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

IVth PLENARY ASSEMBLY

MAR DEL PLATA, 23 SEPTEMBER - 25 OCTOBER 1968

WHITE BOOK

VOLUME VIII

Data transmission

Published by
THE INTERNATIONAL TELECOMMUNICATION UNION
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APPLICABLE AFTER THE IVth PLENARY ASSEMBLY (1968)

(WHITE BOOK)

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SERIES V RECOMMENDATIONS

Data transmission

LIST OF RECOMMENDATIONS

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.

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

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

5. In accordance with C.C.I.T.T. Recommendation R.35 (*White 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)

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 Npm0 (-10 dBm0). When transmission of data is discontinued for any appreciable time, the power level should preferably be reduced to -2.3 Npm0 (-20 dBm0) or lower.

3. For systems not transmitting tones continuously, for example amplitude-modulation systems, higher level up to -0.69 Npm0 (-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 Npm0 (-15 dBm0) on each direction of transmission simultaneously). Further, the level of any tones above 2400 Hz 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 Npm0 (-10 dBm0) (simplex systems) or -1.5 Npm0 (-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 Npm0 (−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 (*White 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.3

INTERNATIONAL ALPHABET No. 5

(Mar del Plata, 1968)

Introduction

A new alphabet 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 telephone network or on telegraph networks has been established by a common work of the C.C.I.T.T. and the International Organization for Standardization.

This 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. In such cases it is considered as the alphabet to be used as common basic language for data transmission and for elaborated message systems.

This new Alphabet No. 5 does not exclude the use of any other alphabet that might be better adapted to special needs.

The main requirements to be satisfied by Alphabet No. 5 were:

— small letters and capital letters of the Latin alphabet;

— diacritical signs;

— controls for transmission and data processing;

— tabulation, back space;

— allocation of combinations for national uses.

As no standardized shift (no upper-case and lower-case for each combination) was asked for, these requirements will be satisfied with a seven-unit " naked code " (128 combinations). By " naked code " it is understood the code reduced to its unit of pure information, without considering units for protection against errors or start-stop operation.

Alphabet No. 5 offers a lot of auxiliary possibilities, as described in the body of the Recommendation; among these possibilities special attention is to be given to the " shift out " possibility. By the use of the " shift out " combination, it is possible to operate the connection with another seven-unit alphabet where the meaning of the combinations pertaining to columns 2, 3, 4, 5, 6, 7 (except " delete ") of the standard table could be different from the basic meaning. When return to the basic alphabet is desired, the combination " shift-in " has to be used.

With the use of the " shift-out ", " shift-in " possibilities, auxiliary alphabets for special needs (graphical arts, non-Latin alphabet, mathematical signs, etc.) can be used in the general frame of the standard Alphabet No. 5. Of course, users of the " shift " possibility have to agree between themselves on the auxiliary alphabet they desire to operate in supplement to the basic alphabet.

1. *Code table of Alphabet No. 5*

1.1 The recommended alphabet and the corresponding two-condition " naked " code is given by the following table:

1.2 *Numbering of the positions in the code table*

Within any one character the bits (or units) are identified by b_7, b_6 . . . b_1, where b_7 corresponds to the highest order, or most significant bit, and b_1 to the lowest order, or least significant bit.

Any one position in the code table may be identified either by its bit pattern, or by its column and row numbers. For instance, the position containing the figure 1 in the table may be identified:

— by its bit-pattern, e.g. 011 0001

— by its column and row numbers, e.g. 3/1.

1.3 *Controls designation*

ACK	Acknowledge	ENQ	Enquiry
BEL	Bell	EOT	End of transmission
BS	Backspace	ESC	Escape
CAN	Cancel	ETB	End of transmission block
CR	Carriage return	ETX	End of text
DC	Device control	F	Function
DEL	Delete	FE	Format effector
DLE	Data link escape	FF	Form feed
EM	End of medium	FS	File separator

Bits												
b_7					0	0	0	0	1	1	1	1
b_6					0	0	1	1	0	0	1	1
b_5					0	1	0	1	0	1	0	1
b_4	b_3	b_2	b_1	Row \ Column	0	1	2	3	4	5	6	7
0	0	0	0	0	NUL	(TC_7)DLE	SP	0	(@) [3]	P	` [4]	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	" [6]	2	B	R	b	r
0	0	1	1	3	(TC_3) ETX	DC_3	£ [3] [7]	3	C	S	c	s
0	1	0	0	4	(TC_4) EOT	DC_4	$ [3] [7]	4	D	T	d	t
0	1	0	1	5	(TC_5) ENQ	(TC_8)NAK	%	5	E	U	e	u
0	1	1	0	6	(TC_6) ACK	(TC_9)SYN	&	6	F	V	f	v
0	1	1	1	7	BEL	(TC_{10})ETB	' [6]	7	G	W	g	w
1	0	0	0	8	FE_0 (BS)	CAN	(	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]	SUB	*	: [8]	J	Z	j	z
1	0	1	1	11	FE_3 (VT)	ESC	+	; [8]	K	([) [3]	k	[3]
1	1	0	0	12	FE_4 (FF)	IS_4 (FS)	,	<	L	[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	^ [4] [6]	n	‾ [4] [5]
1	1	1	1	15	SI	IS_1 (US)	/	?	O	_	o	DEL

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GS	Group separator	SO	Shift-out
HT	Horizontal tabulation	SOH	Start of heading
IS	Information separator	SP	Space
LF	Line feed	STX	Start of text
NAK	Negative acknowledge	SUB	Substitute
NL	New line	SYN	Synchronous idle
NUL	Null	TC	Transmission control
RS	Record separator	US	Unit separator
SI	Shift-in	VT	Vertical tabulation

1.4 *Graphical symbols*

Graphical representation	Name	Position in the code table
(Space)	A normally non-printing graphic character	2/0
!	Exclamation mark	2/1
"	Quotation mark. Diaeresis (Note 6)	2/2
£	Currency symbol £ (Note 2) (Note 7)	2/3
$	Currency symbol $ (Note 2) (Note 7)	2/4
%	Per cent	2/5
&	Ampersand	2/6
'	Apostrophe, acute accent (Note 6)	2/7
(	Left parenthesis	2/8
)	Right parenthesis	2/9
*	Asterisk	2/10
+	Plus sign	2/11
,	Comma	2/12
—	Hyphen, minus sign	2/13
.	Full stop (period)	2/14
/	Solidus	2/15
:	Colon	3/10
;	Semi-colon	3/11
<	Less than	3/12
=	Equals	3/13
>	Greater than	3/14
?	Question mark	3/15
@	Commercial at	4/0
[	Left square bracket	5/11
]	Right square bracket	5/13
^	Upwards arrow, circumflex accent (Note 6)	5/14
_	Underline	5/15
`	Grave accent	6/0
‾	Overline (Note 5)	7/14

1.5 *Notes to the code 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 a combined carriage-return and line feed operation, the function FE_2 will have the meaning of "new line" (NL).

These substitutions require agreement between the sender and the recipient of the data.

The use of this function NL is not allowed for international transmission on general switched telecommunication networks (telegraph and telephone networks).

② For international information interchange, $ and £ symbols do not designate the currency of a given country. The use of these symbols combined with other graphic symbols to designate national currencies may be the subject of other Recommendations.

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 in parenthesis in some cases.

④ Positions 5/14, 6/0 and 7/14 of the seven-bit set table are normally provided for the diacritical signs " circumflex ", " grave accent " and " overline ". However, these positions may be used for other graphical symbols when it is necessary to have 8, 9 or 10 positions for national use.

⑤ For international information interchange, position 7/14 is used for the graphical symbol ‾ (overline), the graphical representation of which may vary according to national use to represent ∾ (tilde) or another diacritical sign provided that there is no risk of confusion with another graphical symbol included in the table.

⑥ 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 the diacritical signs " diaeresis ", " acute accent " and " circumflex accent " when they precede or follow the " backspace " character.

⑦ For international information interchange, position 2/3 of the 7 bit-code table has the significance of the symbol £ and position 2/4 has the significance of the symbol $.

By agreement between the countries concerned where there is no requirement for the symbol £, the symbol "number sign" (#) may be used in position 2/3. Likewise, where there is no requirement for the symbol $, the symbol " currency sign " (¤) may be used in position 2/4.

⑧ If 10 and 11 as single characters are needed (for example, for Sterling currency subdivision), they should take the place of " colon " (:) and " semi-colon " (;) respectively. These substitutions require agreement between the sender and the recipient of the data. On the general telecommunication networks, the characters " colon " and " semi-colon " are the only ones authorized for international transmission.

1.6 *Diacritical signs*

In Alphabet No. 5, some printing symbols may be designed to permit their use for the composition of accented letters when necessary for general interchange of information. A sequence of three characters, comprising a letter, "backspace" and one of these symbols, is needed for this composition, and the symbol is then regarded as a diacritical sign. It should be noted that these symbols take on their diacritical significance only when they are preceded or followed by the " backspace " character; for example, the symbol corresponding to the code combination 2/7 (') normally has the significance of " apostrophe ", but becomes the diacritical sign " acute accent " when it precedes or follows the " backspace " character.

In order to increase efficiency, it is possible to introduce accented letters (as single characters) in the positions marked by Note 3 in the code table. According to national requirements, these positions may contain special diacritical signs.

1.7 *Interpretation of graphics*

The meaning of the graphics is not limited by this Recommendation. However, no interpretation may be chosen which is contradictory to the customary meaning. A graphical symbol can have more than one meaning, e.g. the graphical symbol — (minus) also can have the meaning of hyphen or separation mark.

1.8 *Dual allocations*

A character allocated to a position in the code table may not be placed elsewhere in the table. In the case of positions having two characters allocated to them (2/3, 2/4, 3/10 and 3/11), the character not used cannot be placed elsewhere. This applies also to positions showing a preferred graphic. If such a position is filled with a national character the preferred graphic cannot be placed in another code position.

2. *Functional characteristics related to control characters*

Some definitions in this section are stated in general terms and more explicit definitions of use may be needed for specific implementation of the code tables on recording media or on transmission channels. These more explicit definitions may become the subject of other Recommendations.

A control character may be identified by a general designation, by a specific designation or by a combination of both.

2.1 *General designations of control characters*

The general designations of control characters involve a specific name followed by a subscript number.

They are defined as follows:

TC — *Transmission control* — A functional character intended to control or facilitate transmission of information over telecommunication networks.

The use of the TC characters on the general telecommunication networks may be the subject of future Recommendations.

FE — *Format effector* — A functional character which controls the layout or positioning of information in input/output media.

DC — *Device control* — A functional character for the control of an ancillary device associated with a data processing or telecommunication system, for example an " on " or " off " switching device.

The following is an example of how device control characters could be used in a specific tape system employing two ancillary tape punches and one ancillary tape reader:

DC_1 — First punch " on "

DC_2 — Second punch " on "

DC_3 — Tape reader " on "

The DC_4 character has a specific function, defined in clause 2.2.

IS — *Information separator* — A functional character which is used to separate and qualify information blocks logically. There is a group of four such characters, which are intended to be used in a hierarchical order.

2.2 *Specific designations of control characters*

These are defined as follows:

ACK — *Acknowledge* — A transmission control character transmitted by a receiver as an affirmative response to the sender.

BEL — *Bell* — A character for use when there is a need to call for human attention; it may control alarm or attention devices.

BS — *Backspace* — A layout character which controls the movement of the printing position one printing space backward on the same printing line.

CAN — *Cancel* — A character used to indicate that the information it accompanies is in error.

CR — *Carriage return* — A layout character which controls the movement of the printing position to the first printing position on the same printing line.

DC_4 — A device control character used to interrupt or turn off ancillary devices (STOP).

DEL — *Delete* — This character is used primarily to erase or obliterate erroneous or unwanted characters in punched tape. DEL characters may be inserted into or removed from a stream of data without affecting the information content of that stream. DEL characters may serve to accomplish media-fill or time-fill but then the addition or removal of these characters may affect the information layout and/or the control of equipment.

DLE — *Data link escape* — A transmission control character which will change the meaning of a limited number of contiguously following characters. It is used exclusively to provide supplementary data transmission control functions. Only graphics and transmission control characters can be used in DLE sequences.

EM — *End of medium* — A control character 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.

ENQ — *Enquiry* — A transmission control character used as a request for a response from a remote station. The response may include station identification and/or station status. When a " who are you " function is required on the general switched transmission network, the first use of ENQ after the connection is established shall have the meaning " who are you " (station identification). Subsequent use of ENQ may, or may not, include the function " who are you ", as determined by agreement by the users.

EOT — *End of transmission* — A transmission control character used to indicate the conclusion of the transmission of one or more texts.

* ESC— *Escape* — A functional character which may be used to extend the standard character set of the code table. It is a warning or non-locking shift character which changes the meaning of the next single following code combination. The precise meaning of the character following " escape " requires prior agreement between the sender and the recipient of the data. Where required the character following " escape " may extend the " escape " sequence.

" Escape " sequences are used primarily to obtain additional control functions which may provide amongst other things graphics or graphic sets outside the standard set. Such control functions must not be used as additional transmission controls.

" Null " and " delete " and the ten transmission controls must not be used in defining " escape " sequences. Where they appear in an actual " escape " sequence they shall retain their standard meaning and be disregarded in the interpretation of the " escape " sequence.

The use of certain " escape " sequences will be the subject of further Recommendations.

ETB — *End of transmission block* — A transmission control character used to indicate the end of a transmission block of data where data are divided into such blocks for transmission purposes.

ETX — *End of text* — A transmission control character which terminates a text.

FF — *Form feed* — A layout character which controls the movement of the printing position to the first predetermined printing line on the next form.

FS — *File separator* — See unit separator (US) for definition.

GS — *Group separator* — See unit separator (US) for definition.

HT — *Horizontal tabulation* — A layout character which controls the movement of the printing position to the next in a series of predetermined positions along the printing line.

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* Still subject to special consideration by I.S.O.

LF — *Line feed* — A layout character which controls the movement of the printing position to the next printing line.

NAK — *Negative acknowledge* — A transmission control character transmitted by a receiver as a negative response to the sender.

NL — *New line* — A layout character which controls the movement of the printing position to the first printing position of the next printing line (see Note 1 to the code table).

NUL — *Null* — A character whose sole purpose is to accomplish media-fill or time-fill. NUL characters may be inserted into or removed from a stream of data without affecting the information content but then the information layout and/or the control of equipment may be affected.

RS — *Record separator* — See unit separator (US) for definition.

* SI — *Shift-in* — The shift-in character means that the code combinations which follow shall be interpreted according to the standard code table.

* SO — *Shift-out* — The shift-out character means that the code combinations which follow shall be interpreted as outside of the standard code table until a shift-in character is reached. However, all the control characters (columns 0 and 1) and delete will retain their standard interpretation. The shift-out character is reserved primarily for extension to the graphics.

SOH — *Start of heading* — A transmission control character used as the first character of a heading of an information message.

SP — *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.

STX — *Start of text* — A transmission control character which precedes a text and which is used to terminate a heading.

SUB — *Substitute* — A substitute character used to replace a character which is determined to be invalid or in error.

SYN — *Synchronous idle* — A transmission control character 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 data terminal equipments.

US — *Unit separator* — Terminates an information block called a " unit ". Similarly, " record separator " (RS), " group separator " (GS), " file separator " (FS)

* Still subject to special consideration by I.S.O.

terminate information blocks called " record ", " group ", " file ", respectively.

The four information separators are in the hierarchical ascending order US, RS, GS, FS.

An information block must not be split by a higher order separator, e.g. a " record " may contain a number of complete " units ", but may not contain a part of a " unit ".

VT — *Vertical tabulation* — A layout character which controls the movement of the printing position to the next in a series of predetermined printing lines.

RECOMMENDATION V.4

GENERAL STRUCTURE OF SIGNALS OF ALPHABET No. 5 CODE

(Mar del Plata, 1968)

The C.C.I.T.T.

I. *considering, firstly*

the agreement between the International Organization for Standardization (I.S.O.) and the C.C.I.T.T. on the main characteristics of a seven-unit alphabet (International Alphabet No. 5) to be used for data transmission and for telecommunications requirements that cannot be met by the existing five-unit International Alphabet No. 2;

the interest, both to the users and to the telecommunication services, of an agreement concerning the chronological order of transmission of bits in serial working,

declares the view

that the agreed rank number of the unit in the alphabetical table of combinations should correspond to the chronological order of transmission in serial working on telecommunication circuits;

that, when this rank in the combination represents the order of the bit in binary numbering, the bits should be transmitted in serial working with the low order bit first;

that the numerical meaning corresponding to each information unit considered in isolation is that of the digit:

0 for a unit corresponding to condition A (travail=space); and

1 for a unit corresponding to condition Z (repos=mark);

in accordance with the definitions of these conditions for a two-condition transmission system.

II. *considering, moreover,*

that it is often desirable, in data and messages transmission, to add an extra " parity " unit to allow for the detection of errors in received signals;

the possibility offered by this addition for the detection of faults in terminal telegraph equipment;

the need to reserve the possibility of making this addition during the transmission itself, after the seven information units proper have been sent;

that for the majority of data users the 50-baud telex network is not suitable for this seven-unit alphabet, the data transfer rate being too low, and therefore the code limitations due to this network need not extensively be considered,

declares the view

that signals of the telegraph code using combinations of the seven-unit alphabet for data and messages transmission should in general include an additional " parity " unit;

that the rank of this unit and, hence, the chronological order of the transmission in serial working should be the eighth of the combination thus completed.

III. *considering*

that, in start-stop systems working with electromechanical equipment, the margin of such equipment and the reliability of the connection are considerably increased by the use of a stop element corresponding to the duration of two unit intervals of the modulation;

that for transmissions over telephone circuits via modems installed on the user's premises, the latter must be able to use the connections at the highest possible practical rate in characters per second, and that in such a case a single stop unit leads to a gain of about 10% as regards this practical rate;

that, however, it does not appear that the production of electronic devices capable of working at will with start-stop signals having a stop element equal to one or two unit intervals should lead to costly complications and that such an arrangement can have the advantage of appreciably limiting the error rate without greatly reducing the practical efficiency of the connection,

declares the view

that in start-stop systems using combinations of the new seven-unit alphabet normally followed by a parity unit, the first information unit of the transmitted combination should be preceded by a start element corresponding to condition A (space);

that the duration of this start element should be one unit interval for the modulation rate under consideration, at transmitter output;

that the combination of seven information units, normally completed by its parity unit, should be followed by a stop element corresponding to condition Z (mark);

that for start-stop systems using the seven-unit code on switched telegraph and telephone networks, a two-unit stop element should be used with electromechanical data terminal equipments operating at modulation rates up to and including 200 bauds. In other cases, the use of a one-unit stop element is preferable. However, this is subject to a mutual agreement between administrations and/or recognized private operating agencies concerned;

that similar situations when a one-unit stop element can be used may apply to leased circuits;

that the start-stop receivers should be capable of correctly receiving start-stop signals comprising a single-unit stop element, whose duration will be reduced by a time interval equal to the deviation corresponding to the degree of gross start-stop distortion permitted at receiver input. However, for electromechanical equipment which must use a two-unit stop element (eleven-unit code signal) with a modulation rate of 200 bauds or less, receivers should be capable of correctly receiving signals with a stop element reduced to one unit.

IV. *considering, finally,*

that the direction of the parity unit can only be that of the even parity on the perforated tapes, particularly owing to the possibility of deletion (combination 7/15 of the alphabet) which causes a hole to appear in all tracks;

that, on the other hand, the odd parity is considered essential in the equipment which depends on transitions in the signals to maintain synchronism (in cases where combination 1/6 (SYNC) of the alphabet does not permit of an economical solution),

declares the view

that the parity unit of the signal should correspond to the even parity in links or connections operated on the principle of the start-stop system,

that this parity should be odd on links or connections using end-to-end synchronous operation,

that arrangements should be made when necessary to reverse the direction of the parity unit at the input and output of the synchronous equipment connected either to apparatus working on the start-stop principle or receiving characters on perforated tape.

RECOMMENDATION V.10

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

(Geneva, 1964, amended at Mar del Plata, 1968)

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 five-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

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 B and U.1—Volume VII, *White 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 changeover 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 order to avoid any misunderstanding between the stations concerned, the calling operator should first check the equipment of the distant station (whether manual changeover or automatic changeover).

Case a) — Manual changeover at both terminals

Once the connection has been set up, the following procedure should be followed:

1. The operator of the calling station sends the sequence of four No. 19 combinations. This sequence should not connect the data equipment locally.

2. Upon reception of the SSSS (or """") sequence, the operator of the called station likewise sends the sequence of four No. 19 combinations, and then connects his data equipment to the line.

3. Upon reception of this answer sequence, the calling operator connects his data equipment to the line.

Case b) — Automatic changeover at both terminals

Once the connection has been set up, the following procedure should be followed:

1. The calling station sends the sequence of four No. 19 combinations, which must connect its data equipment to the line automatically within less than 500 milliseconds, starting from the end of transmission of the last signal of this sequence.

2. Reception of the sequence at the other terminal of the connection connects the called station to the data equipment line automatically within less than 500 milliseconds, starting from the end of reception of the last signal of this sequence.

3. The data transmission should not commence before the end of the 500-millisecond delay.

Case c) — Manual changeover at one terminal and automatic changeover at the other

i) *Calling station with manual changeover and called station with automatic changeover*

Once the connection has been set up, the following procedure is followed:

1. The operator of the calling station sends the sequence of four No. 19 combinations, and then immediately connects his data equipment to the line.

2. Upon reception of the sequence of four No. 19 combinations at the called station, the data equipment must be connected to the line within less than 500 milliseconds, starting from the end of reception of this sequence.

3. The data signals should not be transmitted before the end of the 500-millisecond delay.

ii) *Calling station with automatic changeover and called station with manual changeover*

Once the connection has been set up, the following procedure should be followed:

1. The calling station invites its called correspondent, by a brief preliminary message, to send the sequence of four No. 19 combinations. This message must not include within itself the sequence of four No. 19 combinations. If the calling station is not equipped with teleprinter attended by an operator, this preliminary message must be sent automatically.

2. The operator of the called station then sends the sequence of four No. 19 combinations and immediately connects his data equipment to the line.

3. Upon reception of this sequence at the calling station, connection of the data equipment to the line must be effected within less than 500 milliseconds, starting from the end of reception of the last No. 19 signal of the sequence.

4. Transmission of the data signals should not begin before the end of the 500-millisecond delay.

Note. — The arrangements envisaged by Recommendation V.10, point I.b, run counter to the inclusion to the sequence of four No. 19 combinations in the answer-back code of telex lines equipped with a simulator and at the same time in the answer-back of teleprinters equipped with an automatic device for changeover to data transmission. (This fact should be borne in mind in the further study of point F of the study programme.)

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, *White Book*, Volume VII);

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

I.d) Data transmission should be made by means of a start-stop 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 I.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) As soon as the telex connection has been transferred to the data transmission equipment, the transmission must be controlled by the data equipment at each terminal.

If it is useful, for some reason, to return to telex operation, the data terminal equipment must control the transfer back to telex.

This possibility of returning to the telex condition is used by a subscriber who considers it useful, after a data transmission, to return to teleprinter operation for a telex connection, instead of sending the clearing signal as mentioned in paragraph I.f above. This return should be accompanied by the re-entry into service of the answer-back device.

This control may be caused:

1. by the transmission of a special data signal over the line, causing the receiving installation to return to the telex position. The received data terminal equipment must send the same signal in the reverse direction to the opposite terminal before it causes transfer to the telex condition. This mutual signalling identifies the situations at the two terminals;

2. by a local control causing return to the telex situation, set off if no data or supervisory signal is transmitted or received during a given time interval agreed upon by the users.

Note. — Telex connections which include error-correcting synchronous radio systems often insert long pauses into the message and due attention should be paid to this in selecting the agreed interval.

For these control operations, a special circuit should be set aside in the interface connecting the data terminal equipment to the transfer device.

Note. — The above provisions of paragraph I.g could be applied with advantage to the case of telex lines not equipped with teleprinter equipment but simply with answer-back unit simulators.

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 (*White 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

The attention of administrations is drawn to the fact that it is impossible to send signals other than those of a five-unit start-stop code over international connections via time-division multiplex sections specially designed for a five-unit code.

However, telex connections for data transmission may be set up over such relations in the conditions set out in part I of this Recommendation for the transmission of messages composed of signals different from those of the five-unit start-stop code. A service of this nature may be obtained by regrouping the units of these signals in the form of five-unit signals. Such regrouping calls for the use of additional code converters at the sending and receiving terminals.

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 (*White 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.

RECOMMENDATION V.11

AUTOMATIC CALLING AND/OR ANSWERING ON THE TELEX NETWORK

(Mar del Plata, 1968)

1. *General*

1.1 This Recommendation provides a method of automatically originating and answering calls on the telex network. Manual calling or answering with automatic facilities is also provided. At this stage the application of that part of this Recommendation for automatic origination of calls is limited to those networks where route selection is by means of dial pulses. The automatic answering procedure, however, is generally applicable.

1.2 In order to simplify and reduce the cost of overall data terminal systems in the particular cases where calls are originated for data transmission over the telex as well as the switched telephone networks, the interface and data terminal procedures have been arranged to be similar (see Recommendation V.25 for automatic calling and/or answering on the general switched telephone network).

1.3 The object is to standardize the call establishment and answering procedures between the data terminal equipment and the data communication equipment. In addition, the essential interchange circuits, to Recommendation V.24, have been defined between the two equipments.

1.4 It is recognized that other simpler methods may be possible for automatic calling by data terminals connected only to the switched telegraph networks, and these are still under study.

Ultimately it may be technically possible to extend the application of this Recommendation to route selection signals of the telegraph character type (commonly known as keyboard selection). This requires further study.

2. *Interchange circuits*

2.1 *Automatic answering*

These circuits are required to be connected between the data terminal equipment and the control unit within the data communication equipment.

Circuit numbers

CT 101 *	Protective ground
CT 102	Signal ground
CT 103	Data (transmitted)
CT 104	Data (received)
CT 106	Ready for sending (see Note)
CT 107	Data set ready (see Note)
CT 108/2	Data terminal ready
CT 109	Data channel received line signal detector (see Note)
CT 125	Calling indicator
CT 132	Return to non-data mode

Note. — In so far as the operation of the data communication equipment is concerned, circuits CT 106 and CT 109 are not required. But to provide an interface common to that for the telephone case these two circuits are included and are both coupled to CT 107 at the data communication equipment.

2.2 *Automatic calling*

For automatic calling by data terminals connected to telex networks with dial pulse selection, in addition to the above circuits, the following circuits should also be employed between the data terminal equipment and the automatic calling equipment within the data communication equipment:

Circuit numbers

CT 201	Signal ground
CT 202	Call request
CT 203	Data line occupied
CT 204	Distant station connected
CT 205	Abandon call
CT 206	Binary digit signal 1

* May be excluded if so required by local safety regulations

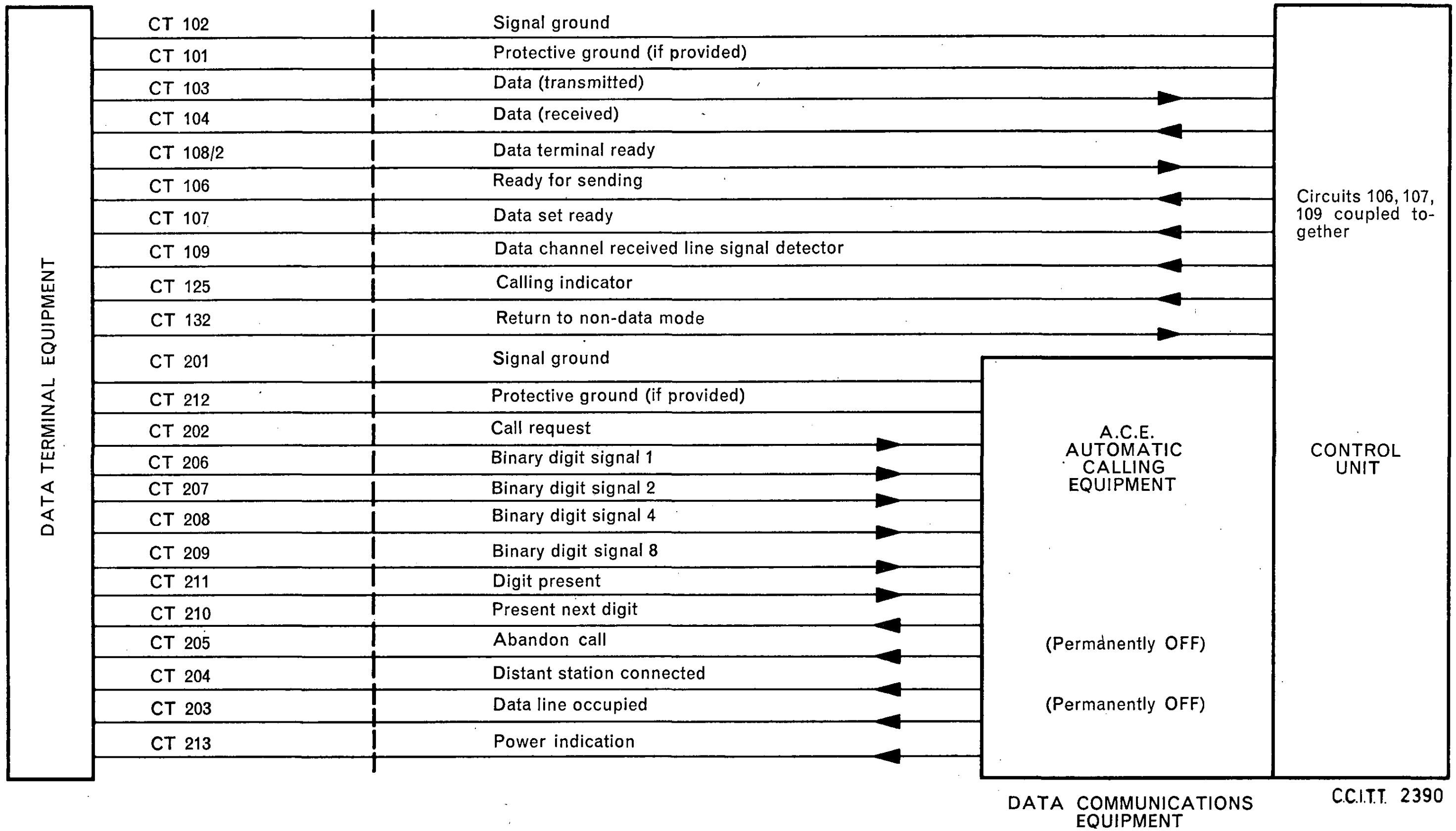

FIGURE 1. — **Proposed interface for telex A.C.E.**

Circuit numbers

CT 207	Binary digit signal 2
CT 208	Binary digit signal 4
CT 209	Binary digit signal 8
CT 210	Present next digit
CT 211	Digit present
CT 212 *	Protective ground
CT 213	Power indication

The above sets of interface circuits are applicable only to a single data terminal equipment connected to a single telex terminal.

The impedances of these interface circuits and the signals (voltages, etc.) on the circuits must be as specified in Recommendation V.24.

Note 1. — The use of a common auto-calling equipment for the connection of several data terminal equipments to one, or more, telex lines and any additional interchange circuits thus required remains for further study.

Note 2. — It may be technically possible to extend the application of this automatic calling interface for " route selection " signals of the telegraph character type (commonly known as keyboard selection). This possibility requires further study.

3. *Procedures*

The connections between various types of terminals covered by the following recommended procedures are:

I)	Telex terminal, with manual calling	⟶	Data terminal, with automatic answering
II)	Telex and data terminal, with manual calling, manual or automatic changeover to data	⟶	Data terminal, with automatic answering
III)	Data terminal, with automatic calling	⟶	Telex terminal, with teleprinter answering
IV)	Data terminal, with automatic calling	⟶	Telex and data terminal, with teleprinter answering, automatic changeover
V)	Data terminal, with automatic calling	⟶	Data terminal with automatic answering

The recommended procedures are:

* May be excluded if so required by local safety regulations.

CASE I

A Station (calling)	B Station (answering)
TELEX TERMINAL: *Manual calling* 1.A Call telex exchange. 2.A Operator dials, or keys, route selection signals and establishes connection in normal way. 3.A Receives A/B signals, either automatically or after sending WRU signal by the operator. 4.A Data exchange may commence.	DATA TERMINAL: *Automatic answering* 1.B Incoming call indication at control unit switches CT 125 to ON and " Call connected " signal is transmitted. 2.B If DTE is going to accept incoming call, CT 108/2 must be switched ON within 2 seconds after CT 125 is ON. 3.B Upon receipt of WRU signal, then a) if CT 108/2 is OFF, control unit sends " answer-back " (A/B) signals followed by the service signals ABS and then the " clearing " signal, b) if CT 108/2 is ON, control unit sends A/B signals. It then connects CT 103 and CT 104 to line; CT 106, CT 107, CT 109 are switched ON. (A/B unit simulator remains enabled.) 4.B Data exchange may commence.
TELEX OPERATOR CLEARS 5.A i) If the operator requires to clear down, he initiates clearing signal in the normal telex manner.	5.B i) Upon receipt of the clearing signal, the DCE returns the clearing signal, switches CT 125, CT 106, CT 107 and CT 109 to OFF. The line is disconnected from CT 103 and CT 104. 6.B i) DTE might or might not switch CT 108/2 to OFF.
	DATA TERMINAL CLEARS 5.B ii) DTE switches CT 108/2 to OFF. 6.B ii) The control unit switches CT 106, CT 107, CT 109 and CT 125 to OFF.

CASE I (continued)

A Station (calling)	B Station (answering)
	7.B ii) The control unit disconnects the line from CT 103 and CT 104; transmits the clearing signal.
5.A ii) Clearing signal received, terminal clears down automatically in normal telex manner.	8.B ii) DTE might or might not switch CT 108/2 to ON.

CASE II

A Station (calling)	B Station (answering)
TELEX AND DATA TERMINAL: *Manual calling — Manual or automatic changeover*	DATA TERMINAL: *Automatic answering*
1.A Call telex exchange.	
2.A Operator dials, or keys, route selection signals and establishes connection in normal way.	
	1.B Incoming call indication at control unit switches CT 125 to ON and " Call connected " signal is transmitted.
	2.B If DTE is going to accept incoming call, CT 108/2 must be switched ON within 2 seconds after CT 125 is ON.
	3.B Upon receipt of WRU signal, then a) if CT 108/2 is OFF, control unit sends A/B signals followed by the service signals ABS and then the clearing signal,
3.A Receives A/B signals, either automatically or after sending WRU signal by the operator.	b) if CT 108/2 is ON, control unit sends A/B signals. It then connects CT 103 and CT 104 to line; CT 106, CT 107, CT 109 are switched ON. (A/B unit simulator remains enabled.)

A Station (calling)	B Station (answering)
4.A Operator transmits four times signal No. 19 of Alphabet No. 2 and: a) immediately switches the line manually to the DTE, or b) (With automatic changeover). The transfer of the line from the teleprinter to CT 103 and CT 104 is effected automatically within the control unit by the transmission of the last signal of the sequence four times signal No. 19. Control unit switches CT 106, CT 107 and CT 109 to ON. A/B unit disabled.	4.B The receipt of the sequence four times signal No. 19 of the No. 2 Alphabet indicates to the DTE that the data equipment at the A station is connected to line. DCE disables the A/B unit simulator.
5.A Data exchange may commence.	5.B Data exchange may commence.
TELEX OPERATOR CLEARS: *Manual change-over only* 6.A i) Operator manually switches the line to the telex terminal. 7.A i) Operator initiates clearing signal in the normal telex manner, and A/B unit is re-enabled.	6.B i) Upon receipt of the clearing signal, the DCE returns the clearing signal, switches CT 125, CT 106, CT 107 and CT 109 to OFF. The line is disconnected from CT 103 and CT 104. 7.B i) DTE might or might not switch CT 108/2 to OFF. 8.B i) A/B unit simulator is re-enabled by DCE.
DATA TERMINAL CLEARS 6.A ii) DTE switches CT 108/2 to OFF.	

CASE II (continued)

A Station (calling)	B Station (answering)
7.A ii) The control unit of DCE switches CT 106, CT 107 and CT 109 to OFF, transfers the line from CT 103 and CT 104 to the telex terminal and transmits the clearing signal. A/B unit is re-enabled.	6.B ii) Upon receipt of the clearing signal, the DCE returns the clearing signal, switches CT 125, CT 106, CT 107 and CT 109 to OFF. The line is disconnected from CT 103 and CT 104. A/B unit simulator is re-enabled. 7.B ii) DTE might or might not switch CT 108/2 to OFF.
6.A iii) Clearing signal received, terminal clears down automatically in normal telex manner. A/B unit is re-enabled.	DATA TERMINAL CLEARS 6.B iii) DTE switches CT 108/2 to OFF. 7.B iii) The control unit switches CT 106, CT 107, CT 109 and CT 125 to OFF. 8.B iii) The control unit disconnects the line from CT 103 and CT 104; transmits the clearing signal. A/B unit simulator is re-enabled.

