversing echo suppressors for continuation of data transmission	(approx.)	120 ms
4. Twice the communication channel propagation time (for example, 2000-mile carrier circuit)	(approx.)	34 ms
Total	(approx.)	290 ms

If the control message is transmitted over a reverse control channel without having to turn around the direction of transmission, the minimum speeds must be such that 32 bits of control message can be transmitted in 290 milliseconds or approximately 110 bits per second.

If one wished to use the reverse channel (or equivalent) to hold the echo suppressors in the disabled condition, while reversing the direction of transmission of the main data channel—for block-error control, the actual waiting times required would not be significantly different from those shown in the foregoing tabulation. This is because of the necessity for protecting against echoes (which would cause errors in the received data) on a long-haul connection. The minimum protection interval would probably be about 100 ms, which is only slightly less than the 120 ms shown in the foregoing tabulation as the time required to reverse an echo suppressor.

Conclusions which may be drawn from the above discussion are:

1. Very-low-speed simultaneous reverse control signal channels are in high demand for simple control functions and can be implemented relatively inexpensively in both the business machine and the data set equipments.

2. To be useful for digital signalling, a reverse control channel must have a bit speed capability of at least 110 bits per second, just to break even in time spent transmitting control signals as opposed to reversing the direction of transmission and sending control signals at high speeds.

3. Digital control signalling at low-to-medium speeds simultaneous with data transmission is not economically attractive to either the business machine manufacturer or the communication equipment provider at the present time.

Bell System work on reverse channel

With the foregoing in mind, the Bell System is presently planning to offer, as a feature of DATA-PHONE service on the switched telephone network (with data speeds up to 2000 bits per second), a simultaneous very-low-speed control channel effective in the reverse direction to data transmission. Previously, a small quantity of X802A auxiliary reverse channel units had been produced on a model shop basis and used for compatibility testing with various business machine equipments. Instead of going into standard production of the 802A, it is planned to include the reverse channel unit as an option in revised designs of various data sets (viz., the 201C & D and the 202C & D). The data set/business machine interface with the reverse channel will be designed, the same as other interface circuits, to accept and deliver "EIA voltage signals".

To make use of the reverse signalling channel requires first that the echo suppressors be disabled and then kept disabled by ensuring that energy is being transmitted from either end of the connection at all times for the duration of the data call. Action of the echo suppressor disablers currently in use require that a single frequency tone in the 2000-2250-c/s band be transmitted from either or both ends of the connection for approximately 400 milliseconds with no energy at any other frequency. This signal then places the echo suppressor in a neutral condition where it will remain until all energy has disappeared from both ends of the connection for approximately 50 ms. When once again enabled, the suppressor functions normally until the disabling signal is once again received.

Data sets which include reverse channel operation are arranged to transmit a single-frequency signal of 2025 c/s for a period of three seconds when the called station is switched from the voice mode of operation to the data mode of operation, either manually or by automatic answer. This signal performs two functions. It first gives an audible indication to the calling attendant that the called station has been switched to the data mode of operation and simultaneously all echo suppressors in the circuit are disabled. Upon completion of the 2025-c/s transmission the called station will be in the data mode of operation. If the station is to be a transmitter, it will be sending data-carrier signals. If it is to be a receiver, it will be transmitting reverse-channel-carrier signals. Either signal serves to maintain echo suppressors disabled.

When a data call has been established and echo suppressors disabled, the direction of data transmission may remain constant or be reversed as many times as necessary. The only requirement is that the data carrier and reverse channel carrier never be turned off simultaneously for more than 50 milliseconds. In order that the business machine logic be as simple as possible, the reverse-channel carrier is automatically turned on at any time the data set request-to-send is turned off. The receiving business machine can of course turn reverse-channel carrier off as the signal indication to the transmitting business machine. If the signal is used to tell the transmitting business machine to back up and repeat a portion of the data message, marking carrier should be transmitted during the backing-up operation.

The reverse channel being incorporated into Bell System data sets is a simple amplitude-modulation channel. This method was chosen to keep the complexity and cost of the reverse channel modem to a minimum. The carrier frequency has been set at 387 c/s, as this frequency is within the sensitivity band of echo-suppressor disablers and far enough below currently used data-signal frequency spectra to allow effective filtering. The reverse-channel band is 25 c/s wide allowing for $\pm$ 5 c/s frequency off-set in non-synchronous telephone carrier channels. It is possible under usual conditions to operate the channel at speeds up to ten bits per second ; however, the expected use will not exceed five bits per second.

A reverse signalling channel such as described is expected to meet the need for reverse control signalling simultaneous with data transmission at a minimum cost and to do so by making use of a narrow slot from a portion of the voice-frequency circuit not usually suitable for data transmission. With this approach it is not necessary to reduce data-transmission speeds below those currently in use.

One of the first applications of the described channel will be with Teletype Corporation's DATASPEED I. DATASPEED I is a straightforward paper-tape to paper-tape transmission system operating at 1050 bits per second. Without reverse channel signalling, there is no way for the transmitting terminal to know whether the data are actually being received or there is a malfunction in the receiving machine. A simple "break" type of signal is required to start and stop the transmitting terminal thereby avoiding needless transmission of data that are not being received or wasting time on a dialled-up circuit not suitable for data transmission. By using the "on" signal of the reverse channel as a "start" signal for the transmitting machine and the reverse channel "off" signal as a "stop" signal, the reverse channel also serves as a circuit assurance indication.

B. A DISABLER FOR BELL SYSTEM ECHO SUPPRESSORS

Introduction

The telephone network is finding increased use as the transmission medium for data systems. Of particular interest is the teletypewriter exchange service (TWX) and other similar low-speed serial data services. A feature desired by dial TWX and many other data-transmission systems is the ability to send signals in both directions simultaneously, an operation precluded by echo suppressors in the telephone toll plant. It is true that transmission can be forced in a direction opposite to that previously established by sending a signal in the reverse direction of sufficient level to "turn-around" the suppressor. In practice, however, the necessary signal levels to reverse the suppressor may exceed plant limits, and the delays involved in turning the suppressor around would certainly complicate signal sequences. By providing the capability for simultaneous two-way transmission over telephone circuits in the network, more efficient error detection and correction, system control, and circuit assurance schemes can be realized. An echo-suppressor disabler has been developed which will inhibit the suppression function of the echo suppressor upon command of a conditioning tone. The engineering design and operational features of this tone disabler are described herein.

Echo-suppressor operation

One of the most important factors bearing on the transmission design of the telephone plant is the relationship of return loss performance to the customer tolerance of talker echo as a function of delay time. The accepted engineering practice has been to control short-delay (less than 45 ms) echo by the use of appropriately assigned trunk losses (termed via net loss (V.N.L.) design). Although this scheme is generally adequate for calls entirely within one regional area, on the longer interregional telephone connections talker echo delays of greater than 45 ms are common, and echo control must be accomplished by use of an echo-suppressor circuit. In contrast to voice design, talker echo is generally of little importance to data systems since two-way simultaneous transmission is provided by the separation of transmitting and receiving channels. On the other hand, a binary data receiver is usually sensitive to any listener echo delay time greater than a significant fraction of the shortest pulse interval. Echo delay in the telephone plant almost always exceeds this interval for practical data speeds, making V.N.L. design and echo suppressors inappropriate for data systems.

A typical four-wire trunk plan showing the location of the echo suppressor and proposed tone disabler is illustrated in Figure 1. It may be noted that the disabler is on the drop (office) side of the single-frequency (S.F.) signalling equipment and, therefore, free from interference by S.F. tones. The two sides of the suppressor are differentially interconnected, making operation dependent on the difference between the speech levels in the even and odd (outgoing and incoming) directions of transmission. For instance, speech power sensed in the even direction of transmission operates a relay chain which introduces a loss of about 40 db (decibels) in the odd direction of transmission. However, a customer may "break in" and cause the suppressor to turn around

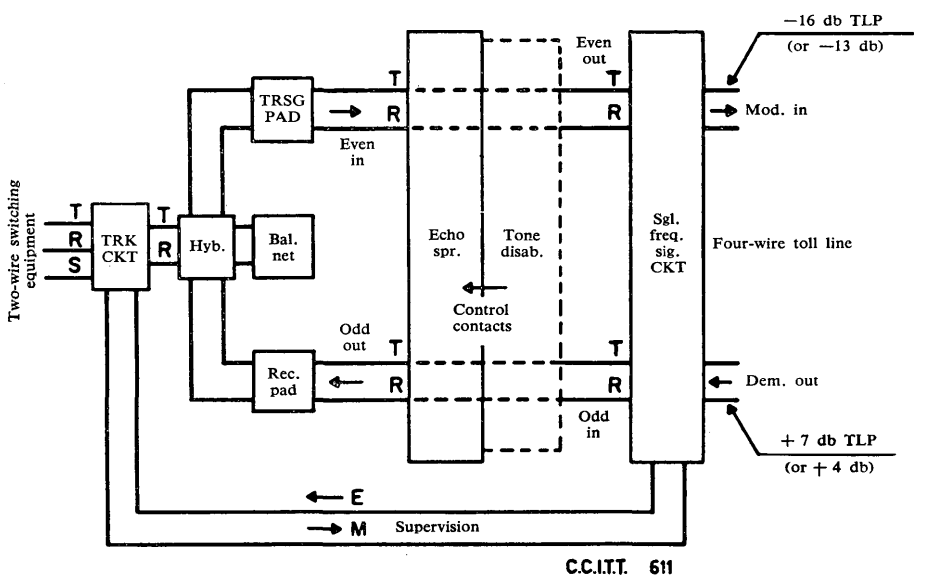


FIGURE 1. — **Typical four-wire trunk equipment arrangement**

if he raises his voice enough to deliver a speech power at the suppressor about 5 db higher than the first talker.

The loss inserted in the transmission path by the suppressor must remain long enough to prevent the return echo from reaching the talker following a pause in his speech. The length of time that transmission loss remains following cessation of speech power in the opposite path is termed "hangover time" and must exceed the round-trip propagation time of the echo path beyond the suppressor. Tests have shown that best operation is obtained when the hangover time is restricted to less than 70 ms. Where transcontinental or transoceanic circuits are involved, a hangover time exceeding 70 ms would be required to retain effective control of echo if only a single suppressor were employed. To overcome this difficulty, an echo suppressor is located at each end of a long trunk with the suppression function disabled in the incoming direction (odd side) of each suppressor. This arrangement, known as "split operation", permits a reduction in hangover adjustment by an amount equal to the round-trip delay of the trunk. It is necessary to retain differential control so as not to affect the ability of the listener to break in.

The important design parameters for the echo suppressor are outlined in Table I. Note that the suppressor turn-around time is indicated as equivalent to hangover time. Actually, a reversal tone would need to be applied for a somewhat longer time interval in order to account for such factors as propagation time and operating variations.

TABLE I

1A echo-suppressor design parameters

Receiver sensitivity (at 0 db transmission level point)	Adjustable between 25 and 35 dbm (db referred to 1 milliwatt) (−31 dbm nominal)
Receiver response	Sensitivity maximum at 1000 c/s (cycles per second), down 20 db at 500 and 2100 c/s
Suppression loss	40 db from 1000 to 1500 c/s, 30 db from 200 to 3500 c/s
Operate time	12 ms
Hangover time (and turn-around time)	Adjustable between 18 and 275 ms (set to round-trip delay time)

It is current operating practice in the Bell System to assign an echo suppressor to all interregional trunks with V.N.L. larger than 2.5 db, and operate two suppressors on a "split" basis on circuits 2500 miles and more in length. The number of suppressors installed in the Bell System is presently estimated at 18 000.

Systems consideration for an echo-suppressor disabler

The systems design of an echo-suppressor disabler took into account the following:

1. Although dial conversion of TWX provides the main impetus for echo-suppressor disabling, the method adopted should allow other data systems which may use the network to disable the

suppressors as well. On the other hand, disabling capability should not require undue complications of these other data systems. The time required for disabling the suppressor should not exceed that normally required to complete a switched connection.

2. The disabler must be compatible with existing telephone plant ; for example, the probability of disabling tone imitation ("talk-off") by either speech, plant tones, or noise must be minimized while the operating characteristics of the echo suppressor (sensitivity, through and suppression loss, even-to-odd side coupling loss, etc.) must not be compromised.

3. For proper operation, the disabler should be relatively insensitive to expected levels of line noise, whether the disabler is accommodating data or speech transmission on an active trunk or an idle trunk.

4. Certain inherent properties of speech, such as its peaked character and the spectral distribution of energy both on a long-term basis and for syllabic intervals, may be advantageously used to distinguish between a legitimate data signal and speech energy.

With these considerations in mind, a number of schemes could be conceived for the operating method of the disabler. Several of these methods are discussed in the next section, followed by a detailed discussion of the plan finally adopted.

Alternative methods of disabling

One plan might be to disable upon receipt of a particular sequential binary code ; however, since data sets will far outnumber echo suppressors, it is desirable to avoid complicating these data sets and those of other large customer volume data systems where simplicity and economy are relatively important.

Another plan, termed coincidence disabling, would require the simultaneous receipt of steady data tones from both originating and terminating data sets. The required disabler sensitivity and the return loss performance of the telephone plant, however, make it impossible reliably to differentiate between the reception of two data tones of identical frequency and a single data tone plus its echo. The judicious choice of dissimilar data tones for coincidence disabling would solve this problem and at the same time offer a marked degree of "talk-off" protection. A disadvantage, however, is that during the interval when data tone coincidence is being established, the direction of transmission through the echo suppressor remains ambiguous, thus complicating the control signal sequence required to establish any duplex connection.

Disabling upon detection of a signal from either even or odd side of the line (i.e., unilateral disabling), employing a sensitivity and response characteristic commensurate with that of the echo suppressor and deriving talk-off protection from timing alone, is another possibility. An analysis of the statistical distribution of talk spurt lengths of subscribers indicates that to ensure against false operation 99.9% of the time during busy hour would require a protective delay time of from 7 to 10 s. This delay is unacceptable for many data services in view of the relatively short holding time expected and, furthermore, the bandwidth of the detection signal would cause vulnerability to noise.

Basic plan : tone disabler

The method operation selected for the tone disabler incorporates the following features : (1) unilateral disabling control, (2) guard action, (3) talk-off protection interval (attack time), (4) broadband holding, and (5) timed release.

Disabling of the echo suppressor is accomplished by the continuous and exclusive transmission of energy in a narrow conditioning band (2000 to 2250 c/s) for a period exceeding the "attack time". As implied, attack time specifies the minimum conditioning tone period required before the disabler transfers to the holding mode and disables the echo suppressor. The principles of guard action, successfully used by single-frequency signalling equipment, utilizes spectral energy in the voice channel outside of the narrow conditioning band to oppose disabling. Once executed, disabling may be continued by providing energy anywhere in the useful voice band, provided signal absence does not exceed release time. This broadband holding characteristic will allow broad-voice-band data systems to initiate and maintain disabling, provided only that the proper conditioning signal precedes normal data.

Table II relates the two stable operating states of the disabler to the trunk condition.

TABLE II

Operating states of disabler and relative trunk condition

Disabler state	Trunk condition
Guard mode (normal suppressor operation) . .	Idle (disconnect) Voice transmission Plant signalling and test tones
Holding mode (suppressor disabled).	Data transmission

General design considerations

The successful operation of the tone disabler is contingent upon compromise among several conflicting design factors. To gain a better understanding of the logical selection of design parameters for the tone disabler, several factors are considered for detailed discussion: (1) choice of the conditioning tone frequency and signal bandwidth, (2) guard action, and (3) "attack" and "recycle" times. These factors constitute the principal elements employed to achieve immunity against signal imitation. Other factors such as (4) conditioning band sensitivity, (5) holding mode release characteristics, (6) release time, and (7) noise susceptibility warrant equal attention. Disabler design parameters are summarized in Table III and discussed in detail in the following.

TABLE III

Tone disabler design parameters

Conditioning band sensitivity . . . . . (at 0 db transmission level point)	Adjustable between 25 and 37 dbm (−31 dbm nominal)
Conditioning band	(2000 to 2250 c/s)
Guard band	(500 to 1775) and (2475 to 3000) c/s
Guard-signal ratio	5 db
Attack time	300 ms
Recycle time	10 ms
Release time	100 ms

1. *Conditioning tone : signal bandwidth*

The dial TWX data set transmits data on a frequency-shift basis, the mark and space frequencies having a 200-c/s separation. It is necessary that the standard voice band accommodate at least two narrow-band channels to allow for duplex operation. The centre frequencies (F_1 = 1170 c/s and F_2 = 2125 c/s) of the narrow-band data channels have been carefully selected (a) to avoid the greater transmission losses and distortion encountered at the upper edge of the voice band, (b) to be compatible with the direction of propagation of existing plant supervisory and signalling tones, and (c) to avoid the S.F. signalling frequencies of 1600, 2400, and 2600 c/s. The originating (calling) station will always send on the F_1 channel, while the terminating (called) station will send on the F_2 channel. The data set connect sequence requires that the mark frequency of the upper channel (F_{2M}) generated at the called station shall be the first data signal transmitted during the course of call establishment and the presence of F_{2M} at the echo suppressor will cause disabling of that device.