CASE III

A Station (calling)	B Station (answering)
DATA TERMINAL: *Automatic calling* 1.A DTE checks CT 213 is ON and CT 203 is OFF. 2.A DTE switches CT 202 to ON. 3.A DTE switches CT 108/2 to ON.	TELEX TERMINAL: *Teleprinter answering—No data equipment*

CASE III (continued)

A Station (calling)		B Station (answering)	
4.A	DCE (auto-calling equipment and control unit) calls telex exchange and switches CT 203 to ON.		
5.A	" Proceed to select " signal is received from telex exchange.		
6.A	DCE switches CT 210 to ON.		
7.A	DTE presents first digit on CT 206, CT 207, CT 208, CT 209.		
8.A	DTE switches CT 211 to ON.		
9.A	DCE sends the " route selection " signals for the first digit, then switches CT 210 to OFF.		
10.A	DTE switches CT 211 to OFF.		
11.A	The sequence of steps 6.A–10.A are repeated for each digit, including the last to be transmitted.		
12.A	DCE switches CT 210 to ON.	1.B	Call received in normal way.
13.A	DTE presents the " end of number " (EON) signals on CT 206, CT 207, CT 208 and CT 209, then switches CT 211 to ON.	2.B	" Call connected " signal sent.
14.A	DCE switches CT 210 to OFF and holds it in this condition for the remainder of call.		
15.A	DTE switches CT 211 to OFF.		
16.A	" Call connected " signal received, DCE connects CT 103 and CT 104 through to line and switches CT 106, CT 107 and CT 109 to ON. (A/B unit simulator in DCE still connected.)		
17.A	If A/B signals are not automatically received within 3 seconds * (see Note 2) DTE transmits signals No. 30 and No. 4 of the No. 2 Alphabet (WRU). The A/B unit simulator of the DCE must not respond to this WRU signal.	3.B	Upon receipt of the WRU signals, the A/B signals are transmitted.
18.A	The A/B signals having been received by the DTE data exchange may commence.	4.B	Data exchange may commence.
	* See Notes 1 and 5		

CASE III (continued)

A Station (calling)	B Station (answering)
DATA TERMINAL CLEARS 19.A i) DTE switches CT 108/2 and CT 202 to OFF. 20.A i) CT 108/2 being switched OFF, the control unit of DCE switches CT 106, CT 107 and CT 109 to OFF. Disconnects CT 103 and CT 104 from the line. Transmits the clearing signal.	5.B i) Clearing signal received, telex terminal clears down in normal telex way.
19.A ii) On receipt of clearing signal monitored from line by control unit, DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line and returns the clearing signal to line. 20.A ii) DTE switches CT 202 and CT 108/2 to OFF.	TELEX OPERATOR CLEARS 5.B ii) If operator requires to clear down, he initiates the clearing signal in the normal telex way.

CASE IV

A Station (calling)	B Station (answering)
DATA TERMINAL: *Automatic calling* 1.A DTE checks CT 213 is ON and CT 203 is OFF. 2.A DTE switches CT 202 to ON. 3.A DTE switches CT 108/2 to ON. 4.A DCE (auto-calling equipment and control unit) calls telex exchange and switches CT 203 to ON. 5.A " Proceed to select " signal is received from telex exchange.	TELEX + DATA TERMINAL: *Automatic change-over*

CASE IV (continued)

A Station (calling)	B Station (answering)
6.A DCE switches CT 210 to ON.	
7.A DTE presents first digit on CT 206, CT 207, CT 208, CT 209.	
8.A DTE switches CT 211 to ON.	
9.A DCE sends the route selection signals for the first digit, then switches CT 210 to OFF.	
10.A DTE switches CT 211 to OFF.	
11.A The sequence of steps 6.A-10.A are repeated for each digit, including the last to be transmitted.	
12.A DCE switches CT 210 to ON.	1.B Incoming call indication received, DCE switches CT 125 to ON.
13.A DTE presents the end of numbers (EON) signals on CT 206, CT 207, CT 208 and CT 209, then switches CT 211 to ON.	2.B " Call connected " signal sent.
14.A DCE switches CT 210 to OFF and holds it in this condition for the remainder of call.	
15.A DTE switches CT 211 to OFF.	
16.A " Call connected " signal received, DCE connects CT 103 and CT 104 through to line and switches CT 106, CT 107 and CT 109 to ON. (A/B unit simulator in DCE still connected.)	
17.A If A/B signals are not automatically received within 3 seconds * (see Note 2) DTE transmits signals No. 30 and No. 4 of the No. 2 Alphabet (WRU). The A/B unit simulator of the DCE must not respond to this WRU signal.	3.B Upon receipt of the WRU signals, the A/B signals are transmitted.
18.A Upon receipt of A/B signals (all coded in accordance with Alphabet No. 2), the DTE transmits over CT 103 the signal sequence of four times signal No. 19 of Alphabet No. 2.	
* See Notes 1 and 5	

CASE IV (continued)

A Station (calling)	B Station (answering)
19.A Upon recognition of the sequence of four times signal No. 19 by the DCE the A/B unit simulator is disabled.	4.B Upon receipt of the signal sequence of four times signal No. 19 a) if CT 108/2 is ON, DCE transfers the line from the telex terminal to CT 103 and CT 104; CT 106, CT 107 and CT 109 are switched ON. A/B is disabled. b) If CT 108/2 is OFF, the call is cleared by DCE.
20.A After a minimum delay of 500 ms data exchange may commence.	5.B Data exchange may commence.
21.A If the DTE requires to communicate with the telex terminal at the B station after data have been exchanged a special data signal should be sent from the DTE to effect this.	6.B Upon receipt of the appropriate data signal, DTE switches CT 132 to ON.
22.A After 2 seconds " telegraph " characters, encoded in accordance with the No. 2 Alphabet, may now be exchanged.	7.B The control unit of the DCE restores the line connection to the telex terminal and switches CT 106, CT 107, CT 109 and CT 125 to OFF.
	8.B Normal telegraph transmission may commence.
	9.B DTE switches CT 132 to OFF.
DATA TERMINAL CLEARS	DATA TERMINAL STILL CONNECTED
23.A i) DTE switches CT 108/2 and CT 202 to OFF.	
24.A i) CT 108/2 being switched OFF, the control unit of DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line, and transmits the clearing signal. A/B unit simulator is re-enabled.	10.B i) On receipt of the clearing signal monitored from the line by the control unit of DCE, CT 106, CT 107, CT 109 and CT 125 are switched OFF.

CASE IV (continued)

A Station (calling)	B Station (answering)
	Line is transferred from CT 103 and CT 104 to the telex terminal and the clearing signal is returned to line in the normal telex manner. A/B unit is re-enabled.
DATA TERMINAL CLEARS 23.A ii) DTE switches CT 108/2 and CT 202 to OFF. 24.A ii) CT 108/2 being switched OFF, the control unit of DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line. Transmits the clearing signal. A/B unit simulator is re-enabled.	TELEX TERMINAL RECONNECTED 10.B ii) Clearing signal received, terminal clears down in normal telex manner. A/B unit is re-enabled.
23.A iii) On receipt of the clearing signal monitored from line by control unit, DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line and returns the clearing signal to line. A/B unit simulator is re-enabled. 24.A iii) DTE switches CT 202 and CT 108/2 to OFF.	DATA TERMINAL CLEARS (not having returned to telex terminal) 10.B iii) DTE switches CT 108/2 to OFF. 11.B iii) The control unit switches CT 106, CT 107, CT 109, and CT 125 to OFF. 12.B iii) The control unit transfers the line from CT 103 and CT 104 to the telex terminal and transmits the clearing signal. A/B unit is re-enabled.

CASE IV (continued)

A Station (calling)	B Station (answering)
	TELEX OPERATOR CLEARS (having returned to telex terminal)
23.A iv) On receipt of the clearing signal monitored from line by control unit, DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line and returns the clearing signal to line. A/B unit simulator is re-enabled.	10.B iv) If operator requires to clear down, he initiates the clearing signal in the normal telex way.
24.A iv) DTE switches CT 202 and CT 108/2 to OFF.	

CASE V

A Station (calling)	B Station (answering)
DATA TERMINAL: *Automatic calling*	DATA TERMINAL: *Automatic answering*
1.A DTE checks CT 213 is ON and CT 203 is OFF.	
2.A DTE switches CT 202 to ON.	
3.A DTE switches CT 108/2 to ON.	
4.A DCE (auto-calling equipment and control unit) calls telex exchange and switches CT 203 to ON.	
5.A " Proceed to select " signal is received from telex exchange.	
6.A DCE switches CT 210 to ON.	
7.A DTE presents first digit on CT 206, CT 207, CT 208, CT 209.	
8.A DTE switches CT 211 to ON.	
9.A DCE sends the " route selection " signals for the first digit, then switches CT 210 to OFF.	

	A Station (calling)		B Station (answering)
10.A	DTE switches CT 211 to OFF.		
11.A	The sequence of steps 6.A-10.A are repeated for each digit, including the last to be transmitted.		
12.A	DCE switches CT 210 to ON.		
13.A	DTE presents the end of numbers (EON) signals on CT 206, CT 207, CT 208 and CT 209, then switches CT 211 to ON.	1.B	Incoming call indication at control unit switches CT 125 to ON and " call connected " signal is transmitted.
14.A	DCE switches CT 210 to OFF and holds it in this condition for the remainder of call.	2.B	If DTE is going to accept incoming call, CT 108/2 must be switched ON within 2 seconds after CT 125 is ON.
15.A	DTE switches CT 211 to OFF.		
16.A	" Call connected " signal received, DCE connects CT 103 and CT 104 through to line and switches CT 106, CT 107 and CT 109 to ON. (A/B unit simulator in DCE still connected.)	3.B	Upon receipt of WRU signal, then a) if CT 108/2 is OFF, control unit sends A/B signals followed by the service signals ABS and then the clearing signal, b) if CT 108/2 is ON, control unit sends A/B signals. It then connects CT 103 and CT 104 to line; CT 106, CT 107, CT 109 are switched ON. (A/B unit simulator remains enabled.)
17.A	If A/B signals are not automatically received within 3 seconds * (see Note 2) DTE transmits signals No. 30 and No. 4 of the No. 2 Alphabet (WRU). The A/B unit simulator of the DCE must not respond to this WRU signal.		
18.A	Upon receipt of A/B signals (all coded in accordance with Alphabet No. 2), the DTE transmits over CT 103 the signal sequence of four times signal No. 19 of Alphabet No. 2.		
19.A	Upon recognition of the sequence of four times signal No. 19 by the DCE the A/B unit simulator is disabled.		
		4.B	The receipt of the sequence four times signal No. 19 of the No. 2 Alphabet indicates to the DTE that the data equipment at the A station is connected to line. DCE disables the A/B unit simulator.

* See Notes 1 and 5

A Station (calling)	B Station (answering)
20.A After a minimum delay of 500 ms data exchange may commence.	5.B Data exchange may commence.
DATA TERMINAL CLEARS 21.A i) DTE switches CT 108/2 and CT 202 to OFF. 22.A i) CT 108/2 being switched OFF, the control unit of DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line and transmits the clearing signal. A/B unit simulator is re-enabled.	6.B i) Upon receipt of the clearing signal, the DCE returns the clearing signal, switches CT 125, CT 106, CT 107 and CT 109 to OFF. The line is disconnected from CT 103 and CT 104. A/B unit simulator is re-enabled. 7.B i) DTE might or might not switch CT 108/2 to OFF.
21.A ii) On receipt of the clearing signal monitored from line by control unit, DCE switches CT 106, CT 107 and CT 109 to OFF, disconnects CT 103 and CT 104 from the line and returns the clearing signal to line. A/B unit simulator is re-enabled. 22.A ii) DTE switches CT 202 and CT 108/2 to OFF.	DATA TERMINAL CLEARS 6.B ii) DTE switches CT 108/2 to OFF. 7.B ii) The control unit switches CT 106, CT 107, CT 109 and CT 125 to OFF. 8.B ii) The control unit disconnects the line from CT 103 and CT 104; transmits the clearing signal. A/B unit simulator is re-enabled.

Note 1. — In some national telex systems the " answer-back " from the A station is automatically tripped by a WRU signal from the telex exchange. Therefore the DTE at both A and B stations must be prepared for these signals to be transmitted before data can be reliably exchanged. The telex operator at the A station in the above cases I and II must also be prepared for these signals to be transmitted before attempting to send the sequence of four times signal No. 19.

Note 2. — This is a provisional time of three seconds. It is considered that a finally agreed figure may be in excess of this.

Note 3. — On the occasions when a call, for various reasons, is not connected through to a B station, service signals (e.g. OCC, DER) will be sent from the telex exchange to the A station. Since those service signals are always prefixed by a continuous signal of " Stop " polarity for minimum duration of 200 milliseconds, the lines at the A station will have been extended to the DTE and the service signals will pass straight to the DTE. The call will be cleared automatically by the telex exchange. The general question of service signals and the responses which the DTE should make to them (e.g. number of attempts to establish the call and timing between successive attempts) is to be studied.

Note 4. — The particular case of a data terminal with automatic calling working into a telex + data terminal with manual changeover has not been included as the indefinite delay of an operator in switching to data could be unacceptable.

Note 5. — Because of the possible reception by data terminal equipments of normal telegraph supervisory and/or service signals (see Notes 1 and 3) it might be advisable for the data terminal equipments at either end of the connection to exchange " message prefix " signals before commencing to exchange data. The coding and decision to use such message prefixes remains the responsibility of the data user.

RECOMMENDATION V.13

ANSWER-BACK UNIT SIMULATORS

(Mar del Plata, 1968)

a) The answer-back code must be released by a device capable of recognizing the " who are you? " signal of Alphabet No. 2 (five-unit code). Hence, this device must keep in a store unit the " figures " situation indicated by combination No. 30 received before combination No. 4 of this alphabet.

b) In view of the procedure adopted for the use of the sequence of four No. 19 combinations as the signal for passage from the " telex " position to the " data " position in terminal equipment, the introduction of this sequence (four times combination No. 19) in the 20 signals of the simulator answer-back code is to be avoided, since it is incompatible with the procedure already adopted.

Note. — It should be noted that, for the same reason of procedure, this four times combination No. 19 sequence should not be introduced in the answer-back code signals of a teleprinter associated with a manual or automatic call-transfer device.

c) The composition of the signals of the answer-back unit simulator can obviously be used for identification of the station obtained by the station which requests the call. If the identification is negative, it is up to this calling station to interrupt the unwanted connection.

Note. — On the other hand, it was agreed that identification in the opposite direction could not be achieved in a simple way by the answer-back unit simulator, since the answer-back code to be checked in this direction is that of the opposite station which is normally the one which has requested the connection.

d) In a telex installation intended for data transmission and equipped with an answer-back unit simulator instead of a teleprinter, the device for changeover from telex to data working—by the passage of the sequence of four No. 19 combinations—must be automatic.

e) The characteristics of the answer-back unit simulator should conform with Recommendation S.6, paragraphs 2 to 6, of Volume VII.

RECOMMENDATION V.21

200-BAUD MODEM STANDARDIZED FOR USE IN THE GENERAL SWITCHED TELEPHONE NETWORK

(Geneva, 1964, amended at Mar del Plata, 1968)

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:

a) manually by agreement between the operators, or

b) automatically.

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).

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 Hz.

For channel No. 2, it is 1750 Hz.

The frequency deviation is ± 100 Hz. In each channel, the higher characteristic frequency corresponds to the symbol 0.

The characteristic frequencies * as measured at the modulator output must not differ by more than ± 6 Hz from the nominal figures.

A maximum drift of frequency of ± 6 Hz is assumed for the line. Hence the demodulation equipment must tolerate drifts of ± 12 Hz 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. 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 echo suppressor tone disabler have been standardized by the C.C.I.T.T. (Recommendation G.161, section C) and the disabling tone should have the following characteristics:

disabling tone transmitted: 2100 ± 15 Hz at a level of -12 ± 6 dBm0,

the disabling tone to last at least 400 ms, the disabler should hold in the disabled mode for any single frequency sinusoid in the band from 390-3000 Hz having a level of -27 dBm0 or greater,

the tolerable interruptions by the data signal to last not more than 100 ms.

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 Npm0 (-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 are 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

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* The nominal characteristic frequencies:

Channel No. 1 (F_A = 1180 Hz and F_Z = 980 Hz)

Channel No. 2 (F_A = 1850 Hz and F_Z = 1650 Hz).

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).

Note. — In the case of multipoint operation the use of the alarms is the subject of further study.

9. *Interchange circuits*

a) *Interchange circuits essential for the modems when used on the general switched telephone network or non-switched leased telephone circuits*

The configurations of interchange circuits are those essential for the particular switched network or leased circuit requirement indicated. Where one or more of such requirements

Interchange circuit		General switched telephone network including terminals equipped for manual calling manual answering automatic calling automatic answering	Non-switched leased telephone circuits	
Number	Designation		Point to point	Multipoint
101*	Protective ground or earth	x	x	x
102	Signal ground or common return	x	x	x
103	Transmitted data	x	x	x
104	Received data	x	x	x
105	Request to send	–	x (Note 3)	x
106	Ready for sending	x	x	x
107	Data set ready	x	x	x
108/1	Connect data set to line	x (Note 1)	x	x
108/2	Data terminal ready	x (Note 1)	–	–
109	Data channel received line signal detector	x	x	x
125	Calling indicator	x	–	–
126	Select transmit frequency	–	–	x (Note 2)

* May be excluded if so required by local safety regulations.

are provided in a modem, then all of the appropriate interchange circuit facilities should be provided.

Note 1. — This circuit shall be capable of operation as circuit 108/1 " Connect data set to line " or circuit 108/2 " Data terminal ready " depending on its use. For automatic calling it shall be used as 108/2 only.

Note 2. — Circuit 126 controls the functions of circuits 126 and 127 as defined in Recommendation V.24.

Note 3. — Circuit 105 is not required when alternate voice/data is used on non-switched leased point-to-point circuits.

Note 4. — Interchange circuits indicated by x must be properly terminated according to Recommendation V.24 in the data terminal equipment and data communication equipment.

b) *Response times of circuits 106 and 109*

Definitions

i) Circuit 109 response times are the times that elapse between the connection or removal of a tone to or from the modem receive line terminals and the appearance of the corresponding ON or OFF condition on circuit 109.

The test tone should have a frequency corresponding to the characteristic frequency of binary 1 and be derived from a source with an impedance equal to the nominal input impedance of the modem under test.

The level of the test tone should fall into the level range between 1 dB above the actual threshold of the received line signal detector and the maximum admissible level of the received signal. At all levels within this range the measured response times shall be within the specified limits.

ii) Circuit 106 response times are the times from the connection of an ON or OFF condition on:

— circuit 105 (where it is provided) to the appearance of the corresponding ON or OFF condition on circuit 106;

— circuit 109 (where circuit 105 is not provided) to the appearance of the corresponding ON or OFF condition on circuit 106.

c) *Response times*

	Note 1	Note 2
Circuit 106		
OFF to ON	20-50 ms	400-1000 ms
ON to OFF	⩽ 2 ms	⩽ 2 ms
Circuit 109		
OFF to ON	⩽ 20 ms	300-700 ms
ON to OFF	20-80 ms	20-80 ms

Note 1. — These times are used on leased point-to-point networks without alternate voice-data facilities and on leased multipoint facilities.

Note 2. — These times are used on general switch network service and on leased point-to-point circuits with alternate voice-data.

d) *Threshold of data channel received line signal detector*

Level of received line signal at receive line signal terminals of modem for all types of connection, i.e. general switched telephone network or non-switched leased telephone circuit:

greater than	—43 dBm	circuit 109 ON
less than	—48 dBm	circuit 109 OFF

The condition of circuit 109 for levels between —43 dBm and —48 dBm is not specified except that the signal detector shall exhibit a hysteresis action such that the level at which the OFF to ON transition occurs shall be at least 2 dB greater than for the ON to OFF transition.

e) *Clamping to binary 1 condition of circuit 104 (received data)*

Two options shall be provided in the modem:

i) When clamping is not used there is no inhibition of the signals on circuit 104 (received data). There is no protection against noise, supervisory and control tones, switching transients etc., appearing on circuit 104.

ii) When clamping is used, circuit 104 (received data) is held in a marking condition (binary 1) when circuit 109 (received line signal detector) is in the OFF condition. When circuit 109 is ON the clamp is removed and circuit 104 can respond to the input signals of the modem.

10. 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 Hz), assuming up to 35 dB attenuation at the frequency 1750 Hz.

b) The data modem should have no adjustment for send level or receive sensitivity under the control of the operator.

11. In case of interruption of a leased circuit, the use of a non-standardized modem over the switched connection established as a substitute for the leased circuit is not recommended.

RECOMMENDATION V.22

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

(Geneva, 1964, amended at Mar del Plata, 1968)

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.

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. Modulation rates and data signalling rates should in no case deviate from the nominal values by more than $\pm 0.5\%$.

When the timing of the transmission is controlled by the data communication equipment the mean modulation or data signalling rate should be within the limit of $\pm 0.03\%$ with respect to the nominal value.

This stricter limit ensures that synchronism is not lost as a result of short line breaks.

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

Note 2. — For parallel transmissions of multi-condition serial modulations and their modulation rates, see Recommendation V.30.

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.

* See Notes 1 and 2.

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 $m\frac{1}{T}\log_2 n$ $\left(\frac{m}{T}\text{ in case of a two-condition modulation}\right)$.

RECOMMENDATION V.23

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

(Geneva, 1964, amended at Mar del Plata, 1968)

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 Hz	1300 Hz	1700 Hz
Mode 2: up to 1200 bauds	1700 Hz	1300 Hz	2100 Hz

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 Hz 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 ± 10 Hz 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 ± 10 Hz 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 ± 20 Hz.

A maximum frequency drift of $\pm$ 6 Hz 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 Hz.

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 Hz	450 Hz

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 v.f. telegraph-type channel, the frequency tolerances should be as recommended in Recommendation R.35 (*White Book*, Volume VII) for frequency-shift voice-frequency telegraphy.

The $\pm$ 6 Hz 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 Hz), 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. *Interchange circuits*

The configurations of interchange circuits are those essential for the particular switched network or leased circuit requirement as indicated in Tables a and b. Where one or more of such requirements are provided in a modem, then all the appropriate interchange circuits should be provided.

a) *Interchange circuits essential for the modems when used on the general switched telephone network, including terminals equipped for manual calling or answering or automatic calling or answering*

Interchange circuit		Forward (data) channel one-way system				Forward (data) channel either way system	
		Without backward channel		With backward channel		Without backward channel	With backward channel
No.	Designation	Transmit end	Receive end	Transmit end	Receive end		
101*	Protective ground or earth	x	x	x	x	x	x
102	Signal ground or common return	x	x	x	x	x	x
103	Transmitted data	x	–	x	–	x	x
104	Received data	–	x	–	x	x	x
105	Request to send	–	–	–	–	x	x
106	Ready for sending	x	–	x	–	x	x
107	Data set ready	x	x	x	x	x	x
108/1 or 108/2 (Note 1)	Connect data set to line / Data terminal ready	x	x	x	x	x	x
109	Data channel received line signal detector	–	x	–	x	x	x
111	Data signalling rate selector (DTE)	x	x	x	x	x	x
118	Transmitted backward channel data	–	–	–	x	–	x
119	Received backward channel data	–	–	x	–	–	x
120	Transmit backward channel line signal	–	–	–	–	–	x
121	Backward channel ready	–	–	–	x	–	x
122	Backward channel received line signal detector	–	–	x	–	–	x
125	Calling indicator	x	x	x	x	x	x

* May be excluded if so required by local safety regulations.

Note 1. — This circuit shall be capable of operation as circuit 108/1 (connect data set to line) or circuit 108/2 (data terminal ready) depending on its use. For automatic calling it shall be used as 108/2 only.

Note 2. — Interchange circuits indicated by x must be properly terminated according to Recommendation V.24 in the data terminal equipment and data communication equipment.

b) *Interchange circuits essential for the modems when used on non-switched leased telephone circuits*

Interchange circuit		Forward (data) channels one-way system				Forward (data) channel either way or both ways simultaneously system	
		Without backward channel		With backward channel			
No.	Designation	Transmit end	Receive end	Transmit end	Receive end	Without backward channel	With backward channel
101*	Protective ground or earth	x	x	x	x	x	x
102	Signal ground or common return	x	x	x	x	x	x
103	Transmitted data	x	–	x	–	x	x
104	Received data	–	x	–	x	x	x
105	Request to send	x	–	x	–	x	x
106	Ready for sending	x	–	x	–	x	x
107	Data set ready	x	x	x	x	x	x
108/1	Connect data set to line	x	x	x	x	x	x
109	Data channel received line signal detector	–	x	–	x	x	x
111	Data signalling rate selector (DTE)	x	x	x	x	x	x
118	Transmitted backward channel data	–	–	–	x	–	x
119	Received backward channel data	–	–	x	–	–	x
120	Transmitted backward channel line signal	–	–	–	x	–	x
121	Backward channel ready	–	–	–	x	–	x
122	Backward channel received line signal detector	–	–	x	–	–	x

* May be excluded if so required by local safety regulations.

c) *Response times of circuits 106 and 109*

Definitions

i) Circuits 109 and 122 response times are the times that elapse between the connection or removal of a tone to or from the modem receive line terminals and the appearance of the corresponding ON or OFF condition on circuits 109 and 122.

The test tone should have a frequency corresponding to the characteristic frequency of binary 1 and be derived from a source with an impedance equal to the nominal input impedance of the modem.

The level of the test tone should fall within the level range between 3 dB above the actual threshold of the received line signal detector and the maximum admissible level of the received signal. At all levels within this range the measured response times shall be within the specified limits.

ii) Circuit 106 response times are from the connection of an ON or OFF condition on:

— circuit 105 (where it is provided) to the appearance of the corresponding ON or OFF condition on circuit 106;

— circuit 107 (where circuit 105 is not provided) to the appearance of the corresponding ON or OFF condition on circuit 106.

iii) Circuit 121 response times are from the connection of an ON or OFF condition on:

— circuit 120 (where it is provided) to the appearance of the corresponding ON or OFF condition on circuit 121;

— circuit 109 (where circuit 120 is not provided) to the appearance of the corresponding ON or OFF condition on circuit 121.

d) *Response times*

	Note 1	Note 2
Circuit 106		
OFF to ON	750 ms to 1400 ms	a) 20 ms to 40 ms b) 200 ms to 275 ms
ON to OFF	$\leqslant$ 2 ms	$\leqslant$ 2 ms
Circuit 109		
OFF to ON	300 ms to 700 ms	10 ms to 20 ms
ON to OFF	5 ms to 15 ms	5 ms to 15 ms

Circuit 121	
OFF to ON	80 ms to 160 ms
ON to OFF	$\leqslant$ 2 ms
Circuit 122	
OFF to ON	$<$ 80 ms
ON to OFF	15 ms to 80 ms

Note 1. — For automatic calling and answering, the longer response times of circuits 106 and 109 are to be used during call establishment only.

Note 2. — The choice of response times depends upon the system application:

a) no protection given against line echoes;
b) protection given against line echoes.

Note 3. — The above parameters are provisional and are the subject of further study.

e) *Threshold of data channel and backward channel received line signal detectors*

Level of received line signal at receive line terminals of modem for all types of connections, i.e. general switched telephone network or non-switched leased telephone circuits:

greater than −43 dBm	Circuits 109/122 ON
less than −48 dBm	Circuits 109/122 OFF

The condition of circuits 109 and 122 for levels between −43 dBm and −48 dBm is not specified except that the signal detectors shall exhibit a hysteresis action such that the level at which the OFF to ON transition occurs is at least 2 dB greater than that for the ON to OFF transition.

f) *Clamping to binary condition 1 of circuit 104 (Received data) and circuit 119 (Received backward channel data)*

Two options shall be provided in the modem:

i) When clamping is not used there is no inhibition of the signals on circuits 104 and 119. There is no protection against noise, supervisory and control tones, switching transients etc. from appearing on circuits 104 and 119.

ii) When clamping is used, circuit 104 is held in a marking condition (binary 1) under the conditions defined below. When these conditions do not exist the clamp is removed and circuit 104 can respond to the input signals of the modem:

— when circuit 109 is in the OFF condition;

— when circuit 105 is in the ON condition and the modem is used in half duplex mode (turnaround systems). To protect circuit 104 from false signals a delay device shall be provided to maintain circuit 109 in the OFF condition for a period of 150 ± 25 ms after circuit 105 has been turned from ON to OFF. The use of this additional delay is optional.

iii) When clamping is used, circuit 119 is held in a marking condition (binary 1) under the conditions defined below. When these conditions do not exist the clamp is removed and circuit 119 can respond to the input signals of the modem:

— when circuit 122 is in the OFF condition.

9. *Equipment for the disablement of echo suppressors*

(See Recommendation V.21, section 5.)

10. *Inclusion of a clock in the modem*

A clock is not an essential item in the standardized modem. However, the modem may conveniently include a clock when used primarily for synchronous transmission.

11. *Temporary use of non-standardized modems over the general switched network*

In case of interruption of a leased circuit, the use of a non-standardized modem over the switched connection established as a substitute for the leased circuit is not recommended.

RECOMMENDATION V.24

FUNCTIONS AND ELECTRICAL CHARACTERISTICS OF CIRCUITS AT THE INTERFACE BETWEEN DATA TERMINAL EQUIPMENT AND DATA COMMUNICATION EQUIPMENT

(Geneva, 1964, amended at Mar del Plata, 1968)

CONTENTS

Section

I. SCOPE

I–1 This Recommendation applies to the interconnecting circuits, being called interchange circuits, between data terminal equipment and data communication equipment* for the transfer of binary data, control and timing signals. This Recommendation also applies to both sides of separate intermediate equipment, which may be inserted between these two classes of equipment.

In any type of practical equipment a selection will be made from the range of interchange circuits defined in this Recommendation, as appropriate. When by mutual arrangement other circuits are to be used, these additional circuits should conform to the electrical characteristics specified in this Recommendation.

The actual interchange circuits to be used in a particular data communication equipment are those indicated in the appropriate C.C.I.T.T. Recommendation, e.g., the usage of circuits 108/1, 108/2, 126 and 127 in modems for 200 and 600/1200 bits per second are indicated in Recommendations V.21 and V.23. However, it is intended in addition to formulate, at a later stage, operational guide lines for the selection of interchange circuits for data communication equipment which are not currently covered by a C.C.I.T.T. Recommendation.

* See definition 53.05.

I–2 The data communication equipment may include signal converters, timing generators, pulse generators, and control circuitry together with equipment to provide other functions such as error control, automatic calling and automatic answering.

Some of this equipment may be separate intermediate equipment or located in the data terminal equipment.

I–3 The range of interchange circuits defined in this Recommendation is applicable, for example:

a) to synchronous and asynchronous data communications;

b) to data communication on leased-line service, either two-wire or four-wire, either point-to-point or multipoint operation;

c) to data communication on switched network service, either two-wire or four-wire;

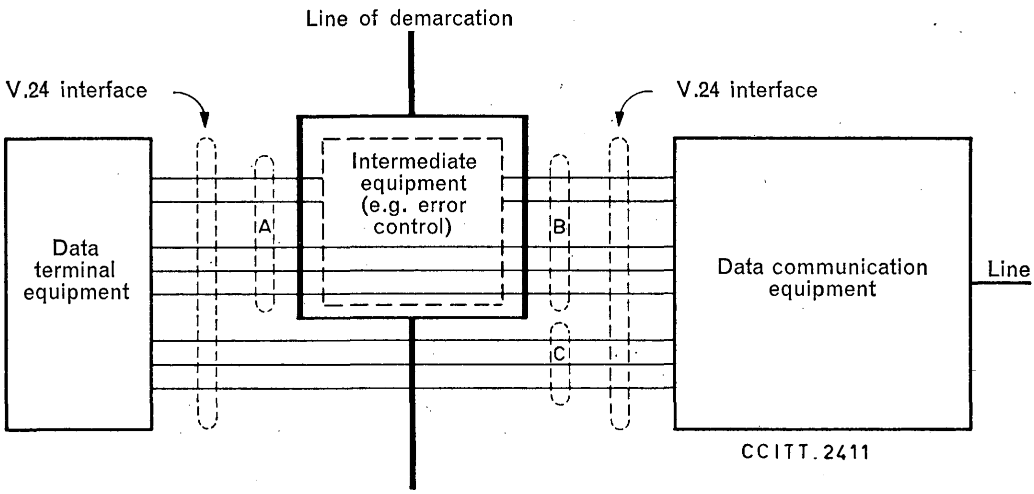
d) only where short interconnecting cables are used between data terminal equipment and data communication equipment. An explanation of short cables is given in section II.

I–4 The electrical characteristics defined in this Recommendation only apply to:

a) interchange circuits, on which the signalling rate does not exceed the limit of 20 000 bits per second;

b) interchange circuits, which can be represented by the equivalent circuit in section V.1.

II. Line of demarcation



Without intermediate equipment the selections A and B are identical.
Selection C may be a selection specifically for automatic calling.

FIGURE 1. — **Illustration of general layout of communication equipment**

The interface between data terminal equipment and data communication equipment is located at a connector, which is the interchange point between these two classes of equipment. Separate connectors may be provided for the interchange circuits associated with the signal-conversion or similar equipment and those associated with the automatic calling equipment.

The connector(s) will not necessarily be physically attached to the data communication equipment and may be mounted in a fixed position near the data terminal equipment.

An interconnecting cable or cables will normally be provided with the data terminal equipment. The use of short cables is recommended with the length solely limited by the load capacitance and other electrical characteristics, specified in section V.

III. Definitions of interchange circuits

III–1 *100-series. General application*

A list of these interchange circuits is presented in tabular form in Figure 2.

Circuit 101—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 102—Signal ground or common return

This conductor establishes the common reference potential for all interchange circuits in the 100-series, except circuit 101 (protective ground or earth). Within the data communication equipment, this circuit shall be brought to one point, and it shall be possible to connect this point to circuit 101 by means of a wire strap inside the equipment. This wire strap can be connected or removed at installation, as may be required to meet applicable regulations or to minimize the introduction of noise into electronic circuitry.

Circuit 103—Transmitted data

Direction: TO data communication equipment.

The data signals originated by the data terminal equipment to be transmitted via the data channel to one or more remote data stations are passed on this circuit to the data communication equipment.

The data terminal equipment shall hold circuit 103 in the binary 1 condition during any time interval between characters or words, and at all other times when no data are to be transmitted via the data channel. The data terminal equipment shall not transfer data on circuit 103 unless an ON condition is present on all of the following four circuits, where implemented:

1. Circuit 105 — Request to send
2. Circuit 106 — Ready for sending
3. Circuit 107 — Data set ready
4. Circuit 108.1/108.2 — Connect data set to line/data terminal ready

All data transferred on circuit 103 during the time an ON condition is present on all of the above four circuits, where implemented, shall be transmitted by the data communication equipment.

Circuit 104—Received data

Direction: FROM data communication equipment.

The data signals generated by the data communication equipment in response to data channel line signals received from a remote data station are passed on this circuit to the data terminal equipment.

For operation with clamping of circuit 104 refer to section IV.

Circuit 105—Request to send

Direction: TO data communication equipment.

Signals on this circuit control the data channel transmit function of the data communication equipment.

The ON condition causes the data communication equipment to assume the data channel transmit mode. This mode may also include the transmission of line signals required for data channel conditioning (equalization, synchronization, clamp removal, etc.), provided that circuit 107 (data set ready) is ON. The ON condition must be maintained as long as the data terminal equipment desires to transmit or is transferring data on circuit 103 (transmitted data).

The OFF condition causes the data communication equipment to assume the data channel non-transmit mode, when all data transferred on circuit 103 (transmitted data) have been transmitted. When circuit 105 is turned OFF it shall not be turned ON again until circuit 106 (ready for sending) is turned OFF by the data communication equipment.

For operation with clamping of circuit 104 (received data), refer to section IV.

Circuit 106—Ready for sending

Direction: FROM data communication equipment.

Signals on this circuit indicate whether the data communication equipment is conditioned to transmit data on the data channel.

The ON condition indicates that the data communication equipment is conditioned to transmit data on the data channel.

The OFF condition indicates that the data communication equipment is not prepared to transmit data on the data channel.

Where circuit 105 (request to send) is provided, the ON and OFF conditions on circuit 106 shall be responses to the ON and OFF conditions on circuit 105. For the appropriate response times of circuit 106, and for the operation of circuit 106, when circuit 105 (request to send) is not provided refer to the relevant Recommendation for data communication equipment.

Circuit 107—Data set ready

Direction: FROM data communication equipment.

Signals on this circuit indicate whether the data communication equipment is ready to operate.

The ON condition indicates that the signal-conversion or similar equipment is connected to the line and that the data communication equipment is ready to exchange further control signals with the data terminal equipment to initiate the exchange of data.

The conditioning of a data channel, such as equalization and clamp removal, will not take place before circuit 107 is turned ON.

The OFF condition indicates that the data communication equipment is not ready to operate. The OFF condition on this circuit shall not impair the operation of circuit 125 (calling indicator).

Circuit 108/1—Connect data set to line

Direction: TO data communication equipment.

Signals on this circuit control switching of the signal-conversion or similar equipment to or from the line.

The ON condition causes the data communication equipment to connect the signal-conversion or similar equipment to the line regardless of the condition on any other interchange circuit.

The OFF condition causes the data communication equipment to remove the signal-conversion or similar equipment from the line, when the transmission of all data previously transferred on circuit 103 (transmitted data) has been completed. The OFF condition shall not disable the operation of circuit 125 (calling indicator).

When circuit 108/1 is turned OFF, it shall not be turned ON again until circuit 107 (data set ready) is turned OFF by the data communication equipment.

Note. — A wiring option shall be provided within the data communication equipment to select the circuit 108/1 or circuit 108/2 operation.

Circuit 108/2—Data terminal ready

Direction: TO data communication equipment.

Signals on this circuit control switching of the signal-conversion or similar equipment to or from the line.

The ON condition, indicating that the data terminal equipment is ready to operate, prepares the data communication equipment to connect the signal-conversion or similar equipment to the line and maintains this connection after it has been established by external means.

When the data communication equipment is conditioned for automatic answering of calls, connection to the line occurs only in response to a combination of the calling signal and an ON condition on circuit 108/2.

The data terminal equipment is permitted to present the ON condition on circuit 108/2 whenever it is ready to transmit or receive data.

The OFF condition causes the data communication equipment to remove the signal-conversion or similar equipment from the line, when the transmission of all data previously transferred on circuit 103 (transmitted data) has been completed. The OFF condition shall not disable the operation of circuit 125 (calling indicator).

When circuit 108/2 is turned OFF, it shall not be turned ON again until circuit 107 (data set ready) is turned OFF by the data communication equipment.

Note. — A wiring option shall be provided within the data communication equipment to select the circuit 108/1 or circuit 108/2 operation.

Circuit 109—Data channel received line signal detector

Direction: FROM data communication equipment.

Signals on this circuit indicate whether the received data channel line signal is within appropriate limits, as specified in the relevant Recommendation for data communication equipment.

The ON condition indicates that the received signal is within appropriate limits.

The OFF condition indicates that the received signal is not within appropriate limits.

For operation with clamping of circuit 104 (received data), refer to section IV.

Circuit 110—Data signal quality detector

Direction: FROM data communication equipment.

Signals on this circuit indicate whether there is a reasonable probability of an error in the data received on the data channel.

The ON condition indicates that there is no reason to believe that an error has occurred.

The OFF condition indicates that there is a reasonable probability of an error.

Circuit 111—Data signalling rate selector

(data terminal equipment source)

Direction: TO data communication equipment.

Either circuit 111 or circuit 112 may be used, but not both.

Signals on this circuit are used to select one of the two data signalling rates of a dual rate synchronous data communication equipment, or to select one of the two ranges of data signalling rates of a dual range asynchronous data communication equipment.

The ON condition selects the higher rate or range of rates.

The OFF condition selects the lower rate or range of rates.

Circuit 112—Data signalling rate selector

(data communication equipment source)

Direction: FROM data communication equipment.

Either circuit 111 or circuit 112 may be used, but not both.

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

The ON condition selects the higher rate or range of rates.

The OFF condition selects the lower rate or range of rates.

Circuit 113—Transmitter signal element timing

(data terminal equipment source)

Direction: TO data communication equipment.

Either circuit 113 or circuit 114 may be used, but not both.

Signals on this circuit provide the data communication equipment with signal element timing information.

The condition on this circuit shall be ON and OFF for nominally equal periods of time, and the transition from ON to OFF condition shall nominally indicate the centre of each signal element on circuit 103 (transmitted data).

Timing information on circuit 113 shall be provided at all times when circuit 107 (data set ready) is in the ON condition. In addition, it is permissible to provide timing information when circuit 107 is OFF.

During periods when timing information is not provided, circuit 113 shall be held in the OFF condition.

Circuit 114—Transmitter signal element timing

(data communication equipment source)

Direction: FROM data communication equipment.

Either circuit 113 or circuit 114 may be used, but not both.

Signals on this circuit provide the data terminal equipment with signal element timing information.

The condition on this circuit shall be ON and OFF for nominally equal periods of time. The data terminal equipment shall present a data signal on circuit 103 (transmitted data) in which the transitions between signal elements nominally occur at the time of the transitions from OFF to ON condition of circuit 114.

Timing information on circuit 114 shall be provided at all times when circuit 107 (data set ready) is in the ON condition. In addition, it is permissible to provide timing information when circuit 107 is OFF.

During periods when timing information is not provided, circuit 114 shall be held in the OFF condition.

Circuit 115—Receiver signal element timing

(data communication equipment source)

Direction: FROM data communication equipment.

Either circuit 115 or circuit 128 may be used, but not both.

Signals on this circuit provide the data terminal equipment with signal element timing information.

The condition on this circuit shall be ON and OFF for nominally equal periods of time, and a transition from ON to OFF condition shall nominally indicate the centre of each signal element on circuit 104 (received data).

Timing information on circuit 115 shall be provided at all times when circuit 109 (data channel received line signal detector) is in the ON condition. This timing information may be present following the transition from ON to OFF condition on circuit 109 for a period of time consistent with the stability of the timing circuitry in the data communication equipment.

During periods when timing information is not provided, circuit 115 shall be held in the OFF condition.

Circuit 116—Select stand-by

Direction: TO data communication equipment.

Signals on this circuit are used to select the normal or stand-by facilities, such as signal converters and communication channels, provided within the data communication equipment.

The ON condition selects the stand-by mode of operation, causing the data communication equipment to replace predetermined facilities by their reserves.

The OFF condition causes the data communication equipment to replace the stand-by facilities by the normal. The OFF condition on this circuit shall be maintained whenever the stand-by facilities are not required for use.

Circuit 117—Stand-by indicator

Direction: FROM data communication equipment.

Signals on this circuit indicate whether the data communication equipment is conditioned to operate in its stand-by mode with the predetermined facilities replaced by their reserves.

The ON condition indicates that the data communication equipment is conditioned to operate in its stand-by mode.

The OFF condition indicates that the data communication equipment is conditioned to operate in its normal mode.

Circuit 118—Transmitted backward channel data

Direction: TO data communication equipment.

This circuit is equivalent to circuit 103 (transmitted data) except that it is used to transmit data via the backward channel.

The data terminal equipment shall hold circuit 118 in the binary 1 condition during any time interval between characters or words, and at all other times when no data are to be transmitted via the backward channel. The data terminal equipment shall not transfer data on circuit 118 unless an ON condition is present on all of the following four circuits, where implemented:

1. Circuit 120 — Transmit backward channel line signal
2. Circuit 121 — Backward channel ready
3. Circuit 107 — Data set ready
4. Circuit 108/1-108/2 — Connect data set to line/data terminal ready.

Circuit 119—Received backward channel data

Direction: FROM data communication equipment.

This circuit is equivalent to circuit 104 (received data), except that it is used for data received on the backward channel.

For operation with clamping of circuit 119, refer to section IV.

Circuit 120—Transmit backward channel line signal

Direction: TO data communication equipment.

This circuit is equivalent to circuit 105 (request to send), except that it is used to control the backward channel transmit function of the data communication equipment.

The ON condition causes the data communication equipment to assume the backward channel transmit mode. This mode includes the transmission of line signals required for backward channel conditioning, provided that circuit 107 (data set ready) is ON.

The ON condition must be maintained as long as the data terminal equipment desires to transmit via the backward channel or is transferring data on circuit 118 (transmitted backward channel data).

The OFF condition causes the data communication equipment to assume the backward channel non-transmit mode, when all data transferred on circuit 118 (transmitted backward channel data) have been transmitted. When circuit 120 is turned OFF, it shall not be turned ON again until circuit 121 (backward channel ready) is turned OFF by the data communication equipment.

For operation with clamping of circuit 119 (received backward channel data), refer to section IV.

Circuit 121—Backward channel ready

Direction: FROM data communication equipment.

This circuit is equivalent to circuit 106 (ready for sending), except that it is used to indicate whether the data communication equipment is conditioned to transmit data on the backward channel.

The ON condition indicates that the data communication equipment is conditioned to transmit data on the backward channel.

The OFF condition indicates that the data communication equipment is not conditioned to transmit data on the backward channel.

Where circuit 120 (transmit backward channel line signal) is provided, the ON and OFF conditions on circuit 121 shall be responses to the ON and OFF conditions on circuit 120. For the appropriate response times of circuit 121, and for the operation of circuit 121, when circuit 120 (transmit backward channel line signal) is not provided, refer to the relevant Recommendation for data communication equipment.

Circuit 122—Backward channel received line signal detector

Direction: FROM data communication equipment.

This circuit is equivalent to circuit 109 (data channel received line signal detector), except that it is used to indicate whether the received backward channel line signal is within appropriate limits, as specified in the relevant Recommendation for data communication equipment.

For operation with clamping of circuit 119 (received backward channel data), refer to section IV.

Circuit 123—Backward channel signal quality detector

Direction: FROM data communication equipment.

This circuit is equivalent to circuit 110 (data signal quality detector), except that it is used to indicate the signal quality of the received backward channel line signal.

Circuit 124—Data channel receiver cut-off

Direction: TO data communication equipment.

This circuit may be used only when the clamping option, as defined in section IV, is not used.

The ON condition causes the data communication equipment to clamp circuit 104 (received data) to the binary 1 condition.

The OFF condition permits received data to be transferred to the data terminal equipment on circuit 104 (received data).

Circuit 125—Calling indicator

Direction: FROM data communication equipment.

Signals on this circuit indicate whether a calling signal is being received by the data communication equipment.

The ON condition indicates that a calling signal is being received.

The OFF condition indicates that no calling signal is being received, and shall appear also approximately coincident with any interruptions within a pulse-modulated calling signal.

Circuit 126—Select transmit frequency

Direction: TO data communication equipment.

Signals on this circuit are used to select the required transmit frequency of the data communication equipment.

The ON condition selects the higher transmit frequency.

The OFF condition selects the lower transmit frequency.

Circuit 127—Select receive frequency

Direction: TO data communication equipment.

Signals on this circuit are used to select required receive frequency of the data communication equipment.

The ON condition selects the lower receive frequency.

The OFF condition selects the higher receive frequency.

Circuit 128—Receiver signal element timing
(data terminal equipment source).

Direction: TO data communication equipment.

Either circuit 128 or circuit 115 may be used, but not both.

Signals on this circuit provide the data communication equipment with signal element timing information.

The condition on this circuit shall be ON and OFF for nominally equal periods of time. The data communication equipment shall present a data signal on circuit 104 (received 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 circuit 128.

During periods when timing information is not provided, circuit 128 shall be held in the OFF condition.

Circuit 129—Backward channel receiver cut-off

Direction: TO data communication equipment.

This circuit is equivalent to circuit 124 (data channel receiver cut-off), except that it controls a clamp on circuit 119 (received backward channel data). This circuit may be used only when the clamping option, as defined in section IV, is not used.

Circuit 130—Transmit backward tone

Direction: TO data communication equipment.

Signals on this circuit control the transmission of a backward channel tone.

The ON condition causes the data communication equipment to transmit a backward channel tone.

The OFF condition causes the data communication equipment to stop the transmission of a backward channel tone.

Circuit 131—Received character timing

Direction: FROM data communication equipment.

Signals on this circuit provide the data terminal equipment with character timing information, as specified in the relevant Recommendation for data communication equipment.

Circuit 132—Return to non-data mode

Direction: TO data communication equipment.

Signals on this circuit are used to restore the non-data mode provided with the data communication equipment, without loosing the line connection to the remote station.

The ON condition causes the data communication equipment to restore the non-data mode. When the non-data mode has been established, this circuit must be turned OFF.

Circuit 133—Ready for receiving

Direction: TO data communication equipment.

Signals on this circuit control the transfer of data on circuit 104 (received data), indicating whether the data terminal equipment is capable to accept a certain amount of data (e.g. a block of data), specified in the appropriate Recommendation for intermediate equipment, for example error control equipment.

The ON condition must be maintained, whenever the data terminal equipment is capable to accept a block of data, and causes the intermediate equipment to transfer the received data to the data terminal equipment.

The OFF condition indicates that the data terminal equipment is not capable to accept a block of data, and causes the intermediate equipment to retain that block.

Circuit 134—Received data present

Direction: FROM data communication equipment.

Signals on this circuit are used to separate the information message from the supervisory message, transferred on circuit 104 (received data).

The ON condition indicates the data, which represent information message.

The OFF condition shall be maintained at all other times.

Interchange circuit No.	Interchange circuit name	Ground	Data		Control		Timing	
		Ground	From DCE **	To DCE **	From DCE **	To DCE **	From DCE **	To DCE **
1	2	3	4	5	6	7	8	9
101	Protective ground or earth	x						
102	Signal ground or common return	x						
103	Transmitted data			x				
104	Received data		x					
105	Request to send					x		
106	Ready for sending				x			
107	Data set ready				x			
108/1	Connect data set to line					x		
108/2	Data terminal ready					x		
109	Data channel received line signal detector				x			
110	Signal quality detector				x			
111	Data signalling rate selector (DTE *)					x		
112	Data signalling rate selector (DCE **)				x			
113	Transmitter signal element timing (DTE *)							x
114	Transmitter signal element timing (DCE **)						x	
115	Receiver signal element timing (DCE **)						x	
116	Select stand-by					x		
117	Stand-by indicator				x			
118	Transmitted backward channel data			x				
119	Received backward channel data		x					
120	Transmit backward channel line signal					x		
121	Backward channel ready				x			
122	Backward channel received line signal detector				x			
123	Backward channel signal quality detector				x			
124	Data channel receiver cut-off					x		
125	Calling indicator				x			
126	Select transmit frequency					x		

Interchange circuit No.	Interchange circuit name	Ground	Data		Control		Timing	
		Ground	From DCE **	To DCE **	From DCE **	To DCE **	From DCE **	To DCE **
1	2	3	4	5	6	7	8	9
127	Select receive frequency					X		
128	Receiver signal element timing (DTE *)							X
129	Backward channel receiver cut-off					X		
130	Transmit backward tone					X		
131	Received character timing						X	
132	Return to non-data mode					X		
133	Ready for receiving					X		
134	Received data present				X			

* Data terminal equipment
** Data communication equipment

FIGURE 2. — **100-series interchange circuits by category**

III–2. *200-series. Specifically for automatic calling*

A list of these interchange circuits is presented in tabular form in Figure 3.

For the proper procedures, refer to the relevant Recommendation for automatic calling procedures.

Circuit 201—Signal ground or common return

This conductor establishes the common reference potential for all 200-series interchange circuits except circuit 212 (protective ground or earth). Within the automatic calling equipment, this circuit shall be brought to one point, and it shall be possible to connect this point to circuit 212 (protective ground or earth) by means of a wire strap inside the equipment. This wire strap can be connected or removed at installation as may be required to meet applicable regulations or to minimize the introduction of noise into electronic circuitry.

Circuit 202—Call request

Direction: TO data communication equipment.

Signals on this circuit are used to condition the automatic calling equipment to originate a call and to switch the automatic calling equipment to or from the line.

The ON condition causes the data communication equipment to condition the automatic calling equipment to originate a call and to connect this equipment to the line. Circuit

202 must be turned OFF between calls or call attempts and shall not be turned ON again before circuit 203 (data line occupied) is turned OFF.

The OFF condition causes the automatic calling equipment to be removed from the line and indicates that the data terminal equipment has completed its use of the automatic calling equipment.

Circuit 203—Data line occupied

Direction: FROM data communication equipment.

Signals in this circuit indicate whether the communication channel is in use (e.g. for automatic calling, data or voice communication, test procedures).

The ON condition indicates that the communication channel is in use.

The OFF condition indicates that the communication channel is not in use, and that the data terminal equipment may originate a call, provided that circuit 213 (power indication) is ON.

Circuit 204—Distant station connected

Direction: FROM data communication equipment.

Signals on this circuit indicate whether a connection has been established to a remote data station.

The ON condition indicates the receipt of a signal from a remote communication equipment, that a connection to that equipment has been established. The ON condition on this circuit must be maintained until the data terminal equipment has completed its use of the automatic calling equipment, i.e. until circuit 202 (call request) is turned OFF.

The OFF condition shall be maintained at all other times.

Circuit 205—Abandon call

Direction: FROM data communication equipment.

Signals on this circuit indicate whether a pre-set time has elapsed between successive events in the calling procedure.

The ON condition indicates that a pre-set time has elapsed.

The OFF condition indicates that call origination can be proceeded. The OFF condition shall be maintained after circuit 204 (distant station connected) comes ON.

The initial time interval starts when circuit 202 (call request) comes ON. Subsequent time intervals start each time circuit 210 (present next digit) is turned OFF.

Digit signal circuits:

Circuit 206 — Digit signal (2^0)
Circuit 207 — Digit signal (2^1)
Circuit 208 — Digit signal (2^2)
Circuit 209 — Digit signal (2^3)

Direction: TO data communication equipment.

On these circuits the data terminal equipment presents the following code combinations, being the digits to be called and associated control characters.

The conditions on these four circuits shall not change while circuit 211 (digit present) is ON.

Information	*Binary states*			
	209	208	207	206
Digit 1	0	0	0	1
Digit 2	0	0	1	0
Digit 3	0	0	1	1
Digit 4	0	1	0	0
Digit 5	0	1	0	1
Digit 6	0	1	1	0
Digit 7	0	1	1	1
Digit 8	1	0	0	0
Digit 9	1	0	0	1
Digit 0	0	0	0	0
Control character EON	1	1	0	0
Control character SEP	1	1	0	1

The control character EON (end of number) causes the data communication equipment to take the appropriate action to await an answer of the called data station.

The control character SEP (separation) indicates the need for a pause between successive digits, and causes the automatic calling equipment to insert the appropriate time interval.

Circuit 210—Present next digit

Direction: FROM data communication equipment.

Signals on this circuit indicate whether the automatic calling equipment is ready to accept the next code combination on digit signal circuits 206, 207, 208 and 209.

The ON condition indicates that the automatic calling equipment is ready to accept the next code combination.

The OFF condition indicates that the automatic calling equipment is not ready to accept signals on the digit signal circuits. When circuit 210 is turned OFF, it shall not be turned ON again before circuit 211 (digit present) is turned OFF.

Circuit 211—Digit present

Direction: TO data communication equipment.

Signals on this circuit control the reading of the code combination presented on the digit signal circuits 206, 207, 208 and 209.

The ON condition causes the automatic calling equipment to read the code combination presented on the digit signal circuits.

Circuit 211 shall not be turned ON when circuit 210 (present next digit) is in the OFF condition, and not before the data terminal equipment has presented the required code combination on the digit signal circuits.

The OFF condition on this circuit prevents the automatic calling equipment from reading a code combination on the digit signal circuits.

Circuit 211 shall not be turned OFF before circuit 210 (present next digit) is turned OFF.

Circuit 212—Protective ground or earth

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

Circuit 213—Power indication

Direction: FROM data communication equipment.

Signals on this circuit indicate whether power is available within the automatic calling equipment.

The ON condition indicates that power is available within the automatic calling equipment and that the automatic calling equipment is ready to operate.

The OFF condition indicates that power is not available, and shall be detected as specified in paragraph V–6.

Interchange circuit No.	Interchange circuit name	Ground	Data		Control	
			From DCE*	To DCE*	From DCE*	To DCE*
201	Signal ground or common return	x				
202	Call request					x
203	Data line occupied				x	
204	Distant station connected				x	
205	Abandon call				x	
206	Digit signal (2^0)			x		
207	Digit signal (2^1)			x		
208	Digit signal (2^2)			x		
209	Digit signal (2^3)			x		
210	Present next digit				x	
211	Digit present					x
212	Protective ground or earth	x				
213	Power indication				x	

* DCE = Data communication equipment

FIGURE 3. — **200-series interchange circuits by category. Specifically for automatic calling**

IV. CLAMPING

When clamping is used the following clamping conditions shall be provided by the data communication equipment:

1. In all applications the data communication equipment shall hold, where implemented:

a) Circuit 104 (received data) in the binary 1 condition when circuit 109 (data channel received line signal detector) is in the OFF condition, and

b) Circuit 119 (received backward channel data) in the binary 1 condition when circuit 122 (backward channel received line signal detector) is in the OFF condition.

2. In addition the data communication equipment, arranged for half duplex (C.C.I.T.T. definition: simplex) operation (turn-around systems), shall also hold, where implemented:

a) circuit 104 (received data) in the binary 1 condition and circuit 109 (data channel received line signal detector) in the OFF condition when circuit 105 (request to send) is in the ON condition, and for a short interval (to be specified in Recommendations for data communication equipment) following the ON to OFF transition on circuit 105, and

b) circuit 119 (received backward channel data) in the binary 1 condition and circuit 122 (backward channel received line signal detector) in the OFF condition when circuit 120 (transmit backward channel line signal) is in the ON condition, and for a short interval (to be specified in Recommendations for data communication equipment) following the ON to OFF transition on circuit 120.

Without these clamping conditions, there is no inhibition of signals, due to excessive noise, supervisory and control signals, switching transients, etc., from appearing on circuit 104, circuit 119, circuit 109 and circuit 122.

V. Electrical characteristics of interchange circuits

The electrical characteristics defined in this section apply to all interchange circuits defined in sections III–1 and III–2. These are represented by the equivalent circuit of Figure 4 and the signalling rate must not exceed the limit of 20 000 bits per second.

V–1 *Interchange equivalent circuit*

Figure 4 shows the interchange equivalent circuit with the electrical parameters, which are specified in this section.

This equivalent circuit applies to all interchange circuits regardless of the category (data, timing or control) to which they belong.

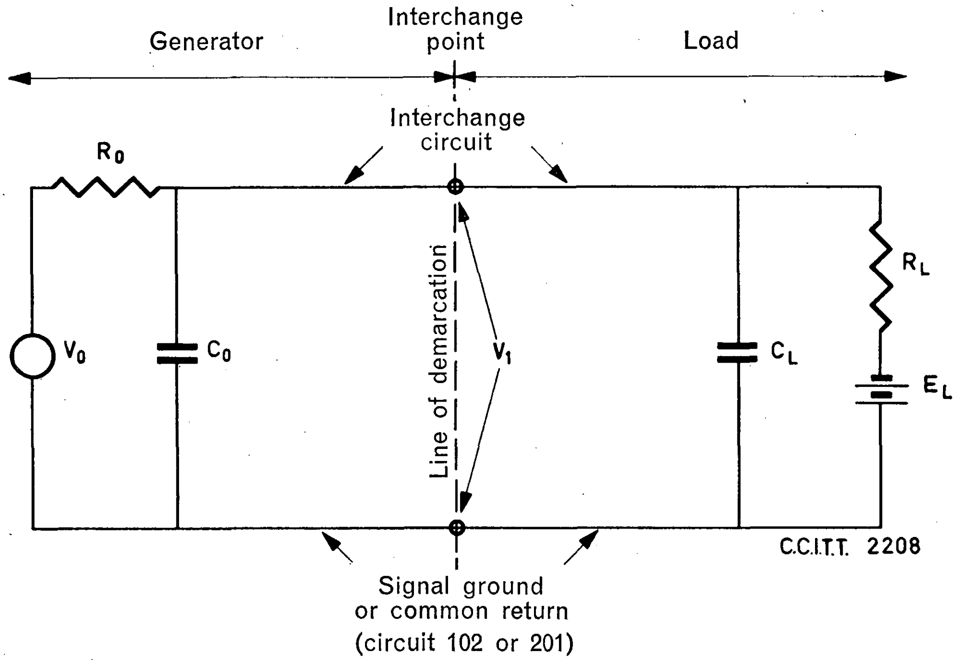


FIGURE 4. — **Interchange equivalent circuit**

This equivalent circuit is independent of whether the generator is located in the data communication equipment and the load in the data terminal equipment or vice versa.

The impedance associated with the generator (load) includes any cable impedance at the generator (load) side to the interchange point.

V_0 is the open-circuit generator voltage;

R_0 is the total effective d.c. resistance associated with the generator, measured at the interchange point;

C_0 is the total effective capacitance associated with the generator, measured at the interchange point;

V_1 is the voltage at the interchange point with respect to signal ground or common return;

C_L is the total effective capacitance associated with the load, measured at the interchange point;

R_L is the total effective d.c. resistance associated with the load, measured at the interchange point;

E_L is the open-circuit load voltage (bias).

V-2 *Load*

The impedance on the load side of an interchange circuit shall have a d.c. resistance (R_L) of not less than 3000 ohms, nor more than 7000 ohms, measured with an applied voltage (either positive or negative polarity) of 3 to 15 volts in magnitude.

The effective shunt capacitance (C_L) of the load, measured at the interchange point, shall not exceed 2500 picofarads.

To avoid inducing voltage surges on interchange circuits the reactive component of the load impedance shall not be inductive.

The open-circuit load voltage (E_L) shall not exceed 2 volts in magnitude.

The load on an interchange circuit shall not impair continuous operation with any input signals within the voltage limits specified in paragraph V–3.

V–3 *Generator*

The generator on an interchange circuit shall withstand an open-circuit and a short circuit between itself and any other interchange circuit (including generators and loads) without sustaining damage to itself or its associated equipment.

The open-circuit generator voltage (V_0) on any interchange circuit shall not exceed 25 volts in magnitude. The impedance (R_0 and C_0) on the generator side of an interchange circuit is not specified; however, the combination of V_0 and R_0 shall be selected such that a short circuit between any two interchange circuits shall not result in any case in a current in excess of one-half ampere.

Additionally, when the load open-circuit voltage (E_L) is zero, the voltage (V_1) at the interchange point shall not be less than 5 volts and no more than 15 volts in magnitude (either positive or negative polarity), for any load resistance (R_L) in the range between 3000 ohms and 7000 ohms.

The effective shunt capacitance (C_0) at the generator side of an interchange circuit is not specified. However, the generator shall be capable of driving all of the capacitance at the generator side (C_0), plus a load capacitance (C_L) of 2500 picofarads.

Note. — Relay or switch contacts may be used to generate signals on an interchange circuit, with appropriate measures to ensure that signals so generated comply with paragraph V–5.

V–4 *Significant levels (V_1)*

For data interchange circuits, the signal shall be considered in the binary 1 condition when the voltage (V_1) on the interchange circuit measured at the interchange point is more negative than −3 V. The signal shall be considered in the binary 0 condition when the voltage (V_1) is more positive than +3 V.

For control and timing interchange circuits, the circuit shall be considered ON when the voltage (V_1) on the interchange circuit is more positive than +3 V, and shall be considered OFF when the voltage (V_1) is more negative than −3 V.

$V_1 < -3$ volts	$V_1 > +3$ volts
1	0
OFF	ON

FIGURE 5. — **Correlation table**

Note. — In certain countries, in case of direct connection to d.c. telegraph-type circuits only, the voltage polarities in Figure 5 may be reversed.

The region between +3 V and −3 V is defined as the transition region. The signal state or circuit condition is not uniquely defined when voltage (V_1) is in the transition region. For an exception to this, see paragraph V–6.

V–5 *Signal characteristics*

The following limitations to the characteristics of signals transmitted across the interchange point, exclusive of external interferences, shall be met at the interchange point when the interchange circuit is loaded with any receiving circuit, which meets the characteristics specified in paragraph V–2.

These limitations apply to all (data, control and timing) interchange signals unless otherwise specified.

1. All interchange signals entering into the transition region shall proceed through this region to the opposite signal state and shall not re-enter this region until the next significant change of signal condition.
2. There shall be no reversal of the direction of voltage change while the signal is in the transition region.
3. For control interchange circuits, the time required for the signal to pass through the transition region during a change in state shall not exceed one millisecond.
4. For data and timing interchange circuits, the time required for the signal to pass through the transition region during a change in state shall not exceed 1 millisecond or 3% of the nominal duration of a signal element on that interchange circuit, whichever is the lesser.
5. To reduce crosstalk between interchange circuits the maximum instantaneous rate of voltage change will be limited. A provisional limit will be 30 volts per microsecond.
6. When electromechanical devices are used on interchange circuits, paragraphs 1 and 2 above do not apply to data interchange circuits.

V–6 *Circuit failures*

The following interchange circuits, where implemented, shall be used to detect either a power-off condition in the equipment connected through the interface or the disconnection of the interconnecting cable:

Circuit 105	(request to send)
Circuit 107	(data set ready)
Circuit 108/1-108/2	(connect data set to line/data terminal ready)
Circuit 120	(transmit backward channel line signal)
Circuit 202	(call request)
Circuit 213	(power indication)

The power-off impedance of the generator side of these circuits shall not be less than 300 ohms, measured with an applied voltage (either positive or negative polarity) not greater than 2 volts in magnitude referenced to signal ground or common return.

The load for these circuits shall interpret the power-off condition or the disconnection of the interconnecting cable as an OFF condition on these circuits.

RECOMMENDATION V.25

AUTOMATIC CALLING AND/OR ANSWERING EQUIPMENT ON THE GENERAL SWITCHED TELEPHONE NETWORK

(Mar del Plata, 1968)

1. *Scope*

1.1 This Recommendation is concerned with the setting-up of a data connection when automatic calling and/or answering equipment is used over international circuits.

Automatic calling and answering systems used within any single administration's area or between two administrations by bilateral agreement are not necessarily constrained by these proposals. In particular the use of 2100 Hz answering tone, as described in the text, could be substituted by another tone when the equipment is used over circuits not equipped with echo suppressors. Similarly the calling tone could be omitted by bilateral agreements, but attention is drawn to sections 8 and 9.

1.2 This standard describes the sequences of events involved in establishing a connection between an automatic calling data terminal and an automatic answering data terminal for the cases of both V.21 and V.23 modems. The system configuration proposed is shown in Figure 1.

Consideration is given only to: a) the events which affect the interfaces between the data terminal equipment and the data communication equipment; and b) the events on the line during establishment of a data call. Interactions within the data communications equipment are not considered, since such consideration is unnecessary for purposes of international standardization.

1.3 The proposed procedures are intended to be suitable for the three types of calls, namely:

a) automatic calling terminal to automatic answering terminal;

b) manual data terminal to automatic answering terminal;

c) automatic calling terminal to manual data terminal.

1.4 The data terminal equipment is responsible for:

- a) during call establishment:
  - i) ensuring that the data communication equipment is available for operation;
  - ii) providing the telephone number;
  - iii) deciding to abandon the call if it is unsuccessfully completed.

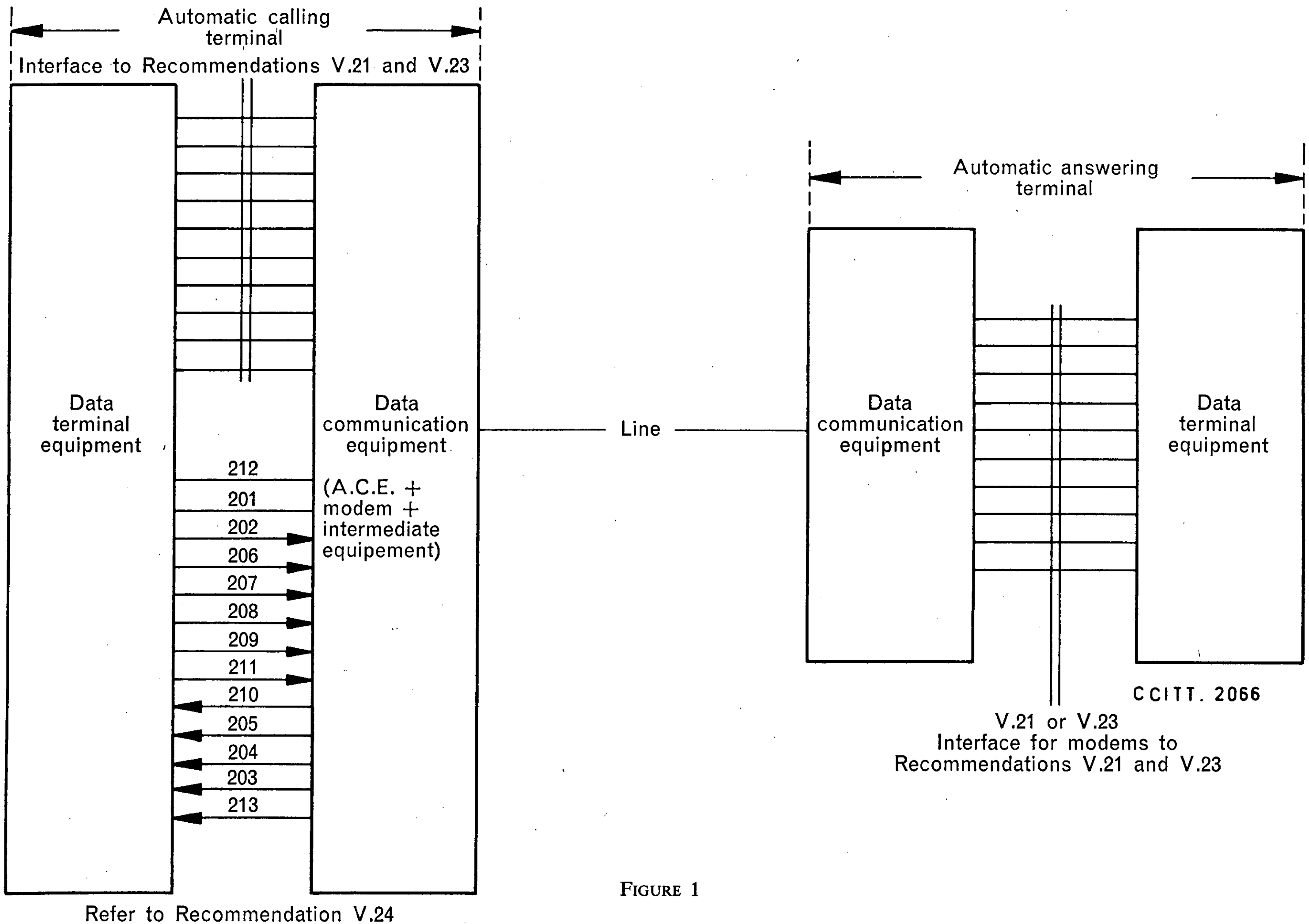

Automatic calling terminal
Interface to Recommendations V.21 and V.23
Data terminal equipment
Data communication equipment
(A.C.E. + modem + intermediate equipement)
212
201
202
206
207
208
209
211
210
205
204
203
213
Refer to Recommendation V.24
Line
Automatic answering terminal
Data communication equipment
Data terminal equipment
CCITT. 2066
V.21 or V.23
Interface for modems to Recommendations V.21 and V.23

FIGURE 1

b) after call is established:

i) establish identities;

ii) exchange such traffic as is appropriate;

iii) to initiate disconnect at calling and answering terminals.

2. *Abbreviations and definitions*

The following abbreviations are used in this Recommendation:

CT 104	Circuit 104, " Received data " interchange circuit
CT 105	Circuit 105, " Request to send " interchange circuit
CT 106	Circuit 106, " Ready for sending " interchange circuit
CT 107	Circuit 107, " Data set ready " interchange circuit
CT 108	Circuit 108, *either* 108/1 " Connect data set to line " interchange circuit *or* 108/2 " Data terminal ready " interchange circuit
CT 109	Circuit 109, " Data channel received line signal detector " interchange circuit
CT 119	Circuit 119, " Received backward channel data " interchange circuit
CT 120	Circuit 120, " Transmit backward channel line signal " interchange circuit
CT 121	Circuit 121, " Backward channel ready " interchange circuit
CT 122	Circuit 122, " Backward channel received line signal detector " interchange circuit
CT 125	Circuit 125, " Calling indicator " interchange circuit
CT 201	Signal ground
CT 202	" Call request " interchange circuit
CT 203	" Data line occupied " interchange circuit
CT 204	" Distant station connected " interchange circuit
CT 205	" Abandon call " interchange circuit
CT 206	" Digit signal " interchange circuit
CT 207	" Digit signal " interchange circuit
CT 208	" Digit signal " interchange circuit
CT 209	" Digit signal " interchange circuit
CT 210	" Present next digit " interchange circuit
CT 211	" Digit present " interchange circuit
CT 212	Protective ground
CT 213	" Power indication " interchange circuit
ACE	Automatic calling equipment
DCE	Data communications equipment
DTE	Data terminal equipment
EON	" End of number " control character
SEP	" Separation " control character
FES	Echo suppressor at the answering terminal (" Far end suppressor ")
NES	Echo suppressor at the calling terminal (" Near end suppressor ")

The following definitions apply to this Recommendation:

Calling tone — The tone transmitted from the calling end.
Answering tone — The tone transmitted from the called end.

3. *Interface procedures at call-originating station*

Event

1. DTE checks if CT 213 ON, and the following circuits OFF: CT 202, CT 210, CT 205, CT 204, CT 203.

2. DTE puts CT 202 ON.

3. DTE puts CT 108/2 ON (CT 108/2 can be placed in the ON condition at any time up to and including event 16).

4. For modem to Recommendation V.23 with backward channel, DTE puts CT 105 ON if the calling end wishes to transmit first, and DTE puts CT 120 ON if the calling end wishes to receive. CT 105 and CT 120 can be placed ON at any time up to and including event 20.