The conditioning tone is thus set at F_{2M} = 2225 c/s. The probability of talk-off is very strongly influenced by the choice of conditioning tone frequency, decreasing rapidly as this frequency is increased. Data obtained from tests made during the development of in-band S.F. signalling circuits are reproduced in Figure 2. It is apparent that 2225 c/s is a fortunate selection of conditioning tone for talk-off minimization.

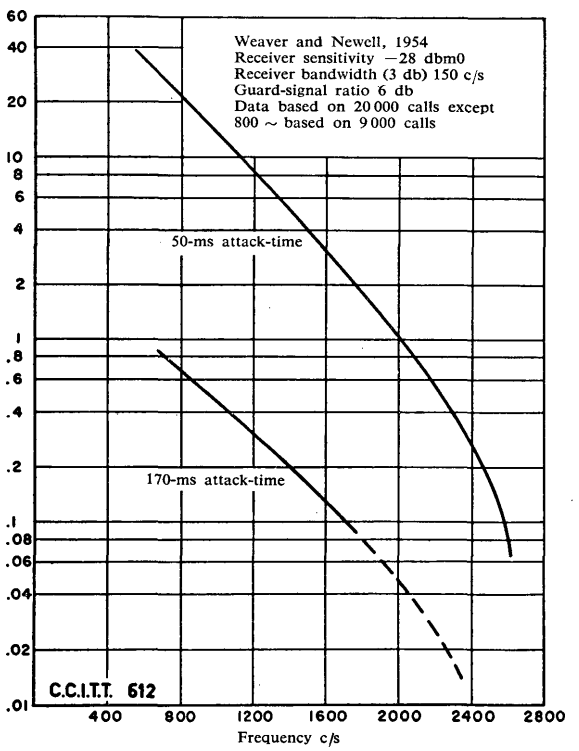


FIGURE 2. — **Conditioning-tone-imitation data**

Freedom from talk-off is also enhanced by using the least possible signal bandwidth, thus effectively reducing the power of speech and noise falling within the conditioning band. Adequate bandwidth must be provided, however, to encompass the signal frequencies of existing and proposed data systems which will benefit from the suppressors disabling feature. Additional latitude must be allowed for frequency shift encountered in non-synchronous single-sideband carrier systems, and variation of generated signal frequencies and component values in the tuned circuits of the disabler.

A conditioning band that brackets the range (2000 to 2250 c/s) at nominal sensitivity has been selected.

2. *Guard action*

The most significant deterrent to talk-off is the use of practically all voice-band frequencies outside of the narrow conditioning band to generate a voltage which is used to oppose that derived from the conditioning band.

The vowel sounds contribute the major portion of total speech power. The phonetic specification of a vowel involves the fundamental frequency (100 to 250 c/s) and the "formants" which define the regions of energy concentration produced by vocal resonances. These vowel properties are subject to wide variations caused by the word or sentence context in which the sound is spoken, and the age, sex or dialectal background of the speaker. Figure 3 illustrates the average position of the formants and their relative magnitudes for vowel sounds contributing significant second formant energy within the conditioning band. Formant energy tends to diminish with increasing frequency; therefore, response shaping at the low end of the guard band must be tailored to utilize fully first formant energy to prevent false disabling by these sustained vowel sounds.

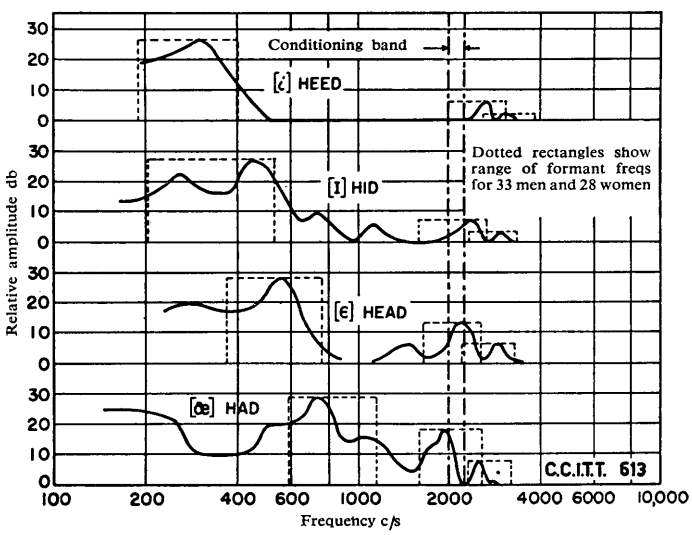


FIGURE 3. — **Typical vowel spectra contributing significant energy in the conditioning band (2000 to 2250 c/s)**

Guard action above 2500 c/s may take advantage of the upper formants and also protect against very-high-level speech by sensing modulation products caused by amplifier overloading. A plot of the resultant direct voltage as measured at the output of the disabler detector is shown in Figure 4 as a function of frequency; the several contours represent constant S.F. input powers referred to the 0 transmission level point.

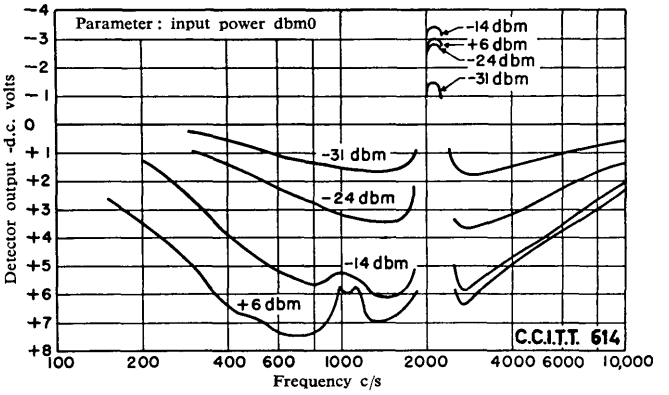


FIGURE 4. — **Detector characteristic**

3. *Attack time and recycle time*

Talk-off protection can be improved by providing the longest possible attack time consistent with data-set timing and switching sequences required to establish the connection. Basically, this suggests a time greater than the average syllabic period (125 ms) and noise burst lengths. An attack time of 300 to 400 ms meets this requirement, while providing a margin for disabling, even under noisy conditions, within the minimum timing interval required by the data set connect sequence. Talk-off advantage is enhanced and the reliability of tone disabling improved by making the attack-time interval independent of the input-signal magnitude.

To satisfy the disabling criterion, a comparison must be made between the actual time interval during which only conditioning tone has appeared and the required attack time as derived above. Any deviation from the "tone-only" condition should immediately reset the timing mechanism. Resolution is a function of the recycle time, which should be brief enough to prevent integration over a time period spanning normal consonant or vowel intervals, otherwise a train of tone bursts or speech syllables could cause false disabling even though no single burst could do so independently.

4. *Conditioning band sensitivity*

The transmitting levels of data sets are set as high as possible to secure an adequate signal-to-noise margin at the receiver; however, crosstalk, intermodulation distortion, and repeater overloading impose practical limitations. The data-set output level is adjusted to compensate for most transmitting loop-loss variations, and provides a −10 dbm signal (at 2125 c/s) at the switch of the serving central office, although for the longest loops this level is reduced to as low as −15 dbm.

The design sensitivity of the disabler in the conditioning band must account for the probability of multilink connections, the expected variation in data-set output levels, tolerance in the performance of the disabler, the higher transmission loss at 2125 c/s, the design and operating variations in trunk loss, and flexibility to adapt to future changes in the plant. A sensitivity of −31 dbm referred to the 0-db transmission level point (or −31 dbm0) within the conditioning band should permit a data set at the end of the longest loop to initiate disabling for 99% of the switched connections reaching an echo suppressor on final routes in the network.

5. *Holding mode : release characteristic*

Once disabling has been established by the recognition of a legitimate conditioning tone, it must be sustained by energy anywhere in the useful voice band. Holding sensitivity must therefore be designed to accommodate expected data-signal levels at the echo suppressor. In addition, it must be made at least equal to that of the conditioning band to prevent hunting (i.e. mode instability).

The disabler must release from the "holding mode" upon sensing an idle trunk condition, in the face of the maximum line noise that may be expected consistent with commercial-grade telephone service. This places an upper limit on holding sensitivity and a compromise must be sought between the two opposing considerations mentioned above. Fortunately, such a compromise has been found ; a plot of the minimum S.F. tone level needed to sustain holding is plotted versus frequency in Figure 5. Special attention must be given to the holding response where a concentration of noise power can be expected from harmonics of 60 c/s or stray plant supervisory tones.

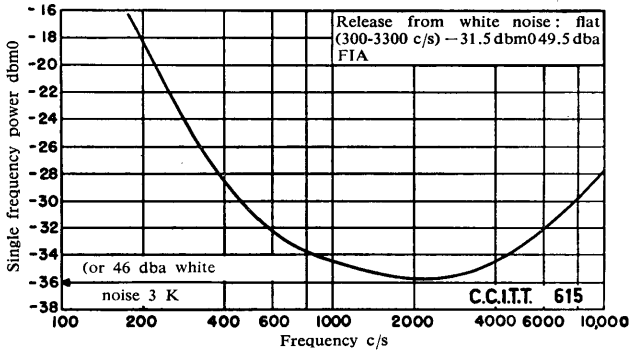


FIGURE 5. — **Release characteristic (tone applied to only one direction of transmission)**

6. *Release time*

The release time should span any momentary transmission drop-out (signal loss in both even and odd sides of the four-wire line is necessary for release) and provide sufficient latitude so that future data systems can easily adapt their signal sequences to sustain disabling. The release must be fast enough to cause a minimum of impairment upon accidental speech disabling or false holding on a noisy trunk. A timed release of 100 ms is the design objective for the tone disabler.

7. *Noise considerations*

Having derived the minimum required sensitivity for the conditioning band and related the sensitivity of the guard band in a manner that will minimize "talk-off", the probability that noise power failing in the guard band might delay or prevent conditioning tone recognition must be explored. With a signal-guard characteristic as shown in Figure 6, total random noise power (flat noise in a 3-kc/s band) of −34 dbm0 will extend attack time by 5%, while −31.5 dbm0 of noise will inhibit entirely disabling of a 2125-c/s conditioning tone −29 dbm0. As mentioned previously, the disabler must also release from the holding mode in the presence of the maximum expected trunk noise. The holding release characteristic of Figure 5 will permit the disabler to transfer to the guard mode with a total noise power input of −31.5 dbm0.

Available telephone plant noise statistics are generally given in terms of dba or dbrn (units which employ frequency weighting designed to measure relative interfering effect to a listening subscriber) and do not indicate the spectral distribution of noise energy or even the r.m.s. noise power. Assuming flat noise within a 3-kc/s voice band, a line noise of −32 dbm0 or greater can be expected at the disabler only 1% of the time. However, since both even and odd transmission paths are sampled, the disabler will incur some noise penalty.

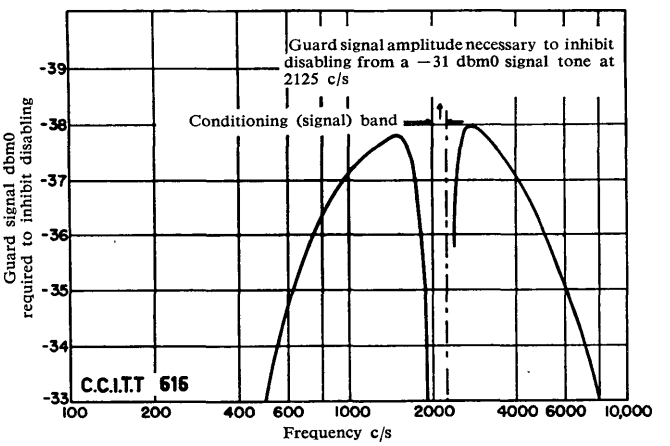


FIGURE 6. — **Guard-signal characteristic**

Circuit design description

The tone disabler includes a hybrid mixing circuit, a four-stage transistorized a.c. amplifier, a tuned differential detector, a Schmitt trigger, an R-C timing circuit, and a transistor switch controlling a relay which disables the suppressor. A simplified diagram is shown in Figure 7.

The hybrid combiner couples the signals from both even and odd sides of the transmission path to the a.c. amplifier, while maintaining an isolation loss between even and odd sides of at least 50 db in the voice band. The line-bridging impedance is very high so as not to interfere

with echo-suppressor operation. Two pad values are available to compensate for the different transmission level points that may exist at the even and odd paths.

The a.c. transistor amplifier consists of two two-stage feedback amplifiers in tandem, and has a thermistor-compensated feedback loop to improve temperature stability. A gain-control potentiometer allows sensitivity adjustment of about ±6 db. A ripple filter is used to render the amplifier insensitive to battery noise.

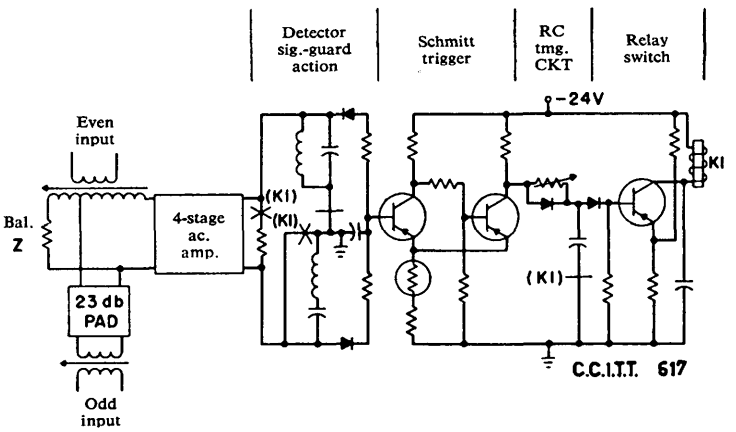


FIGURE 7. — **Tone disabler simplified schematic**

The detector circuit employs an anti-resonant tuned network to develop conditioning signal voltage and a series tuned network to generate guard voltage. These voltages are half-wave rectified, filtered, and polarized so as to oppose each other. A direct relationship between disabler operation and multiple input signals is shown in Figure 6 where the minimum S.F. guard signal necessary to inhibit disabling is plotted against frequency for a constant conditioning tone (2125 c/s) applied at −31 dbm0. Relay contacts control the broadband holding feature by effectively replacing both tuned circuits with a resistive load. Choice of this load resistance determines the holding mode release characteristic which allows disabling to cease when S.F. energy in the voice band drops below −35 dbm0.

The Schmitt trigger has a negative threshold which is temperature-compensated by a thermistor. An input signal in the conditioning band (2000 to 2250 c/s) of −31 dbm0 or greater will cause the Schmitt to operate. On the other hand, with a 31-dbm0 tone at the centre of the conditioning band, and a −36-dbm0 tone applied in the guard band from 700 to 1775 c/s or from 2475 to 3000 c/s, the Schmitt will not trigger.

The operation of the Schmitt trigger initiates a charging cycle in the R-C timing circuit. When the voltage developed across the capacitor exceeds the breakdown voltage of the avalanche diode plus the reverse bias on the transistor, the switch will conduct and close the relay. The attack time can be varied by adjusting the timing resistance in series with the capacitor. If the trigger should revert back to normal before the operation of the relay, the capacitor is quickly discharged through the diode shunting the adjustable timing resistor, thus securing a minimum recycle time.

The release time of 100 ms is achieved by the following : A back contact of relay K1 removes the capacitor from the timing circuit, thus preventing loss of charge upon a momentary signal drop-out that would necessitate recycling through the attack-time interval to re-establish disabling, and the collector of the relay switch transistor is bypassed to ground by a large capacitor to delay the relay release as well as minimize the inductive kick of the relay.

The disabler has been designed to operate from the standard 24-volt positive ground signal battery, used also by the echo suppressor. The maximum current drain is approximately 30 milliamperes. The disabler will operate successfully over an ambient temperature range of 30 to 140 degrees Fahrenheit.

The existing echo suppressor test set can be used to line up and maintain the tone disabler. In addition, attack and release times can be measured.

Initial tests of the disabler, covering the range of signal levels that can be reasonably expected at the suppressor, produced no signal imitations from normal speech or programme material. A more extensive testing programme is in progress which will give a broad statistical sampling of disabler performance as it relates to both voice and data transmission.