5. Line goes " off-hook ".

6. DCE puts CT 203 ON.

7. Telephone system puts dial tone on line.

8. DCE puts CT 210 ON.

9. DTE presents the first or appropriate digit on circuits 206, 207, 208 and 209.

10. DTE puts CT 211 ON.

11. DCE dials first digit; then takes CT 210 OFF.

12. DTE takes CT 211 OFF.

13. Steps 8 to 12 are repeated (but this process may be interrupted by SEP) until the last digit is presented and used. Step 8 is then repeated but step 14 follows.

14. DTE presents EON on circuits 206, 207, 208 and 209, it then puts CT 211 ON.

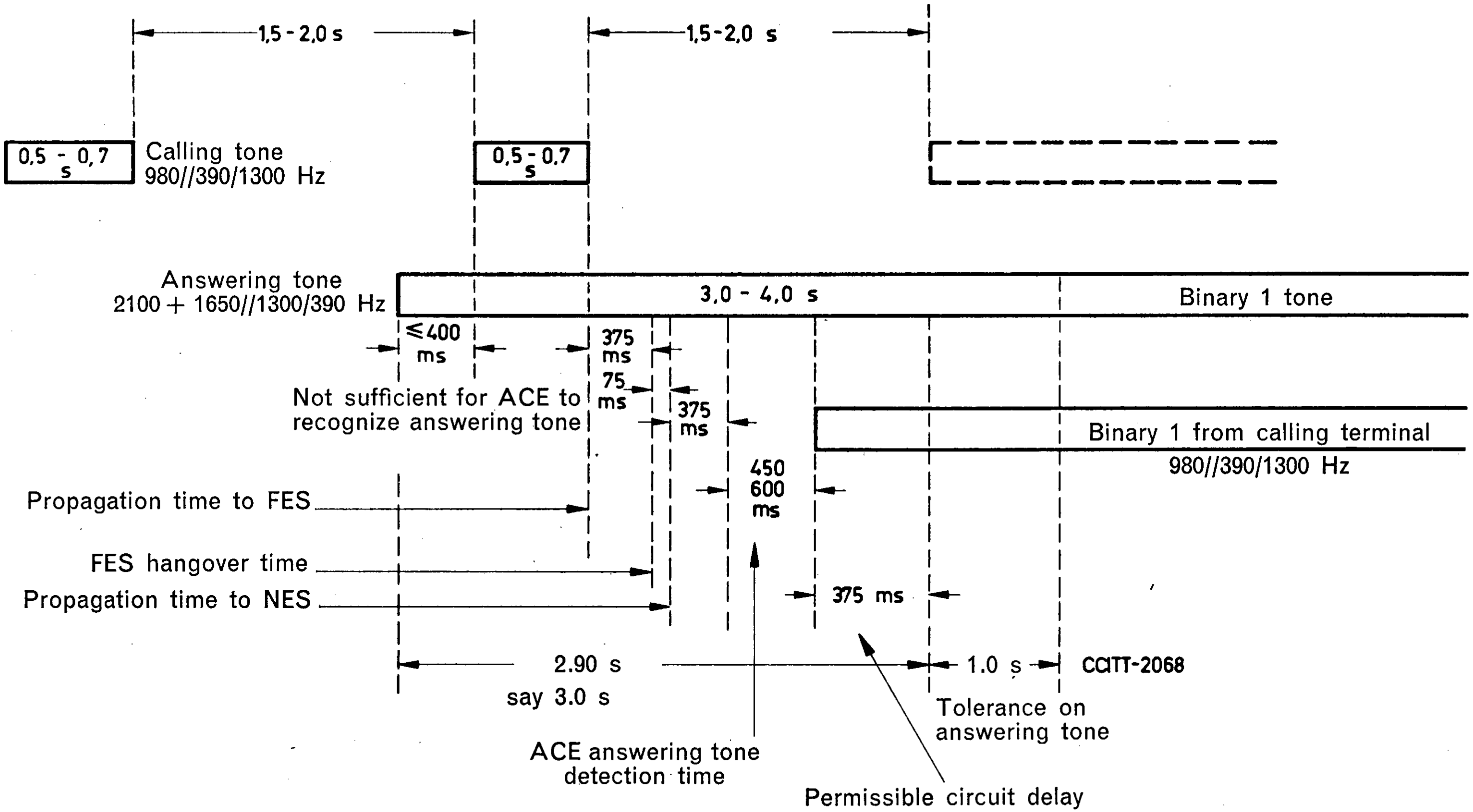
15. DCE takes CT 210 OFF.

16. DTE takes CT 211 OFF and puts CT 108/2 ON, if not previously ON.

17. Interrupted tone, as shown in Figure 2, is transmitted to line from the calling DCE terminal.

18. a) If the call is answered by data terminal, then 2100 Hz tone is received by calling DCE. Echo suppressors are disabled during coincidence of a silent period in the calling tone (17) and 2100 Hz.

b) If the call is not answered, or is answered by a non-data station then no 2100 Hz is received at the calling terminal. If no answering tone is received after an elapsed time



Abbreviations used

Hz = hertz
s = second
ms = millisecond

ACE = automatic calling equipment
FES = echo suppressor at the answering terminal
NES = echo suppressor at the calling terminal

FIGURE 2. — **Timing of line signals under "worst case" tolerance conditions, where all timing is related to the calling terminal in the duplex mode using either V.21 modems or V.23 modems which are equipped with both forward and backward channel**

CT 205 comes ON. This time is measured from event 15 and selectable in the range of 10–40 seconds.

19. When 2100 Hz has been recognized by the DCE for a period of 450 to 600 ms, the interrupted calling tone is discontinued by the DCE, as shown in Figure 2.

a) DCE transfers control of the telephone line from CT 202 to CT 108/2 and turns ON CT 204.

b) DTE may then turn OFF CT 202 without disconnecting the call.

Duplex case V.21 and V.23 modems	*Half duplex case V.23 modem only*
20-A. a) For V.21 modem, DCE puts 980 Hz signal on the line. b) For V.23 modem i) If CT 105 is ON, DCE puts 1300 Hz signal on the line. ii) If CT 120 is ON, DCE puts 390 Hz signal on the line. The 2100 Hz answering tone must not activate circuits 104, 109, 119 or 122.	20-B No action.
21-A Simultaneously with 20-A.a, b i) or b ii) above the DCE puts CT 107 ON.	21-B The DCE examines the line to determine the end of the 2100 Hz answering tone. i) If CT 105 is ON and the DCE detects an absence of 2100 Hz tone for 50 to 100 ms, CT 107 comes ON and at the same time 1300 Hz is put on the line. After its normal delay, CT 106 comes ON and the terminal can then transmit data. ii) If CT 105 is OFF and the DCE detects an absence of 2100 Hz tone for 50 to 100 ms, CT 107 comes ON. When the 1300 Hz tone is recognized and after its normal delay, the DCE puts CT 109 ON to allow the examination of CT 104 by the DTE.
22-A. a) For the V.21 modem case, the DCE waits to receive 1650 Hz. It then puts ON CT 109 and CT 106 after normal delays. The DTE can then transmit data.	22-B No action.

Duplex case V.21 and V.23 modems

b) For the V.23 modem case

i) if CT 105 is ON the DCE puts CT 106 ON after its normal delay. The DTE can then transmit data;

ii) if CT 120 is ON the DCE puts CT 121 ON after its normal delay and the DTE awaits for CT 109 to come ON in the expectation of receiving data.

Note. — Prior to event 19 a, CT 202 is turned OFF to disconnect the call regardless of the condition of CT 108/2. After event 19 a, both CT 202 and CT 108/2 must be turned OFF to disconnect. The ON condition of CT 205 is an indication to DTE to disconnect.

4. *Interface procedure at called station*

1. Ringing received on line. DCE puts CT 125 ON.

2. a) If CT 108/2 is ON, DCE goes " off-hook ".

b) If CT 108/1 or CT 108/2 is OFF, the DCE waits for CT 108/1 or CT 108/2 to come ON, and then goes " off-hook ". If CT 108/1 or CT 108/2 does not turn ON, then the call is not answered.

3. After DCE goes " off-hook ", it transmits 2100 Hz for a period as shown in Figure 2. In the case of the V.23 modem operated in the half-duplex mode the timings are as shown in Figure 3.

4. At the end of the 2100 Hz transmission, the DCE puts CT 107 ON. (See also section 6.)

In the duplex case

a) In the V.21 modem case, the DCE immediately transmits 1650 Hz (binary 1). When the DCE receives 980 Hz, it puts CT 109 ON and, after its normal delay, puts CT 106 ON.

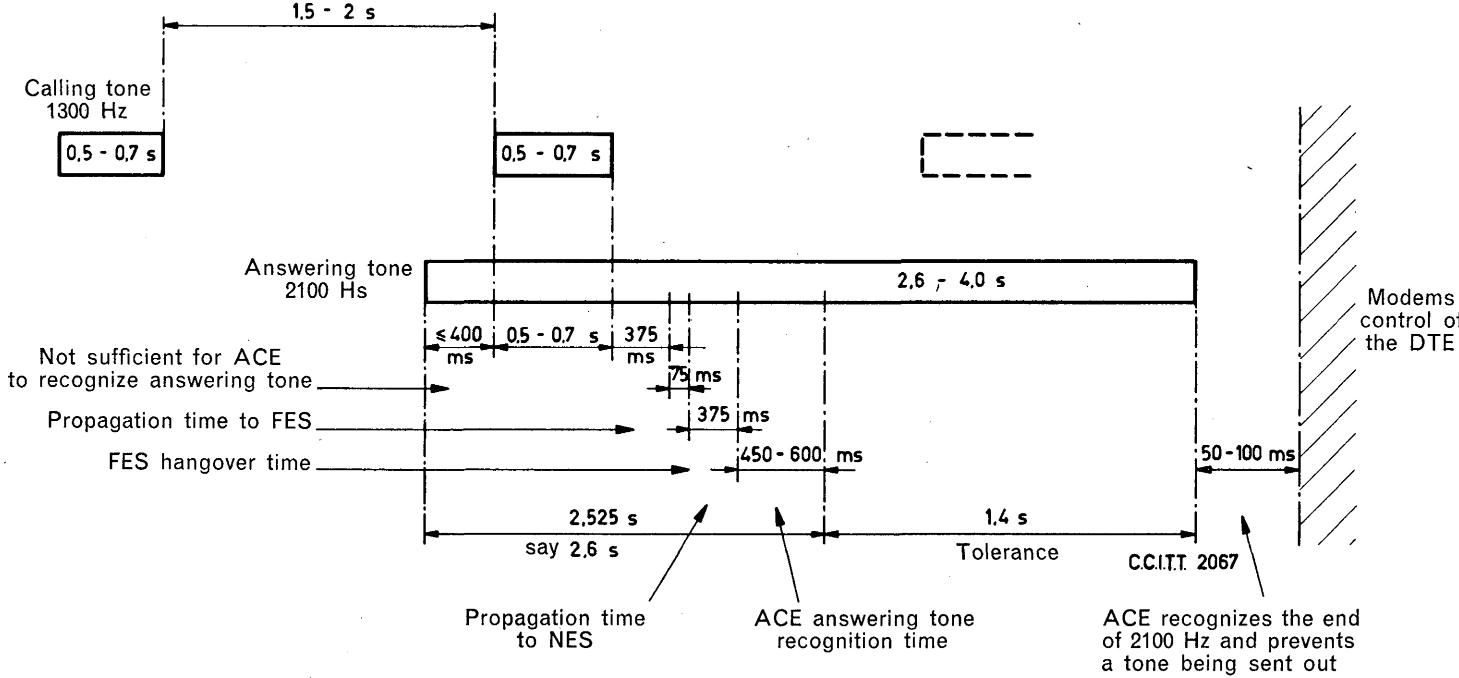
The DTE can then transmit data.

b) In the V.23 modem case:

i) If CT 105 is ON, the DCE transmits 1300 Hz (binary 1). After its normal delay, the DCE puts CT 106 ON. When the DCE receives the 390 Hz signal from the calling terminal, it switches CT 122 ON after its normal delay.

The DTE can then transmit data.

ii) If CT 120 is ON, the DCE transmits 390 Hz (binary 1) and after its normal delay puts CT 121 ON. When the DCE receives the 1300 Hz from the calling terminal, it switches CT 109 ON after its normal delay, in expectation of receiving data.



Abbreviations used

Hz = hertz
s = second
ms = millisecond
ACE = automatic calling equipment
FES = echo suppressor at the answering terminal
NES = echo suppressor at the calling terminal
DTE = data terminal equipment

FIGURE 3. — **Timing of line signals under "worst case" tolerance conditions where all the timing is related to the calling terminal in the half duplex (simplex) mode using V.23 modems without a backward channel**

In the half duplex case

Where V.23 modems are used without a backward channel, the DCE times out the 2100 Hz answering tone.

i) If CT 105 is ON, the DCE waits 50 to 100 ms before putting 1300 Hz (binary 1) on the line, and after its normal delay CT 106 is switched ON.

The DTE can then transmit data.

ii) If CT 105 is OFF, the DCE waits for 1300 Hz on the line. After receiving 1300 Hz and its normal delay, CT 109 is switched ON to allow the DTE to receive data.

5. If within an appropriate time after CT 107 has been turned ON, no recognizable signals are received the Data Terminal Equipment should turn OFF circuit 108/1 or 108/2.

5. *Proposed line procedure for duplex modems*

Figure 2 shows the timings of line signals when automatic calling and answering are employed in a system employing duplex modems. The sequence of operations is as follows:

After the ACE has pulsed out the digits for calling the answering station and control has been extended to the modem and calling signals are sent to line. These calling signals comprise interrupted binary 1 tones which are ON for not more than 0.7 second and not less than 0.5 second and OFF for not more than 2.0 seconds and not less than 1.5 second. These interrupted tones are repeated until an answering tone is received from the called station. The answering tone must be a 2100-Hz signal to enable echo suppressors to be disabled. Its duration should be not less than 3.0 seconds and not more than 4.0 seconds. This answering tone must be followed immediately by the binary 1 tone from the called station modem.

At the calling station, the DCE must examine the line, and if, during the silent periods in the calling tone, it receives the answering tone for not less than 0.45 second (with an upper tolerance on this timing circuit of 0.6 second) the calling terminal modem must send the binary 1 condition to the called terminal. On receipt of binary 1 conditions at both terminals, station identification and data transmission can commence.

To keep the echo suppressors disabled it is necessary to ensure that following the transmission of the 2100-Hz answering signal from the called station, which also serves to disable the echo suppressor during the silent period in the calling tone, energy is maintained in the frequency spectrum 800–3000 Hz with gaps no longer than 100 milliseconds. Thus, if the called station is required to initially receive data in the V.23 case, the binary 1 condition will be 390 Hz, which is inadequate for holding the echo suppressor disabled. However, if binary 1 from the calling terminal is applied immediately the recognition time of 0.45 to 0.6 second has expired, the circuit will be able to tolerate a permissible loop delay of 0.75 second under " worst case " conditions. Reference to Figure 2 shows that this figure is derived from summing all the signal durations prior to the receipt of binary 1 at the answering terminal. The initial period of $\leqslant$ 400 ms represents a condition where the answering tone duration is insufficient to disable the NES. The propagation delay following the calling tone burst is the time taken for the " trailing edge " of the calling tone to reach the FES. The FES will then take up to 75 ms to disable. The answering tone has a propagation delay to reach the NES. The DCE then requires 0.4 to 0.65 second to detect this answering tone. A further allowance must be made for the binary 1 condition to reach the answering terminal, if the FES is not to be allowed to drop out again. Thus a minimum answering tone duration of 2.9 seconds is determined. This is rounded to give safe figures of 3.0 to 4.0 seconds for the answer tone duration.

6. *Proposed line procedures for half duplex modems*

The line procedures outlined consider the case of the V.23 modem used in half duplex mode.

Systems which operate in the half duplex mode and which employ automatic calling equipment shall determine by pre-arrangement which of the two stations—calling or answering—shall first transmit to the other upon the establishment of the connection. As indicated in sections 3 and 4, the data terminal equipment at the terminal which is to transmit first must put circuit 105 ON, at the appropriate point in the call-establishment sequence.

Figure 3 shows the timings of line signals when automatic calling and automatic answering are employed in a system employing half duplex modems. The sequence of operation is as follows:

After the DCE has dialled the digits of the directory number for the automatic answering station followed by the EON character, the DCE sends calling tone to the answering terminal. The calling tone consists of a series of interrupted bursts of binary 1 signal (1300 Hz) ON for a duration of not less than 0.5 second and not more than 0.7 second and OFF for a duration of not less than 1.5 second and not more than 2.0 seconds.

As soon as the called station is connected to the line (i.e. circuits 125 and 108 are ON and the station is " off-hook "), it sends a continuous 2100 Hz answering tone for a duration of not less than 2.6 seconds and not more than 4.0 seconds.

This answering tone propagates towards the calling station and, during the course of one or two interruptions between bursts of calling tone, causes any echo suppressors in the circuit to disable. The answer tone is recognized by the calling station for a period of between 0.45 and 0.60 second after its arrival. The calling station terminates the calling tone burst sequence and recognizes the end of the answer tone for a period of between 50 and 100 milliseconds after its arrival at the calling terminal.

At the end of this delay, the DCE puts CT 107 ON.

Similarly, the answering station DCE delays for a period of between 50 and 100 milliseconds after terminating the answer tone before putting CT 107 ON.

The data communication equipment at the station at which circuit 105 had been turned ON (by pre-arrangement) commences to send binary 1 (1300 Hz). Data communication can commence after circuit 106 is put ON at that station.

During the automatic calling and answering procedures the echo suppressors will be disabled. They may become re-enabled during any modem turn-around.

7. *Manual data terminal calling automatic answering station*

The procedure for establishment of a call from a manual data terminal to an automatic answering data station is similar to that from an automatic calling data terminal, except that no tone is transmitted from the calling terminal until the called station has answered. The manual operator dials the required number, hears 2100 Hz returned from the automatic answering terminal and then presses his data button to connect the data communication equipment to the line during the period that 2100 Hz is being received. Circuit 107 (data set ready) comes ON at the time as specified in event 21.

8. *Automatic calling terminal calling manual data station*

An operator answering a call from an automatic calling equipment hears an interrupted tone of 0.5 to 0.7 second ON and 1.5 to 2.0 seconds OFF. The data button must be depressed to connect the modem to line. A period of about 3 to 4 seconds of 2100 Hz tone is transmitted to the calling terminal to disable echo suppressors and notify the calling station that the connection is being established. This sequence is followed by data transmission as required.

9. *Protection of ordinary telephone users*

As both automatic calling and automatic answering data terminals transmit tones to line during call establishment, a normal telephone user who becomes inadvertently connected to one will receive tone signals for a period of sufficient duration to indicate clearly to him that he is incorrectly connected.

2400 BITS PER SECOND MODEM FOR USE ON FOUR-WIRE LEASED POINT-TO-POINT CIRCUITS

(Mar del Plata, 1968)

On leased circuits, considering that there exist and will come into being many modems with features designed to meet the requirements of the administrations and users, this Recommendation in no way restricts the use of any other modems.

1. The principal characteristics for this recommended modem for transmitting data at 2400 bits per second on four-wire leased point-to-point circuits conforming to Recommendation M.102 are as follows:

a) it is capable of operating in a full duplex mode;

b) four-phase modulation with synchronous mode of transmission;

c) inclusion of a backward (supervisory) channel at modulation rates up to 75 bauds in each direction of transmission, the use of these channels being optional.

2. *Line signals*

2.1 The carrier frequency is to be 1800 ± 1 Hz. No separate pilot frequencies are provided. The power levels used will conform to Recommendation V.2.

2.2 *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	—∞
—1	—7
—2	—4
—3	—3

provisionally the backward channel shall be 6 dB lower in power level than the data channel.

2.3 The data stream to be transmitted is divided into pairs of consecutive bits (dibits). Each dibit is encoded as a phase change relative to the phase of the immediately preceding signal element. At the receiver the dibits are decoded and the bits are reassembled in correct order. Two alternative arrangements of coding are listed in the table below. The left-hand digit of the dibit is the one occurring first in the data stream.

TABLE 1

Dibit	Phase change *	
	Alternative A	Alternative B
00	0°	+ 45°
01	+ 90°	+135°
11	+180°	+225°
10	+270°	+315°

* The phase change is the actual on-line phase shift in the transition region from the end of one signalling element to the beginning of the following signalling element.

2.4 The main differences between the two alternatives given in paragraph 2.3 are as follows:

Consideration	Alternative A	Alternative B
Code sensitivity	Loses synchronism if a long string of binary 0's is transmitted. This can be overcome by several methods	Non-sensitive except for a slight loss of margin on certain repetitive dibits
Backward channel allocation	No loss of margin	A backward channel has not yet been proved to be a practical possibility

3. *Data signalling and modulation rates*

The data signalling rate shall be 2400 bits per second $\pm$ 0.01 %, i.e. the modulation rate is 1200 bauds $\pm$ 0.01 %.

4. *Received signal frequency tolerance*

Noting that the carrier frequency tolerance allowance at the transmitter is $\pm$ 1 Hz and assuming a maximum frequency drift of $\pm$ 6 Hz in the connection between the modems, then the receiver must be able to accept errors of at least $\pm$ 7 Hz in the received frequencies.

5. *Backward channel*

The modulation rate, characteristic frequencies, tolerances, etc., to be as recommended for backward channel in Recommendation V.23.

6. *Interchange circuits*

Interchange circuit		Forward (data) channel half duplex or full duplex	
No.	Designation	Without backward channel	With backward channel
101*	Protective ground or earth	x	x
102	Signal ground or common return	x	x
103	Transmitted data	x	x
104	Received data	x	x
105	Request to send	x	x
106	Ready for sending	x	x
107	Data set ready	x	x
108/1	Connect data set to line	x	x
109	Data channel received line signal detector	x	x
113	Transmitter signal element timing (DTE source)	x	x
114	Transmitter signal element timing (DCE source)	x	x
115	Receiver signal element timing (DCE source)	x	x
118	Transmitted backward channel data	–	x
119	Received backward channel data	–	x
120	Transmit backward channel line signal	–	x
121	Backward channel ready	–	x
122	Backward channel received line signal detector	–	x

* May be excluded if so required by local safety regulations.

7. *Threshold and response times of circuit 109*

A fall in level of the incoming line signal to -30 dBm or lower for more than 10 ± 5 ms will cause circuit 109 to be turned OFF. An increase in level to -27 ± 1 dBm or higher will within 10 ± 5 ms turn this circuit ON.

8. *Timing arrangements*

Clocks should be included in the modem to provide the data terminal equipment with transmitter signal element timing (Recommendation V.24, circuit 114) and receiver signal element timing (Recommendation V.24, circuit 115). Optionally, the transmitter signal element timing may be originated in the data terminal equipment and be transferred to the modem via the appropriate interchange circuit (Recommendation V.24, circuit 113).

9. The following information is provided to assist equipment manufacturers:

The data modem should have no adjustment for send level or receive sensitivity under the control of the operator.

RECOMMENDATION V.30

PARALLEL DATA TRANSMISSION MODEMS STANDARDIZED FOR UNIVERSAL USE IN THE GENERAL SWITCHED TELEPHONE NETWORK

(Mar del Plata, 1968)

There is a need for one-way data transmission systems where a large number of low-cost sending stations (outstations) transmit to a central receiving station (instation) over the switched telephone network.

The following systems are desired:

a) transmitting 16-character combinations;

b) transmitting 64-character combinations;

c) transmitting 256-character combinations.

In most cases a character signalling rate of 20 characters per second will be sufficient; 40 characters per second may be required for some applications of the 16-character combination system.

The transmission from the instation to the outstations is limited either to simple acknowledgement signals (data collection systems) or to analogue signals (voice-answering systems).

The use of normal push-button telephone sets in the outstation for some of these applications may be of advantage for the user. However, it is recognized that for the time being on some telephone systems there exist certain limitations in the frequency band 600 to 900 Hz. This is due to the characteristics of the telecommunication path, such as signalling frequencies and metering pulses. Therefore, for a universal system the frequency band of the data channel is 900 to 2000 Hz, which excludes the use of the normal push-button telephone set.

A so-called parallel data-transmission system using two or three times one out of four frequencies can fulfil the above requirements.

For these reasons, the C.C.I.T.T. *unanimously declares the following view*:

1. Parallel data-transmission systems can be used economically when a large number of low-cost sending stations (outstations) wish to transmit to a central receiving station (instation) over the switched telephone network (or on leased telephone circuits).

Apart from the possibility of the use, on a restricted scale, of a system that is compatible with multifrequency push-button telephone signalling devices, the following system is recommended as a universally applicable system for the switched telephone circuits.

2. *Facilities*

2.1 *Data channel*

The basic system has a maximum of 16-character combinations and a modulation rate of up to 40 bauds. This permits a character signalling rate of up to 20 characters per second when an inter-character rest condition is used, or up to 40 characters per second with the use of a binary timing channel. This basic system consists of two groups of four frequencies, one frequency from each group being transmitted simultaneously (two times one out of four).

The basic system includes provision for expansion up to 64-character combinations by the addition of a third four-frequency group (three times one out of four). No use is foreseen for the system with 64-character combinations at character signalling rates above 20 characters per second, within this class of inexpensive parallel transmission equipment.

An expansion of the basic system to cater for 256 characters (up to 20 characters per second) is achieved by using only two groups for the conveyance of data, each character being transmitted in two sequential parts. The two half characters are positively identified by the two different conditions of a binary channel. The timing channel mentioned above is recommended to be used for this purpose.

Where an inter-character rest condition is required the full number of frequency combinations in the modem will not be available to the user as character combinations:

a) with the 16-frequency combination system, only 15 characters will be available unless a timing channel is used from frequency group B;

b) with the 64-frequency combination system only 63 characters are available.

These recommended systems have an inherent transmission error-detecting capability.

2.2 *Backward channel*

Provision is made for the following facilities:

a) a speech channel non-simultaneous with forward data;

b) a backward channel for audible signalling;

c) a backward channel for electrical signalling purposes.

Facilities b and c are provided, either non-simultaneous or optionally simultaneous with the forward data channels.

A loudspeaker will be provided in the outstation modem. On an optional basis a d.c. signalling output will be provided. If national regulations permit, a voice-answering output will also be provided on an optional basis.

3. *Frequency allocations*

3.1 *Data channels*

The following frequency allocations and designations are recommended:

TABLE 1

Group \ Channel No.	1	2	3	4
A	920 Hz	1000 Hz	1080 Hz	1160 Hz
B	1320 Hz	1400 Hz	1480 Hz	1560 Hz
C	1720 Hz	1800 Hz	1880 Hz	1960 Hz

For the basic 16-character system only groups A and C are used.

If an inter-character rest condition is used, during the time no input data circuits are operated, rest frequencies are sent to line. The highest frequency in each group is recommended to be the rest frequency.

3.2 *Timing channel*

If a timing channel is provided in the 16-character system this should consist of a selected pair of group B frequencies. The recommended frequencies are $F_{B2} = 1400$ Hz and $F_{B3} = 1480$ Hz.

In the case where this timing channel will be used to identify the two halves of the character in the 256-character system, the higher frequency is transmitted simultaneously with the first half of the character.

No timing channel is provided in the 64-character combination system.

3.3 *Backward channel*

The frequency of the backward channel for audible and electrical signalling shall be 420 Hz. This tone may be amplitude modulated at rates up to say, 5 bauds.

The question of whether there is a requirement for higher modulation rates or for a frequency-modulated backward channel and whether it can be used simultaneously with the forward data frequencies is a matter for further study.

3.4 *Tolerances*

The tolerances on both data and backward frequencies should be $\pm$ 4 Hz.

The receiver should cater for $\pm$ 6 Hz difference due to carrier systems in addition to the transmitter tolerance of $\pm$ 4 Hz.

4. *Power levels*

Based on Recommendation V.2 the following maximum power levels measured at the zero relative level point are recommended for each transmitted frequency:

4.1 *Data and timing channels*

4.1.1 16-character system without timing channel and with a non-simultaneous backward channel: -13 dBm0.

4.1.2 All other cases: -16 dBm0.

4.2 *Backward channel*

4.2.1 Non-simultaneous: −10 dBm0.

4.2.2 Simultaneous: −16 dBm0.

In systems where either the simultaneous or the non-simultaneous backward channel is used, all power levels should be −16 dBm0.

The maximum difference between any data tone at the transmitter terminal should be 1 dB.

5. *Threshold levels of the data channel received signal detector*

When the level of the received signal in group C exceeds −49 dBm, circuit 109 shall be ON. When the level of this received signal is less than −54 dBm, circuit 109 shall be OFF. The detector circuit which causes circuit 109 to turn ON or OFF shall exhibit a hysteresis action such that the level at which the OFF to ON transition occurs shall be at least 2 dB greater than that for the ON to OFF transition.

Group C was chosen for this purpose because it is the most critical from a received level point of view.

6. *Minimum level of received signal on the backward channel*

The expected minimum level is −45 dBm for the 420-Hz tone. This information is provided to assist equipment manufacturers.

7. *Instation modem interface*

7.1 List of essential interchange circuits:

101	Protective ground (Note 1)
102	Signal ground or common return
104	Received data (12 or 8 circuits depending on whether group B is provided or not. These received data circuits are designated A1, A2 . . . C4, each corresponding to its relevant frequency (see Table 1)
107	Data set ready
108/1	Connect data set to line (Note 2)
108/2	Data terminal ready (Note 2)
109	Data channel received line signal detector
125	Calling indicator
130	Transmit backward tone
— a)	Voice-answering input (Note 3)
— b)	Voice-answering input (Note 3)

The following optional interchange circuits may be provided:

110	Data signal quality detector
131	Received character timing

Note 1. — May be excluded if so required by local safety regulations.

Note 2. — This circuit shall be capable of use as circuit 108/1 "connect data set to line" or circuit 108/2 "data terminal ready", depending upon its use. For automatic calling it shall be used as 108/2 only.

Note 3. — New interface circuit, this circuit needs to be defined.

7.2 The electrical characteristics of the interchange circuits comply with Recommendation V.24 (with the exception of the voice-answering input circuit).

Data circuits: when the frequency corresponding to the circuit is ON, the appropriate interchange circuit will be negative. When the frequency in this channel is OFF, the interchange circuit will be positive.

For timing purposes in the 256-character system, a single interchange circuit is selected from group B such that positive polarity indicates the first half of the character period and a negative polarity indicates the second half of the character.

8. *Outstation modem interface* (Note 1)

8.1 *List of essential interchange circuits*

301	Protective ground (Note 2)
302	Data common return
303	Transmitted data (nine or six circuits depending on whether group B is provided or not. These transmitted data circuits are designated A1, A2 ... C3 each corresponding to its relevant frequency (see Table 1)
304 a) b)	Activate transmitter
305 a) b)	Activate receiver

8.2 *The following optional interchange circuits may be provided*

306 a) b)	Received backward signal
307 a) b)	Voice-answering output (if national regulations permit)
308 a) b)	Data set connected

When the optional timing channel is used then the appropriate data leads are operated.

Note 1. — All the numbers in 300-series are provisional.

Note 2. — May be excluded if so required by local safety regulations.

8.3 *Electrical characteristics*

The data and control interchange circuits at the outstation will be operated by the opening or closing of contacts carrying only direct current.

9. *Table of correspondence for each group*

TABLE 2

At outstation closing of circuit	Channel No. on line	At instation negative polarity on circuit
1	1	1
2	2	2
3	3	3
None	4	4

Not more than one circuit per group may be closed at a time.

10. *Character set*

This Recommendation includes the allocation of transmission frequencies to the interchange circuits.

The allocation of interchange circuits to the code combinations to be transmitted, i.e. definition of a character set, must conform to the conditions defined in this Recommendation and must take into account the application requirements and the type of input media (paper tape, punched cards, keyboards, etc.).

For this reason the recommendation for a character set is primarily for I.S.O. in collaboration with C.C.I.T.T.

Note. — Examples of Alphabets and coding methods are given in Supplements Nos. 20 and 21 to the *White Book*, Volume VIII, and in Supplements Nos. 56 and 57 to the *Blue Book*, Volume VIII.

RECOMMENDATION V.35

TRANSMISSION OF 48-KILOBITS-PER-SECOND DATA USING 60- TO 108-KHz GROUP BAND CIRCUITS

(Mar del Plata, 1968)

On leased circuits, considering that there exist and will come into being other modems with features designed to meet the requirements of the administrations and users, this Recommendation in no way restricts the use of any other modems.

This is a particular system using a group reference pilot at 104.080 kHz.

Principal recommended characteristics to be used for simultaneous both-way operation are the following:

1. *Input/output*

Rectangular polar serial binary data.

2. *Transmission rates*

Preferred mode is synchronous at 48 000 ± 1 bits/second, with the following exceptions permissible:

a) Synchronous at 40 800 ± 1 bits/second when it is an operational necessity, or

b) Non-synchronous transmission of essentially random binary facsimile with element durations in the range 21 microseconds to 200 milliseconds.

Note. — Operation at half data signalling rate shall be possible when the line characteristics do not permit the above data signalling rates.

3. *Encoding/decoding*

Synchronous data should be encoded to avoid restrictions on the data input format. Such restrictions would be imposed by the need to have sufficient transitions for receiver clock stability, without short repetitive sequences of data signals which would result in high level discrete frequency components in the line signal. Synchronous data should be encoded and decoded by means of logical arrangements similar to those shown in Appendix 1.

4. *Modulation technique*

The baseband signal (see section 5) should be translated to the 60- to 104-kHz band as an asymmetric sideband suppressed carrier A.M. signal with a carrier frequency of 100 kHz. A pilot carrier will be necessary to permit homochronous demodulation. To simplify the problem of recovery of the pilot carrier for demodulation the serial binary data signal should be modified as stated in section 5. The transmitted signal should correspond with the following:

a) The data carrier frequency should be 100 000 ± 2 Hz.

b) The nominal level of a frequency translated suppressed carrier 48 kilobits/second encoded data baseband signal in the 60- to 104-kHz band should be equivalent to −5 dBm0.

c) A pilot carrier at −9 ± 0.5 dB relative to the nominal level of the signal in 4 b above should be added such that the pilot carrier would be in phase, to within ± 0.04 radian, with a frequency translated continuous binary 1 input to the modulator.

d) The modulator should be linear, and the characteristics of the transmit bandpass filter should be such that the relative attenuation distortion and the relative envelope delay distortion in the range 64 to 101.5 kHz are less than 0.2 dB and 4 microseconds respectively.

5. *Baseband signal*

a) The encoded synchronous or random non-synchronous serial binary data signal should be modified by the following transform:

$$\frac{pT_1}{1+pT_1}, \text{ to remove the low frequency components,}$$

where p is the complex frequency operator, and

T_1 is $\frac{25}{2\pi}$ times the minimum binary element duration, i.e. 83 microseconds.

The value of T_1 shall have an accuracy of $\pm 2\%$.

In this form the signal is referred to as the baseband signal.

b) The baseband signal resulting from the transformation should not suffer impairment greater than that resulting from relative attenuation distortion or relative envelope delay distortion of 1.5 dB or 4 microseconds respectively, *and*

i) distortion due to modification of the baseband signal by the transform

$$\frac{pT_2}{1+pT_2}$$

where T_2 is 3.18 milliseconds; *or*

ii) distortion due to modification of the baseband signal by the transform

$$\left[\frac{pT_3}{1+pT_3}\right]^2$$

where T_3 is 6.36 milliseconds.

c) Paragraphs 5 a and 5 b should apply in the range 0 to 36 kHz.

6. *Voice channel*

A 4-kHz speech channel provided as an integral part of this system should correspond to channel 1 of a 12-channel system, i.e. as a lower sideband SSB signal in the 104- to 108-kHz band.

a) The characteristics of this channel should correspond with and conform to Recommendation G.232 (C.C.I.T.T. *White Book*, Volume III).

b) This voice channel is optional.

7. *Group reference pilots*

a) Means should be provided to facilitate the injection of a group reference pilot at 104.080 kHz.

b) The sections of Recommendation G.232 which refer to interference with and from supergroup reference pilots apply to the group reference pilots used with this equipment.

8. *Adjacent channel interference*

a) When transmitting encoded synchronous serial binary data at 48 kilobits/second on the data channel, the out-of-band energy in a 3-kHz band centred at any frequency in the range 1.5 to 58.5 kHz or 105.5 to 178.5 kHz should not exceed −60 dBm0. Additionally, the crosstalk produced in the range 103.680 to 104.480 kHz should not exceed the levels stated in Appendix 2.

b) When a signal at 0 dBm0 at any frequency in the range 0 to 60 or 104 to 180 kHz is applied to the carrier input terminals, the resulting crosstalk measured in the demodulated data baseband should not exceed a level equivalent to −40 dBm0.

9. *Line characteristics*

The characteristics of a channel over which this equipment can be expected to operate satisfactorily are given in Appendix 3.

10. *Interface*

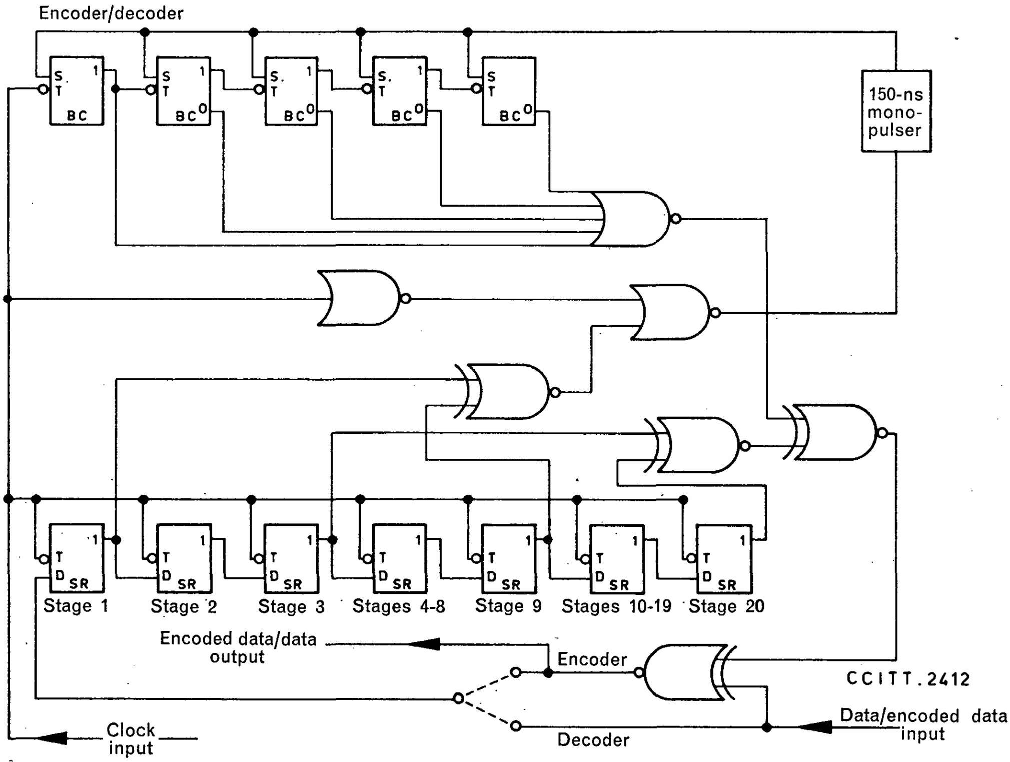
a) The interchange circuits should be as follows:

No.	Function
101 (Note 1)	Protective ground or earth
102	Signal ground or common return
103 ∅	Transmitted data
104 ∅	Received data
105	Request to send
106	Ready for sending
107	Data set ready
109	Data channel receive line signal detector
114 ∅	Transmitter signal element timing
115 ∅	Receiver signal element timing

Note 1. — May be excluded if so required by local safety regulations.

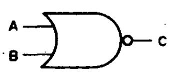
b) The electrical characteristics of the interchange circuits marked ∅ should be as described in Appendix 4; the circuits not marked should conform to Recommendation V.24.

APPENDIX 1 (Illustrative only)



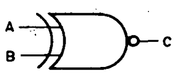
Symbol truth tables

Or / Or - Not



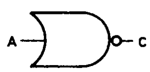
A	B	C
0	0	1
0	1	0
1	1	0
1	0	0

Exclusive or - not



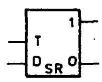
A	B	C
0	0	1
0	1	0
1	1	1
1	0	0

Inverter



A	C
1	0
0	1

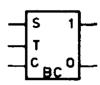
Shift register



T	D	1 output	0 output
1	Q	—	—
0	—	Q	$\overline{Q}$

Q is either 0 or 1

Binary counter



T	1 output	0 output
1	Q	$\overline{Q}$
0	$\overline{Q}$	Q

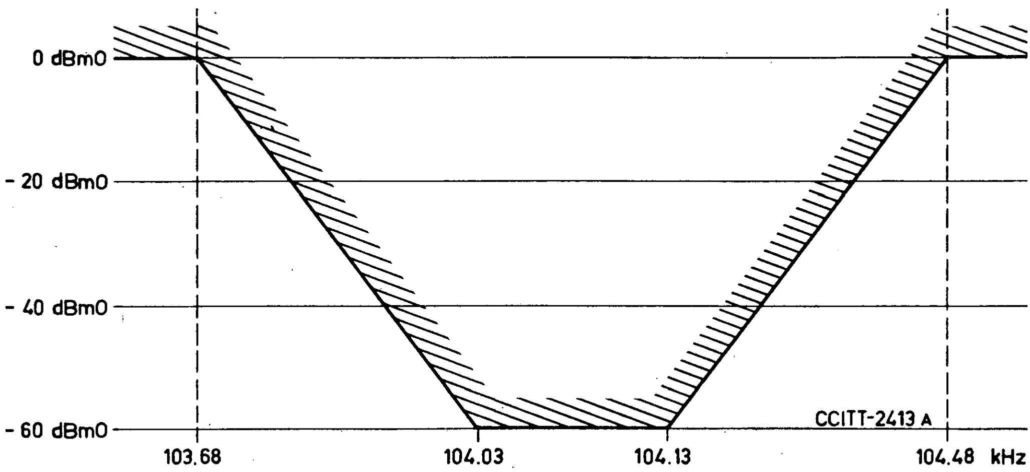
S	1 output	0 output
0	Q	$\overline{Q}$
1	1	0

(Reset)

Note. — Negative-going transitions of clocks (i.e. 1 to 0 transitions) coincide with data transitions. This is self-synchronizing.

Appendix 2

Maximum permissible level of interference in the region of the group reference pilot(s)



Appendix 3

Provisional characteristics of corrected wideband group circuits

(Extract from Recommendation H.14—Characteristics of wideband group circuits for the transmission of wide-spectrum signal (data, facsimile, etc.))

It is assumed that there are not more than two sections in tandem and not more than two through-group filters corrected with regard to group delay, the group delay distortion of each through-filter being less than 10 microseconds in the 64.2-103.8 kHz band. It is also assumed that the group and supergroup pilots at 104.08 kHz and 547.92 kHz are used and that they are blocked at the end of the link by filters with reduced group delay distortion in the 64.2-103.8 kHz band.

a) *Group delay distortion*

The group delay distortion of such a circuit, between modems, is shown in Figure 2. To respect these limits, it may be necessary to avoid using groups 1 and 5.

b) *Attenuation distortion*

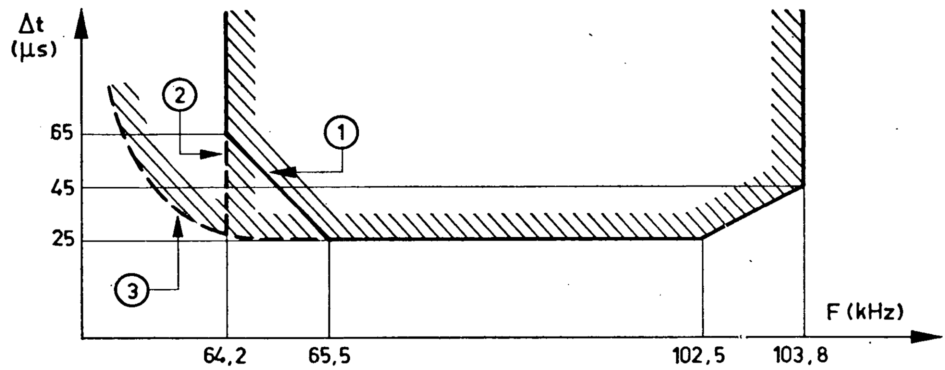
The attenuation distortion of such a circuit, between modems, is shown in Figure 3.

c) *Carrier leaks*

The leak from any carrier in the wanted band should not exceed —4.6 Npm0 (—40 dBm0). This value is provisional and may be incompatible with the pertinent value specified in Recommendation G.223, j, for a single modulation stage.

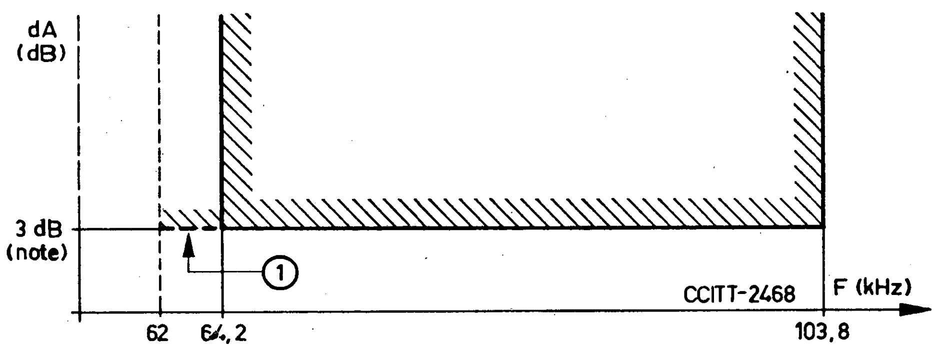
This point is the subject of a question under study.

Note. — The question of characteristics of non-corrected wideband group circuits of complex constitution is under study by Study Group XV.



① = Circuit including one group 5 section.
② = Circuit without any group 5 section.
③ = Between 62 and 64.2 kHz, the characteristics may be fixed by agreement between administrations (circuit without any group 5 section).

FIGURE 2. — **Maximum group delay distortion in a corrected wideband group circuit**



① = Between 62 and 64.2 kHz, the characteristics may be fixed by agreement between administrations (circuit without any group 5 section).

Note. — A lower value (for example, 1.5 dB) may be obtained in simple routings.

FIGURE 3

APPENDIX 4

The electrical characteristics of interchange circuits for data transmission at rates above 20 kilobits/second

1. *Cable*

The interface cable should be a balanced twisted multipair type with a pair characteristic impedance within 10% of the associated terminator input impedance of the timing waveform frequency.

2. *Generator*

This circuit should comply with the following requirements:

a) source impedance in the range $100 \pm 50\%$ ohms;

b) resistance to earth * of either short-circuit terminal $150 \pm 10\%$ ohms;

c) when terminated by 100-ohm resistive load the terminal to terminal voltage should be $0.55 \pm 20\%$ volts so that the A terminal is positive to the B terminal when binary 0 is transmitted, and the conditions reversed to transmit binary 1;

d) the rise time between the 10% to 90% points of any change of state when terminated as in 2. c should be less than 1% of the nominal duration of a signal element or 40 nanoseconds, whichever is the greater;

e) the arithmetic mean of the voltage of the A terminal with respect to the common return and the B terminal with respect to the common return (d.c. line offset) should not exceed 0.6 volt, again when terminated as in 2. c above.

3. *Load*

The load should comply with the following:

a) input impedance in the range $100 \pm 10\%$ ohms;

b) resistance to common return of $150 \pm 10\%$ ohms, measured from either short-circuit terminal.

4. *Electrical safety*

A generator or load should not be damaged by connection to earth potential, short-circuiting, or cross-connection to other interchange circuits.

5. *Performance in the presence of noise*

A generator as in section 2 above connected to a load as in section 3 above should operate without error in the presence of longitudinal noise or d.c. common return potential differences (common return offset) as follows:

a) with $\pm$ 2 volts (peak) noise present longitudinally, i.e. algebraically added to both load input terminals simultaneously with respect to the common return, or

b) with $\pm$ 4 volts common return offset;

c) if common return offset and longitudinal noise are present simultaneously, satisfactory operation should be achieved when:

$$\frac{\text{common return offset}}{2} + \text{longitudinal noise (peak)} = 2 \text{ volts or less.}$$

* i.e. interchange circuit No. 102 " Signal ground or common return ".

RECOMMENDATION V.40

ERROR INDICATION WITH ELECTROMECHANICAL EQUIPMENT

(Mar del Plata, 1968)

If use is made of a code providing for the introduction into each character signal of an extra unit for the parity check, it is possible with electromechanical equipment to detect errors not only in the transmission channel but also in part of the mechanical translation or transmission equipment.

It might be possible therefore, when an error is detected in a character signal, to arrange for an error indication to be given on the position where the error is found.

This indication could take the form of an extra perforation in the tapes of the perforated tape equipment or a special printout with direct printing equipment.

Such devices would however be either very costly or only partially effective (for example, many character signals of the No. 5 Alphabet do not correspond to any printout so that for these characters the normal sign cannot be replaced by an " error " sign).

For these reasons, the C.C.I.T.T. *unanimously recommends*
that use of an alarm or error-counting device is the best method if a local indication is required for an error detected in a character signal.

RECOMMENDATION V.41

CODE-INDEPENDENT ERROR CONTROL SYSTEM

(Mar del Plata, 1968)

1. *General*

This Recommendation is primarily intended for error control when implemented as an intermediate equipment which may be provided either with the data terminal equipment or with the data communication equipment. The appropriate interfaces are shown in Figures 1 and 2. The system is not primarily intended for use with multi-access computing systems. The Recommendation does not exclude the use of any other error-control system that may be better adapted to special needs.

The modems used must provide simultaneous forward and backward channels. The system uses synchronous transmission on the forward channel and asynchronous transmission on the backward channel. When modems to Recommendation V.23 are used with data signalling rates of 1200 or 600 bits/s in the general switched telephone network, Recommendation V.22 applies, the error-control equipment being classed as communication equipment. The margin of the synchronous receiver should be at least ±45%.

The system employs block transmission of information in fixed units of 240, 480 or 960 bits and is therefore most suited to the transmission of medium or long data messages, but a fast starting procedure is incorporated to improve the transmission efficiency for shorter messages.

Error control is achieved by means of automatic repetition of a block upon request (ARQ) from the data receiver. If storage is provided at the receiver, detected errors can be removed before the system output (clean copy). Storage for at least two data blocks must be provided at the transmitter.

The forward bit stream is divided into blocks each consisting of four service bits, the information bits, and 16 error-detection (or check) bits in that order, the check bits being generated in a cyclic encoder. Thus each block transmitted to line contains 260, 500 or 980 bits.

The system will detect:

a) all odd numbers of errors within a block;

b) any error burst not exceeding 16 bits in length and a large percentage of other error patterns.

Assuming a distribution of errors as in *Blue Book*, Volume VIII, Supplement No. 22, the error-rate improvement factor has been indicated by a computer simulation to be of the order of 50 000 for a block size of 260 bits.

The fixed block system employed limits the use of the system to those lines having a loop propagation time not greater than the figures given in Table 1. Allowances of 40 ms for total modem delay and 50 ms for the detection of the RQ signal have been made.

TABLE 1

Maximum permissible line loop propagation times (ms)

Block size (bits) \ Data signalling rate (bits/s)	200	600	1200	2400	3600	4800
260	1210	343	127	18	–	–
500	2410	743	327	118	49	14
980	4810	1543	727	318	182	114

2. *Encoding and checking process*

The service bits and information bits, taken in conjunction, correspond to the coefficients of a message polynomial having terms from x^{n-1} (n = total number of bits in a block or sequence) down to x^{16}. This polynomial is divided, modulo 2, by the generating polynomial $x^{16} + x^{12} + x^5 + 1$. The check bits correspond to the coefficients of the terms from x^{15} to x^0 in the remainder polynomial found at the completion of this division. The complete block, consisting of the service and information bits followed by the check bits, corresponds to the coefficients of a polynomial which is integrally divisible in modulo 2 fashion by the generating polynomial.

At the transmitter the service bits and information bits are subjected to an encoding process equivalent to a division by the generator polynomial. The resulting remainder is transmitted to line immediately after the information bits, commencing with the highest order bits.

At the receiver, the incoming block is subjected to a decoding process equivalent to a division by the generator polynomial which in the absence of errors will result in a zero remainder. If the division results in other than a zero remainder, errors are indicated.

The above processes may conveniently be carried out by a 16-stage cyclic shift register with appropriate feedback gates (see Figures 3 and 4) which is set to the all 0 position before starting to process each block; at the receiver the all 0 condition after processing a block indicates error-free reception.

3. *The service bits*

3.1 *Block sequence indication*

The four service bits at the beginning of each block transmitted to line indicate the block sequence and convey control information external to the message information. One of these control functions is to ensure that the information block order can be checked during repetitions, thus ensuring that information is not lost, gained or transposed. Three block sequence indicators A, B and C are used cyclically in that order.

Once a sequence indicator has been attached to an information block it remains with that block until the block is received correctly. Examination of the sequence indication is an additional part of the checking process.

3.2 *Allocation of service bits*

The allocation of the 16 possible combinations of the four service bits is given in Tables 2 and 3. Table 2 lists essential and therefore mandatory combinations and Table 3 optional combinations.

TABLE 2

Essential combinations

Note. — The digit on the left occurs first.

Group	Combination	Function
a	0011	Block A sequence indicator
b	1001	Block B sequence indicator
c	1100	Block C sequence indicator
d	0101	Synchronizing sequence prefix

TABLE 3

Optional combinations

Group	Combination	Function
e	0110	Hold block
f	1000	End of transmission (this block contains no data)
g	0001	Start of message 1 (five-unit codes)
h	1010	Start of message 2 (six-unit codes)
j	1011	Start of message 3 (seven-unit codes)
k	0010	Start of message 4 (eight-unit codes)
l	0100	End of message (this block contains no data)
m	0111	Data link escape (general control block)
n	1101	To be allocated by bilateral agreement
p	1110	
q	1111	
r	0000	

3.3 *Control functions*

Synchronization is the only essential control function catered for in the service bits.

The optional data link escape (general control) block contains data which are special in some way agreed by the users.

Additional optional functions are start of message 1 (or for five-unit codes), start of message 2 (or for six-unit codes), start of message 3 (or for seven-unit codes), start of message 4 (or for eight-unit codes), end of message, and end of transmission.

Four additional service bit combinations are available for allocation by bilateral agreement.

The message information part of the non-data blocks (hold, end of transmission and end of message) is of no significance, but such blocks will still be checked at the receiver.

When the optional facilities groups g to k are not used, the first data block following the OFF to ON transition of " ready for sending " is automatically prefixed block A sequence indicator, group a. Data blocks BCABC etc. then follow sequentially unless one (or more) of the other types of block are inserted.

When the optional facilities groups g to k are used, the first data block is prefixed by one of the start of message indicators 1, 2, 3 or 4 (groups g to k), depending on the number of bits per character which will be used during transmission. Data blocks ABCAB etc. then follow. Should an interruption to a leased type connection occur during transmission or should an operator interrupt the transmission to change on to the speech mode, the transmission will be resumed with the sequence indicator following that of the last block to be accepted before the interruption. A start of message indicator should not be used after such an interruption.

In the case of switched connections special measures may be necessary to ensure that an interrupted message is not continued by a new message without appropriate indication.

4. *Correction procedure*

A binary 1 condition on the backward channel (the supervisory channel) indicates the need for repetition of information (RQ). Conversely, a binary 0 implies acceptance of the transmitted information. The rules governing the transmission and reception of these conditions are given below and in sections 5 and 6.

4.1 *Data transmitter sequence*

Starting and resynchronizing conditions are given in sections 5 and 6, this section only dealing with normal operations.

Data are transmitted block by block, but the contents of each transmitted block together with its service bits are held in store at the transmitter until correct reception has been ensured. Storage for at least two blocks must be provided.

During transmission of a block the condition of the backward channel (circuit 119) is monitored for a period of 45-50 milliseconds immediately prior to transmission of the last check bit. If any RQ is found within this period the block is rendered invalid by inverting this last bit. The transmitter then recommences transmission from the beginning of the previous block by reference to the store. During the retransmission of the block which follows the detection of the RQ signal, the state of the backward channel is ignored.

4.2 *Receiver procedure*

In normal operation a binary 0 is maintained on the backward channel as long as blocks are received with correct check bits and permissible service combinations. Any data contained in these blocks are passed to the receiver output. If a " clean copy " output is required data storage for at least one block should be provided, since a block cannot be checked until it has been completely received.

When a block has been received which does not meet the error check condition, binary 1 is transmitted on the backward channel and the expected service bit combination is noted in the receiver.

Usually, the first received data block in the repetition cycle having correct check bits also will have an acceptable service bit combination and any data within it will be processed. Occasionally the first block which checks correctly may bear an abnormal service bit combination due to a line transmission error in the backward channel (causing either a mutilated or imitated binary 0 signal). In either case the data in this first block are discarded. In the case that the block checks correctly but the service bit combination indicates the block preceding the expected block, a binary 0 should be applied to the backward channel.

If the next block checks correctly and bears an acceptable service bit combination, its data should be processed and normal operation resumed. In the case that the service bit combination indicates an invalid block, a binary 1 should be applied; moreover, if the service bit combination indicates the block following the expected block, it is implied that a binary 0 has been imitated for the whole of the 45 ms period specified in sub-section 4.1 and an alarm must be given since it is not possible to recover from this (rare) condition automatically.

5. *Starting procedures*

5.1 *Transmitter procedures and synchronizing pattern*

During the delay between " request to send " and " ready for sending ", line idle conditions (binary 1) are emitted by the modem. The first data signals, after the modem is ready for sending, are the synchronizing sequence prefix (0101) followed by the synchronizing filler followed by the synchronizing pattern. The filler may be of any length provided it includes at least 28 transitions and does not include the synchronizing pattern. The synchronizing pattern is 0101000010100101 starting from the left-hand digit (see Appendix for possible derivation). The 28 transitions are provided for bit synchronization purposes. These synchronizing signals are followed by block A or a start of message block (groups g to k in Table 3). During the whole of this sequence from the beginning of the synchronizing prefix the transmitter ignores the condition of the backward channel, acting as though binary 0 were present. The condition of the backward channel then assumes its normal significance (see section 4). Should this be binary 1 during the examination period of the second block, this block must be completed with the last bit inverted and the starting procedure must be recommenced from the beginning of the synchronizing sequence prefix.

5.2 *Receiver procedures*

Binary 1 is emitted on the backward channel at the receiving terminal until the synchronizing pattern (0101000010100101) is detected at which time binary 0 is emitted and block timing is established. The only acceptable service bit-combinations to follow the synchronizing pattern are the block A sequence indicator or a start of message indicator (when used). If other service bit combinations are received, binary 1 is returned and the search for the synchronizing pattern is resumed.

6. *Resynchronization procedure*

6.1 *Recovery of synchronization*

Should the receiver fail to recognize an acceptable block within a reasonable time, then it must examine the incoming bit stream continuously to find the synchronization pattern. When this pattern is found, block timing is re-established and the binary 0 condition applied to the backward channel; the procedure is identical to the starting procedure except that the expected service bit combination is that following the last sequence indicator to have been accepted.

6.2 *Emission of synchronization pattern*

If the normal repetition cycle has continued for a number of times consecutively (typically 4 or 8) the transmitter must assume that resynchronization is necessary. The normal repetition cycle is replaced by a three-block cycle including a synchronization block and the two blocks previously repeated. The synchronization " block " contains the synchronization sequence prefix, filler and pattern as described in paragraph 5.1.

Note. — A short filler should result in quicker resynchronization particularly when long blocks are used. However, the short filler has the disadvantage that correct synchronization can be lost if the prefix is imitated or disturbed by noise or should the synchronization pattern be disturbed. The use of the longer filler, making the block the same length as the data block, overcomes this difficulty. There is the option to choose either length, both lengths being compatible.

6.3 *Use of synchronization block for delay in transmission*

The information flow may be suspended by the insertion of a synchronizing " block ". In the case of the short filler it is essential that the receiving terminal should recognize the synchronizing prefix and change itself immediately into the synchronizing search mode, otherwise synchronization will be lost. In the case of the filler which produces a normal block length it is desirable to change into the search mode without abandoning block timing, a backward binary 0 being returned at the end of the block if the prefix is recognized and the check bits correspond to the synchronization pattern.

It may happen that the transmitter emits a resynchronization cycle before the receiver has changed into the synchronization search condition. The procedure at the receiver is identical to that just described for the use of a synchronization block for suspending the information flow.

7. *Interfaces*

7.1 *Modem interfaces*

In the normal case where the modems are not an integral part of the data terminal, the modem interfaces are as shown at points A-A in Figures 1 and 2. Where synchronous modems are employed, the appropriate signal element timing circuits will also be included in these interfaces.

7.2 *Data terminal interfaces*

Where the error control equipment (including stores) is not an integral part of the data terminal, the error control equipments are interposed between the data terminals and the modem. The data terminal interfaces are then as indicated at B-B and C-C in Figures 1 and 2 respectively. A signal element timing circuit is included in each of these interfaces.

7.2.1 In the case of the transmitting terminal all the interchange circuits perform their usual functions but " ready for sending " also takes advantage of the final paragraph of its definition in Recommendation V.24 and performs in the following manner:

Ready-for-sending circuit (see Figure 1)

This circuit, in conjunction with the signal element timing circuit, will inform the data terminal equipment when data are required in response to the request-to-send circuit. The ready-for-sending circuit will go to the ON condition when data are required and to the OFF condition when data are not required (in general this will be during the service and check bit transmissions and any repetition). This circuit will not go to the ON condition until the request-to-send circuit has gone to the ON

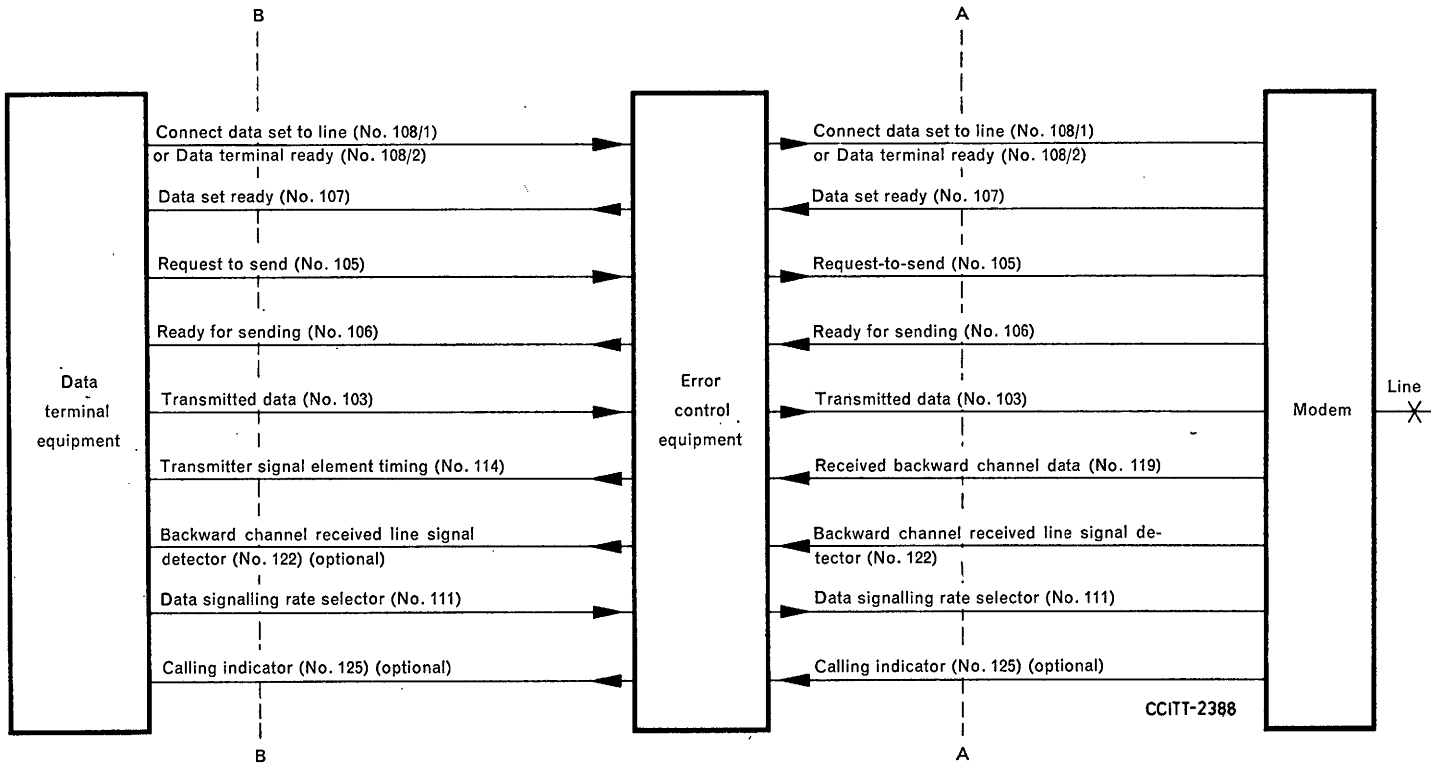


FIGURE 1. — **Interchange circuits—Transmitting terminal**

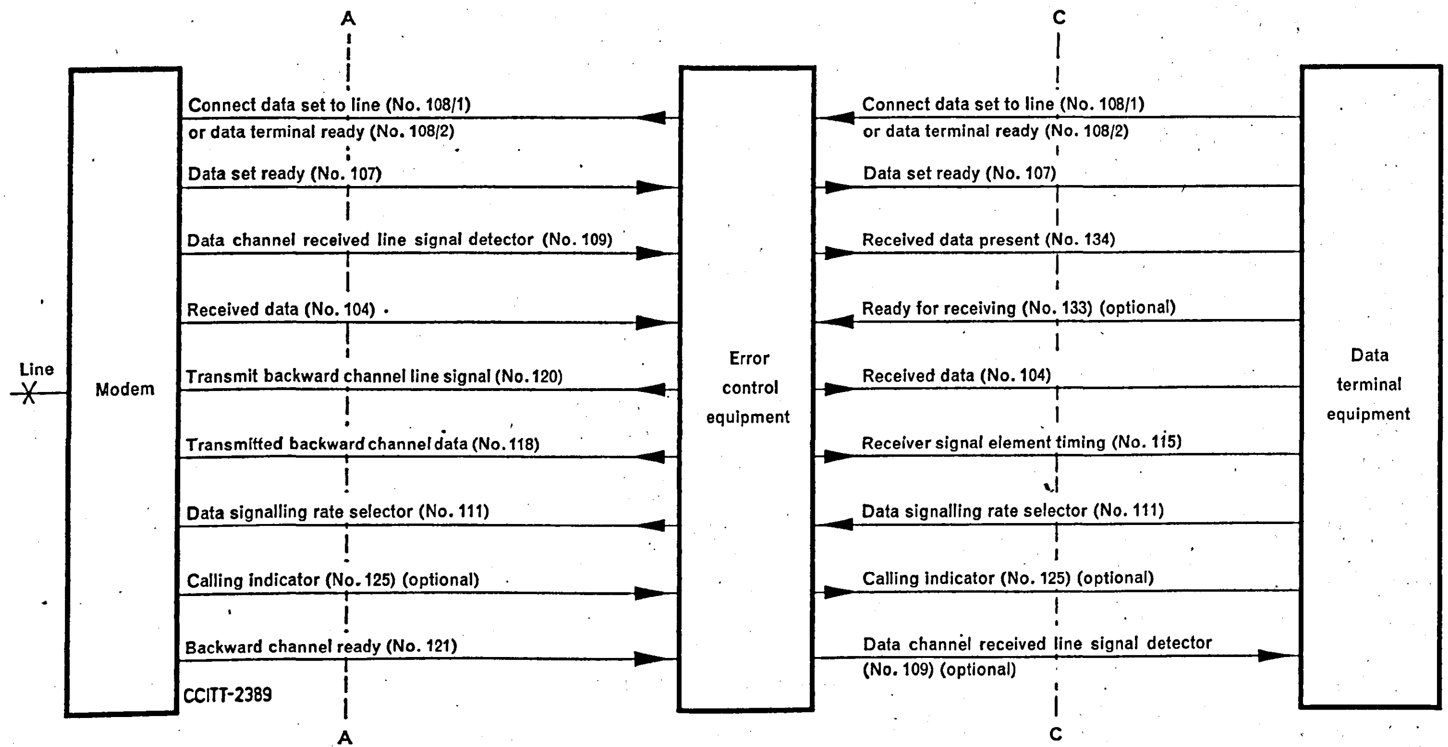


FIGURE 2. — Interchange circuits—Receiving terminal

condition. All transitions of this circuit will coincide with the signal element timing transition from ON to OFF. The transition from ON to OFF will thus coincide with the signal element timing transition from ON to OFF during the 240th, 480th or 960th bit of the information within a block, as appropriate.

7.2.2 In the case of the receiving terminal two new circuits are introduced, but since two (or more) of the modem interface circuits are not used in this interface, the number of circuits is not increased. Circuit 118 (transmitted backward channel data) is not available at this interface.

A " ready-for-receiving " function must be provided to inform the error control equipment of the status of the data terminal. This function may be performed by circuit 108, in which case a connection on the switched telephone network will be released when the circuit goes from ON to OFF. Alternatively, a separate function control circuit may be provided in order to retain the line connection for short periods when the data terminal is unable to accept data. This new circuit may be assumed to take the place of circuit 120 and functions in the following manner:

Ready-for-receiving (see Figure 2)

Direction: to error control equipment from data terminal equipment

The data terminal equipment shall maintain the ON condition on this circuit when the data terminal equipment is ready to receive data. Since the error control equipment will receive data in blocks, the data terminal equipment must be capable of receiving data also in blocks. Therefore, the data terminal equipment shall only change this circuit to the ON condition if the data terminal equipment is capable of accepting a block of data (240, 480 or 960 elements) and shall return to the OFF condition if the data terminal equipment cannot accept another block within 15 element intervals after the end of the previous block of transferred data.

Note. — If this ready-for-receiving circuit is OFF at the end of this 15-element period, an RQ condition will be generated.

The other new circuit performs the function of responding to the ready-for-receiving function and is therefore analogous to circuit 121 (backward channel ready). This new circuit functions as below:

Received data present (see Figure 2)

Direction: from error control equipment to data terminal equipment

This circuit, in conjunction with the signal element timing circuit, will inform the data terminal equipment when data are going to be output in response to the receive data terminal's connect data set to line (and separate ready-for-receiving circuit when provided) and the incoming data from the distant end being adjudged correct. The received data present circuit will go to the ON condition when data are going to be output and to the OFF condition at all other times. All transitions of this circuit will coincide with the signal element timing transition from ON to OFF. The transition from ON to OFF will thus coincide with the signal element timing transition from ON to OFF during the 240th, 480th or 960th bit of the information within a block as appropriate.

7.2.3 Additional interchange circuits may be provided at the data terminal interface by bilateral agreement of the users. These additional circuits may be used to introduce service bit control functions other than those provided as a basic necessity. Such circuits should not interfere with the operation of the recommended circuits.

8. *Use of service functions*

Data link escape is included in Table 3 as an optional indicator and its use is left to be agreed between operators. It may, for instance, be used to signal to a receiving station that the sending station wishes to speak over the connection. In this case the receiving equipment would operate a bell or similar calling device and transfer the line from the modem to a telephone. Alternatively, it may cause a short message to be printed on a teleprinter for the attention of an operator.

End of transmission is envisaged as giving a positive indication to the receiver that the transmission has ended and that the connection may be released. This is an alternative to the data terminal equipment interpreting the received data to know when to release the connection.

The optional start of message indicators together with the end of message indicator may be used to route messages to different destinations or terminal equipment at the receiving end, which may include the selection of equipment appropriate to the code used.

The hold block need not be used at a transmitter since synchronization sequences may be used as packing between data blocks in the event of data not being ready at the transmitting data terminal equipment, but if required a hold block may be used for this purpose.

APPENDIX

Encoding and decoding realization for cyclic code system

1. *Encoding*

Figure 3 shows an arrangement for encoding using the shift register. To encode, the storage stages are set to zero, gates A and B are enabled, gate C is inhibited and k service and information bits are clocked into the input. They will appear simultaneously at the output.

After the bits have been entered, gates A and B are inhibited and gate C enabled, and the register is clocked a further 16 counts. During these counts the required check bits will appear in succession at the output.

Generation of the synchronizing pattern may be achieved by making $k = 4$, the four bits being 0101. Clocking is suspended for the duration of the synchronizing filler.

2. *Decoding*

Figure 4 shows an arrangement for decoding using the shift register. To decode, gates A, B and E are enabled, gate D is inhibited and the storage stages are set to zero.