Supplement No. 86

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — August 1963)*

DEVICES TO AUTOMATICALLY ORIGINATE AND ANSWER TELEPHONE CALLS FOR DATA TRANSMISSION

1. *Unattended automatic answering equipment*

1.1 In the idle condition there must be no d.c. path between the telephone line connection terminals. The insulation resistance between the terminals and between each terminal and earth should not be less than 5 megohms at 250 volts d.c.

1.2 Before answering, the input impedance to ringing current ($16^2/_3$-25 c/s) should be about 5000 ohms and should include a series capacitor of 2 µF.

1.3 A ringing current detector circuit at the input of the device should operate reliably with any applied voltage between 30 and 70 volts r.m.s. at $16^2/_3$-25 c/s ; and should not be falsely operated by voltage surges—e.g., when the telephone line is momentarily disconnected.

When ringing current is detected, answering should be delayed to allow the telephone bell to ring for at least 10 seconds so that the call may be answered, if desired, by any person who may be present, also so that a caller will experience a similar sequence of events to that on an ordinary call.

1.4 The incoming call should be answered by applying to the line a d.c. circuit not exceeding 150 ohms and having a return loss against 600 ohms of not worse than 15 db. The holding circuit may be part of the input circuit to the data-transmission equipment and not necessarily part of the answering equipment. At the same time a signal should be extended to the data-processing equipment to start readers, punches, etc.

1.5 After applying the answering loop the device should pause for three seconds during which it should be able to detect any telephone supervisory tone used with pay-on-answer call offices (pay tone).

1.5.1 Where no "pay tone" is detected, an identifying message in the form described in 1.6 should be transmitted without further time interval.

1.5.2 Where "pay tone" is detected, the identifying message should be delayed for 12 seconds or until there has been a period of 0.25 second without tone, whichever is the less, and then transmitted.

1.6 A recorded speech message identifying the called telephone should be transmitted as required by 1.5.1 or 1.5.2. The speech transmission should be of good quality, free from noticeable background noise, and should be in the frequency band 300-3500 c/s. A mean output level of the order of 0.25 to 0.5 volt r.m.s. when connected across a 600-ohm non-inductive load should be considered as the optimum for the purposes of design. The identifying message may include a tone in the range 900-2100 c/s to be used as an acknowledge signal by an automatic call originating device as described in 2. This tone to be at the end of the message and to persist for at least 14 seconds and the voltage transmitted to line should not exceed 2 volts peak-to-peak across 600 ohms non-inductive load connected in place of the telephone line as measured on an oscilloscope.

1.7 After transmitting the identifying message, the equipment should connect the data-transmission equipment to line within 5 seconds.

1.8 No indication will be given from the telephone line at the end of a message. The device must therefore be able to determine when transmissions have ended, been interrupted due to a line disconnection or no signals are received within 15 seconds of the end of the identifying message, and revert to the unanswered condition (1.1) within 15 seconds. The recognition of end of message or no signal must not be upset by the presence of noise or telephone supervisory tones—e.g., dial tone at the line connection terminals.

2. *Unattended automatic call origination*

The facilities given below apply to a device to automatically originate calls via automatic telephone exchanges. The equipment would be unsuitable for calls via manual exchanges. The variable time delays on manual exchange calls together with the lack of electrical signals over the line to indicate call progress make it difficult to define facilities to be provided by equipment to automatically set up calls via manual exchanges.

2.1 In the idle condition the electrical conditions at the line terminals should be the same as given in 1.1.

2.2 When a call is to be set up, an electrical indication from the automatic data-processing equipment should cause the sequence of events described in 2.2.1 to 2.2.7 to be followed.

2.2.1 The line should be looped by a circuit of d.c. resistance not greater than 150 ohms measured at the line terminals of the equipment.

2.2.2 Five to 10 seconds after the connection of the d.c. circuit as required by 2.2.1 the called telephone number should be signalled to the exchange by loop disconnect pulses.

The pulse speed should be between 9 and 11 pulses per second and each pulse should consist of a make and break period where the break period should be 63 to 70% of the time of the whole pulse. The loop resistance measured at the line terminals of the apparatus should be less than 1 ohm during the make period and greater than 5 megohms during the break period. The pulsing contacts must be free from bounce, and be provided with a spark quench circuit consisting of 2 μF in series with 30 ohms.

Between each digit forming the called number there should be an interval of not less than 800 ms or greater than 4 seconds.

2.2.3 The apparatus should be capable of transmitting a number consisting of any required number of digits.

2.2.4 On completion of pulsing out the required number the device should provide a line hold loop not exceeding 150 ohms d.c. resistance, and having a return loss against 600 ohms of not worse than 15 db, and prepare to recognize an acknowledgement signal. The holding loop may be provided by the data-transmission equipment as detailed in paragraph 1.4 if required.

The equipment should also commence transmitting a recorded speech message, conforming to the transmission requirements of paragraph 1.6, identifying the calling telephone line. This message should not last more than 10 seconds and should be repeated until an acknowledgement signal is detected, or for 60 seconds, whichever is the lesser time. An interval of not more than 2 seconds should occur between each repetition of the identifying message to allow the equipment to detect an acknowledgement signal.

Note. — In paragraph 1.6 it is suggested that the identifying message from automatic answering equipment should include a tone to be used as an acknowledge signal to an automatic call originating device. Where an automatically originated call is to be answered manually, the receive terminal equipment will need to include facilities to transmit an acknowledge signal that may be identified by the calling station.

2.2.5 The equipment should be capable of distinguishing between line or switching noises and also telephone supervisory tones viz. ringing tone, number unobtainable, engaged tone, and the expected acknowledge signal even when a frequency error of up to 5 c/s has occurred over the line.

2.2.6 In the event of no acknowledge signal being detected within 60 seconds of completion of pulsing, the equipment should release the call by reverting to the idle condition described in paragraph 2.1. Not more than three successive attempts should be made to set up a call to a particular telephone number.

2.2.7 The apparatus should also revert to the idle condition within 15 seconds of the indication from the data-processing equipment of end of transmission or line interrupted.

Supplement No. 87

FEDERAL REPUBLIC OF GERMANY — *(Extract from contribution COM Sp. A/No. 78 — August 1963)*

FULLY AUTOMATIC ORIGINATING AND ANSWERING OF CALLS

A. *Attended operation*

Calls are established in the same way as in normal telephone traffic. In order to prevent the simultaneous beginning of data transmission at both stations, one of them must be the controlling station. This should be the calling station, because it is paying the charges and is mostly attended. The called station should be as passive as possible. Upon communication by telephone it shall be the first to change over to "data". When the receiver is replaced, it sends a "ready" signal to the calling subscriber, which enables the latter to recognize the change-over. The "ready" signal should be sent only upon the replacement of the receiver so as to prevent that this will be forgotten at the end of the data transmission and the line remain engaged.

Upon reception of the "ready" signal the calling subscriber also changes over to "data" and can begin with the transmission of data or can cause the called station to commence with sending data by transmitting an agreed identification signal.

At the end of data transmission both stations should inevitably switch back to telephony and release the connection.

After completion of data transmission and prior to the release of the connection, there should be the possibility of sending an audible signal for several seconds so as to allow the subscribers to talk with each other.

B. *Unattended called station*

Considering that a data station can be dialled by all subscribers of the network, even by non-data stations, the unattended station must be provided with an automatic answering device which supplies information by a text being comprehensible to humans: e.g. number of the station, station at present switched to "data", number to be dialled, etc.

According as the answering unit is connected immediately after the incoming of the call or only when data transmission has not taken place, one can discriminate between the following two procedures:

1. *Answering device answers call*

The terminal equipment must be ready for operation before the station can be changed over to unattended operation. The automatic answering device answers any incoming call. At the end of the announcement the called station switches to "data" while sending the "ready" signal as in the case of attended operation (see under item A). If, within a specified period of time, no signal is received from the calling station, the unattended station shall disconnect itself.

2. *Answering device answers only in exceptional cases*

The terminal equipment must be ready for operation before the station can be changed over to unattended operation. An incoming call is first answered by the "ready" signal (as under A), inviting the calling data subscriber to replace the receiver. After a specified period of time, the "ready" signal is disconnected and an agreed identification signal, only interpretable by data equipment, is sent, allowing the calling station to recognize the establishment of the desired communication. Then the procedure is the same as that outlined under A—i.e., the calling station begins with the transmission of data or causes the called station to commence with sending the data by transmitting an agreed identification signal.

If, within a certain period of time, no signal is received from the calling station upon transmission of the identification signal, the answering device is switched in by the called station. Upon transmission of the announcement, the unattended station disconnects itself.

The advantage in case B1 is that it requires less equipment since the announcement of the answering device allows the calling subscriber to identify the called station. A subscriber who dialled the wrong number will immediately be informed of his error.

Method B2 is useful in all cases where an exchange of data immediately upon the establishment of the connection is essential. The station set up according to B2 can straightaway interwork with an unattached calling station.

At manual P.B.X.s each incoming call is answered by the operator, so that no automatic answering device is required.

Supplement No. 88

FRANCE — *(Contribution COM Sp. A/No. 80 — August 1963)*

DATA-TRANSMISSION EQUIPMENT WITH AUTOMATIC RESPONSE OPERATING OVER A SWITCHED TELEPHONE NETWORK

The telex network already offers the possibility of sending a message to a correspondent in the absence of anyone at the receiving end. It would be useful if a data-transmission service, using the general telephone network, were to offer similar facilities to its subscribers. In this case, it would be necessary to equip the telephone lines associated with data-transmission apparatus with control and switching devices similar to the control units in teleprinters connected to the telex network.

A. *Basic requirements*

1. First, these systems should be so conceived that no change whatsoever is required in the telephone exchanges or network. They should be adapted to the network as it now exists.

2. The subscriber's equipment in the data-transmission service should, in principle, include a telephone station which can be used for telephone calls in the usual way when the data-transmission apparatus is not in use.

3. It should be possible to set up telephone calls in the normal way, the caller making the same operations as are required to reach an ordinary subscriber and proceeding to special operations only after the call is established.

4. These special operations for the transmission of data should have no effect on the behaviour of the exchanges, which play only a passive role until the call is cleared. The clearing of these calls should be received by the exchanges in the same way as the clearing of an ordinary call.

B. *Desirable facilities*

1. *Automatic response*

The receipt of an ordinary telephone call current should automatically produce, so far as the exchange is concerned, the same operation at the called station as the unhooking of an ordinary telephone handset, connecting the called station with the caller and keeping it under his control.

This "automatic response", of course, sets the charge meter in motion even when the caller tries to make a telephone call in the absence of the called person.

Note. — "Automatic response" systems used to be forbidden in the French telephone network. This is no longer the case, and the French Administration has already authorized devices enabling spoken messages to be registered in the called person's absence. An automatic response system is being considered also for a facsimile transmission service in the telephone network. This question received special examination by C.C.I.T.T. Study Group XIV (see Volume VII). This point deserves attention, for the problem raised presents many analogies with the one with which we are concerned in this contribution.

2. *Identification*

This problem is the logical consequence of automatic response : the sending operator should be able to ascertain the identity of the called station to be sure that the exchange of data does take place with the proper correspondent. The problem was solved in telex by introducing the answer-back unit, and in this case also it is essential that the data-transmission service afford its subscribers the same safeguards.

We might consider (as was done in the facsimile service—see note above) an audible answer-back signal, previously recorded and emitted automatically by the called station and set off by the automatic response. This is a simple system, but has the disadvantage that, unlike telex, it leaves no material record for identification purposes.

If this material record is considered essential, it would be necessary to provide for receipt of the answer-back signal on a special stand enabling it to be displayed immediately, which in most cases (unlike telex) would exclude the use of the stand provided for the receipt of data.

3. *Telephone switching to data transmission and vice versa*

The sending operator should be able to transfer her own equipment and the equipment at the other end to the data-transmission position.

It should then be possible for an operator at either end to interrupt the transmission of data in order to revert to the telephone position and to actuate at the other end a device permitting an operator to be called.

It should be possible for an operator to be called automatically in case of breakdown or when any of the data-transmission equipment is not available.

Finally, it should be possible, following a telephone call or conversation, to resume the transmission of data by making the original operations.

It is understood that all these operations must be possible without clearing the established telephone connection and without having to repeat the telephone call and the selections involved.

4. *Switching of data transmission to data reception and vice versa*

Some users may wish to have either transmitting equipment or receiving equipment only. The manual operation for switching to data transmission should thus be designed either to start the transmission of data from local equipment to the corresponding equipment or to start reception by the local equipment of data already recorded in the corresponding equipment.

Simplified equipments may be considered for these users.

Usually, stations have to be able to transmit and receive data alternately. In that case, the operator at either end will have to be able to switch manually, as desired, from transmission to reception, or vice versa. For instance, he should be able to obtain replies to several different successive questions.

5. *Secrecy*

This question is bound to arise from the introduction of data reception in a public switched network. This kind of operation requires special secrecy precautions, since subscribers must be prevented from taking advantage of the procedure for the fraudulent reception of information which is not intended for them. For this purpose, steps have to be taken to prevent transmission in the absence of an operator unless it is guaranteed by the reception of a special set of signals which is allocated to the station in question and kept secret.

6. *Clearance*

It should be possible for a telephone connection to be cleared from either end by manual means, whether the position is telephonic or data transmission.

C. *Technical characteristics to be standardized—Signalling*

If we want to exploit the possibilities of the general network to the full—i.e., to enable any two subscribers in the data-transmission service to exchange information on the sole condition that the modems with which they are equipped are compatible—the basic requirement is to standardize the signalling used to control the operations mentioned above.

The French Administration has made a study of a data-transmission system with automatic response which permits the sending of information in both directions alternately. This system has a sending and a return frequency-division multiplex channel.

The system is based on the following principles :

1. Once the telephone connection has been established in the usual way at the initiative of the caller, receipt of the call current sent by the exchange to the called station automatically ensures looping of the line and, if necessary, emission of the answer-back signal.

2. If an operator wishes to transmit data, he makes a manual "transmission" operation which switches his station to the data-transmission position ; the replacing of the telephone handset then sets off a characteristic signal on the line in the sending channel. At the other end, receipt of this signal causes the equipment to switch to the data-reception position.

3. When an operator wishes to receive information, he makes a manual "reception" operation which prepares his station to receive data ; the replacement of the telephone handset then sets off a characteristic signal on the line in the return channel. At the other end, receipt of this signal causes the equipment to switch to the data-transmission position.

Both latter operations may be made by an operator at either the sending or the receiving end.

The characteristic signals may be either a characteristic (low) frequency, if the channel is frequency-modulated, or the carrier frequency, if the channel is phase-modulated.

4. So long as the transmission of data continues, a signal must be emitted constantly on the sending and the return channels.

This happens in any case on the sending channel, whether it is frequency- or phase-modulated, and on the return channel when it is used to transmit control signals.

5. *Return to the telephonic position*

Removal of the telephone handset at either end causes the signals emitted at the same end to disappear.

a) Disappearance of the signals on the return channel causes the operator of the equipment in the data-transmission position to call.

b) Disappearance of the signals on the sending channel causes the operator of the equipment in the data-reception position to call.

This call signalling ceases when the telephone handset is removed.

Either end may thus interrupt the transmission of data at any moment and, by cutting off its own emission, cause reversion to the telephonic position. Similar signalling by the disappearance of frequencies is provided in the event that anything goes wrong.

6. *Changing the direction of data transmission*

The direction of data transmission is changed by carrying out at one of the extremities the operation which transfers the equipment to the telephonic position and then the "transmission" or "reception" operation described in 2 and 3 ; it is not necessary to lift the handset at the other end.

7. *Clearance*

Clearance is obtained :

— in a telephone call, by simply hanging up ;

— in the operator's calling cycle, when the caller hangs up and thus sets off a timing-switch at the other end.

In the data-transmission position, clearance may be obtained by transferring to the telephonic position and using one of the above-mentioned procedures. To avoid the delay caused by the timing-switch, clearance must be effected by a special manual operation which can be made at either end and which sends a special signal in line, to eliminate the loop at the other end ; the controlling station is automatically cleared when it observes that the signals received have disappeared. In this case, a special clearance signal must be selected which differs from the signals used to transfer to data transmission or data reception.

Supplement No. 89

PHILIPS TELECOMMUNICATION INDUSTRIES — *(Contribution COM Sp. A/No. 95 — October 1963)*

DATA TRANSMISSION VIA THE PUBLIC TELEPHONE NETWORK

This supplement describes the principles upon which we have based the design of a device linking data equipment to the switched telephone network. The device, which has been labelled "DIGITPHONE", is to be located at telephone subscribers' premises. It permits, after establishment of a normal telephone connection in the normal manner, switching the connection to the data equipment at both ends in order to exchange data.