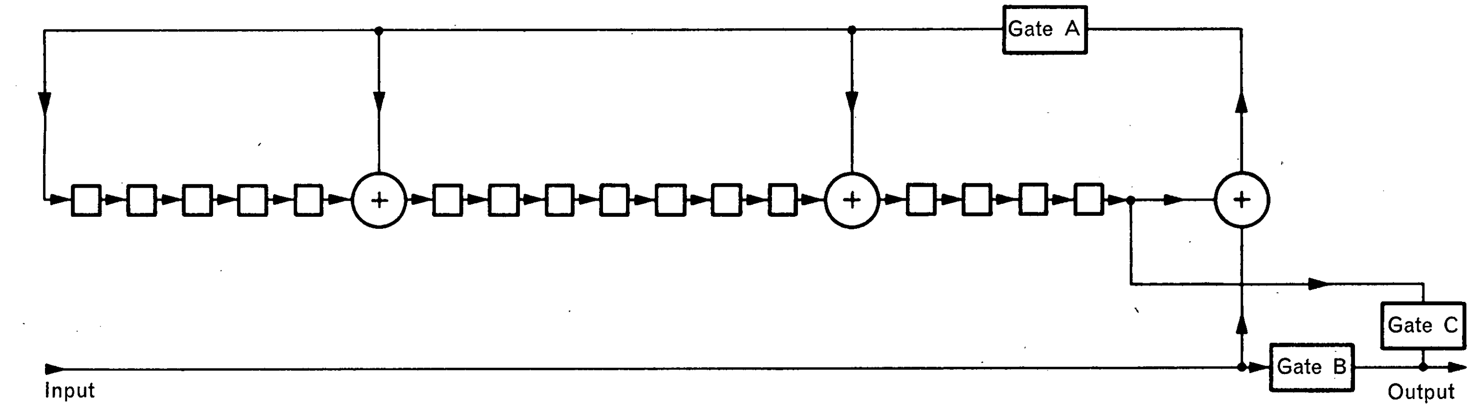


FIGURE 3. — **Encoder**

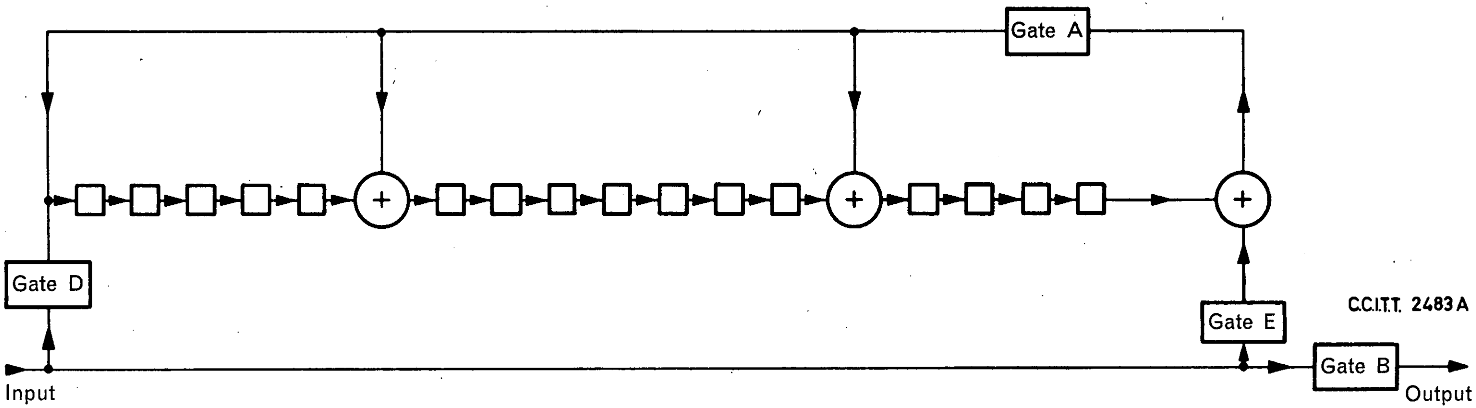


FIGURE 4. — **Decoder**

The k information or prefix bits are then clocked into the input and after k counts gate B is inhibited, the 16 check bits are then clocked into the input and the contents of the storage stages are then examined. For an error-free block the contents will be zero. A non-zero content indicates an erroneous block.

3. *Synchronizing at receiver*

For block synchronizing gate D is enabled (Figure 4) and gates A, B and E are inhibited and the register is examined in successive bit intervals for the required 16-bit pattern. When the pattern is recognized the register and bit counter are set to zero and decoding proceeds normally.

RECOMMENDATION V.50

STANDARD LIMITS FOR TRANSMISSION QUALITY OF DATA TRANSMISSION

(Mar del Plata, 1968)

One of the most important factors affecting data transmission quality—similarly to telegraph transmission quality—is the distortion in time of the significant instants (known as " telegraph distortion ": Definition 33.04 of the *List of Definitions*); the degree of signal distortion must be kept within certain limits, the ultimate objective being that the degree of distortion on received signals should be compatible with the margin of the receiving equipment.

This distortion on received signals arises from the composition of:

a) the sending distortion;

b) the distortion introduced by the transmission channel.

Hence, limits must be fixed for the degree of sending distortion and for the degree of distortion due to the transmission channel.

The limits contemplated for the transmission channel are specified in Recommendation V.53; these limits, which are not yet final, are recalled below:

Channel with modem V.21 : 20-25%

Channels with modem V.23

600 bauds—leased circuits : 20-30%

1200 bauds—leased circuits : 25-35%

600 bauds—switched circuit: 25-30%

1200 bauds—switched circuit: 30-35%
(when this mode of operation
is possible)

These figures are expressed provisionally in maximum degrees of individual distortion and apply to the circuit including the modems.

The limits for the degree of sending distortion must be fixed so that a reasonable margin is left for the receiving equipment, making allowance for the distortion introduced by the circuit.

In view of the foregoing, the C.C.I.T.T. *unanimously issues the recommendation that:*

1. With regard to the *quality of transmission signals* (signals at point A—Recommendation V.51), it is preferable, given the wide range of possible modulation rates, to adopt a single standard for each type of modem.
2. When a V.21 modem is used, the duration of a unit element should be at least 90% of the duration of the unit element at 200 bauds (i.e. $\frac{1}{200} \times \frac{90}{100}$ second, or 4.5 milliseconds).
3. When a V.23 modem is used, the duration of a unit element should be at least 95% of the duration of the unit element either at 1200 bauds ($\frac{1}{1200} \times \frac{95}{100}$ second, or 0.791 millisecond) or at 600 bauds ($\frac{1}{600} \times \frac{95}{100}$ second, or 1.583 millisecond).
4. If a system sends signals of which the sending distortion is systematically well below the limits specified above for the category concerned, the permissible margin for receivers of that system may be reduced.
5. The values indicated above could be revised when a more accurate plan for transmission quality has been drawn up.

Note. — The receive margin limits will be studied in liaison with the I.S.O. (see point W).

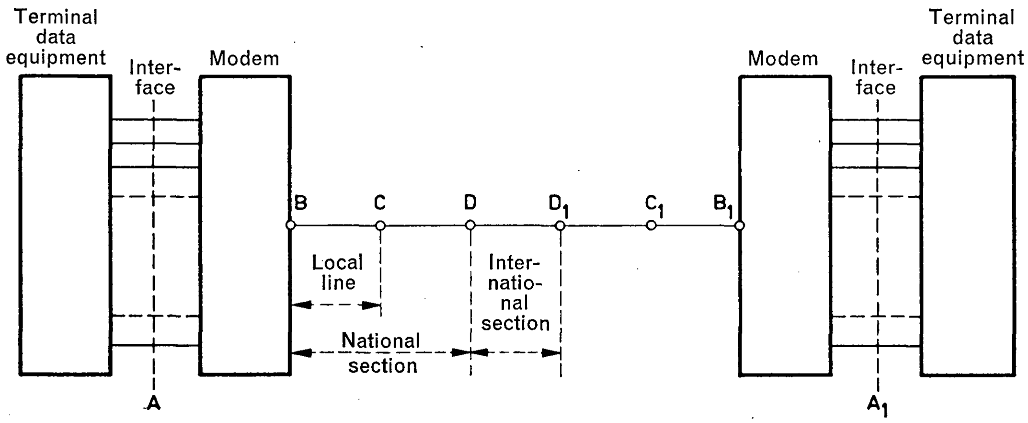
RECOMMENDATION V.51

ORGANIZATION OF THE MAINTENANCE OF INTERNATIONAL TELEPHONE-TYPE CIRCUITS USED FOR DATA TRANSMISSION

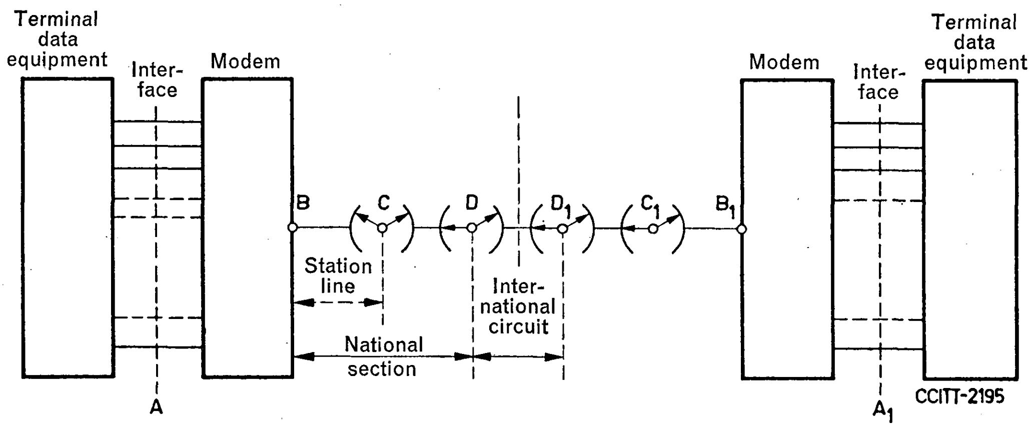
(Mar del Plata, 1968)

In order to ensure satisfactory co-operation between administrations and private agencies interested in the maintenance of international telephone-type circuits used for data transmission, and in order to ensure the maintenance of satisfactory data communication, it is necessary to unify the essential action to be taken for the establishment and maintenance of leased and switched telephone-type circuits used for data transmission.

The composition of a data transmission communication is as follows:



i) Leased circuit



ii) Switched connection

Taking into account these considerations, the C.C.I.T.T. *unanimously declares the view:*

1. That maintenance measurements should consist of:

   a) *Telegraph maintenance measurements;*

   b) *Telephone maintenance measurements.*

2. *Telegraph maintenance measurements*

Telegraph maintenance measurements consist of:

a) measuring the degree of distortion in time of significant instants;

b) measuring error rates.

2.a)1 *Maintenance method by measurements of significant instants distortion in time* (known as " Telegraph distortion ": Definition 33.04 of the *List of Definitions)*

Provisionally, the distortion of significant instants should be evaluated by measuring the degree of isochronous distortion in the absence of any appreciable disturbance which might produce error bursts.

Note 1. — This measurement should be made by evaluating the degrees of early and late individual distortion, the degree of isochronous distortion being the sum, in absolute terms, of the maximum degrees of early and late individual distortion.

Note 2. — It is recognized that measurements of isochronous and individual distortion are both quite important, but that for lack of an exact definition of the ideal reference instant, the non-uniformity of individual distortion measurements makes them difficult to apply in an agreed manner.

2.a)2 Measurements of distortion should be made using the apparatus recommended in Recommendation V.52 placed at points A and A1.

2.a)3 The duration of the measurement should be 20 seconds.

2.b) *Maintenance method by measurement of the error rate*

2.b)1 Measurements should be taken of the error rate on the bits and the error rate on the blocks (or sequence); the measurements should be made with the apparatus described in Recommendation V.52 using the pseudo-random sequence at the appropriate signalling rate. The apparatus should be placed at points A and A1.

2.b)2 The duration of the measurements shall be 15 minutes. Every measurement shall be a gross measurement, i.e. without any deduction of errors and without stopping the measurements if there is, for any reason, a brief period of high error-rate. However, if an incident of appreciable importance (e.g. prolonged interruption of the circuit or loss of synchronization on several pseudo-random sequences) upsets the measurement, the measurement should be started again.

3. *Telephone maintenance methods*

3.1 Telephone maintenance methods consist of :

i) measurement of the attenuation at 800 Hz;

ii) measurement of the attenuation distortion as a function of the frequency;

iii) measurement of the delay/frequency distortion;

iv) measurement of noise with the psophometer;

v) measurement of impulsive noise using the instrument described in Recommendation V.55.

3.2 Impulse noise levels should be expressed in dBm0 because a) the difference between the various national transmission plans is taken into account and b) the level value is related to the value of data signal level to a close degree.

3.3 Bearing in mind the following two points,

— that Recommendation V.2 demands a maximum data signal level of −10 dBm0 for a simplex transmission and −13 dBm0 for duplex transmission,

— that there has been considerable experience of using the threshold −18 dBm0 and −22 dBm0,

the threshold settings should be −18 dBm0 and −21 dBm0 respectively for the ordinary and special quality circuits mentioned in Recommendation M.102, the standard measuring instrument being adjustable to thresholds 3 dB apart.

Owing to lack of experience, no external filter should be used for present maintenance purposes.

3.4 At the time of measurement the line should be terminated at both ends on impedances of 600 ohms each. The modem may be used for this purpose if it complies with this impedance.

4. *Maintenance procedures* should be carried out as follows:

4.1 *Maintenance measurements before a leased circuit is put into service*

Before a circuit is put into service the following measurements should be made:

i) *between points B and B1:*
telephone measurements as specified in section 3;

ii) *between points A and A1:*
telegraph measurements as specified in section 2.a;

iii) so far as possible, these measurements should be made at peak periods;

iv) on the modems, modem test measurements.

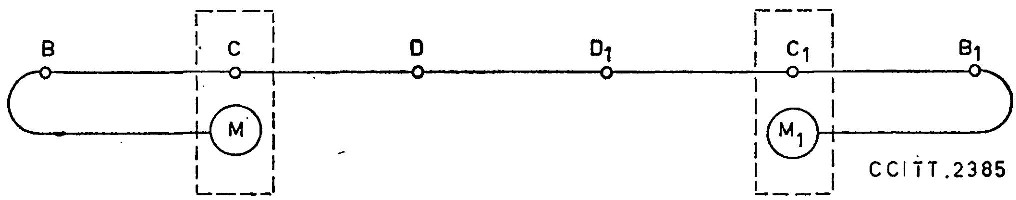
For this purpose the modems must be equipped, if possible, with looping devices on interface or line sides, so that the loop test can be performed either from the data terminal equipment or from a test centre.
The design of these loops and the tests to be made for modem checks cannot be specified for general application since the deciding factors for these tests are the type of modem and the type of interface signal.

4.2 *Routine maintenance of leased circuits*

On a leased point-to-point connection, maintenance of the B-B1 circuit is organized in the same way as prescribed for leased telephone circuits, which follows as closely as possible the organization prescribed for the maintenance of circuits in the general telephone service; on each international group of circuits there is, in each country, an international maintenance centre (I.M.C.) and one of them is the controlling centre. Circuit information is kept in each I.M.C. An I.M.C. has to inform the controlling I.M.C. of any intervention or incident on the circuit.

4.3 Telegraph measurements requiring restitution of the characteristic instants are practical only between the interfaces.

On the other hand, by using a supplementary telephone pair between B and C and between B1 and C1, it might be possible to take " telephone " measurements from the connecting centres, on the lines of the following diagram:



It is not recommended that measurements between M and M1 be substituted for those to be made between B and B1; however, this change in the measuring point might be acceptable when the measurements made from M to M1 can be regarded as representative of the measurements from B to B1. This is a question of local circumstances.

4.4 *Maintenance measurements before data transmission equipment for use on the switched network is put into service*

It is recommended that data test centres be set up in one or more countries; each subscriber to the switched data service should come under one of these centres. Before putting into service, this centre should be called and, once the communication has been set up, telegraph-type measurements only should be taken (distortion, error rate).

4.5 *Routine maintenance measurements for switched connections*

Between C and C1, the circuit should receive the routine maintenance for telephone circuits, in accordance with the Recommendations in the *White Book,* Volume IV.

Since means are available to the users for checking the quality of data transmissions:

— it is unnecessary to arrange for routine telephone-type maintenance measurements between B and B1;

— it is unnecessary to arrange for routine telegraph-type maintenance measurements between A and A1.

However, when there are signs of trouble developing, preventive maintenance measurements should be taken, but they will not be routine or systematic.

Administrations should be in a position to carry out modem tests.

5. *Fault location and clearance* should be effected as follows:

In each country an *international data co-ordinating centre* (I.D.C.C.) shall be set up.

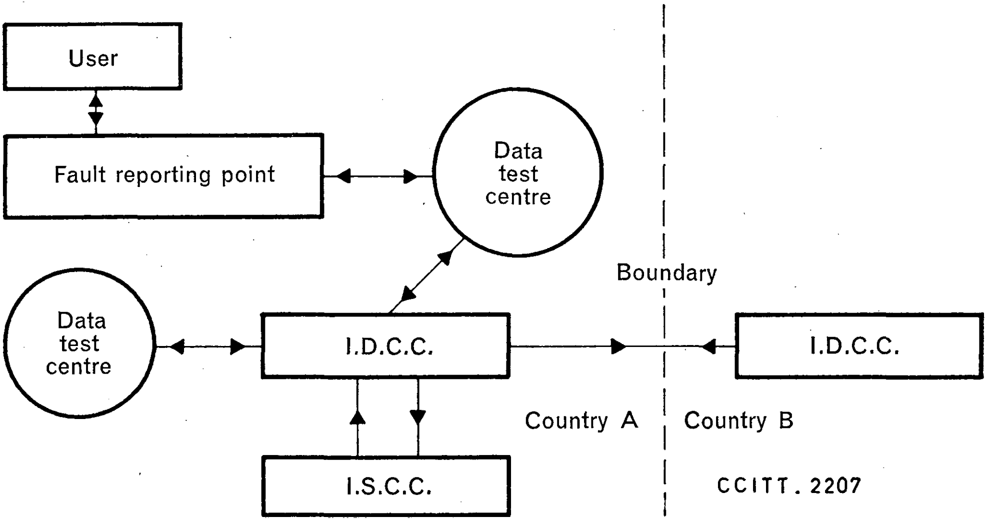
This centre:

— shall keep a record of international data transmission links (by leased circuit or switching);

— shall direct operations for putting international links for data transmission into service;

— shall act as a liaison point between countries for locating faults.

This centre is not a test centre, but it orders and co-ordinates the operation of test centres. It has no direct contact with users.



International service co-ordination centre

The above diagram shows the links between the various services which may take part in locating and clearing faults on an international link.

The procedure for reporting and clearing faults would thus be as follows:

a) The user tests the line by telephone with the user at the other end, if the data transmission system permits;

b) the user checks that the equipment for which he is responsible is in normal working order;

c) the user reports the fault to his fault-reporting point according to the instructions issued by his administration;

d) tests are made on the national network with the intervention, if necessary, of the data test centre;

e) if these tests are abortive, the I.D.C.C. is informed; this I.D.C.C. contacts the I.D.C.C. in the other country;

f) the I.D.C.C.s guide the search for the fault and, if necessary, call on the services of the maintenance centres on the international circuits involved.

If there is no data test centre in a country, the data test centres of a neighbouring country may be used with the consent of the administrations concerned.

By agreement between the administrations concerned, a user in country A may carry out tests with the aid of a data test centre in country B, and vice versa.

RECOMMENDATION V.52

CHARACTERISTICS OF DISTORTION AND ERROR-RATE MEASURING APPARATUS FOR DATA TRANSMISSION

(Mar del Plata, 1968)

Considering

that distortion and error rate measurements are of interest in data transmission and that measuring instruments must have compatible characteristics for international interoperation,

the C.C.I.T.T. *unanimously declares the view that:*

1. *Modulation rates*

a) The nominal *modulation rates* of the measuring apparatus to be used in the tests are:

50, 75, 100, 200, 600, 1200, 1800, 2000, 2400, 3000, 3600 and 4800 bauds.

b) The accuracy of these rates should be $\pm 0.03\%$. Their short-term stability should be better than 0.01% during a period of 20 seconds.

c) To obtain these rates a time-base external to the instrument may be used.

2. *Emission of test signals*

a) In order to test circuits for data transmission on an international basis it is necessary to standardize the test patterns to be used. Such a pattern should be a pseudo-random one having the following characteristics:

— it should contain all or at least the majority of eight-bit sequences likely to be met in the transmission of actual data;

— it should contain sequences of zeros and ones as long as possible compatible with ease of generation;

— the pattern should be of sufficient length such that at modulation rates higher than 1200 bauds its duration is significant compared with line noise disturbances.

Accordingly a 511-bit test pattern is recommended. The pattern may be generated in a nine-stage shift-register whose fifth and ninth stage outputs are added together in a modulo-two addition stage, and the result is fed back to the input of the first stage. The modulo-two adder would be such that the output produces a 0 output when the two inputs are similar and 1 output when the two inputs are dissimilar.

The following table shows the state of each stage of such a shift register during the transmission of the first 15 bits. The pattern over a longer period is as under:

11111111100000111101111100010111001100 etc.

Shift register stages during pseudo-random pattern generation

1	2	3	4	5	6	7	8	9	Output
1	1	1	1	1	1	1	1	1	1
0	1	1	1	1	1	1	1	1	1
0	0	1	1	1	1	1	1	1	1
0	0	0	1	1	1	1	1	1	1
0	0	0	0	1	1	1	1	1	1
0	0	0	0	0	1	1	1	1	1
1	0	0	0	0	0	1	1	1	1
1	1	0	0	0	0	0	1	1	1
1	1	1	0	0	0	0	0	1	1
1	1	1	1	0	0	0	0	0	0
0	1	1	1	1	0	0	0	0	0
1	0	1	1	1	1	0	0	0	0
1	1	0	1	1	1	1	0	0	0
1	1	1	0	1	1	1	1	0	0
1	1	1	1	0	1	1	1	1	1

It is clear from the table that this pattern is the sequence of bits in stage 9 of the shift register but it also represents the sequence in any other stage shifted in time. The choice of stage to be connected to the output is therefore a matter of circuit convenience.

b) Other test signals recommended are: permanent space, permanent mark, 1:1, 3:1, 1:3, 7:1 and 1:7, all of which may be sent over the line for an unlimited time.

c) A tolerance of $\pm 1\%$ is the maximum permissible for the transmitter distortion of the test signals.

d) The form of signals shall be as prescribed in Recommendation V.24.

3. *Synchronization of transmitting with receiving apparatus*

The receiving measuring set shall be synchronized with the transitions of the test signals received; these signals to be the recommended 511-bit sequence.

4. *Measurement of distortion*

a) The apparatus should measure the degrees of early and late individual distortion.

b) The receiver should measure bias distortion on reversals (1:1 signals), with a $\pm 2\%$ measurement accuracy.

c) The tolerance of the measurement of the individual distortion of pseudo-random signals shall be $\pm 3\%$.

d) The impedance of the receiving apparatus shall be as recommended in Recommendation V.24.

e) Provisionally, the margin of the measuring receiver should be measured in terms of the " margin of a synchronous receiving set " (Definition 34.09), under the following measurement conditions: the signals entering the receiving measuring set shall be those defined in paragraph 2 above, with the transitions in one direction only subject to a delay equal to $\Delta\%$ of the theoretical duration of a significant unit. The modulation rate may be fixed at the nominal value, and at a value in the range: nominal value $\pm 0.03\%$. The receiving measuring set should not indicate any error after the synchronizing period as long as Δ is less than 90%; this applies to both directions of the transitions subject to the delay Δ. Under these conditions the margin of the measuring apparatus shall be said to be over 90%.

5. *Measurement of error rate*

Both bit error rate and block error rate measurements should be possible with the apparatus.

Information on the error rate for sequences of 511 bits should be given in a form similar to that for the bit error rate, the two measurements being made simultaneously.

RECOMMENDATION V.53

LIMITS FOR THE MAINTENANCE OF TELEPHONE-TYPE CIRCUITS USED FOR DATA TRANSMISSION

(Mar del Plata, 1968)

For data transmission maintenance purposes, the following limits are recommended for the essential parameters indicating the quality of a transmission channel.

1. *Telegraph distortion limits*

Limits for the *degree of distortion on a transmission channel* between the interfaces (i.e. including the modems) vary with the data transmission system. The following values are recommended, these same limits applying to the backward channel:

System with modem V.21 : 20-25%
Systems with modem V.23
600 bauds—leased circuits : 20-30%
1200 bauds—leased circuits : 25-35%
600 bauds—switched circuit: 25-30%
1200 bauds—switched circuit: 30-35%
(when this mode of operation is possible).

These figures express provisionally maximum degrees of individual distortion. They will be converted into degrees of isochronous distortion once a method for determining the reference ideal instant has been studied, specifying a synchronization procedure for the distortion-measuring receiver.

2. *Limits for error rates*

2.1 *Bit error rate*

The following limits are recommended; when they are exceeded the maintenance services should consider the transmission channel defective.

Modulation rate (bauds)	*Connection*	*Maximum bit error rate*
1200	switched (when possible)	10^{-3}
1200	leased	$5 \cdot 10^{-5}$
600	switched	10^{-3}
600	leased	$5 \cdot 10^{-5}$
200	switched	10^{-4}
200	leased	$5 \cdot 10^{-5}$

the period of measurement being about 15 minutes (more precisely, the period corresponding to the transmission of the total number of sequences which is closest to 15 minutes).

Note. — These values are not intended for use in planning circuits, but for the information of maintenance services.

2.2 *Block error rate*

Information on the error rate for sequences of 511 bits would be given in a form similar to that for the bit error rate, the two measurements being made simultaneously. However, no limit for the sequence error-rate can be recommended for the time being.

Note. — To enable administrations to appreciate the value of sequence error rate measurement, the following table shows the *maximum and minimum theoretic values* of error rates for sequences of 511 bits corresponding to different values of bit error rate.

These theoretic values do not depend on the modulation rate. For the purposes of this table, a modulation rate of 1200 bauds has been taken as an example.

Modulation rate	: 1200 bauds
Period of measurement	: 15 minutes = 900 seconds
Number of bits transmitted	: 1 080 000
Length of sequence	: 511 bits
Number of sequences transmitted	: 2113

Bit error rate	Number of erroneous bits	Erroneous sequences			
		Maximum number	Maximum rate in %	Minimum number	Minimum rate in %
1	2	3	4	5	6
$2 \cdot 10^{-3}$	2160	2113	100	5	0.24
10^{-3}	1080	1080	51.1	3	0.15
$5 \cdot 10^{-4}$	540	540	25.5	2	0.10
10^{-4}	108	108	5.1	1	0.05
$5 \cdot 10^{-5}$	54	54	2.5	1	0.05

Note 1. — The *maximum* number of erroneous sequences given in column 3 corresponds to a *uniform* distribution of erroneous bits (one bit per sequence).

Note 2. — The *minimum* number of erroneous sequences given in column 5 corresponds to a *grouped* distribution of erroneous bits (sets of 511 bits affecting the sequences).

Note 3. — It will be seen that for a bit error rate of 10^{-3}, the sequence error rate can vary between 0.15% and 51.1%. (This shows the value of sequence error rate measurement, not only for users, but also for administrations, which can thus obtain useful information on the causes of bit and sequence error).

3. *Limit of uniform-spectrum random noise*

See Recommendation G.153 (*White Book,* Volume III).

4. *Limits for impulsive noise*

4.1 Since the number of noise impulses is counted with the instrument in the flat bandwidth condition, the following limit is recommended:

On a leased circuit the admissible limit is 70 impulse counts per hour; but in view of the fact that error rate measurements are conducted for periods of 15 minutes each, the recommended maintenance limit is 18 counts in 15 minutes for leased circuits. The measurements should be made during peak hours.

Note. — These values are given as an indication. The question of the duration of the measurement and permissible maximum standards for impulsive noise forms the subject of future studies.

4.2 For the general switched telephone network, no maintenance limits are recommended for impulse counts, although it is recognized that measurements of impulsive noise are useful as a diagnostic aid at the discretion of the administration. This is because the impulse count results taken on any one connection vary considerably with time and even greater differences appear between various connections.

RECOMMENDATION V.55

IMPULSIVE-NOISE-MEASURING INSTRUMENT FOR DATA TRANSMISSION

(Mar del Plata, 1968)

Considering

that impulsive noise is of interest to data transmission and that first consideration should be given to a simple pulse counter suitable for field use,

the C.C.I.T.T. *unanimously declares the view:*

that the instrument for impulsive noise measurement should have the following characteristics:

a) it should register a count whenever the voltage applied at the input exceeds an adjustable threshold;

b) it should operate independently of the sense (or polarity) of the applied impulse;

c) the nominal input impedance should be 600 ohms within the range of 200 to 3400 Hz or a switchable high impedance giving a bridging loss not exceeding 0.1 dB. The input circuit should be balanced in relation to earth, with a degree of balance such that a pulse whose level is 60 dB higher than the threshold, applied between the mid-point of the source impedance and the earth terminal of the instrument, should not operate it;

d) the threshold should be adjustable in steps of 3 dB (with a tolerance of ± 0.5 dB) from -50 dB to 0 dB with respect to 1.1 volt, which is the peak voltage of a sine wave having a power of 1 mW in 600 ohms.

The thresholds for the two polarities should be within 0.5 dB of each other.

e) after the instrument has been calibrated against a 1000-Hz sine wave signal at a level of 0 dBm and with the weighting control network in the " flat " condition, rectangular pulses of either polarity, of 50 milliseconds duration having a peak amplitude of 1.1 volt, and with an interval between pulses in excess of the operating time of the counter (dead-time, see f), shall be applied to the input of the instrument and cause the counter to operate at the correct rate. When the operating level control is set at —1 dBm, and the duration of these pulses is gradually reduced, the counter shall count at the correct rate when the pulses have a duration of 50 microseconds but shall not count when the pulses have a duration of 20 microseconds;

f) dead-time is defined as the time after which the counter is ready to register another pulse after the start of the preceding pulse. Several values for this dead-time have been proposed. Whatever range of values may be adopted for a particular instrument, the value of 125 ± 25 milliseconds should be provided in all cases;

g) in the flat bandwidth condition, the response should be within ±1 dB in the frequency range 275 Hz to 3250 Hz. Outside this range, the response curve should be compatible with the sensitivity requirement in clause e above;
The instrument may provide other optional bandwidths;

h) to enable the instrument to be used for other than maintenance measurements, it should be so designed that external filters may be added.

One of these filters shall have the following characteristics:

3 dB points at 600 Hz and 3000 Hz;

Characteristic from 750 Hz to 2300 Hz flat to within ± 1 dB;

Response falling off at about 18 dB per octave:
- below 600 Hz
- above 3000 Hz

i) calibration should be possible from the peaks of a 1-mW standard test tone;

j) a built-in timer, continuously adjustable from 5 to 60 minutes, should be provided. Significant testing intervals will be 5, 15, 30 and 60 minutes;

k) all the preceding clauses shall be satisfied when the ambient temperature varies between +10° C and + 40° C;

l) the capacity of the counter shall be at least 999.

ANNEX

(to Recommendation V.55)

Use of the impulsive noise counter for data transmission

1. Levels should be expressed in dBm0, because a) the difference between the various national transmission plans is taken into account and b) the level value is related to the value of data signal level to a close degree.

Bearing in mind the following two points:

- that Recommendation V.2 demands a maximum data signal level of −10 dBm0 for a simplex transmission; and −13 dBm0 for duplex transmission;
- that there has been considerable experience of using the threshold −18 dBm0 and −22 dBm0;

the threshold settings should be −18 dBm0 for telephone type circuits and −21 dBm0 for the special quality circuits mentioned in Recommendation M.102, the standard measuring instrument being adjustable to thresholds 3 dB apart.

Owing to lack of experience, no external filter should be used for present maintenance purposes.

However, the study of the use of external filters should continue; one of these filters should be the one described in point h of Recommendation V.55; the United Kingdom Administration uses an impulse counter that also includes a filter having the following characteristics:

3 dB points at 300 Hz and 500 Hz;

Response falling off at about 18 dB per octave:

— below 300 Hz,
— above 500 Hz.

At the time of measurement the line should be terminated at both ends by impedances of 600 ohms. The modem may be used for this purpose if it complies with this impedance.

2. For counting the number of impulses, the instrument shall be used in the " flat " bandwidth condition.

On a leased circuit, the admissible limit should be 70 impulse counts per hour; but in realizing that error rate measurements are conducted for periods of 15 minutes each, the recommended maintenance limit should be 18 counts in 15 minutes for leased circuits. The measurements should be made during peak hours.

3. For the general switched telephone network, there should be no recommended maintenance limits for impulse counts, but the instrument might be useful as a diagnostic aid at the discretion of the administrations. This is because the impulse count results taken on any one connection vary considerably with time and even greater differences appear between various connections.

4. The correlation between the bit error rate and the number of impulsive counts thus determined has not yet been established.

IMPORTANT NOTICE

1. An asterisk indicates that a question is urgent, i.e. that the study of the question has to be completed before the Vth Plenary Assembly.

2. Since Special Study Group D was set up by the Plenary Assembly, all questions relating to pulse code modulation (p.c.m.) have been assigned to this Study Group for the time being.

The Chairman of Special Study Group D will make arrangements with the other Chairmen for effecting liaison with the other Study Groups concerned as work progresses.

3. When a question is of interest to more than one Study Group and no joint study group has been set up to deal with it, the mention of the other Study Group(s) concerned is intended for the information of the members of the Study Group to which the question has been assigned, to enable them to arrange for the necessary co-ordination *within their national* Administrations, in accordance with a decision of the IVth Plenary Assembly.

QUESTION ON DATA TRANSMISSION ENTRUSTED TO SPECIAL STUDY GROUP A FOR THE PERIOD 1968-1972

Chairman: Mr. J. Rhodes (United Kingdom)

Vice-Chairman: Mr. V. N. Vaughan (United States of America—American Telephone and Telegraph Company)

Question 1/A — Data transmission

(continuation of Question 43 of Study Groups 1 and 8, 1957-1960, amended at Geneva, 1964, and at Mar del Plata, 1968)

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 test to be made

Point	Title	Remarks
B	Supplement to the vocabulary for data transmission	
C *	Complement to standardization of International Alphabet No. 5	See Questions 6/I and 9/VIII; to be studied by the ALP Joint Working Party
F *	Automatic originating and answering of calls in the telex network	
H *	New telegraph-type network for message and data transmission	See Question 9/X; to be studied by the NRD Joint Working Party
I	Error control in the telegraph network	

* Urgent question.

Point	Title	Remarks
J	Use of telephone-type circuit for the simultaneous transmission of: 1) data and telegraph signals; 2) data and speech; 3) data and facsimile signals	To be studied with LTG Joint Working Party
K	Duplex data transmission at 200 bauds, alternating with telephony	
L	Tests for data transmission systems on telephone-type circuits	
M *	Complement to standardization of the modem at 600/1200 bauds	
N *	Standardization of data signalling rates on telephone-type circuits	
O *	Modem for data signalling rates above 1200 bits per second on the general switched telephone network and 2400 bits per second on telephone-type leased circuits	
Q *	Further study of parallel data transmission for universal use on telephone circuits	
Q *bis* *	Study of parallel data transmission systems using the push-button telephone signalling frequencies	
R	Error control in the general telephone network	
S + T	Measurement of phase distortion and transmission loss between subscribers	Results to be reported to S.G.s IV and XIV
U *	Specification of frequency/loss characteristics, phase distortion and impulsive noise limits for circuits leased for data transmission; measuring techniques to check these specifications	Of concern to S.G. IV; to be studied with LTG Joint Working Party
V *	Specification of impulsive noise limits for data transmission over switched telephone circuits	Of concern to S.G. XI (Question 8/XI) and Study Group IV (Question 8/IV)
W *	Maintenance methods	Study results to be reported to S.G. IV
Y *	Automatic originating and answering of calls in the telephone network	
Z *	Data transmission over 48-kHz and 240-kHz circuits	
AA *	Acoustic coupling of data terminals to the telephone instrument	
AB *	Use of digital transmission (or pulse code modulation)	To be studied initially by Special Study Group D (Questions 11/D and 7/D)

Point	Title	Remarks
AC *	Use of circuits established by means of satellite	
AD *	Comparative tests of modems	
AE *	Continuation of the compilation of the list of interchange circuits in general	
AF *	Planning limits for the characteristics of telephone-type circuits and wideband circuits used for data transmission	
AG *	Transmission constraints on wideband data system	
AH *	Modems for transmission of medical analogue data	
AI *	Transmission of data over intercontinental telephone-type circuits	
AJ	Future integrated networks for data transmission	Point 3 of this Question is to be studied by Special Study Group D (Question 11/D)

* Urgent question.

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

(continuation of point B of the study programme 1964-1968, amended at Mar del Plata, 1968)

This study should be pursued, bearing in mind:

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

— study of definitions for p.c.m. terms by Special Study Group D.

Question 1/A - point C* - Complement to standardization of International Alphabet No. 5

(continuation of point C of the study programme 1964-1968, amended at Mar del Plata, 1968)

1. Study of points arising from the implementation of Recommendations V.3 (International Alphabet No. 5) and V.4 (General structure of signals of International Alphabet No. 5 code).

2. Further study of the definitions and use of certain control characters such as " shift-out ", " shift-in ", ESC (Escape), DLE (Data link escape), etc.

Note 1. — This point, common to Study Group I (Question 6/I) and Study Group VIII (Question 9/VIII), is to be studied by the Joint Working Party ALP.

Note 2. — The study should be pursued in co-operation with the I.S.O.

Note 3. — For this study, see Supplements Nos. 1, 2, 3 and 4 of this volume.

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

(continuation of point F of the study programme 1964-1968, amended at Mar del Plata, 1968)

Apart from the case standardized in Recommendation V.11 (Automatic calling and/or answering on the telex network), the following cases should be examined:

a) keyboard selection system;

b) system of a simpler interface.

Question 1/A - point H* — New telegraph-type network for message and data transmission

(continuation of point H of the study programme 1964-1968, amended at Mar del Plata, 1968)
(Question 9/X of Study Group X: to be studied by the Joint Working Party NRD)

Programme for this study

1. *What should be adopted for modulation rate steps?*

It is understood that the 0-200 baud range is the minimum admitted; upper limits envisaged should be specified.

2. *What signalling should be proposed between international switching centres?*

The aim in view is to seek agreement on a uniform signalling system which would be satisfactory as regards:

— the mean time for setting up calls, which should be short;

— the problem of verification of the obtained subscriber.

As regards signalling to set up and clear down calls, the Study Group feels that the list of signals and functions mentioned in the left-hand column of Table 1 in Recommendation U.11 could serve as a basis for the list of signals and functions to be provided in the new network. This list would of course be adapted to the conditions of the new network. The signalling system would have to allow for the use of communication satellites (long propagation times) and the use of digital transmission for a pulse-code-modulation system.

Certain administrations are of the opinion that the draft signalling plan with the use of a special circuit for signalling between switching centres (as proposed for telephone system No. 6) should be considered.

It is very important that the signals used for selection should be very strictly standardized, whereas the message transmission should be able to be independent of the code. Hence, the selection signals and the message signals could be of a different nature.

With regard to verification of the obtained subscriber, it should be possible to effect this automatically. This verification is bound up with the call-connected signal, it being clearly understood that the call-connected signal and the verification signals (which will be known as the selection verification signals) correspond to two separate functions. Two methods were described: in one of them, the call-connected signal is first returned to the calling subscriber, then the verification is checked by the terminal equipment of the calling subscriber; in the second method, the selection is checked by the outgoing switching centre—if the result is positive, a proceed-to-transmit signal is then sent to the calling subscriber.

The method of checking the selection obtained must be studied bearing in mind that the subscriber terminal equipment may not always include a teleprinter.

3. *Should it be possible to choose freely the code, modulation rate and telegraph transmission method (start-stop or synchronous)?*

— the system should accommodate both the start-stop method and the synchronous method;

— it should be possible to choose the message transmission code freely, provided it is a two-condition code.

It should be noted that the clearing signal may impose some restriction on this principle of the independence of the system from the code. The question of the transmission plan must be studied in relation with the question of independence vis-à-vis the code and speed.

— that any modulation rate included in the range (or ranges) finally admitted must be permitted for message transmission.

4. *If it is not possible to adapt existing national systems soon, should the connection of a subscriber in country A to a switching unit in country B be offered as an interim solution? What signalling should be adopted on the line connecting such a subscriber to the switching centre?*

The reply to the first part of this question is obviously a matter for agreement between the administrations concerned. However, it should be noted that when transfer is made from the interim situation to the definitive situation, i.e. when country A brings its own switching system into service, there is a risk that the subscriber equipment in country A may become useless if the signalling system between a subscriber and his exchange has not been internationally standardized or if this standardization has not been applied. It must be recognized that the introduction of such standardization is conditioned by the prior standardization of signalling between switching centres.

Hence, it is only possible to settle this case by bilateral agreement for the time being.

5. *Which method of working (half-duplex or duplex) should be permitted for users?*

The network (i.e. the switching centres and the links between switching centres) should allow for duplex operation.

The question remains open as regards the subscriber's line and terminal equipment.

As mentioned under point 4, standardization of subscriber lines and of the terminal selection and signalling equipment should not be considered to be a purely national affair. An international standardization would simplify international signalling and would be a great help to developing countries.

6. *Under which rules should the rates be fixed and international accounting organized?*

This question must be studied in liaison with Study Groups I and III. The projected system will be able to supply automatically a large amount of information for the establishment of charges.

7. *What transmission paths should be used for modulation rates higher than 200 bauds?*

The study will have to be carried out in liaison with Study Group IX and, as regards the use of p.c.m., with Special Study Group D.

8. *To what extent are existing national switching systems suitable for transmission at higher rates?*

Certain administrations indicated that it would be difficult to use their existing telex network for the new service.

The administrations which feel able to adapt the existing telex network to higher speeds should indicate to what extent (modulation rates, codes) this adaptation would be possible.

9. *Which basic switching methods (circuit switching or message retransmission) should be used?*

The projected network is a world-wide network. Since many subscribers will wish to have the possibility of duplex operation or the immediate exchange of information, it considers that the circuit-switching method will have to be the basic one.

However, it notes that storing of messages with subsequent retransmission will be necessary for certain special services.

Among these special services, consideration must be given to:

— connections between equipments working at different speeds (in particular, interconnection with the telex networks);

— services with multiple addresses, such as broadcasting, conference;

— storage of the message in case of occupation of trunk circuits or of the called subscriber's lines. This storage should be requested by the calling subscriber; in this case, subsequent routing of the message is controlled by the switching centre where the storage takes place. It is important to pay attention to the choice of the place for this storage, when the called subscriber line is engaged.

It should be noted that this question of auxiliary storage devices will raise problems concerning the alphabet and the speed.

This study programme was prepared with a circuit-switched network in mind, with message retransmission for certain auxiliary services.

Note. — See Supplements Nos. 44 and 45 to this volume.

ANNEX

(to point H)

Data transmission in switched networks

(Extract from the report by Special Study Group A, December 1967)

1. The IIIrd Plenary Assembly of the C.C.I.T.T. (Geneva, 1964), decided that the following Question 9/X should be studied:

Question 9/X — Special circuits used in switched services

Is it necessary to provide facilities in automatic switching centres for the selection of special circuits for data transmission, facsimile, high-speed telegraphy, etc., by means of a code number, for example?

If so, what are the possibilities and what arrangements should be made?

This question was allotted to Study Group X (Telegraph switching).

In addition, the study programme annexed to Question 1/A (Data transmission) included the following point H:

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

1. 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?

2. At its meeting in Brussels in November 1965, Special Study Group A made the following comment:

i) The need for a switched 200-baud telegraph-type network for data transmissions is justified by:

a) the impossibility of using, in a large enough number of cases, a code other than the 5-unit code on the existing telex network, unless signals are regrouped by converters;

b) the slowness of 50-baud transmission;

c) the improvement in the error rate on a telegraph-type network as compared with that obtained if the switched telephone network is used and modems installed at the user's end;

d) the possibility of offering users lower charges than telephone charges.

ii) It appears technically feasible to integrate the 200-baud system into the existing telex network to a certain extent, but this extent requires further study.

If it is considered desirable to integrate a 200-baud system into the existing telex network, this integration should be effected in such a way as to avoid the necessity of subsequent modifications.

iii) Most of the members held the view that a modulation rate of 200 bauds constituted a reasonable limit for the planning of a telegraph network for data transmission. However, if such a network were to be established quite separately from the existing telex network, the question of using it at rates of more than 200 bauds might arise and would deserve more thorough study.

3. Proposals were subsequently made for setting up telegraph-type networks which can be operated at modulation rates higher than 200 bauds.

Thus at its Melbourne meeting in October 1966, Study Group X thought it useful to broaden Questions 9/X and 1/A—point H by combining them into the following question:

a) what new network (or networks) should be provided to take into account the probable demand for an improved message interchange network to allow for the full exploitation of International Alphabet No. 5 at a higher character rate than is at present provided?

b) what network or networks are likely to be required for data interchange at medium and high rates of information transfer?

c) what transmission medium should be employed for such networks?

d) what should be the method of switching?

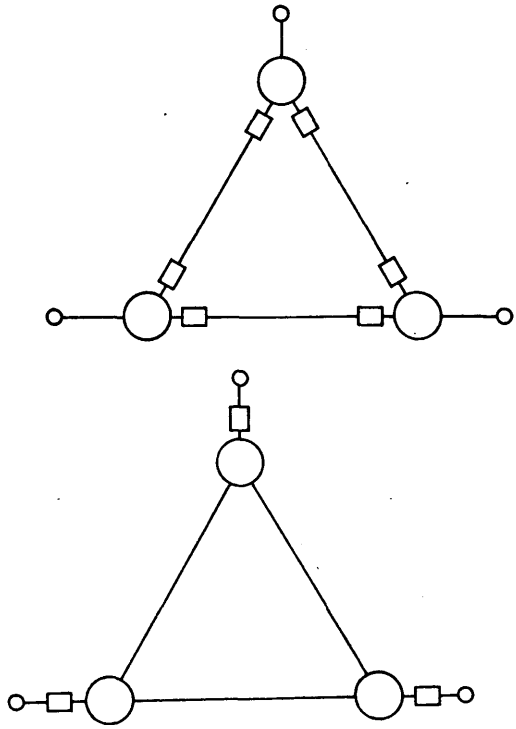
In June 1967, Study Group X met in Geneva to prepare the study of new telegraph-type networks for data and message transmission. Its study programme (Question 9/X = 1/A-point H) was also approved by Special Study Group A at its meeting in December 1967.

In this connection, it is worth pointing out that the distinction between a telegraph-type and a telephone-type network is not based on the modulation rate; in a telegraph-type network, there is no modem at the subscriber's premises; on subscriber's lines or through switching centres, signal-trains are formed by direct-current modulation; in a telephone-type network, the modem is at the subscriber's premises; on the subscriber's lines and through the switching centres, signal-trains are formed by modulation of an alternating current.

4. A special meeting of Study Group A was held from 13-15 December 1967 to prepare the studies to be undertaken on future data transmission networks.

Telegraph-type network

Telphone-type network



CCITT.2541

□ Modem

○ Switching centre

o Subscriber

The question confronting Study Group A was defined by the Chairman, Mr. J. Rhodes, in the following terms:

" Up to the present we have provided facilities for data transmission over existing networks which were provided for the national and international telephone and telegraph services. The fact that these services, in particular the telephone service, were not designed for data transmission, has produced many problems which administrations and private operating companies, in co-operation with the data processing industry, have solved most admirably and, on the whole, have been able to provide the data transmission facilities as the demand has arisen.

" Data processing and the demand for data transmission is, however, growing very rapidly and it is time to pause and consider whether the existing facilities required can continue to be provided on the existing telephone and telegraph networks. We have to-day this unique opportunity to discuss the problem.

" In the past it can generally be said that communication networks have started nationally and then, at a later stage the requirement for international connection has arisen. It is a considerable tribute to the work of the C.C.I.T.T. and its predecessors, the C.C.I.F. and the C.C.I.T., that such interconnection has been made possible. Nevertheless, I am sure you would agree that if the problem of international interconnection had been considered earlier in the national consideration of such networks, our tasks in the past would have been very much easier.

" If I might summarize the task as I see it, the problems before us are:

1) What new facilities for data transmission are to be asked from us in the next 5, 10 and 20 years?

2) Should we continue to try to provide such facilities over the existing telephone and telegraph network?

3) Should we, on the other hand, consider the study and later the setting up, of a specialized data network or networks?

" Big changes are also likely to come in transmission. Pulse code transmission of speech is showing great promise and will bring into being digital transmission systems which provide facilities for data transmission at a far higher rate than can be compressed into the present telephone bands using analogue methods. Unfortunately, like all major changes, it must be introduced slowly and it is not possible to foresee when digital transmission systems will be at least partially universally available."

5. The following conclusions were drawn from the discussions: For the distant future the solution would seem to be an integral network for all the telecommunication services using pulse code modulation (p.c.m.) systems. Such a network will not be universal before a certain period which depends on the progress in the use of p.c.m.; study of the use of p.c.m. should be carried out with a view to this integration *.

For the near future the situation appears as follows:

— there is a proposal for study of a telegraph-type network, by switching of circuits, usable for data and message transmission at a modulation rate up to 200 bauds;

— there is a proposal for studying the possibility of this network in reference to higher data signalling rates; the maximum rate to be envisaged is not yet specified;

— there is a proposal in favour of a switched network for data signalling rates in the range of 40-50 kilobits per second;

— a network with intermediate retransmission (store-and-forward) can be envisaged;

— finally, it should point out a more pronounced use of V.21 and V.23 modems over the existing telephone network.

6. Study Group X, at its meeting in June 1967, concluded that it was useful to study the construction of an international telegraph-type network for traffic (message and data) at a modulation rate in the range of 0-200 bauds, with the possibility of making the network accommodate higher rates.

7. Special Study Group A has noted that Study Group X proposes to embark on a study programme for a new network for message and data interchange at rates up to 200 bauds. Such a network would have useful possibilities for data transmission side by side with the existing facilities for 200 bauds on the telephone network.

Special A also noted the possibility of this network being used at higher transmission rates.

Special A welcomes such a study in view of the merging of data, telephone and telegraphy. Nevertheless at this stage Special A feels that solutions to the transmission of data at rates higher than 200 bits per second are not sufficiently clear for the Study Group to decide the specific type of network which will be suited to the future.

It nevertheless welcomes the outcome on this problem which may come from the Study Group X studies and at the same time it will itself set a question to consider the more broad and general aspect of data transmission network at higher speeds and also the much higher speeds which will be required in the future.

For this purpose Special Study Group A drew up a new question, which is now indicated under point AJ of the study programme 1968-1972.

* See Questions 7/D and 11/D (Volume III of the *White Book*).

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

(continuation of point I of the study programme 1964-1968, amended at Mar del Plata, 1968)

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. — The case of simple arrangements which might be restricted to error indication only should also be taken into consideration in this study (see paragraph 1 to the Annex to this point).

Note 3. — See Supplements Nos. 5 and 6 of this volume.

ANNEX

(to point I)

EXTRACT FROM THE REPORT BY SPECIAL STUDY GROUP A, OCTOBER 1968

1. Contributions so far submitted on this subject make specific proposals on information feedback systems, on systems using character redundancy or on systems using block repetition systems.

These contributions tend to be directed at error control systems of the types referred to in Recommendation V.10, I (e) and II (e). Certain administrations considered that for a number of applications, particularly the telemetering, systems of these kinds are too expensive and offer superfluous facilities. There may be need for simple arrangements which might be restricted to error indication only; improvement of the error rate by up to 10^2 would suffice. The terms of the question were considered broad enough to include the study of these requirements but a note to the question might be worth attention to this aspect.

The further study of this point in conjunction with point R (Error control in the general telephone network) should be made.

2. *Duplex data transmission for error control*

Special Study Group A requested Study Group I that the note of point 4 of Recommendation F.62 be clarified, so that the adoption of error control systems over the telex network and in particular the information feedback systems in data transmission could be facilitated.

Meanwhile Study Group I confirmed that the use of the telex return path for the purpose of error control in data transmission is not considered to be duplex operation and agreed, therefore, to amend the note in question as follows:

Note. — Duplex telex connections used exclusively for data transmission with the purpose of checking errors should not be considered as " duplex operation " (see Volume II B, Recommendation F.62, § 4, Note).

Question 1/A - point J — Use of telephone-type circuit for the simultaneous transmission of: 1) Data and telegraph signals; 2) Data and speech; 3) Data and facsimile signals

(continuation of point J of the study programme 1964-1968, amended at Mar del Plata, 1968)

(to be studied in liaison with the Joint Working Party LTG)

For the most efficient use to be made of a long-haul point-to-point telephone-type circuit, it is in many cases desirable simultaneously to carry data and telegraph signals. In some cases it may also be desirable to carry data and speech simultaneously, and possibly data and facsimile signals. For this reason the following questions should be studied:

I. What type of frequency and/or time-division multiplexing should be used?

II. What type of modulation(s) should be used?

These questions should be studied for the three applications:

1) Data and telegraph transmission;

2) Data and speech transmission;

3) Data and facsimile transmission.

Note 1. — It is envisaged that standardized systems would be of use to both administrations and to the user, but that they would not in any way restrict the user to such systems. However, the study should include the technical limitations placed upon the user by the characteristics of the circuit.

Note 2. — See Supplements Nos. 7 and 8 of this volume.

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

(continuation of point K of the study programme 1964-1968, amended at Mar del Plata, 1968)

— Continuation of the study of duplex data transmission at 200 bauds, alternating with telephony, bearing in mind Recommendation V.21 as far as the use of either interchange circuit 108/1 or 108/2 for point-to-point non-switched leased telephone circuits is concerned.

— The use of the alarm in the case of multipoint operation is also a subject of further study (refer to the note to point 8 of Recommendation V.21).

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

(continuation of point L of the study programme 1964-1968, amended at Mar del Plata, 1968)

It would be difficult and probably undesirable to establish a rigid test specification, but nevertheless some guiding principles can be given for further 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. *Data signalling rate*

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

Study Group A expressed the opinion that two preferred rates can be stated, namely 1200 bits/s and 600 bits/s, 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 bits/s.

On point-to-point circuits, the preferred rates would be 600, 1200 and 2400 bits/s and tests asked for data signalling rates of 3600, 4800, 7200 and 9600 bits/s.

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 using the 511-bit pseudo-random test pattern standardized in Recommendation V.52 should be carried out on a selection of samples totalling not less than the quantities shown in the following table:

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

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. 9 to 15 of this volume.

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

(continuation of point M of the study programme 1964-1968, amended at Mar del Plata, 1968)

The following points should be examined:

— Response time of certain interchange circuits involved in the modem standardized in Recommendation V.23.

— Study of points arising from the implementation of Recommendation V.23.

Note. — See Supplements Nos. 16, 17 and 18 of this volume.

Question 1/A - point N* — Standardization of data signalling rates on telephone-type circuits

(continuation of point N of the study programme 1964-1968, amended at Mar del Plata, 1968)

1. 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	7200 [1]
9600	

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.

2. *For asynchronous transmissions,*

several points of view were expressed during the discussion of this question:

a) the standardized data-signalling rates with synchronous modulation (200 bits/s, 600/1200 bits/s) constitute maximum modulation rates in start-stop operation, and any lower data-signalling rates can be used with start-stop working;

b) the standardized data-signalling rates for synchronous modulation (200 bits/s, 600/1200 bits/s) are indicated as the preferred values for start-stop modulation rates (perhaps with some other values to be specified);

c) the modulation rates permitted on start-stop networks should be standardized and there should be few of them (200 bauds, 600/1200 bauds).

Corresponding to these major viewpoints, the Study Group considers, as it did already in 1963, that it would be necessary to identify two distinct classes of user:

i) the user of simple data-transmission equipment, who has to transmit to several similar users and hence demands a compatible data-transmission facility to facilitate this free and ready interchange of data. It is likely that equipment of this type could be supplied in accordance with the specifications of the telecommunication authority as part of a truly public data-transmission service;

ii) the operator of a complex data-processing system, whose individual operating points are an integral part of a single system and who has little need to communicate with other points

[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.

outside that system. Such terminal equipment will most probably be supplied by the data-processing manufacturer and the system design will be greatly simplified by a free choice of operating speeds. The use of the public network by such systems simply takes advantage of the lower cost for limited use which it provides and does not really use the interconnection flexibility it offers. It can be regarded as a leased circuit type of operation over the public switched network, and the freedom of choice of operating speeds arises from this similarity.

The characteristics of the data sources and sinks of this network may justify the use of a clearly defined modulation rate; hence, freedom should be allowed to choose the modulation rates for these networks (within the maximum limit fixed).

Question 1/A - point O* — Modem for data signalling rates above 1200 bits per second on the general switched telephone network and 2400 bits per second on telephone-type leased circuits

(continuation of point O of the study programme 1964-1968, amended at Mar del Plata, 1968)

Which data transmission characteristics should be standardized for data signalling rates:

a) in excess of 1200 bits/s on the general switched telephone network?

b) in excess of 2400 bits/s on leased point-to-point and multi-point circuits, e.g. 3600, 4800, 7200, 9600 bits/s?

In considering the relative merits of the above data signalling rates the cost effectiveness of the implementation of each rate should be studied with a view to minimizing the numbers of recommended rates and modems.

For the rates in question consideration should be given to:

— type of modulation;
— synchronization;
— necessary bandwidth;
— possibility of half-speed operation;
— ability to transmit all bit sequences;
— ability to establish synchronism quickly;
— ability to adapt the data signalling rate to line quality.

Note. — See Supplement No. 19 of this volume.

Question 1/A - point Q* — Further study of parallel data transmission for universal use on telephone circuits (see Recommendation V.30)

(continuation of point Q of the study programme 1964-1968, amended at Mar del Plata, 1968)

1. Is there a requirement for a frequency modulated backward channel? If so, can it be used simultaneously with the forward data frequencies?

2.[1] Definitions are required for the proposed new interchange circuits, these are:

a) *At the outstation*

1) Essential interchange circuits

[1] All the numbers of the interchange circuits in 300 series are provisional.

301 Protective ground

302 Data common return

303 Transmitted data (nine or six circuits depending on whether Group B is provided or not).

304 a, b } Activate transmitter

305 a, b } Activate receiver

2) Optional interchange circuits

306 a, b } Received backward signal

307 a, b } Voice answering output (if national regulations permit)

308 a, b } Data set connected

b) *At the instation*

1) Essential interchange circuits

130 Transmit backward tone

a, b } Voice answering input

2) Optional interchange circuit

131 Received character timing.

3. The electrical characteristics of the outstation modem interface need to be defined.

4. There is the need to study the question of the allocation of code combinations to the transmission frequencies (channel numbers). Is there a need to standardize the method of coding or should it be left to the user to allocate codes?

5. Is there a requirement for dibit control of the outstation transmitter and dibit decoding at the instation?

6. The subject of operating procedures needs further study.

7. Characteristics of the Data Signal Quality Detector, when used in parallel systems, requires careful study.

8. The methods of deriving timing require study. Timing can be recovered from the received data channels and from the timing channel when it is used.

Note. — See Supplements Nos. 20 and 21 of this volume.

Question 1/A - point Q *bis — Study of parallel data transmission systems using the push-button telephone signalling frequencies**

(Mar del Plata, 1968)

1. *Definition of the nature of the data transmission signals*

a) Characteristics of the generated signals

— frequency allocation and tolerances for other than push-button telephone frequencies;

— level;
— timing.

b) Transmission channel characteristics
— loss;
— carrier frequency drift;
— attenuation/frequency distortion.

c) Interference characteristics
— voice;
— echoes;
— noise.

2. *Instation terminals*

a) Specify the operation of data communication equipment for the demodulation of signals generated by a push-button telephone dial or a compatible parallel modulator (see section 3 b below)

1) Origination and answering of calls—automatic and/or manual.

2) Optional station arrangements, e.g. compatibility with line hunting or call distributing arrangements.

3) Non-simultaneous backward channels and their characteristics:
— tones;
— voice;
— data channel input;
— other.

4) Signal demodulation and error control:
— sensitivity;
— dynamic range;
— response times for signal detection circuits;
— response times for interference protection circuits;
— error performance.

b) Specify the interchange circuits

1) Functional characteristics of interchange circuits:

a) Definition of forward channel data interchange circuits:
— two times one-out-of-four;
— two times one-out-of-five;
— four lead parallel binary;
— serial.

b) Definition of backward channel interchange circuits.

c) Definition of control interchange circuits.

2) Electrical characteristics of interchange circuits:
— digital circuits;
— voice interchange circuits.

3) Response times of interchange circuits.

3. *Outstation terminals*

a) Characterization of push-button telephone sets and/or dials used for data transmission:
- — hand-operated dials;
- — card-operated dials;
- — repertory dials.

b) Parallel data modulators employing push-button signalling frequencies:
- — two-frequency group systems;
- — three-frequency group systems;
- — amplitude or frequency modulation;
- — use of rest tones;
- — character signalling rates;
- — interface and procedures.

c) Backward channels (simultaneous or non-simultaneous):
- — low-speed channels, e.g. 0-20 bauds;
- — high-speed channels, e.g. V.23 data channel.

ANNEX

(to point Q *bis*)

Data transmission using the push-button telephone signalling frequencies

(Contribution COM Sp.A-No. 200, June 1968, American Telephone and Telegraph Company)

This paper discusses the American Telephone and Telegraph Company's push-button dialling system when it is used for data applications. It covers coding, the keyboard, error performance, the technical aspects of the receiving data set, equipment, interfaces, operational requirements, and typical systems.

Figure 1 shows the frequency plan of the audio-frequency tones utilized in the service[1]. Fundamentally, the eight frequencies are arranged in two groups of four frequencies, one low group and one high group. Each character or digit is represented by two frequencies, one from each group as shown in Figure 1. Twelve such symbols are presently assigned *for use in North America*, leaving four for future use (this as opposed to C.C.I.T.T., which designates six spares).

Many factors were considered in the selection of frequencies. Among these factors were transmission characteristics of a typical telephone line, avoidance of certain combinations of frequencies that occur frequently in speech, and the selection of frequencies not harmonically related.

In dialling service a receiver is needed at the central office to detect the frequency combinations and initiate switching functions. These receivers are not adequate for use in end-to-end signalling. A new data set receiver was developed with added sensitivity, wider bandwidths, echo protection, talk-off protection, answer-back channels, line control functions, and facilities for interfacing with a large variety of business machines.

[1] The frequencies of this plan have been adopted for international use by the C.C.I.T.T. (Recommendation Q.23, *White Book*—Volume VI).

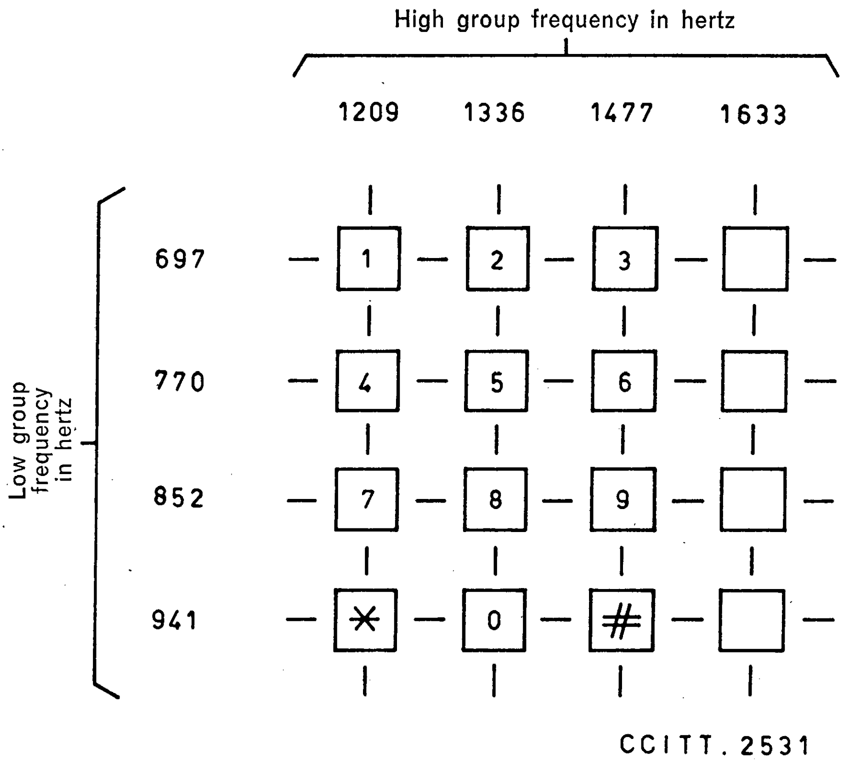


FIGURE 1. — **Frequency assignments**

Description of data set receiver

The receiver responds to multifrequency signals generated by the push-button telephone of 40 ms in duration with an interpulse time of 45 ms and whose total power may vary over a range of 40 dB. The receiver is capable of operating at an overall power level sensitivity of −46.0 dBm. The channel bandwidths of the receiver are ± (0.017 f_0 + 15 Hz), where f_0 is the nominal signalling frequency and the 15 Hz is allowance for carrier offset.

The receiver delivers output indications (data) to the business machine for approximately 37 ms, regardless of input signal duration in excess of 40 ms. A data carrier detector (DCD) indication is delivered 2 to 3 ms after the data indication. The DCD is reset by the disappearance of the input signal or the end of the timed data output whichever occurs last.

The receiver is capable of differentiating between valid signals and speech, noise or echo signals. Worst case echoes are expected with delays of 40 ms at an amplitude 14 dB below the amplitude of the signal.

The data set has three different answer-back channels available to communicate with the transmitting location. These include an input port for pre-recorded voice signals, a three-state oscillator for tone generation, and a high-speed data input port.

Control functions for automatic operation are provided. The data set has a telephone line terminating impedance of 600 or 900 ohms. It has its own protection against lightning surges on the telephone line. It has a return loss of greater than 20 dB within the signalling band of 600 to 2100 Hz.

The data set operates from a 117-V to 60-Hz power source over an ambient temperature range of 4° to 49° C (40° F to 120° F) and relative humidity of 20 to 90%.

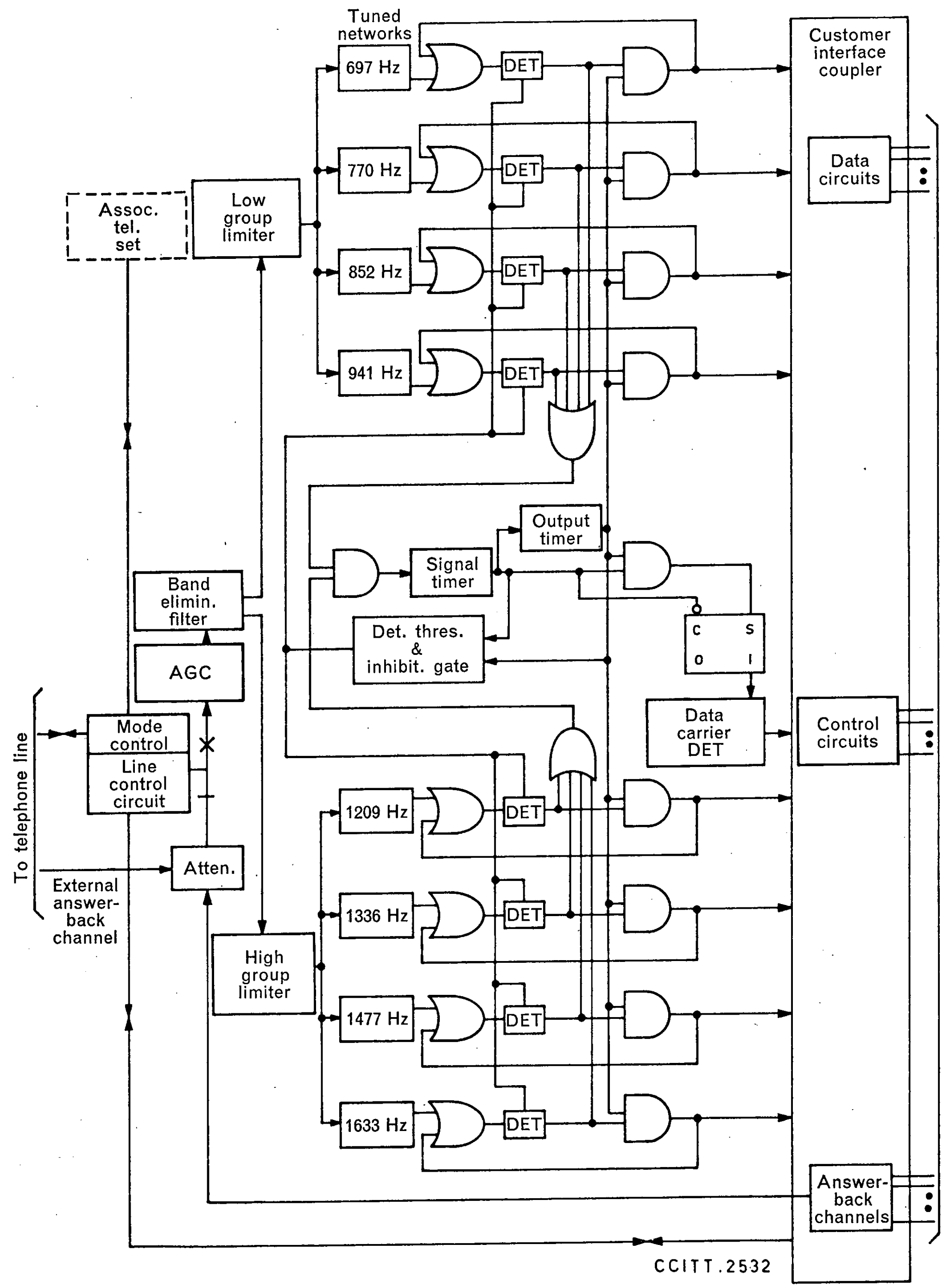


FIGURE 2. — Block diagram of data set

Data set receiver operation

The data set as shown in block diagram form in Figure 2 is divided into two parts, the basic receiver and an interface coupler. The basic receiver may be functionally broken down into three sections: 1) line control circuits, 2) receiver channel, and 3) answer-back channels.

Line control circuit

Figure 3 shows a simplified block diagram of the line control circuit in the basic receiver. The 20-Hz ringing signal from the central office is detected by the ring detection circuit. This signal is rectified and then operates a relay R (not shown). If the customer has signalled the data set on the data terminal ready lead that he is ready to accept incoming calls, the C relay will operate; this permits line current to flow through the input transformer, causing the 20-Hz ringing signal to be removed from the line by the central office. The line holding relay H will be operated by the C relay and hold the C relay operated after the R relay releases.

When the R relay operates, it resets the interval timer. When the relay releases, the timer is started. At the end of a 1.1-second timer interval, the 2025-Hz oscillator is operated. This signals the calling party that the data set has been enabled. The 0.57-second timer resets the SC flip-flop which turns off the 2025-Hz oscillator and operates the data set ready (DSR) relay which indicates that the data set has answered an incoming call and is ready to receive or transmit data. Indication of the original ringing signal is transmitted to the customer during the period that the 20-Hz ringing signal is present.

Receive channel

The data set is placed in the receive mode by an ON condition on the data receive lead which operates the DR relay in Figure 3. When the data receive lead is OFF, the data set is in the answer-back mode. The constituent parts of the receive channel are the AGC amplifier, band-elimination filters, limiters, detectors, signal and output timers, and the data carrier detector (Figure 2).

AGC amplifier

The input signal to the data receiver is amplified to a fixed level by the AGC amplifier. This circuit compensates for transmission loss variations of different connections through the switched telephone network. For an input signal variation of 41 dB, the output varies ± 1.0 dB. The AGC has an attack time of 2 to 4 ms and a long release time. The sensitivity of the receiver is controlled mainly by the AGC and is 0 to -41 dBm. The slow release characteristic of the AGC is used to provide protection against digit simulation due to echo signals.

Band-elimination filter and channel limiters

The band-elimination filter, which is driven by the AGC amplifier, separates the received signals into their respective high and low groups. The signals are then passed to the group channel limiters. Also, any noise received which is outside both the signalling groups will be passed to both group channel limiters.

The limiters supply a constant output square wave, provided the input is above a design threshold. The threshold circuit is used in conjunction with the slow release time characteristic of the AGC to provide protection against duplication of input information caused by echo signals. In addition, through the instantaneous limiter action, some immunity is provided against digit simulation by voice signals. Equalization is also provided in the limiters to compensate for the uneven attenuation characteristic of the telephone plant.

Channel detectors, signal and output timers, and data carrier detector

The output of each channel limiter goes to a group of four series-tuned networks used for recognition of the signal frequencies. Each tuned network output connects to a detector whose operating threshold is approximately 2.5 dB below the peak output from the tuned networks.