The following principles have been observed:

Principle 1. — The device is concerned only with the telecommunication aspects of the transfer process.

Note. — These telecommunication aspects cover an area posing a number of specific problems which are on the one hand quite universal for any switched telephone connection and on the other hand separable from aspects such as error correction, digital identification of the subscribers, etc. The interface of the device has been selected accordingly.

Principle 2. — The device operates in conjunction with a standard type subscriber's telephone set with which the connection can be established, permitting a verbal contact between operators at both ends before data transfer is to be started.

Note 1. — The device avoids using any of the functions available in the subscriber's telephone set so as to permit the use of any type of set.

Note 2. — Both operators should press their "DATA" buttons in order to prepare for the transfer of data. When one of them fails to do so the device will automatically switch the connection back to telephone.

Principle 3. — By the operation of a key the operator may switch the terminal to the unattended mode. It will then answer calls without human action.

Note 1. — The position of the relevant switch indicates the mode of operation.

Note 2. — Both in the attended and in the unattended mode a lamp indicates that the transfer of data is in progress.

Note 3. — No bell will be rung when the unattended station is called as this would invite unauthorized interference with the automatic answering procedure.

Principle 4. — The device permits both-way data traffic, i.e. traffic in either direction alternately.

Note 1. — The data equipment indicates the availability of data to be transferred by lodging a "REQUEST TO SEND" signal with the device.

Note 2. — The device determines when the transmission of data may commence and advises the data equipment accordingly with a "PROCEED TO SEND" signal.

Note 3. — The length and contents of the transmission blocks are determined solely by the sending data equipment and not by the communication equipment.

Note 4. — Upon completion of a block the sending data equipment withdraws its "REQUEST TO SEND" signal whereupon the device invites the far end to send. After a short pause the far end terminal may send either a request for repeat when an error has been detected or a confirmation of correct reception together with or without a message.

Note 5. — This operational procedure obviates the disabling of echo suppressors in the circuit.

Principle 5. — Upon the establishment of the connection the device is responsible for the supervision of the circuit.

Note 1. — It performs the checking function independent of the transfer of data as such.

Note 2. — Before inviting the data equipment to send, the device checks the continuity of the communication facility in both directions.

Note 3. — During transmission the checking is performed in between the blocks of information and their associated confirmations. A block length of 5 seconds is regarded as a practical figure.

Note 4. — In order to permit the device to perform its checking function with continuous transmissions of longer than 5 seconds it is desirable that the sending data equipment interrupts its transmission at regular intervals.

Principle 6. — When no further data are tendered for transmission in either direction the devic should automatically switch back to telephone operation after a specified period.

Note 1. — When neither traffic nor confirmation is to be expected from the receiving-data equipment for a period longer than specified the connection should be maintained by a different means. For this purpose the data-receiving equipment might give a signal to the device indicating the reception of valid data.

Note 2. — Depending on the condition of the telephone sets at both ends (handsets or off the hooks) and the release procedure in the telephone exchanges, the connection will be released or held.

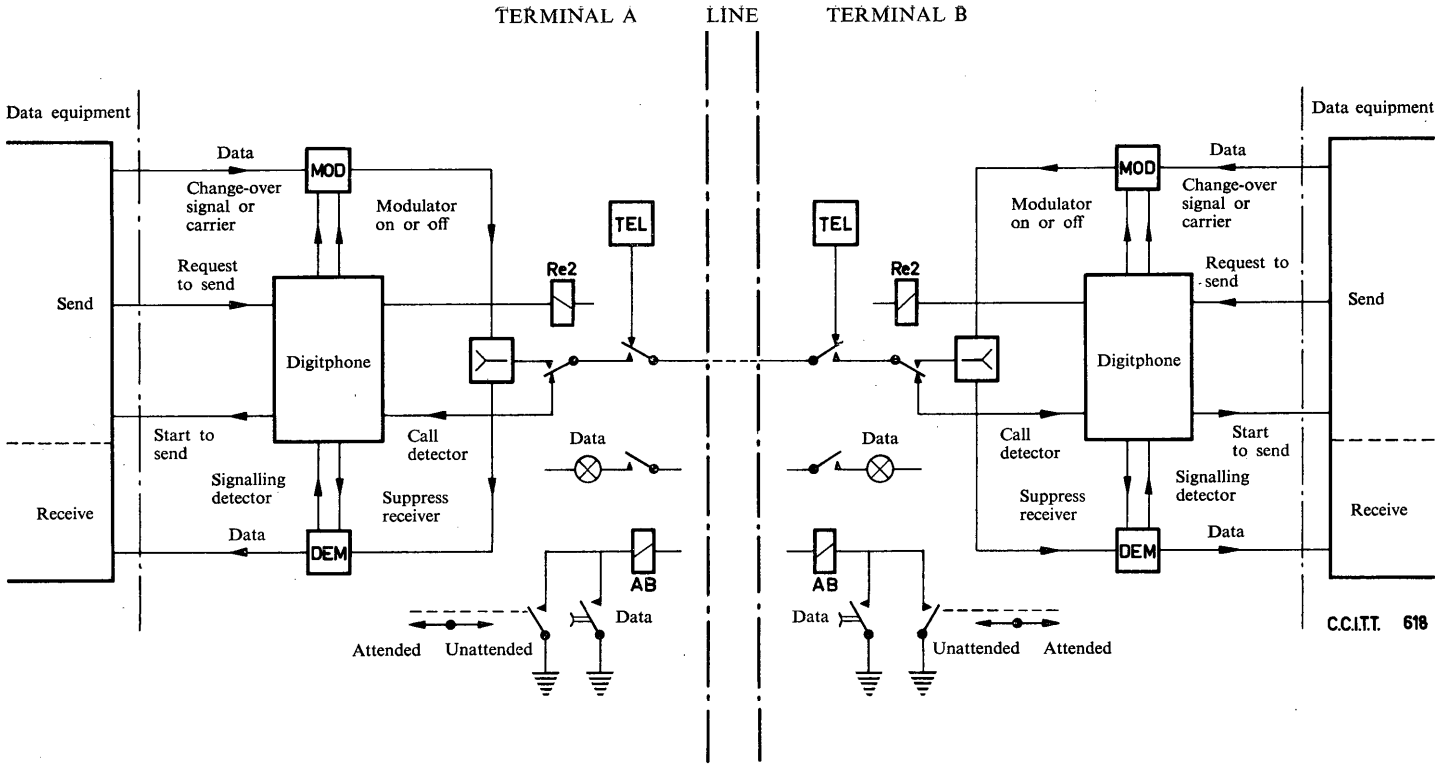
Note 3. — In case the handsets are off the hooks, the operators at both ends will be informed of the switch-back by the extinction of a lamp.

Note 4. — As an option their attention may be called acoustically.

Note 5. — A similar switching back to telephone will occur when the connection is interrupted in some way (see Principle 5) or when some defect occurs in the modem.

Note 6. — The operators may restart the data transfer at will.

A block diagram of the "DIGITPHONE" system is given opposite.



Digitphone unit with data equipment and modem

Supplement No. 90

AMERICAN TELEPHONE AND TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A/No. 75 — August 1963)*

REVIEW OF THE AUTOMATIC CALLING PROGRAMME FOR SWITCHED DATA COMMUNICATION SERVICES

Introduction

Many applications for switched data communication services (data services that permit data communication on the switched telephone network) require an attendant only during the call set-up process. The availability of automatic calling for such applications will permit completely automatic switched data communication services. Data-processing centres may gather data automatically and then process and redistribute these data. In addition to normal work-day operation, this might be done at night or at other times when telephone rates are lower. Automatic calling will also permit use of the switched telephone network by alarm-reporting systems.

The unit intended to automatically originate switched data calls is an automatic calling unit (ACU). It can perform all the functions usually performed by an attendant in originating a data call. This includes putting the station in the "off-hook" condition, detecting a dial go-ahead signal, dialling, determining whether or not the call has been established properly, and placing the data set in the data mode. Also, the ACU transfers the telephone line between itself and the data set in order to exclude the data set during the time that the ACU is performing its functions.

Factors affecting the design of automatic calling units

A major factor which influences the attractiveness of switched data services is the ratio of data-message length to call set-up time—the time from when the calling station goes "off-hook" until it is ready to transmit or receive data. As this ratio becomes larger, the percentage of the time that the line is in a condition for actual data communication is increased; therefore, fewer telephone lines are needed to handle a given amount of data (Figure 1). Minimizing the call set-up time enhances the attractiveness of switched data communication services. This reduction is especially important in applications where short messages are transmitted. Calling units should facilitate keeping the set-up time as short as possible.

Another factor affecting design of calling units is the requirement that an ACU perform the call set-up functions usually performed by an attendant. In addition to functions such as detecting dial tone and dialling, the ACU must handle the following possible call outcomes:

1. Call successfully established.
2. Re-order or intercept.
3. Called station busy.

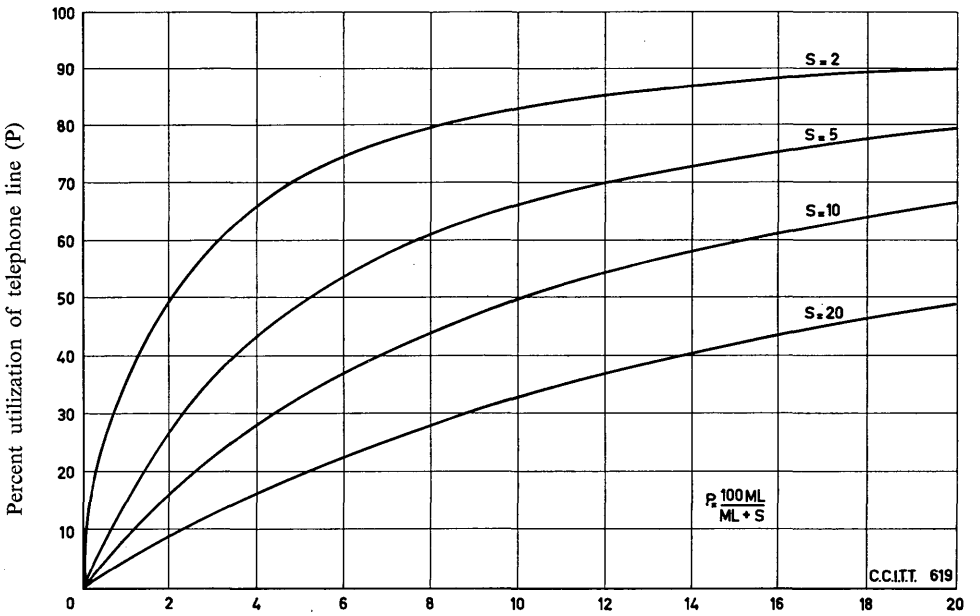


FIGURE 1. — **Utilization of the telephone line for various set-up times (assumes message transmission throughout the time that the line is prepared for it)**

4. Called station does not respond to ringing.
5. Wrong number.
6. Prolonged silence—a "high-and-dry" call.

The steps in the call set-up process along with the possible call outcomes are shown in Figure 2.

Automatic calling unit operation and features

Call originating data stations normally consist of business machine equipment and telephone company equipment. The interconnection of this equipment is shown in Figure 3a. The addition of an ACU yields the station configuration shown in Figure 3b.

To perform the functions usually performed by an attendant in manually originating a data call, the ACU operates as follows :

In response to an order from the business machine, the ACU transfers the telephone line from the associated data set and gives an "off-hook" indication to the central office. After detecting the dial go-ahead signal, the ACU passes the telephone number (which is either internally stored or is presented by a business machine) to the central office. Upon completion of dialling, the ACU waits for the call to be answered properly. If there is a valid answer, the ACU detects

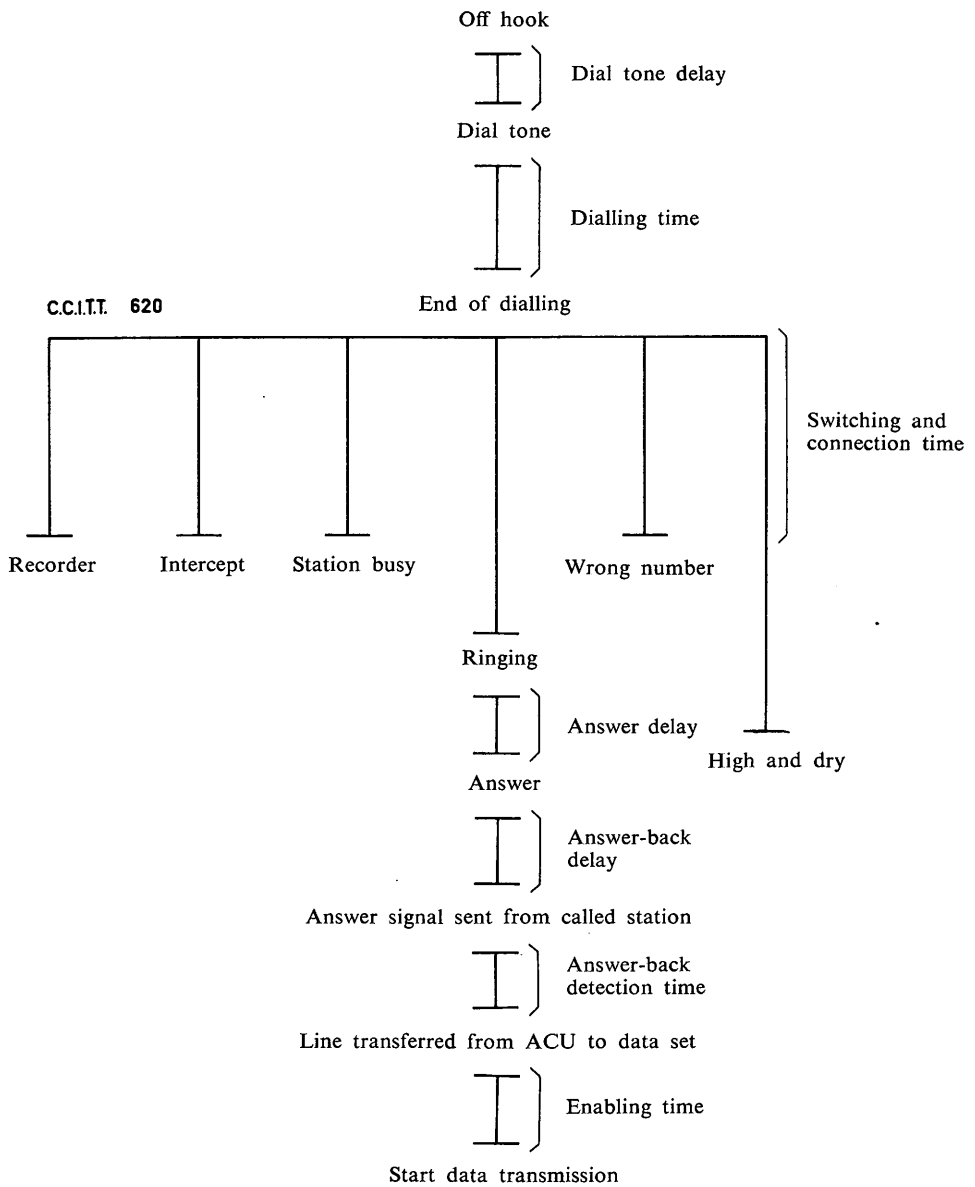


FIGURE 2. — **Set-up time intervals and possible outcomes for a data call**

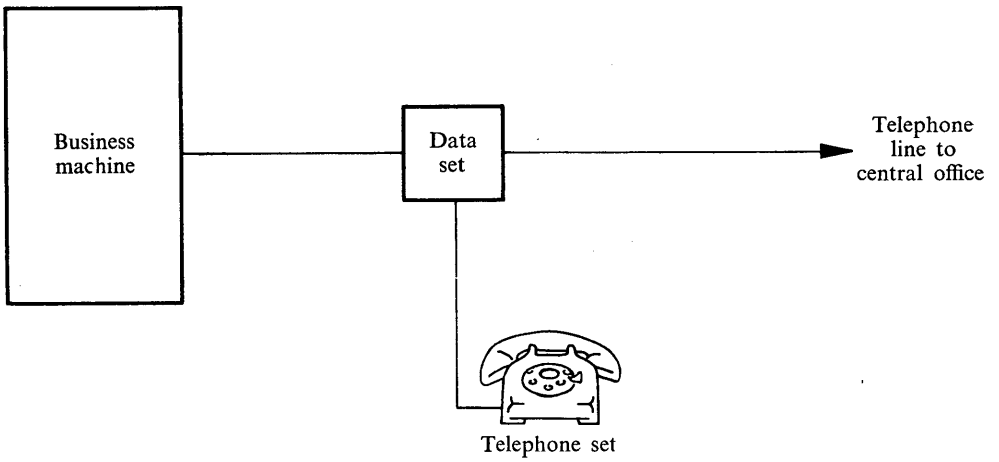


FIGURE 3a. — **Normal data station arrangement**

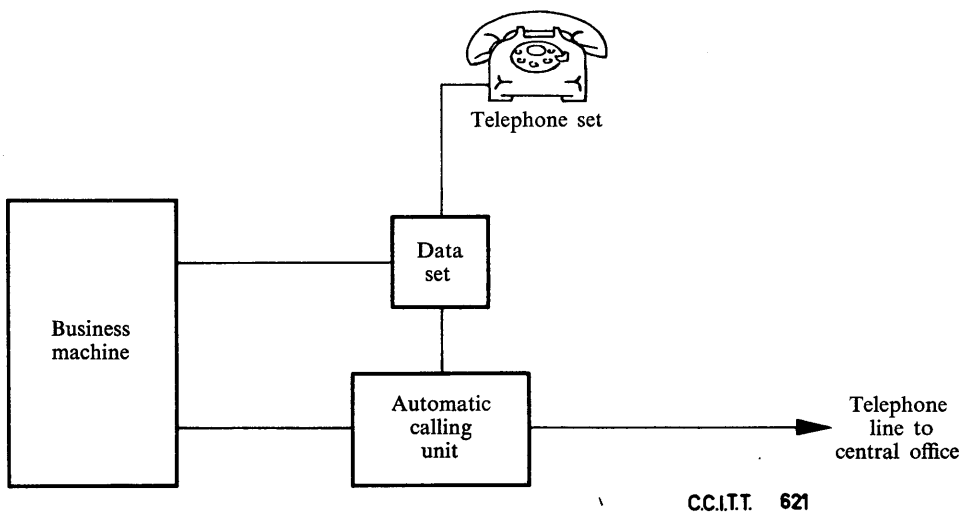


FIGURE 3b. — **Data station arrangement with automatic calling unit**

an answer signal, returns the telephone line to the data set, and places the data set in the data mode. In any other event the ACU gives an indication to the business machine that the call has not been established successfully. Alternately the ACU returns the line to the data set upon receiving an EON (end of number) code immediately after passing the last digit to the central office.

In this mode of operation the data set in conjunction with the business machine must determine whether the call is completed properly.

A calling unit prevents an automatic origination while the associated telephone line is in use on an incoming or manually placed data call. After termination of any call, the calling unit delays a new automatic origination about one second to allow the telephone plant to recognize and take action on the termination.

The telephone plant has been designed to take advantage of conversation and human-ear characteristics. The combination of tones used in dial tone is an example of this. Dial tone is distinctive to the human ear even though the individual tones are not held to close tolerances. Consequently, except in No. 5 offices which have been specially adapted, dial tone varies somewhat through the telephone plant. This variation makes reliable machine detection of dial tone unpractical. Similarly, machine detection of call progress signals (signals such as audible ring and busy, that indicate the end status of the network's attempt to complete the call) is unpractical, except where the call routing is in accordance with the WADS switching plan.

If the EON code is not used, the ACU must determine whether the call has been completed properly; therefore, the called station must answer with a prescribed signal. The need for this signal and the desirability for the called station to answer quickly generally make automatic answering both necessary and attractive at the called station. If the call is not completed properly, the ACU must handle one of the situations listed earlier (also shown in Figure 2) and give an indication to the business machine. This indication is a suggestion to the business machine to abandon the call and try again later.

The ACU must not mistake call progress signals, voice answers, or line noises for valid answers. This requires the use of a very selective answer detector in the ACU.

Types of calling units

Applications where automatic calling will be used may be classified according to the number of telephone numbers it is necessary to call. The data-gathering example cited earlier may require the capability of calling any number on the switched network or some limited repertory of numbers. On the other hand, an alarm-reporting station will call only one number. Availability of these three calling capabilities—any number, limited repertory, or one number—should provide attractive service offerings for most automatic calling applications.

Brief discussions of the three types of ACU follow. A more detailed discussion of ACU operation, including a listing of interchange leads and their functions, is given in the Appendix.

1. *Any number*

It is not feasible to store all possible telephone numbers or even a large variable list of numbers in an ACU. Unlimited calling capability will require number storage external to the ACU. This can best be handled in the business machine using, for example, punched cards, perforated paper tape, or a computer memory.

The dialling information should be passed from the business machine to the ACU by a method that minimizes the over-all data system costs for a broad range of applications. Such a method will best serve customers using the data service supplied jointly by a business machine company and the telephone company. The cost of the ACU will be increased in proportion to the memory

capability it must have. If the ACU delivers one digit of the telephone number to the central office before receiving the next digit from the business machine, minimum memory is required. This digit-by-digit procedure for passing the telephone number from the business machine to the ACU should not affect the cost of the business machine.

It is unpractical to design a specific ACU for each application, so adequate flexibility must be designed into the ACU. Passing the digits in serial form on one lead in the manner that data are passed from business machines through serial data sets would require the ACU to have a rather complex converter if different clocks, codes and data speeds were encountered. It appears that any such conversion could be more economically performed in the business machine. Whereas adoption of any serial method for passing the digits would require some storage in every ACU and parallel-to-serial translation for some business machines, adoption of a serial method using a fixed clock, code, and data speed might require additional translation in some business machines.

To avoid the foregoing difficulties, a method has been adopted that requires the business machine to pass the digits to the ACU in a four-bit binary code in parallel form. It is felt that adequate flexibility in the ACU and optimum system costs are possible with this method. In addition, presenting the digits in four-bit binary permits immediate compatibility with business machines that will use the ASCII (American Standard Code for Information Interchange) seven-level code.

In this method a digit-present lead controlled by the business machine tells the ACU when to read the states of the digit leads. The procedure for passing digits between the business machine and calling unit is :

> The ACU turns its digit-request lead on when it is ready to receive a digit. After setting the states of the four-digit leads, the business machine turns on its digit-present lead. The ACU translates the binary code and delivers the digit to the central office. The ACU turns off its digit-request lead and then the business machine turns off its digit-present lead and waits for the next digit request from the ACU. The states of the digit leads may be changed whenever the digit-present lead is off. This sequence is repeated until all digits in the telephone number have been delivered to the central office.

This action-reaction type operation obviates the need for timing in the information exchange between the calling unit and the business machine. Whereas many, but not all, business machines operate at microsecond speeds, the ACU operates in the milliseconds speed range. This time incompatibility is eliminated in the method of operation discussed above.

2. *Repertory*

Some applications may require automatic calling to only a limited repertory of numbers. For such applications it may be economical to have the calling unit store the numbers and call them sequentially on successive requests by the business machine. Random access to the numbers would require the business machine to provide information comparable in complexity to that

required by the any-number ACU ; therefore, an any-number ACU would most likely be used with machines capable of providing this information. A sequential repertory ACU will permit a relatively simple business machine at a central location to automatically call several outlying data stations.

Although a development programme for a repertory ACU has not been started, preliminary study is in progress.

3. *One number*

This version of the ACU permits storage of a single telephone number. The number is semi-permanently stored, and can be changed by a telephone company installer. In response to each service request by the business machine, the one-number ACU calls the pre-set number.

Future possibilities for automatic calling

There are several ways in which the attractiveness of automatic calling may be enhanced. Among these are : reduction of dialling time, reduction of switching and connection time, detection of call progress signals, and use of one ACU to originate calls on more than one telephone line.

1. *Dialling time*

When dial pulse signalling at the normal 10 pps (pulses per second) rate is used, the dialling time may make up almost half of the time required to set up a call. An ACU operating at 10 pps takes about 13.5 seconds to dial an average ten-digit number. Reduction of the dialling time is possible if TOUCH-TONE calling is used. A ten-digit number is then delivered to the central office in about one second. A TOUCH-TONE ACU is being developed and should find increased application as TOUCH-TONE central office equipment becomes increasingly available in the telephone plant. All of the No. 5 central offices which are utilized in the WADS switching plan (Reference : Section B100) are now equipped for TOUCH-TONE calling.

2. *Switching and connection time*

Decreased switching and connection time is possible if the number of switching points is reduced and/or the switching speed at each switching point is increased. Presently there is wide variation in these times because of alternate routing and different types of switching offices that may be encountered. Development of electronic central offices similar to the experimental office used in Morris, Illinois, is evidence of possible future reduction in switching times.

3. *Standard call progress signals*

It was noted previously that machine detection of call progress signals on the general switched telephone network is not practical. It is now available on calls routed exclusively within the WADS switching plan. Detection of these signals might reduce the time lost on incompleted calls in some applications.

4. *Multiline automatic calling*

Present plans call for using one calling unit with each automatically originating line on a data station. Fewer calling units would be required on a multiline station if one ACU were switched from one line to another. The possibility of providing a multiline automatic calling unit is being studied.

Conclusion

Automatic calling units make possible completely automatic data communication over the switched network. The three types—any number, repertory, and one number—will permit a broad range of applications. This range might be further broadened through (1) use of TOUCH-TONE calling in an ACU, (2) introduction of faster switching offices and standard call progress signals into the telephone system, and (3) use of one ACU to originate calls on more than one telephone line.

APPENDIX

The three types of Automatic Calling Unit (ACU) are discussed in the order : any number, repertory, and one number. A description of the function of each interchange circuit between the business machine and ACU is given. Except for the process of handling the telephone number to be called, the three types of ACU have similar operating procedures. Because of this similarity of operation, the any-number ACU is discussed in greatest detail and the discussions of the repertory and one-number units only indicate how these units differ from the any-number unit.

Where the interchange circuit signals are voltage levels, the electrical characteristics presented in E.I.A. Standard Proposal 808 (same as proposed for C.C.I.T.T. standardization) will be met. An option permitting contact closures to signal ground for interchange circuit signals will be provided. The signals passed across a particular interface will be uniform for all interchange circuits. That is, there will not be a mixture of voltage level and contact closure signals across the same interface.

I. ANY-NUMBER ACU

A. *Interchange circuits and their functions* (Figure A-1)

1. The following interchange circuits are directed from the business machine to the ACU :

a) *Call request*

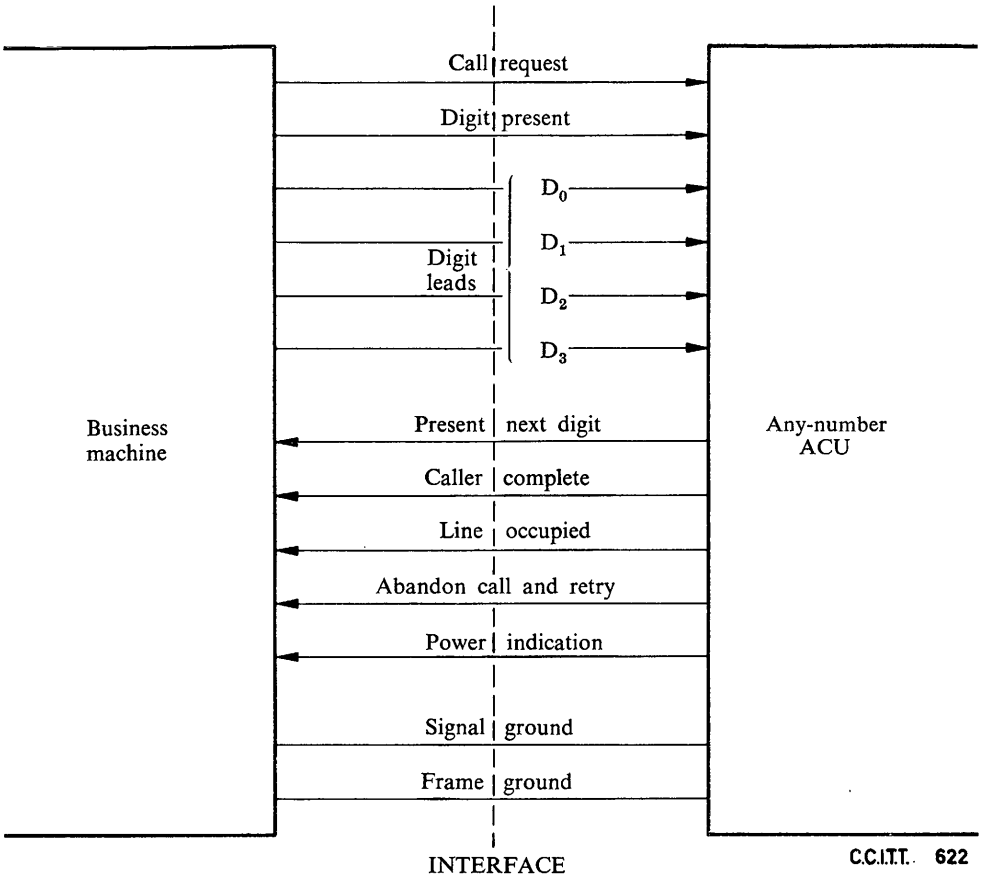
Gives an indication to the ACU that the business machine wants to place a call.

b) *Digit leads*

Give four-bit binary representation of decimal digit to be dialled.

c) *Digit present*

Indicates when the states of the digit leads may be read by the ACU.



Note : Arrows are directed from signal source

FIGURE A-1. — **Interchange circuits — Business machine and any-number ACU**

2. The following interchange circuits are directed from the ACU to the business machine:

a) *Present next digit*

The ON condition indicates that the ACU is ready to receive a digit from the business machine. The OFF condition indicates that the digit has been passed to the central office.

b) *Power indication*

The circuit is ON whenever the ACU is functional from a power standpoint. If this circuit is OFF, the ACU is to be considered inoperative.

c) *Caller complete*

This circuit is ON whenever the data set is in the data mode, and indicates that a call is under control of the business machine through the data set rather than through the ACU.

d) *Line occupied*

An ON indication is given whenever the telephone line associated with the ACU is in use. This is an indication that the ACU will ignore any request for service presented at a time when this lead is already ON. The lead will also be ON when the ACU is in the test mode.

e) *Abandon call and retry*

This circuit is associated with a timer that times out whenever a pre-set interval has elapsed between successive events in the calling procedure. When this lead comes ON, it indicates that an event has not occurred within a reasonable time and the call probably will not go through successfully.

3. Also included are SIGNAL GROUND and FRAME GROUND circuits :

a) *Signal ground*

Provides a common ground reference for interchange circuits. It may be connected to the frame.

b) *Frame ground*

This circuit is to be electrically bonded to the frame of the equipment on each side of the interface. The frames in turn should be appropriately connected to external grounds as required.

B. *Operating procedure*

1. In response to a call request from the business machine, the ACU will :

a) seize the central office line ;

b) recognize a dial go-ahead signal ;

c) request a digit by turning PRESENT NEXT DIGIT ON ;

d) indicate that the associated line is now in use by turning LINE OCCUPIED ON.

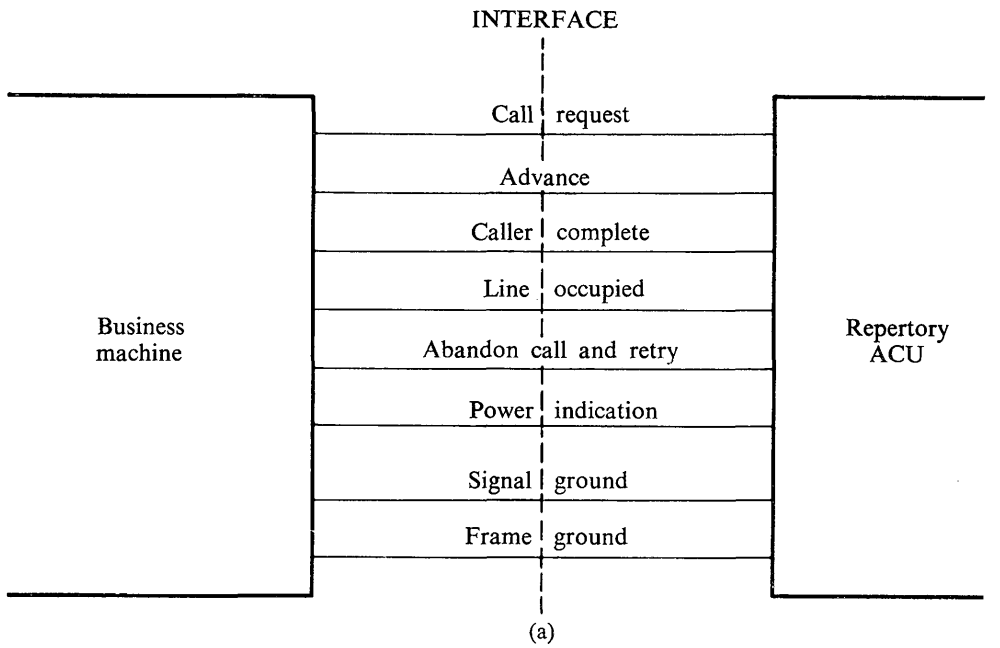
2. In response to DIGIT PRESENT ON the ACU will pulse out the digit indicated by the states of the four-digit leads.

3. The ACU will then turn PRESENT NEXT DIGIT OFF and wait for the business machine to turn DIGIT PRESENT OFF. When DIGIT PRESENT goes OFF, the ACU will request the next digit by turning PRESENT NEXT DIGIT ON again. This sequence continues until all the digits are pulsed out.

4. After each digit is pulsed out, the ACU looks for an answer signal. When the answer signal is detected, the ACU transfers the telephone line back to the data set. Alternately, if the business machine presents an END OF NUMBER (EON) code on the digit leads, the ACU will transfer the telephone line back to the data set.

5. If the data set goes into the data mode when it receives the line, the ACU turns CALLER COMPLETE ON, If not, ABANDON CALL AND RETRY will come ON after time-out.

6. Once the line has been returned to the data set, the call is terminated by the business machine, through the data set in the usual manner. Before the line has been returned to the data set, the call may be terminated by turning CALL REQUEST OFF. Whenever a call is terminated, the ACU ignores further operation of CALL REQUEST until proper inter-call timing has been provided by the ACU.



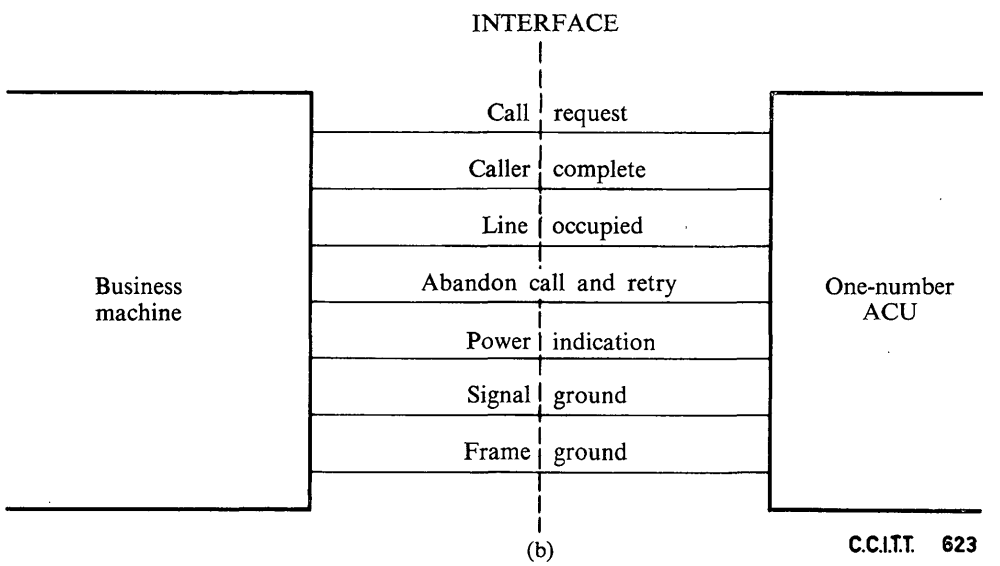


FIGURE A-2. — **Interchange circuits — Business machine and (a) repertory ACU (b) one-number ACU**

II. REPERTORY

With the exception of DIGIT PRESENT, PRESENT NEXT DIGIT, and the four-digit leads, this ACU will probably have the same circuits used in the any-number ACU (Figure A-2a). The internally stored repertory of numbers will be called sequentially. The customer may be provided with an ADVANCE lead which when operated will step the ACU to the next number in the repertory. This will enable the business machine to call the same number again if an attempted call does not go through successfully.

Except for the digit-passing sequence in the any-number ACU (steps 2 and 3 in the operating procedure), the repertory ACU might operate in the same manner. The repertory ACU in response to CALL REQUEST ON will pass one of the stored telephone numbers to the central office. The CALLER COMPLETE ON indication could be used by the business machine as an indication to turn ON the ADVANCE lead and thus prepare the ACU to dial the next number in the repertory on the next call.

III. ONE NUMBER

Again the operation is the same as that of the any-number ACU except for the digit-passing sequence (steps 2 and 3 of the any-number ACU operating procedure). Since there is only one number stored in the ACU, the ADVANCE lead used in the repertory ACU has no utility in this unit. The interchange circuits are shown in Figure A-2b.

Each time CALL REQUEST is made ON, the ACU will call the stored number.

Supplement No. 91

WESTERN UNION TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A /No. 65 — July 1963)*

BROADBAND SWITCHING SERVICE

The Telegraph Company will soon place in operation a switching network of advanced modern design to establish a new and unique service in the field of voice-data communications. This service will be known as the "Western Union Broadband Switching Service". The initial phase will comprise seven switching centres and a number of concentrator units to provide service to about 17 cities. Equipment for this system has been designed for optimum dependability and reliability, and will be capable of automatically connectiong any two subscribers over one of several varieties of transmission channels according to the subscribers' communicative needs. The characteristics of these channels have been chosen to accommodate the various forms of analog and digital communications most likely to be required, and each channel is appropriately equalized at all switching points.

The communications industry determined some decades ago that wide-bandwidth trunk facilities were most efficient for handling heavy traffic, various smaller-sized channel bandwidths being derived from these arterial routes. While these communications paths existed for some time they have not been readily available for general use, especially on a "call-up" basis. The Broadband Switching Service is arranged to permit the customer to choose the type channel he wants for each particular transmission by the simple process of including an extra digit in the push-button selection code utilized to set up the call. When it is desired to initiate a call, the handset is picked up and the subscriber listens for the dial tone. After hearing the dial tone he pushes the digit buttons in proper order corresponding to the voice data number which would be listed in the directory, for example, 437-3-589. In this 7-digit number the first three digits indicate the distant exchange. There may be several exchanges in a large city. The fourth or middle digit selects the bandwidths desired for the particular call. The remaining three digits designate the subscriber in the distant exchange area. Particular subscribers may contact each other by use of an appropriate 3-digit address code instead of the standard 7-digit code. Switching is controlled by means of multifrequency tone combinations. The called subscriber will be alerted to an incoming call by a distinctive tone signal instead of a ringing bell. An indication will be returned to the calling party that the called subscriber is receiving this signal. The party at the distant end will answer in the same manner as he would on a conventional telephone. Transfer to the data mode is accomplished by both parties placing their handsets in the data position and upon completion of the data-transmission portion of each call, customers may return to the voice mode for acknowledgement of transmission or conversation. Billing for this data-voice service will be based upon the bandwidth called up for each connection, distance, and time.

The switching equipment is designed to handle bandwidths of 2000 c/s, 4000 c/s, 8000 c/s, 16 kc/s and 48 kc/s. The channel bandwidths to be provided initially will be the narrower types, the wider bands being made available as demand increases. All channels will be of the "four-wire" or two-path type. The advantages of a four-wire system are elimination of echo and singing distortion, improved quality of the transmission path and, the availability of the return path for parity check, synchronization of data or computer equipment and, if the users so elect, the system will permit simultaneous transmission of data in both directions.

The switching equipment may also be used for "piggy-back" operation. For example, a channel of voice bandwidth could be used for switching and to carry on a conversation while at the same time data would be flowing over a channel of wider bandwidth. For example, a 48-kc/s channel could be supplied between the frequencies of 12 kc/s and 60 kc/s which would be a frequency range convenient for local distribution as well as for switching, and this channel could be switched along with the voice channel and used simultaneously.

Supplement No. 92

UNITED KINGDOM — *(Extract from contribution COM Sp. A/No. 71 — July 1963)*

USE OF CIRCUITS OF VERY LARGE BANDWIDTH

It may be necessary in the future to provide leased circuits of greater bandwidth than that given by a high-quality telephone circuit or by a programme circuit. The use of group and supergroup links has been studied in the United Kingdom and there are already a number of leased group links in use or in course of provision. These links are set up in the basic group B (60-108 kc/s) between group distribution frames in the terminal repeater stations and are extended to the renter's premises on 1.2/4.4-mm coaxial cable specially installed for the purpose.

In the expectation that international leased group links may be required, it would be useful if the C.C.I.T.T. were to frame Recommendations governing their use for data transmission and other applications (e.g. facsimile, multiplex, telephony, etc.).

The following points should be considered:

1. *Bandwidth*

A typical performance for a single group section would give a spread of 3 db over the band 60.6 to 107.7 kc/s.

2. *Group-delay frequency distortion*

This will depend upon the routing of the particular link. Figure 3 shows typical values of group/delay/frequency characteristics for various items of equipment. At least one through group filter, at the input to the main multichannel system, should be assumed. (United Kingdom practice is to fit a through group filter at each terminal g.d.f.)

3. *Noise*

Normal design standards on multichannel systems would lead to wideband noise which may be assumed to have a total hourly-mean power over a 48-kc/s band of:

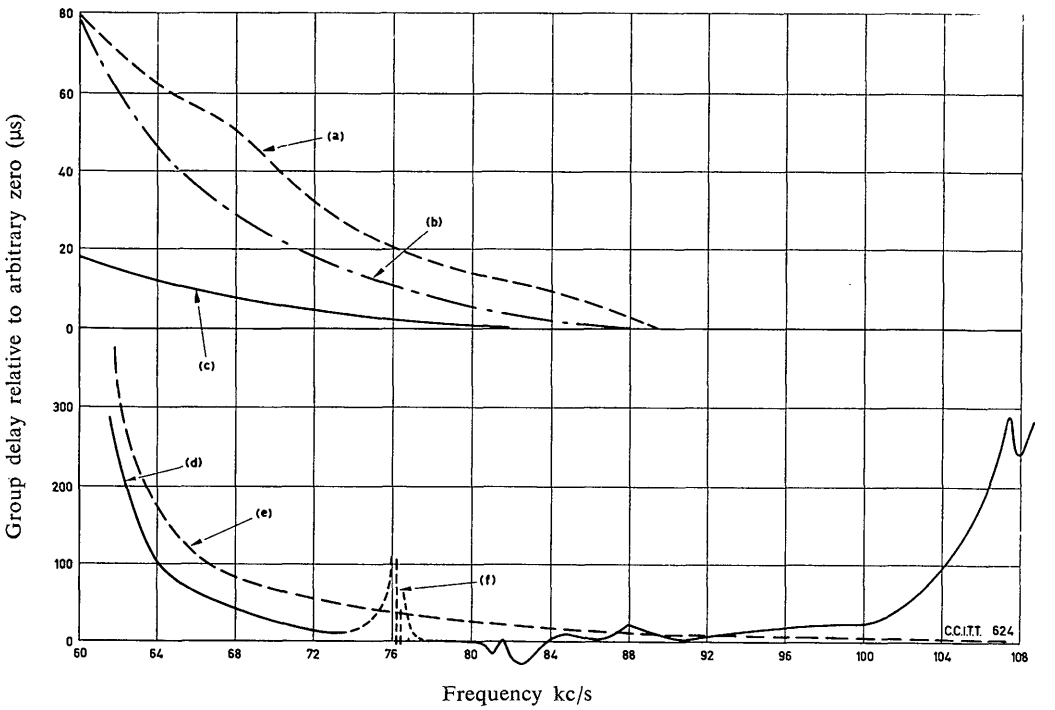
$$-35.5 + 10 \log_{10} \frac{L}{2500} \text{ dbm0}, \qquad L \text{ is length of link in km.}$$

Short-period fluctuations due to the intermodulation component must be expected.

In addition, carrier-leaks and other signals at multiples of 4 kc/s are possible, as a result of incomplete filtration in modulation processes, generally at levels around −50 dbm0, although certain products at much higher levels can occur.

4. *Signal levels*

The group link will have been designed to carry the power corresponding to 12 telephone channels with regard to the statistical distribution of speech and other signal levels; when used for data transmission the power leading should be equivalent. This condition could be assumed



(a) = Effect of 75 miles coaxial line link L.F. cut-off characteristic (transmitting group 5, supergroup 1)
(b) = Effect of one through-supergroup filter on edge group (group 5)
(c) = 1 pair of group and supergroup translations G 5, SG 6
(d) = Example of through-group filter characteristic
(e) = Example of 24-circuit filter characteristic (group 2)
(f) = Dotted section of curve shows effect of a secondary resonance which is present in some filters

FIGURE 3. — **Group delay/frequency characteristics of groups transmitted on typical H.F. cable systems and equipment in the United Kingdom**

to be met if the data signal complied with the following specification (measured at a point of zero transmission relative level) :

— mean power during an hour : −4 dbm0 ;

— instantaneous peak of a complex signal not to exceed the amplitude of a sine-wave of r.m.s. value + 19.5 dbm0 ;

— no sustained single-frequency signal to exceed the amplitude of a sine-wave of r.m.s. value 0 dbm0.

5. *Pilots*

A supergroup reference pilot at 104.08 or 104.14 kc/s could be expected on group 3 in a supergroup. It could be suppressed in the supergroup translating equipment, but the response of the group link would then have a discontinuity. It would seem desirable, in general, to avoid the

use of group 3. A group reference pilot is desirable for maintenance purposes. The C.C.I.T.T. pilot at 84.08 or 84.14 kc/s may not be admissible and some other solution should then be found.

6. *Go-return crosstalk*

The crosstalk ratio between the two directions of transmission may be 50 db; somewhat better values could be achieved with modern equipment throughout.

The possible introduction of PCM transmission systems into the telephone networks has interest for future data-transmission systems. A particular feature which might be considered when preferred modulation rates are discussed is direct access of a data signal to the multiplex path in place of one telephone channel.

Supplement No. 93

CHILE TELEPHONE COMPANY — *(Extract from contribution COM Sp. A/No. 74 — August 1963)*

BROADBAND CIRCUITS (I.E. ABOUT 48 kc/s BANDWIDTH)

It is unlikely that the majority of potential users of broadband circuits will have sufficient traffic to load such a circuit for a large proportion of the working-day; nevertheless, they will probably demand a high information rate to achieve rapid delivery of information or for such a purpose as to avoid storage over long periods, etc. It seems likely, therefore, that users of switched broadband circuits (assuming such circuits to become available) will far exceed users of leased broadband circuits; many of the broadband circuits now leased will be relinquished in favour of a switched broadband service as soon as such a service is established.

For these reasons, proposals for data transmission at high speeds should be planned primarily for application to switched circuits.

This switched service would be used mainly as a means of establishing communication between particular pairs of subscribers or between pairs belonging to definite groups (business organizations, etc.). Some of these groups will require transmission over relatively short distances, others over long distances; also these groups will tend to place a different emphasis on the highest attainable speed, etc., in relation to costly data-processing installations. Nevertheless, the tariffs for broadband switched circuits are likely to be high enough to ensure that subscribers to the switched service will demand at least the information rate obtainable by biternary or CDC transmission, if not the rather greater speeds attainable by VSAM or quaternary phase methods of transmission. It seems, therefore, that simple frequency or phase modulation need not be considered for switched broadband circuits.

The intermodulation problems arising from the concentration of energy into a single frequency when some systems are left in a steady condition will justify keeping the average signal power as low as possible; good noise tolerance will assume even greater importance, therefore, than is the case for medium-speed transmission. For this reason, modulation methods 3 and 5 are likely to be the most suitable. These are:

VSAM with d.c. and low-frequency signals blocked at the input to permit insertion of a carrier frequency pilot to facilitate homodyne reception.

Phase reversal modulation of carrier waves of equal frequency but quadrature phase, these being assembled to provide a quaternary phase signal and separately demodulated by homodyne carriers reconstituted from twin pilot frequencies transmitted with the signal.

However, it is difficult at this time to say that there will be no demand for other systems.

The same methods will serve many users of broadband leased circuits. However, amongst such users some will load their lines so lightly that an information rate of about half the attainable maximum will probably suffice. For these, the all-round merits of simple F.M., with the extra significant conditions attainable, are likely to make it first favourite.

Although it would be premature to make recommendations on the broadband service at the present time, Special Study Group A could usefully come to some tentative conclusions on some of the probable characteristics of that service so as to provide a reasonably firm basis for further study.

The following are our proposals in this direction :

1. Circuits used in this class of service will need to be of two types-:

a) Asymmetric four-wire or six-wire circuits composed of a unidirectional broadband path permanently associated with a bidirectional two-wire or four-wire telephone circuit.

b) Symmetrical six-wire or eight-wire circuits composed of two broadband paths, one in each direction, permanently associated with a bidirectional two-wire or four-wire telephone circuit.

From some points of view it would seem desirable to derive the telephone circuits from the broadband circuits by frequency division, though in the case of asymmetric circuits the broadband and the telephone band paths could not be derived from the same one-way group. Nevertheless, it is considered that they should, in fact, be quite independently produced.

Further, it is considered that switched asymmetric circuits should be provided in two forms so that the broadband path is made available either in the caller-called direction or vice versa.

2. Switched broadband circuits should be designed particularly for unattended operation at the called station.

3. Broadband circuits of the same type and quality should be available for a variety of purposes. Uses foreseen for the switched asymmetric circuit are :

— high-speed data (and telegraph) transmission ;

— high-speed facsimile (and telephoto) ;

— one-way low-speed television ;

and, for the symmetric circuit :

— two-way high-speed data (and telegraph) transmission ;

— two-way high-speed facsimile (and telephoto) ;

— two-way low-speed television ;

— high-grade privately ciphered speech.

The selection of circuits of adequately uniform phase characteristics and the elimination, or replacement, of regulation pilot channels and band-stop filters should be considered in the light of all these and other possible usages of broadband circuits.

Supplement No. 94

AMERICAN TELEPHONE AND TELEGRAPH COMPANY — *(Extract from contribution COM Sp. A/No. 75 — August 1963)*

A DESCRIPTION OF THE BELL SYSTEM'S 40.8-KILOBIT-PER-SECOND DATA-TRANSMISSION SYSTEMS

Abstract

Systems that transmit serial binary digital data at 40.8 kilobits per second are being developed by the Bell System. These are synchronous systems operating under control of a stable timing signal at the transmitter and presenting both recovered data and timing signals at the receiver. They will employ N2-carrier facilities for short-haul transmission and L-carrier facilities for long-haul transmission of 40.8-kb/s data. A voice-frequency co-ordination channel will be provided along with the data channel. These systems are being developed for high-speed data transmission on TELPAK A 48-kc/s bandwidth channels.

Introduction

This supplement describes the Bell System's 40.8-kilobit-per-second wideband data-transmission systems—hereinafter called 40.8-kb/s systems [1]. This is the first such paper and it somewhat precedes the completion of development work. However, its primary purpose is to impart an understanding of both what the systems will comprise and what they will do. Hopefully, the descriptive background established through this paper will be particularly useful if certain arrangements undergo changes in the future and if other papers treat the system design more analytically. The remainder of this introduction notes three things, namely, some factors leading to development of 40.8-kb/s systems, a bandwidth difference between systems, and the first Bell System trial and use of data transmission at approximately 40.8 kb/s. This is followed by a description of the over-all system characteristics and descriptions of system terminals and local loops.

The Bell System uses some of the transmission channels available from its carrier systems for short-haul and long-haul private line services. Until recently, most of these private line channels have been provided as individual voice-bandwidth channels which customers use for transmitting either speech or data. However, because of the techniques of handling voice channels in groups, carrier systems have the potential of providing channels having bandwidths much greater than the voice bandwidth. To utilize this potential, the Bell System has originated the TELPAK concept under which they now provide channels of relatively large bandwidths for private-line use. Channels in the TELPAK A class can have bandwidths up to 48 kc/s and channels in the TELPAK D class can have bandwidths up to 960 kc/s.

[1] The plural (systems) applies because two different carrier systems will be employed.

In making the bandwidth potential of carrier systems available, the Bell System expects to accomplish at least three objectives. These three are : 1) provide customers a saving in the cost of communications ; 2) meet expanding needs for speech and data communications ; and 3) increase the flexibility of private line services by making channels available for simultaneous transmission of speech and high-speed data. As one step toward realizing these objectives, the Bell System is developing standard systems for transmitting speech and digital data together over 48-kc/s bandwidth TELPAK A channels. These systems are becoming known as the 40.8-kb/s data-transmission systems.

Wideband channels for 40.8 kb/s data transmission will be supplied from L-carrier systems for long-haul service and from N2-carrier systems for short-haul service. The maximum number of simultaneous two-way data transmissions possible in each carrier system is 17 in L1-carrier, 59 in L3-carrier and 1 in N2-carrier. From L-carrier a wideband channel will be equivalent to 12 L-carrier voice channels and will be apportioned to accommodate one data channel and one voice-frequency co-ordination channel. From N2-carrier a wideband channel will be equivalent to 7 N-carrier voice channels and will accommodate one data channel. The voice co-ordination channel will be adjacent to the data channel. The wideband N2-carrier channel will cover a frequency band approximately 16% wider than the frequency band covered by the wideband L-carrier channel because of modulation, frequency allotment, and equipment design differences between N and L carrier systems. However, the stated system bandwidth for 40.8-kb/s systems will correspond to the L-carrier channel bandwidth because this is the narrowest of the wideband channels supplied for 40.8-kb/s service.

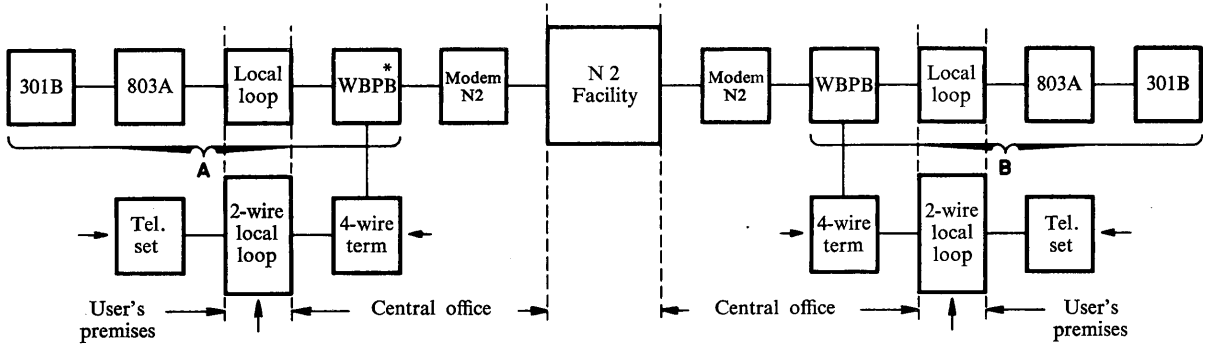
A data link interconnecting the Murray Hill and Holmdel, New Jersey, facilities of Bell Telephone Laboratories was the first trial and application of Bell System service at approximately the 40.8-kb/s speed. This link is still in service and is used to exchange computer data at 42 kb/s through experimental 301A data sets over approximately 20 miles of N2-carrier facilities. Experience gained through this application has influenced the choice of transmission and performance objectives for data transmission at 40.8 kb/s. It has also firmly established the feasibility of data transmission at this speed.

System characteristics

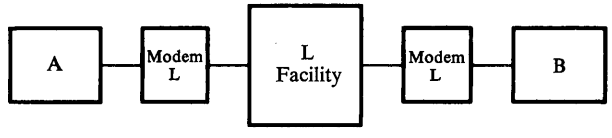
A 40.8-kb/s system provides one voice-frequency co-ordination channel and one wideband data channel for transmission of serial binary digital data between data sources and sinks that operate at 40.8 kb/s. It is characterized by the system layout, speed, timing data input and output, transmit and receive levels, code restrictions, modulation, information encoding, bandwidth, performance, and impairment limits. These are described in the following paragraphs. The characteristics of carrier systems are given in other sections of this present document.

System layout

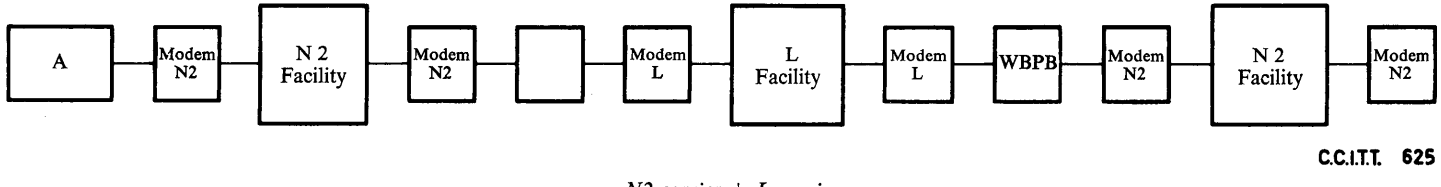
The 40.8-kb/s system layout is similar to other Bell System data system layouts in that data modems are located on a user's premises and local loops connect these modems to central offices where access to transmission facilities is provided. Figure 1 shows three arrangements of terminals and facilities that define in block diagram from 40.8-kb/s systems. It indicates that N2-carrier



N2-carrier only



L-carrier only



N2-carrier + L-carrier

* WBPB = Wideband patch bay

FIGURE 1. — **Block diagrams of 40.8-kb/s systems**

and L-carrier systems can be used in combination and shows that a voice-frequency co-ordination channel is provided along with the wideband data channel.

System lengths vary with the type of carrier facilities used and are limited primarily by the build-up of noise, frequency offset and envelope delay distortion. The maximum length that N-carrier systems are designed for is 200 miles. With careful engineering it should be possible to transmit 40.8-kb/s data over this length. However, it is expected that the longer 40.8-kb/s systems using only N2-carrier will be nearer 100 miles. The number of individual N2 systems used in series may be limited to two or three depending upon the above impairments. When only L-carrier systems are used, no more than five group connectors—meaning no more than six L systems—may be employed in series principally because of envelope delay distortion build-up from the group connectors. Cross-country transmission over L-carrier should be possible under these conditions. Where both N2 and L- carriers are employed in a system, it is expected that no more than two N2-carrier links will be used and these are to be used as end links. Future arrangements may provide for transmitting 40.8-kb/s data on equalized non-loaded cable pairs of certain sizes over distances up to 30 or 50 miles.

The numbers of various terminals and local loops required for a system are as follows. Two 301B data sets, two 803A data auxiliary sets, four local loops, and at least two wideband patch bays are always required for each system. Arrangements where more than two wideband patch bays are required, and the required quantities of modified carrier terminals are described now —first for the case where only L-carrier is employed, second for the case where only N2-carrier is employed, and third for the case where N2-carrier end links are employed in combination with L-carrier. When only L-carrier is employed (one or more systems), two modified L terminals and two wideband patch bays are required. Transfers from one L system to another are achieved by using L group connectors which do not require bringing data signals down to L-carrier baseband. Transfers are made in the group band of frequencies. When only N2-carrier is employed, 2 M modified N2 terminals and $M + 1$ wideband patch bays—where M is the number of N2 systems employed—are required. Transfers between N2 systems require bringing data signals down to N2-carrier baseband. When N2-carrier end links are employed with L-carrier, four modified N2 terminals, two modified L terminals, and four wideband patch bays are required. Transfers between N2 and L systems are made at carrier baseband (10 to 51 kc/s).

For the voice co-ordination channel, two telephone sets, two two-wire local loops, two four-wire terminating sets with signalling equipment, and voice patch bays equal in number to the wideband patch bays are required.

Speed and timing

Each 40.8-kb/s system is a synchronous system and operates only at a speed of 40.8 kb/s. The transmitter is controlled by a stable square-wave timing signal having a frequency of 40.8 kc/s $\pm$ 0.01% and a 50% $\pm$ 1% duty cycle. Through an option arrangement, the system can either generate this timing signal or accept it from the data source. The receiver delivers a 40.8 kc/s square-wave timing signal recovered from the transmitted data signal. Under normal conditions, the system is always in synchronism because the transmitter transmits a particular idle code during idle time. Should the receiver get no signal for some period of time, synchronism will hold for about 10 milliseconds.

Data input and output

The system accepts serial binary d.c. pulses at the transmitter and delivers these pulses on a regenerated basis at the receiver.

Transmit and receive levels

The transmit level is fixed at + 10 dbm and cannot be adjusted. No more than ± 1 db variation is expected among data sets. The receive level will be between −4 and −30 dbm. The levels into and out of the carrier systems will be −10 ± 0.25 dbm.

Code restrictions

There are no restrictions on the format, bit sequence, character organization, or error control that might be employed in most types of information the data source might present for transmission. Long sequences of repeated bit pairs should be avoided, however, because they tend to produce interference in neighbouring channels. If synchronism is to be maintained during idle time, the bit sequence should be a repeated 1000 pattern. The system is arranged to automatically transmit the repeated 1000 pattern if the data source relinquishes control of the data channel.

Modulation and information encoding

Input d.c. pulses are converted by the transmitter to an a.c. signal for transmission. Phase modulation of a carrier signal is employed during this conversion and the information encoded in the d.c. pulses is encoded in phase differences in the a.c. signal. The four phase differences that are employed are described later. The receiver converts the a.c. signal back to d.c. pulses. Differential phase detection is employed during this conversion to re-encode the information in these d.c. pulses.

Bandwidth

The bandwidth of 40.8-kb/s wideband data systems is properly stated as being 48 kc/s. This is the bandwidth of a TELPAK A channel and it is the minimum bandwidth into which is fitted both the data channel and the voice co-ordination channel. The a.c. line signal out of the transmitter requires a transmission bandwidth of 40.8 kc/s as a minimum when it contains random data. The voice co-ordination channel requires the usual nominal 4-kc/s bandwidth as a minimum. The 48-kc/s bandwidth is supplied from L-carrier systems.

Performance

Experience has so far shown that 99% efficiency is acceptable and can be achieved in transmitting digital data in the 40.8-kb/s systems. This efficiency is based on transmitting computer records over the data link previously mentioned. Each record contains 1000 8-bit characters and error control consists of error detection with retransmission of records for error correction. Retransmissions are limited to three for any one record in this application.

A 99% efficiency means that only 1% of the records are retransmitted. If the error occurrence is assumed to be one bit error per block, then this performance equates to an average bit-error rate that approaches one error per million bits transmitted.

Experience supplementing that gained on the data link has been gained in field tests where data transmissions were made over 2000 miles of L-carrier and over 500 miles of L-carrier with

N2-carrier end links. Results of such field tests indicate that the aforementioned performance is obtainable.

Facility and terminal treatment relating to delay and attenuation equalization, level regulation, and control of impulse noise, thermal noise, and interference is employed and directed toward achieving the 99% efficiency. The sections of this supplement describing terminals detail the various treatments.

Impairment limits

Limits must be set on envelope delay and attenuation distortion, thermal and impulse noise, and tone and crosstalk interference or the data signal may become so impaired that performance objectives cannot be met. The various limits are as follows:

1. over-all envelope delay distortion is limited to 50 microseconds in the 12.75 to 48.45 kc/s line signal band of frequences;
2. over-all attenuation slope in the band in 1) above is limited to 6 db;
3. the minimum tolerable signal-to-thermal-noise ratio at the input to the receiver is 20 db;
4. from field test data, a limit has been set on impulse noise counts at 225 counts per section (local loops, carrier links) per half-hour. To be put into service sections must not show an impulse count (under appropriate test conditions which are beyond the scope of this paper) that exceeds this number;
5. the minimum tolerable signal to tone ratio at the receiver input is 20 db;
6. crosstalk between transmit and receive signals at the data set must be 40 db below the receive level minimum which is set at −30 dbm.

The above describes the characteristics of the over-all system. Descriptions of system terminals and local loops are given next.

301B data set

The 301B data set and the 803A data auxiliary set are the data terminals that are located on a user's premises in 40.8-kb/s systems. This section describes the 301B data set and the next section describes the 803A data auxiliary set. Primarily, only the functions of the 301B are described here because the theory and internal organization of data sets similar to the 301B are given elsewhere. [4]

The 301B data set is essentially a data modem with associated timing and control circuits, and interfaces for connecting with a data source or sink and local loops. It is packaged in a grey metal case designed for table top or desk mounting. It weighs approximately 35½ pounds and its size is approximately 18 inches wide, 12 inches deep, and 8 inches high.

The transmitter accepts serial binary unipolar positive d.c. pulses and converts these to a + 10 dbm a.c. output signal suitable for transmission. The d.c. pulses are accepted under the control of a stable 40.8 kc/s ± 0.01% square-wave timing signal that is supplied on an option basis by either the data set or the data source. Input data signal transitions must be coincident with positive-going transitions of the timing signal because the data signals are sampled at the negative-going transitions of the timing signal. The samples are grouped in pairs (dibits) in the 301B forming the combinations 00, 01, 10, and 11 that phase-modulate a carrier signal to produce

the line signal (under the control of a dibit clock). The phase modulation involves establishing a unique phase difference between each two adjacent (line) dibits with the result that information is encoded in carrier signal phase differences. The possible phase differences are four odd multiples (1, 3, 5, 7) of $\pi/4$ radians and phase-difference selection depends on the bit content of both the previous and the present dibits. The existence of a phase difference between every dibit is used in the receiver to recover timing. The line signal spectrum is located in the 10.2-51-kc/s band of frequencies and for repeated combinations of bits shows pairs of spectral lines as shown in Figure 2.

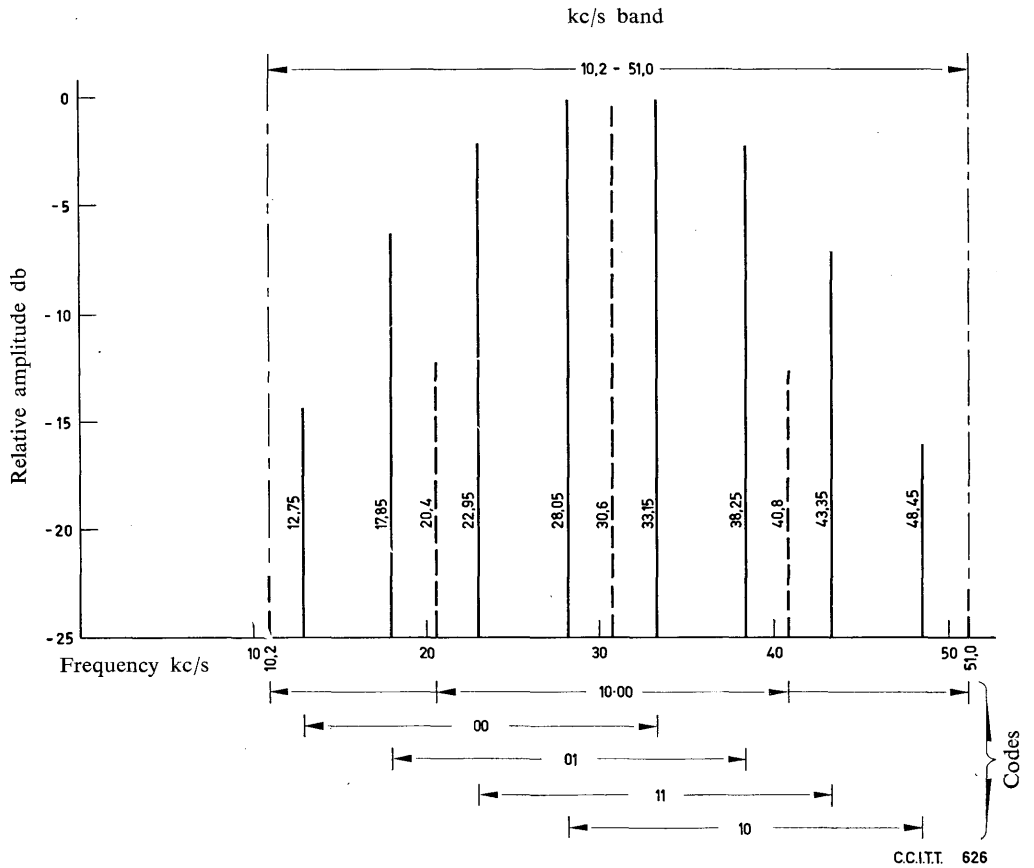


FIGURE 2. — **Data set 301B spectrum for fixed codes**

The receiver accepts the phase-modulated line signal and converts it to d.c. pulses. It has an automatic volume control range of 35 db and can accept signals in a level range of +10 dbm to −50 dbm. To re-encode the information in d.c. pulses thus forming the original sequence of bits, a process of product demodulation is employed. This process essentially determines the particular phase difference between adjacent (line) dibits. The receiver also recovers the

40.8-kc/s timing signal from the steady occurrence of carrier signal phase differences. It presents both the 40.8-kc/s square-wave timing signal and the regenerated d.c. pulses to the data sink. If, for a period of time, no signal appears at the receiver input the receiver output timing signal is reliable for about 10 ms.

The data set interface is a current interface and does not follow EIA interface standards. It presents, to the data source and sink, both data and control leads which are terminated in cable drivers and cable termination. These accept or supply less than 5 milliamperes of current to indicate a binary 1, control "off" or mark and greater than 23 milliamperes to indicate a binary 0, control "on" or space. The data and control leads presented by the interface are listed below and explained in the following paragraph.

Transmit	*Receive*
SD — Send data	RD — Received data
SR — Send request	SCR — Serial clock received
SCT — Serial clock transmit	COO — Carrier on—off
SCTE — Serial clock transmit external	IT — Interlock
DCT — Dibit clock transmit	
CS — Clear to send	

Signals (d.c. pulses or levels) are presented to the data set on the SD, SR, and SCTE leads. They are presented by the data set on the SCT, DCT, CS, RD, SCR, COO, and IT leads. The condition of the IT lead indicates whether or not the data set can be used. IT "on" indicates that data set power is on and the data set can be used. IT "off" indicates that the data set cannot be used. The data source should hold an "off" signal on the SR lead until it is ready to send data. In this condition, the data set is transmitting the repeated 1000 pattern of bits to maintain synchronism and the CS lead is held by the data set in an "off" condition. When the data source turns SR "on" the 1000 pattern is immediately stopped, the CS changes to "on" and stays "on" until SR is turned "off", and at the next positive transition of the 40.8-kc/s clock, the data source may begin sending data. Data signals from the data source should appear on the SD lead. Timing signals for control of "data in" appear from the data set on the SCT lead or from the data source on the SCTE lead. The DCT lead presents a square-wave dibit timing signal at 20.4 kc/s from the data set to the data source. The RD lead and the SCR lead present to the data sink the received data and recovered 40.8-kc/s timing signals respectively. The COO lead is held "off" by the data set until signals are received at which time it is turned "on". It stays "on" whenever signals are being received.

803A data auxiliary set

The 803A data auxiliary set is mounted external to the 301B data set and interconnects the 301B and the local loop. It is provided to permit certain testing of either the 301B or the local loop. Jacks mounted on the 803A permit plugging in test equipment to monitor data signals or to connect separately to the 301B line transformers or the unterminated local loops. For a back-to-back test of the 301B, the 803A may be operated to a test position which breaks connection between the 301B and the local loops, interconnects the local loops, interconnects the

301B transmitter and receiver through a pad, and grounds the IT lead to indicate to the data source that normal data transmission cannot occur. This test feature permits testing isolated sections of the over-all data installation to determine the general area of a trouble condition. The 803A is packaged in a grey metal case and is designed for wall mounting. It weighs approximately 2½ pounds and its size is approximately 7 inches wide, 7 inches high, and 3 inches deep.

Local loops

Local loops connect the data terminals on the user's premises to the wideband patch bays which, in 40.8-kb/s systems, are the carrier system interfaces. These loops consist of cable pairs, terminals, and repeaters if these are required. They handle data signals in the 10.2-51-kc/s frequency band. Separate local loops are required for each direction of transmission.

On local loops, levels are measured at the 30.6-kc/s data set carrier frequency and loss is usually measured at 51 kc/s. On a transmit local loop, the input level is always + 10 ± 1 dbm and the output level is always −10 ± 0.25 dbm. On a receive local loop, the input level is always −10 ± 0.25 dbm and the output level is between −4 and −30 dbm, depending on the local loop length. When amplifiers are required, regulated amplifiers are used and when equalization is required, it is supplied on a post-equalization basis on all loops. Only attenuation slope is equalized.

Local loop cable pairs are non-loaded and are either 19, 22, 24, or 26 gauge. In terms of loss, the maximum length of cable used without intermediate repeaters is 40 db. This equates to approximately 2.7 to 11 miles depending on wire size. For lengths exceeding 40 db loss, repeaters are inserted in the loop at appropriate places.

Repeaters are full duplex and consist of adjustable pads, equalizers, amplifiers, and transformers. Two functions are accomplished by the repeaters. These are : 1) signal level is restored to a regulated level of + 10 ± 0.25 dbm (−30 dbm minimum repeater input level) ; and 2) up to 20 db of attenuation slope is equalized. The number of repeaters that can be used in series varies and is limited by the build-up of envelope delay and attenuation distortion. The functions of level conversion and equalization are also accomplished by the terminal amplifier-equalizer which is one of three terminals designed for use in local loops.

Local loop length—in terms of loss—determine which one of three terminals will be part of the loop. A 0-6-db loop will have a loop equalizer terminal consisting of transformers and an adjustable pad. This loop equalizer is used only to adjust the signal level to −10 ± 0.25 dbm out of a transmit loop and to between −4 and −30 dbm out of a receive loop. A 6-13-db loop will have a loop equalizer terminal consisting of transformers, adjustable pads, and slope equalizers. This loop equalizer is used to equalize up to 20 db of attenuation slope and to adjust loop output levels to those just mentioned. A 13-40-db loop will have a terminal amplifier —equalizer that is essentially like the intermediate repeater. However, the output levels here will be −10 ± 0.25 dbm for transmit loops and between −4 and −30 dbm for receive loops. These loop terminals will be located near the data terminals on receive loops and in a central office on transmit loops. In the central office they will connect with wideband patch bays.

Wideband patch bay

Wideband patch bays are provided as a common interface between local loops and carrier terminals and between N-carrier and L-carrier terminals. It is at these bays that the wideband data signal is at the common level of −10 dbm in the 10.2-51-kc/s frequency band. These bays accommodate loop equalizers, patching jacks, and a distribution frame for six wideband data channels and associated voice-frequency co-ordination circuits.

Carrier terminal modifications

Without being modified, the existing N2 and L carrier terminals cannot provide access to the carrier frequency space required for 40.8-kb/s data transmission. As a part of the 40.8-kb/s systems development, both an N2-carrier and an L-carrier wideband modem have been developed. These modems are designed to replace voice bandwidth channel equipment in existing carrier terminals to modify the terminals as required. The N2 modem is designed for use only in N2-carrier terminals and the L modem is designed for use in L type multiplex terminals for either L1 or L3-carrier systems. Each of these modems is described below.

N2 wideband modem

The N2 wideband modem is full duplex and consists of five units, namely, a transmitting loop terminating unit, a transmitting unit, a high-frequency equalizing unit, a receiving unit, and a receiving loop-terminating unit.

Essentially, the transmit section of the modem accepts the 10.2-51-kc/s data signal, translates this signal to the N2 203.5 to 244.3 kc/s high group band of frequencies, pre-equalizes the N2 data channel and presents the data signal now in the high group band to the N2 group transmitter at the proper level. The choice of the above N2 frequency band for 40.8-kb/s data was governed by considerations of minimizing both single-tone interference into paralleling N systems and envelope delay distortion.

To accomplish the above, the units in the transmit section consist of the following:

1) One transformer and two pads are in the loop-terminating unit. The data signal level is changed from −10 to −25 dbm here.

2) A modulator, an amplifier, and a band-pass filter are in the transmitting unit. Here the data signal is translated to the N2 high group band.

3) A pad and up to six equalizers are in the high-frequency equalizing unit. The required pre-equalization of the N2 data channel occurs here and the data signal is presented from here to the N2 group transmitter.

Essentially, the receive section of the modem accepts the data signal in the 203.5-244.3-kc/s high group band from the N2 group receiver, translates this signal to the 10.2-51-kc/s band of frequencies, adjusts the signal level to -10 ± 0.25 dbm, and presents the data signal now in the 10.2-51-kc/s band to the wideband patch bay.

To accomplish the above, the units in the receive section of the modem consist of:

1) A band-pass filter, a demodulator, a low-pass filter, and an amplifier are in the receiving unit. Here, only the N2 high group data signal frequencies are passed by the band-pass filter. The demodulator translates the data signal to the 10.2-51-kc/s band and the low-pass filter passes only data signal frequencies below 51 kc/s. The amplifier adjusts the data signal level.

2) One transformer and a pad are in the loop-terminating unit. Here, the data signal level is adjusted to -10 ± 0.25 dbm and the data signal is passed to the wideband patch bay.

It is obvious from the above that, in the N2 wideband modem, data signals out of the transmit section and into the receive section are always in the N2 high group band of frequencies. Translations to or from the N2 low group band of frequencies are provided in N2 group transmitters or receivers as required.

Along with the data channel, six voice channels are available for use in an N2 system that is arranged for 40.8 kb/s transmission. The voice channel adjacent to the lower side of the data channel in the N2 high group is the 40.8-kb/s voice co-ordination channel. The modified N2 terminals will be equipped with all six voice channel modems to provide adequate loading for regulation of the N2 system. A potential advantage of the above placement of the voice co-ordination channel is that the observed condition of the voice channel may be indicative of the condition of the data channel. Arrangements for multiplexing the data signal and the signals from the six available voice channels are similar to arrangements used for multiplexing signals from the 12 voice channels available in an N2 system which is not arranged for 40.8 kb/s transmission.

L-carrier wideband modem

The L-carrier wideband modem is full duplex for both the data channel and the voice channel. The transmit section has two inputs and one output and the receive section has one input and two outputs. The signal out of the transmit section is in the 60-108-kc/s group band and is placed in an L-carrier supergroup for transmission. The number of groups useful for 40.8-kb/s data, their placement in the supergroup, and the number of supergroups useful for 40.8 kb/s data are governed by considerations of envelope delay distortion and noise. These considerations limit L1-carrier to 17 and L3-carrier to 59 simultaneous two-way 40.8-kb/s data transmissions.

Essentially, the transmit section of the modem : 1) accepts the 40.8 kb/s data signal and the voice co-ordination signal ; 2) separately translates these into portions of the 60-108-kc/s group band ; 3) multiplexes these translated data and voice signals ; and 4) presents the resultant signal to the L type multiplex high-frequency patch bay.

To accomplish the above, the transmit section consists of the following :

1) The data signal path containing various pads which adjust the signal level, a low-pass filter which excludes signal energy above 51 kc/s, an amplifier-modulator which adjusts the signal level and translates the data signal upwards in frequency, a band-pass filter which passes signal energy only in the 60-104-kc/s band, and a transmit equalizer which post-corrects envelope delay distortion from the transmit section.

2) The voice signal path containing a type A-5 channel bank which translates the voice signal upwards in frequency, a band-pass filter which passes signal energy only in the 104-108-kc/s band, and a buffer amplifier which adjusts the signal level and matches impedances.

3) A hybrid network which multiplexes the translated data and voice signals.

The modem employs a single oscillator for translating the data signal in both the transmit and receive sections.

Essentially, the receive section of the modem: 1) accepts the multiplexed data and voice signals in the 60-108-kc/s group band; 2) separates the data and voice signals; 3) translates them to their respective basebands; and 4) presents the baseband data signal to the wideband patch bay and the baseband voice signal to the voice patch bay.

To accomplish the above, the receive section of the modem consists of the following:

1) A hybrid network which passes the multiplexed signal to both the data signal path and the voice signal path.

2) The data signal path containing a band-pass filter which passes signal energy only in the 60-104-kc/s band, a receive equalizer which pre-corrects envelope delay distortion from the receive section, an amplifier-demodulator which translates the data signal downwards in frequency and adjusts the signal level, a low-pass filter which excludes signal energy above 51 kc/s, and an amplifier and various pads which adjust the signal level.

3) The voice signal path containing a buffer amplifier which adjusts the signal level and matches impedances, a band-pass filter which passes signal energy only in the 104-108-kc/s band, and a type A-5 channel bank (the same one that is in the transmit section) which translates the voice signal to baseband. For routine group level measurements and line-up, L systems are ordinarily supplied with 92-kc/s pilots and band elimination filters. This arrangement, however, cannot be used in 40.8-kb/s channels because the band elimination filter causes serious impairment of the data signal. It is possible to use the 30.6-kc/s data set carrier frequency for alignment purposes but, compared with using the pilot, a loss of accuracy results. Consequently, new arrangements are under development to provide a 104.08-kc/s pilot and band elimination filter for group level measurements and line-up.

Summary

The Bell System is now offering TELPAK channels for private line use. These may be used for transmitting speech and/or data. To transmit high-speed data on TELPAK A 48-kc/s bandwidth channels, the Bell System is developing standard systems which will operate at 40.8 kb/s. These systems will provide a wideband data channel and a voice-frequency co-ordination channel. They will employ N2-carrier facilities for short-haul transmission and L-carrier facilities for long-haul transmission. They will also employ data terminal equipment which will be located on a user's premises, local loops which will connect the data terminals to the carrier facilities, and wideband modems which modify existing carrier terminals for 40.8 kb/s transmission.

A data link connecting the Murray Hill and Holmdel, New Jersey, facilities of Bell Telephone Laboratories is the first Bell System trial and use of data transmission at approximately 40.8 kb/s. Transmission at 40.8 kb/s has been studied in field trials that employed 2000 miles of L-carrier facilities and 500 miles of L-carrier facilities with N2 carrier end links. Several installations employing experimental data terminals and special arrangements for local loops and carrier terminal modification have been made to provide 40.8 kb/s transmission service.

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