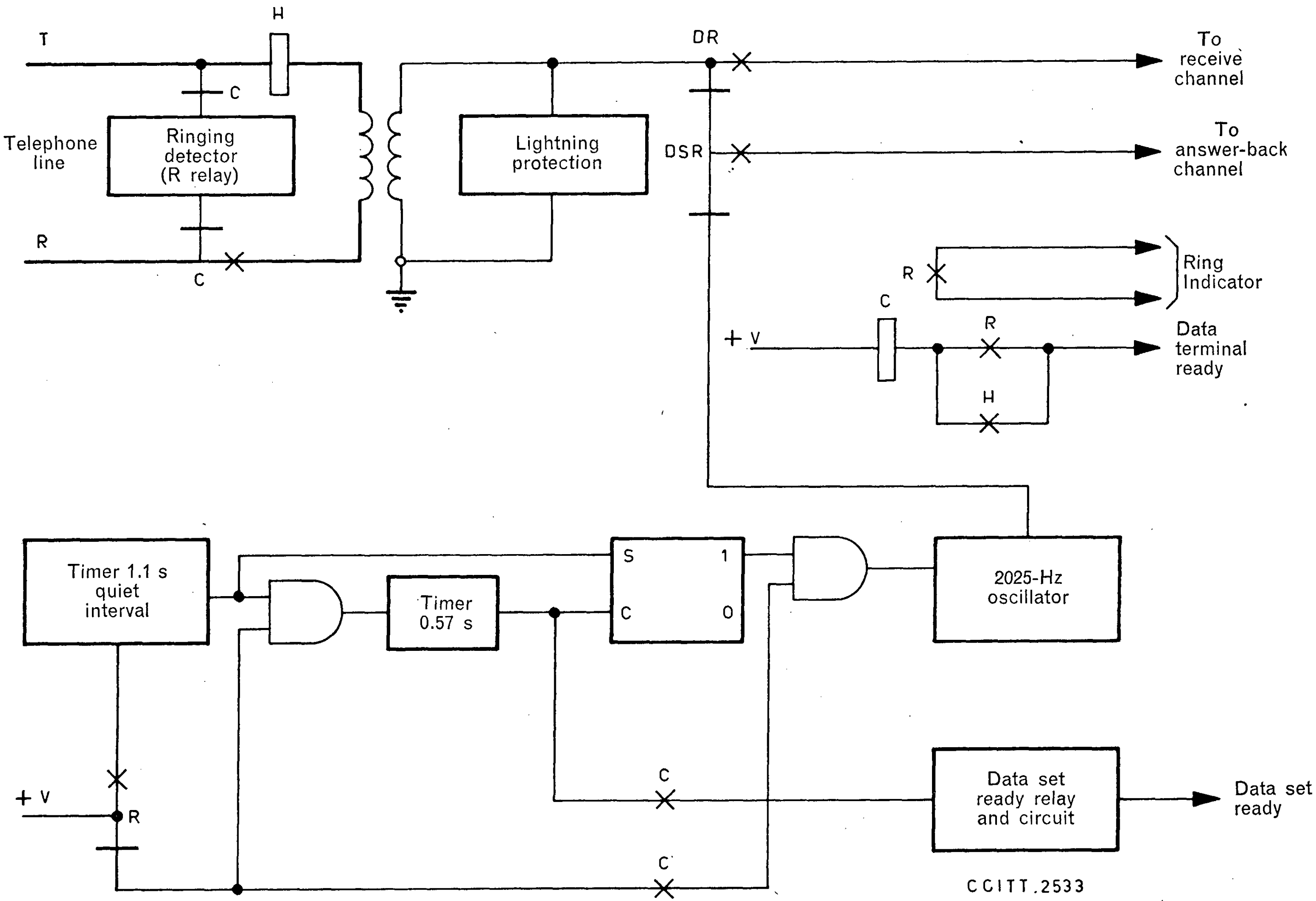


FIGURE 3. — **Block diagram of line control circuit**

This threshold ensures that one and only one detector in each group will operate. Once the detector operates, it will remain operated for as long as the signal is present at the detector. An AND gate is used to verify that one detector is operated in each group. This is verified for a timed interval of 40 ms. Next, the output timer is turned on, which in turn operates and holds the correct output driver in each group for 37 ± 2 ms. The data carrier detector is operated 2 to 3 ms after the data output driver, with an indication passed through the interface coupler. The data carrier detector will be reset by the signal timer at the end of its timing transition or by the ending input data signal, whichever occurs last. Feedback from the output drivers to the operated detectors causes the detectors to remain operated should the input data signal disappear during the 37 ms output period. To prevent the operation of the other detectors, the threshold level is raised 1 dB above the tuned network outputs.

Answer-back channels

Communication from the answer-back channel may take the form of pre-recorded voice signals, tone signalling, or high-speed data exchange. For the exchange of high-speed data, the calling station must have a data set capable of communicating at the speeds in question in addition to the telephone set.

A voice frequency input port is available to the customer to couple pre-recorded or machine-generated voice signals to the telephone line. These signals are limited before they are passed to the telephone line. When a signal is 10 dB above the recommended transmission level of 0 dBm from the connecting business machine, it is peak limited. The input impedance to the port is 600 ohms balanced. Figure 4 shows the input power to the data set from the connecting business machine to the maximum output power level to the telephone line. This curve may be shifted down 3 or 6 dB to match allowable transmission levels.

The answer-back tone transmitter will transmit three different frequencies by manipulation of interface leads A and B as shown:

Leads	Transmitted frequencies (Hz)
AB′	1017
.AB	1785
A′B	2025

Operation of the answer-back transmitter leads pre-empts the voice answer-back channel. The answer-back transmitter may be operated in a frequency shift keying (FSK) mode at a rate of 40 bauds. The answer-back attenuator in Figure 2 is adjusted during installation to ensure that the answer-back signals enter the connecting central office at acceptable level.

Interface

Three interface couplers have been designed and are subsequently described.

Relay contact interface coupler

Each detector in the relay contact interface (Figure 5) causes an output driver to operate a dry-reed type relay. The output to the connecting business machine is a two-out-of-eight code (one relay contact closure in each group) for data indications and a data carrier detector closure to signal ground for sampling and timing indications. Each of the data contacts is protected by a series RC network of 470 ohms and 0.1 microfarad.

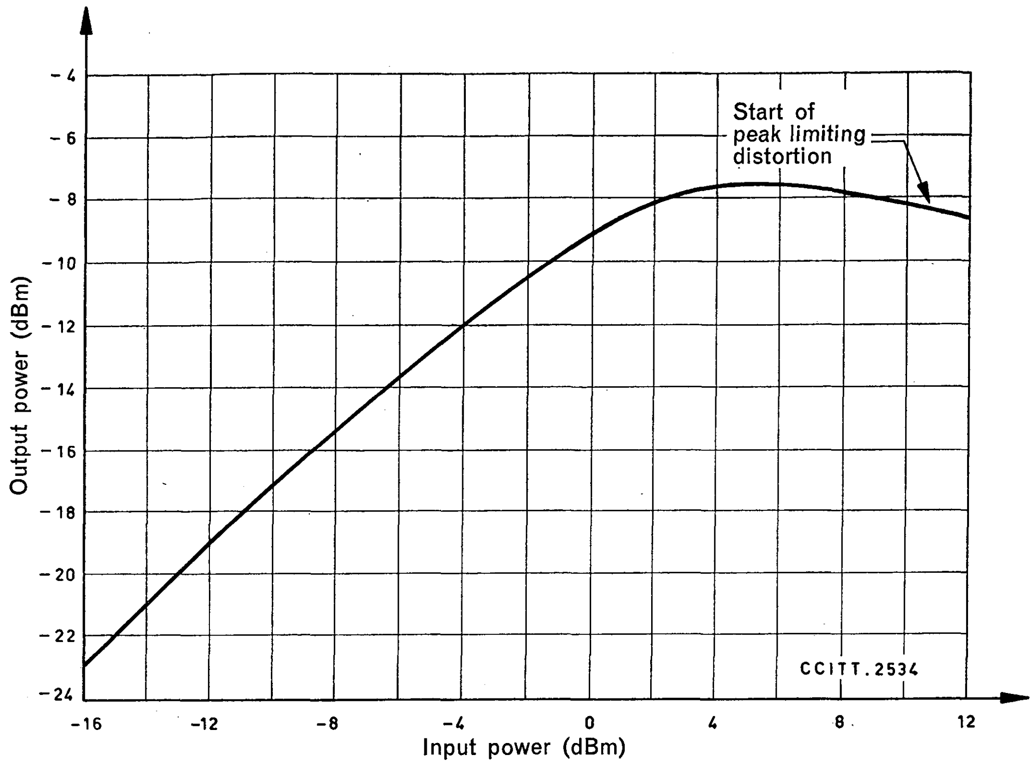


FIGURE 4. — **Power transfer characteristics of voice answer-back limiter**

The following control functions are contained in this coupler:

1) Ring indicator: a ground indication is transmitted to the connecting business machine whenever 20-Hz ringing voltage appears at the input to the data set.

2) Data set ready: a ground indication is passed to the connecting business machine whenever the data set has answered a call and is ready to receive or transmit data.

3) Data terminal ready: a ground indication is received by the data set indicating that the business machine is ready to receive data calls.

4) Attendant: a ground indication is received by the data set from the business machine when a data call should be intercepted by an attendant at the receiving location.

5) Out-of-service: the presence or absence (installer option) of a ground indication on this lead will make the data set appear busy to incoming calls.

6) Data receive: the connecting business machine transmits a ground indication on this lead to transfer the data set to the receive mode or removes the ground indication to transfer the data set to the answer-back mode.

Binary coded matrix voltage interface coupler

To eliminate costly and complex translating equipment between the data set and a central processor, a coupler with a voltage interface using a special four-level binary code was designed.

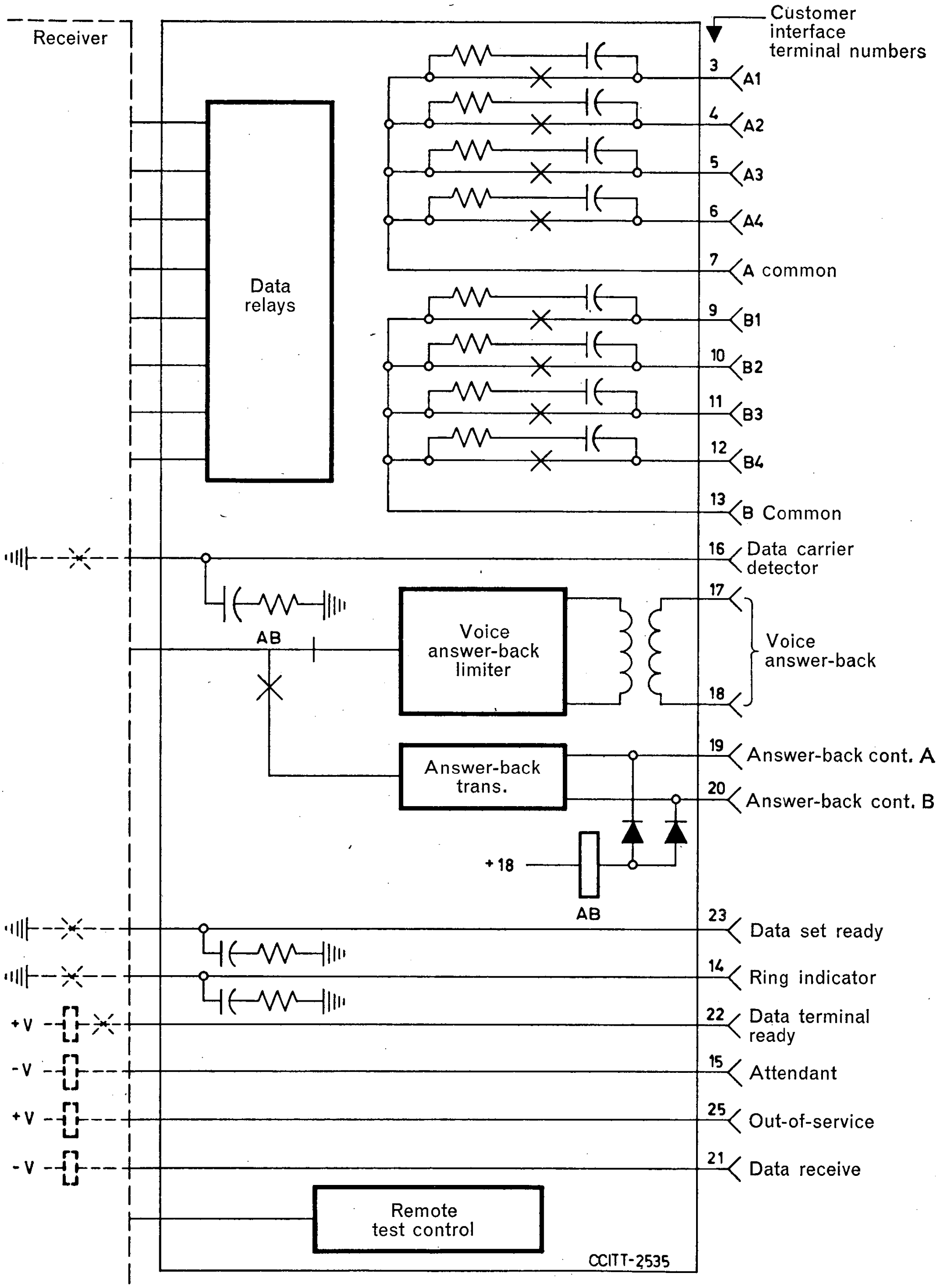


FIGURE 5. — **Customer interface coupler—Relay contact interface**

For simplicity, this code will be referred to as a BCM (binary coded matrix) code. This code resulted from considering the push-button dial matrix and assigning a code to the matrix in such a way as to minimize the cost of translation and the cost of connecting processing equipment. This code assignment is shown in Figure 7.

The data carrier detector function in the data set becomes very important when using this code. The data carrier detector indication is transmitted after the data indications and may be used to sample the data leads permitting the detection of 0000.

A block diagram of the BCM coupler is shown in Figure 6. It contains voltage drivers for the control and data functions, the voice answer-back limiter, the answer-back transmitter, and remote test circuitry. The interface is voltage controlled (except for the voice answer-back channel which is a 600-ohm balance input). The interface functions available, along with terminal assignments, are shown in Figure 8. These functions, except functions RD1, RD2, RD3 and RD4 are the same as those explained in the previous section except that ground indications are replaced by the appropriate voltage indications as shown in Figure 8. Electrical characteristics of the interface leads conform substantially to C.C.I.T.T. Recommendation V.24, section V.

ASCII[1] *voltage interface coupler*

This interface (Figure 9) is intended for use with terminal equipment accepting serial data input and having minimal logic capabilities. Characters representing the digits 0-9 are coded by the receiver as the corresponding ASCII characters 0-9. The frequencies corresponding to the 11th and 12th buttons (* and # keytops, respectively) are coded by the installer to be any character in the first four columns of the ASCII code. The interface unit is capable of detecting a sequence of two successive depressions of the twelfth button. The translation of the first and second depressions can be different. Again, the coding for the second depression can be any code in the first four columns of the ASCII code. If desired, the unit can be arranged to disconnect the call when two consecutive receptions of the twelfth button frequencies are detected. Also, if desired, the unit can be arranged to provide automatic disconnect after a 15-second or 45-second period of inactivity. To ensure message separation, the installer can arrange the interface unit to generate automatically the ASCII character selected for the second depression of the 12th key at the beginning of each call. The proper even-parity bit is added to each 7-bit ASCII code to form an 8-bit character.

Echo protection

The telephone transmission plant, like many other transmission media, contains many discontinuities which manifest themselves as generating echo signals. These echo signals (called listener echoes in this application) directly affect the performance of the data set since they may appear as legitimate data signals or mask successive data signals. Hence, the requirement is imposed on the data set to ignore echo signals and to pass on to the connecting business machine only those signals it believes to be legitimate data information. This discrimination is performed by the AGC amplifier and the channel limiters.

Digit simulation

The digit simulation (talk-off) performance of the data set is affected by many variables: exposure time, noise, type of talker (male or female), characteristics of the transmission channel, etc. The more important variable is exposure time; the shorter the exposure time, the lower the probability of error per call due to digit simulations.

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[1] American Standard Code for Information Interchange: Equivalent Reference C.C.I.T.T. *Blue Book*, Volume VIII, Supplement No. 2, November 1964.

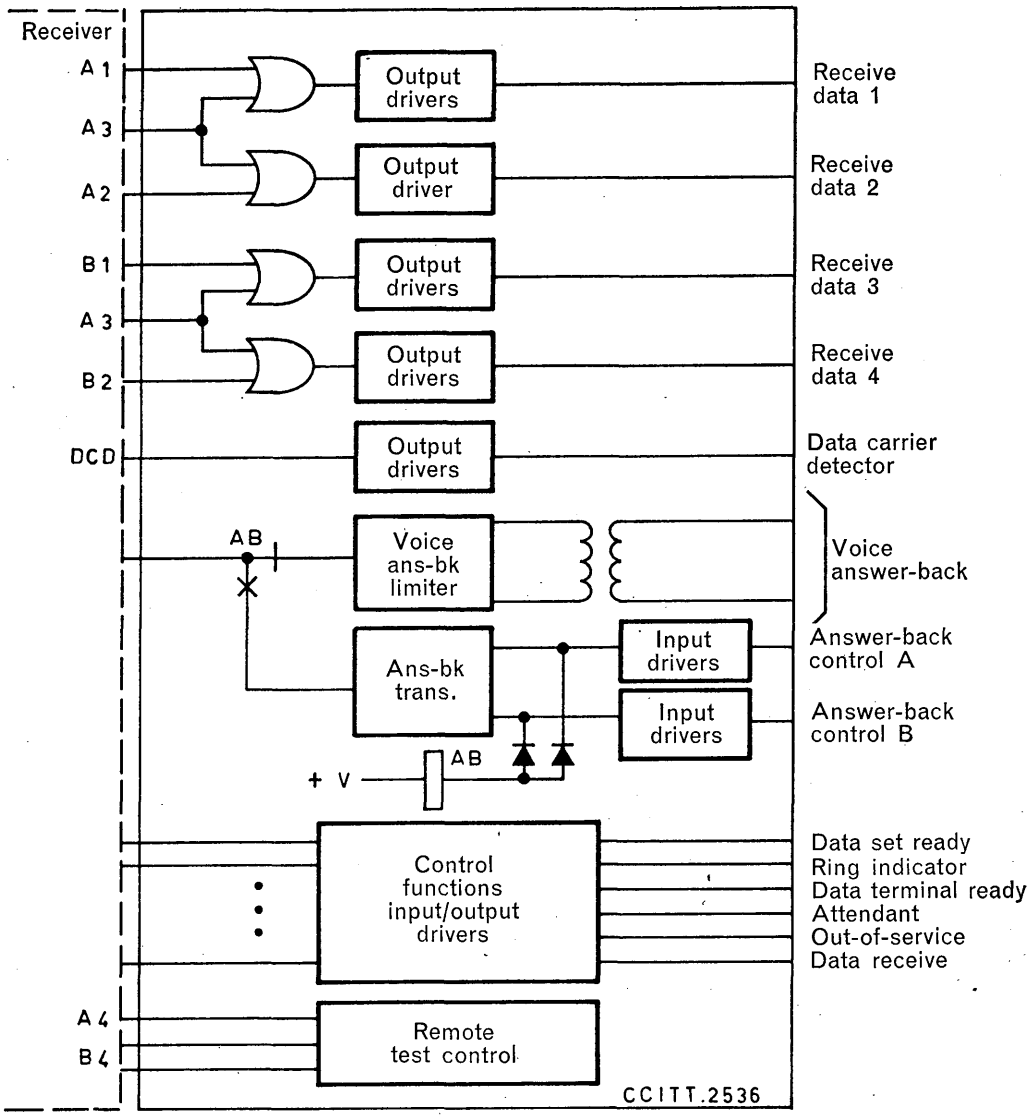
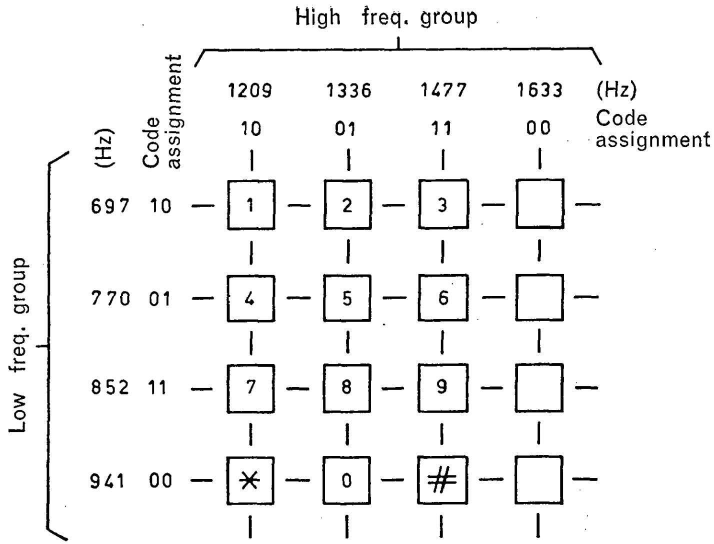


FIGURE 6. — Customer interface coupler—Binary coded matrix voltage interface



(a)

B.C.M. code table

Digit	Receive data leads			
	RD1	RD2	RD3	RD0
1	1	0	1	0
2	1	0	0	1
3	1	0	1	1
4	0	1	1	0
5	0	1	0	1
6	0	1	1	1
7	1	1	1	0
8	1	1	0	1
9	1	1	1	1
0	0	0	0	1
*	0	0	1	0
#	0	0	1	1
Unassigned	1	0	0	0
	0	1	0	0
	1	1	0	0
	0	0	0	0

(b) CCITT.2537

FIGURE 7. — a) Arrangement of dial showing code assignment of the BCM voltage interface coupler

b) BCM code table

Type of circuit	Function	Terminal number	Interface terms		
Data	Receive data 1	3	*Binary state:*	one	zero
	Receive data 2	4	*Signal condition:*	marking	spacing
	Receive data 3	5	*Voltage:*	negative	positive
	Receive data 4	6	*Paper tape:*	hole	no hole
	Data carrier detector	16			
Control	Ring indicator	14			
	Data set ready	23			
	Data receive	21	*Control function:*	on	off
	Data terminal ready	22			
	Out-of-service	25	*Voltage:*	positive	negative
	Attendant	15			
	Answer-back control A	19			
	Answer-back control B	20			
Ground	Signal ground	24	See EIA standard RS-232-B		
	Protective ground	1			
	Voice answer-back A	17	600-ohm balanced pair		
	Voice answer-back B	18	Maximum input level 0 dBm		

FIGURE 8. — **Binary coded matrix (BCM) voltage interface**

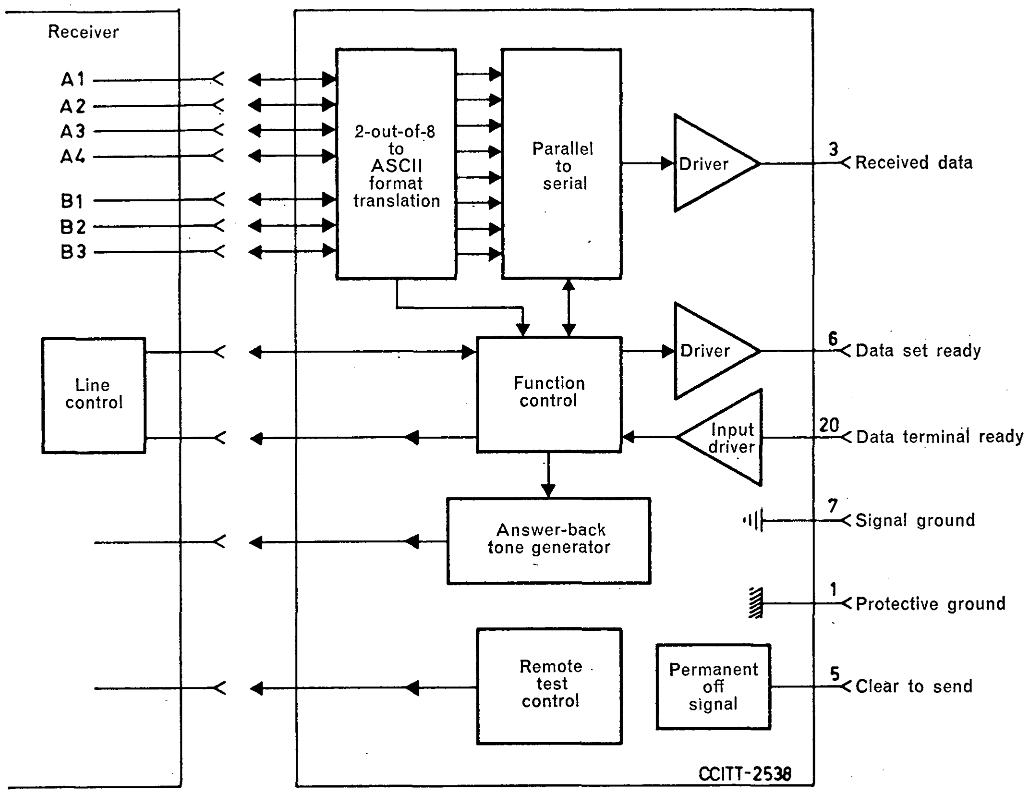


FIGURE 9. — **Customer interface coupler-serial, ASCII voltage interface**

The receiver was tested by exposing it to continuous speech and noise from 40 000 telephone connections for a period exceeding 1600 hours. The distribution of digit simulations among the 16 total available combinations is shown in Figure 10. This figure thus shows the conditional probability of simulation of each of the 16 digits, given that a digit simulation has occurred.

What error rate due to speech and noise may one expect when using the receiver in a system with push-button type telephone sets? A sample of users of this system indicates that the input information will be in blocks of 15 characters or less and will be entered manually. The time to enter 15 characters varies with the user, the grouping of the characters, the environment of the user, etc. Tests indicate that the receiver will have an error rate of one digit simulation per 12.97 hours of exposure time for the 10-decimal digits, one digit simulation per 9.51 hours of exposure for the 10-decimal digits plus * and #, and one digit simulation per 8.56 hours of exposure for the complete 16 digits.

The data set is vulnerable to being talked off only when a call has been answered, and the data receiver is connected to the telephone line. This time normally starts at the end of the 2025-Hz tone which is transmitted to the calling station and ends when the customer leaves the data receive mode. This time will be referred to as access time.

The real exposure time of the receiver is:

$$E(t) = A(t) - [ND \cdot C(t)]$$

where

$E(t)$ = exposure time

$A(t)$ = access time

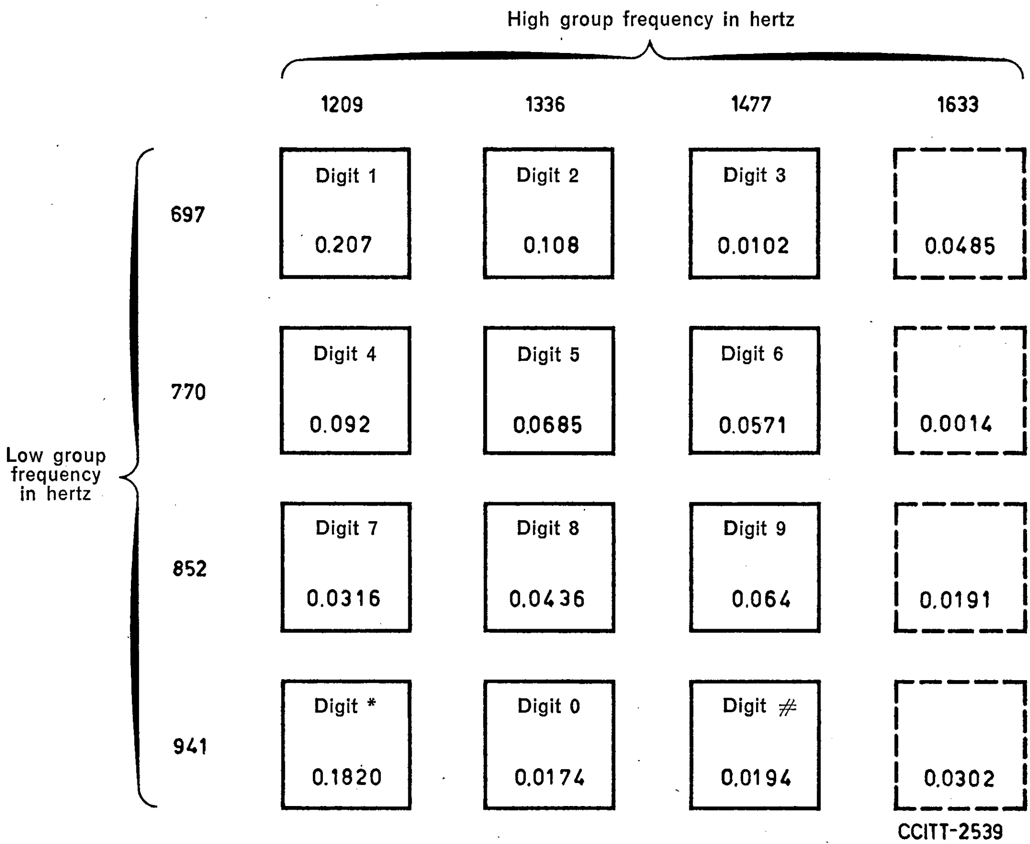


FIGURE 10. — **Probability of simulation of frequency pairs due to noise coupled through the telephone handset for data set 403D**

ND = number of digits dialled

$C(t)$ = average time per digit the carbon microphone is disabled in the telephone handset.

Consider an example which shows the access time $A(t)$ for a 15-digit number to be 12 seconds. From the tests performed, the mean time to dial a digit has been found to be 180 ms with a standard deviation of 115 ms. The carbon microphone in the telephone handset is disabled approximately 10 ms before the tones are transmitted to the telephone line and is enabled 10 ms after the tones have ended. Therefore, the mean time the carbon microphone is disabled is 200 ms.

The exposure time is then:

$$E(t) = 12 - (15 \cdot 0.2),$$

$$E(t) = 9.0 \text{ seconds},$$

and the calls per digit simulation *(ds)*, or error rate *ER*, is then:

$$ER = 12.97 \, \frac{\text{hours}}{ds} \cdot \frac{3600 \text{ seconds}}{\text{hour}} \cdot \frac{1 \text{ call}}{9 \text{ seconds}}$$

$$ER = 5188 \text{ calls}/ds$$

for the 10-decimal digits.

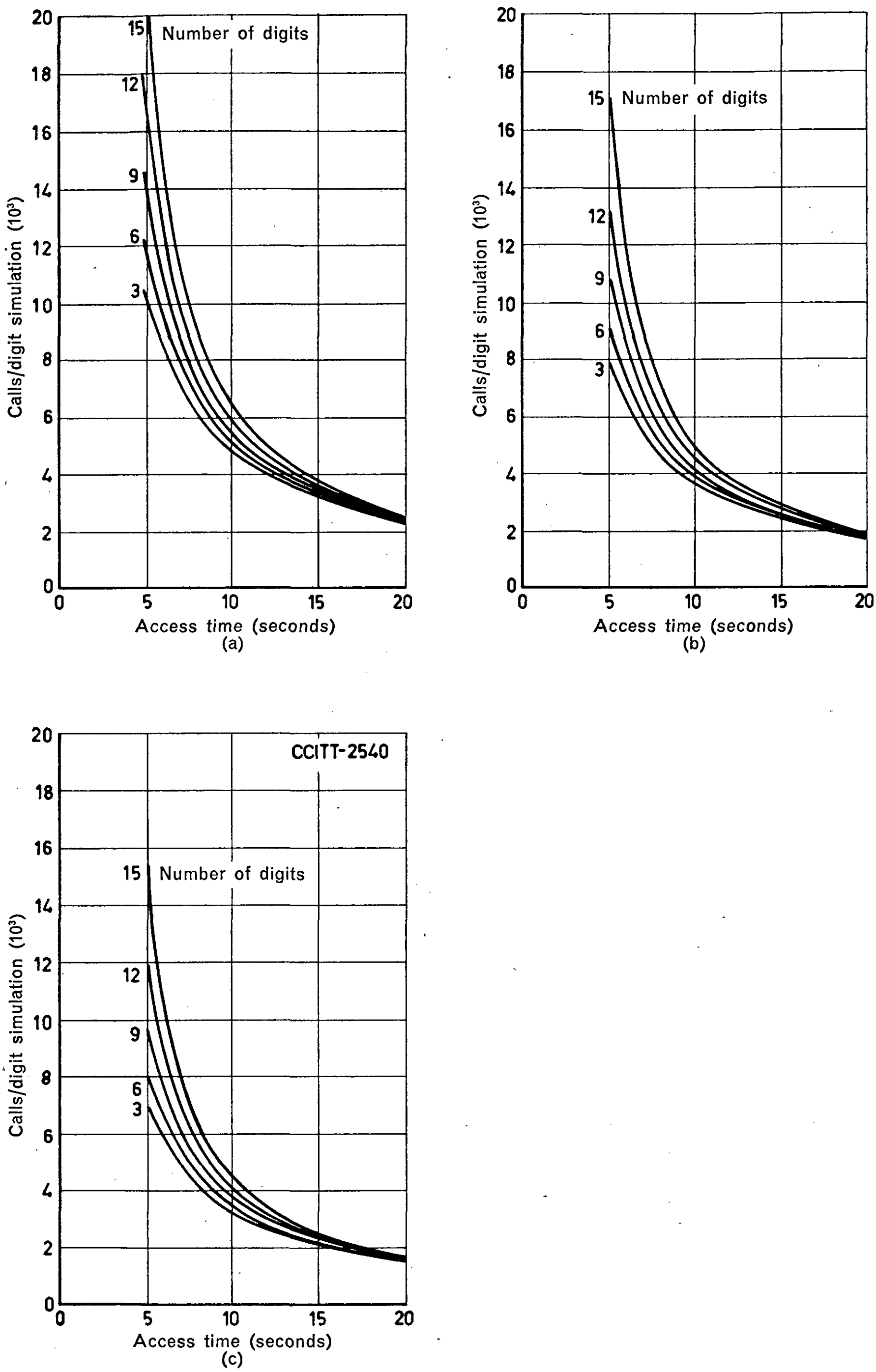


FIGURE 11. — **Performance of data set 403D for hand-entered digits when a) the 10-decimal digits, b) the 10-decimal digits plus * and #, and c) the full 16 digits are sampled**

Since the digital simulation rate is a function of the access time and of the number of digits (total pulse time), a family of curves may be constructed to determine the error rate for any particular system. Using the previous information of error rate and also the mean time $C(t)$ that the carbon microphone is disabled in the telephone handset, the curves in Figure 11 are constructed for hand-pulsed information only. These curves may be adjusted for different values of $C(t)$. For example, suppose we have an application in which the access time is 12 seconds and the number of manually entered digits is 9; a performance rate of 4500 calls per digit simulation is read from the ordinate axis of Figure 11 a.

These curves apply only for systems where there is one inquiry. For multiple access calls these curves must be adjusted accordingly.

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

(continuation of point R of the study programme 1964-1968, amended at Mar del Plata, 1968)

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?

The following points should be examined:

i) character by character error-control system;

ii) small data block error-control system;

iii) efficiency of data signal quality detector;

iv) error control system based on data signal quality detector.

Note. — See Supplements Nos. 6 and 22 to 27 of this volume.

Question 1/A - point S + T — Measurement of phase distortion and transmission loss between subscribers

(continuation of points S and T in the study programme 1964-1968, amended at Mar del Plata, 1968)
(Study results to be reported to Study Groups IV and XIV)

1. *Phase distortion*

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.

2. *Transmission 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 Hz), 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. 14 and 28 to 35 of this volume.

ANNEX

(to point S)

Extract from the report by Special Study Group A, October 1968

1. Special Study Group A took note of a report by Study Group XVI (Geneva Meeting, June 1966) on characteristics of world-wide connections with regard to data transmission (see Supplement No. 31).

Study Group XVI will continue to collect further information from Special Study Group A to amplify its report. Special Study Group A considered that this document contains valuable information and more detailed information on group delay characteristics and quality of the connections would be welcome.

2. The measuring of group delay distortion has proved to be necessary for the assessment of the quality of data transmission lines and for equalizing measures which may be required. Particularly in international maintenance activities, it is possible that the generator and the receiver come from different manufacturers. Therefore it appears to be necessary to specify fundamental characteristics which are required for the actual measurement problem and for the interworking of devices from different manufacturers.

For that reason Special Study Group A proposed the following studies to be made urgently:

1) Which method of measuring group delay distortion shall be proposed?

2) Which synchronizing method shall be used between generator and receiver?

3) Which modulating frequencies shall be used in the various frequency ranges?

4) For which overall frequency range and for which switchable subranges shall the device be designed?

5) Which transmitting and receiving level ranges shall be provided?

6) Which output and input impedances shall be selectable?

7) Which indicator shall be used?

8) Which measuring error shall be permitted in the various frequency ranges?

9) How shall tone signalling receivers, which possibly may be included in voice channels, be prevented from responding and releasing the connection? Would it be necessary to take account of pilot frequencies which could be included in larger transmission bands?

10) Which frequency sweep-rates shall be proposed in the various frequency ranges?

The following comments on these ten points by the Federal Republic of Germany were annexed to this questionnaire:

To 1

For measuring group delay distortion the Nyquist method, which is generally accepted, should be used. As method of modulation both amplitude modulation and frequency modulation are possible.

Since there are no decisive advantages or disadvantages that speak in favour of or against one of the two methods, the German Administration is willing to agree to each method which is proposed by the majority of the Study Group members.

For delay distortion equalization and for supervisory measurements (survey measurements) the sweep method is recommended because of its speed and its greater information capacity; point-by-point measurements are not to be excluded.

To 2

Synchronizing between generator and receiver shall take place without auxiliary leads; that is, in this case the receiver must be synchronized exclusively with the aid of the incoming signal. It must be specified whether the incoming signal shall contain special synchronizing signals; if yes—which?

To 3

The modulation frequency in the various frequency ranges must be in accordance with the required measuring accuracy and the resolving power which is necessary for the operation; in this case the modulation frequency should not be in any relation to the customary mains frequencies which can be expressed by an integral number.

To 4

As overall frequency range we propose:

200 Hz to 600 kHz

For reasons of practicability it should be divided in the following subranges:

Voice channels:	200 Hz to 5 kHz
Programme circuits:	200 Hz to 15 kHz
Broadband carrier systems:	10 kHz to 556 kHz

the latter possibly once more divided in subranges for

Basic primary groups 60 kHz to 108 kHz and
Basic secondary groups 312 kHz to 552 kHz.

To 5

The range for transmitting and receiving levels should be between -35 and $+10$ dBm; the dynamic receiving level range should in this case be at least 30 dB.

To 6

The output and input impedances should be adjustable optionally to 600 ohms, 150 ohms and 75 ohms. A position $R_i \sim 0$ ohm for the generator and a high-impedance input position for the receiver may be favourable. For the frequency range between 12 kHz and 552 kHz a balance to earth > 60 dB appears to be sufficient.

In the case of a connection to the public telephone network, provisions must be made to secure that the customary connecting conditions of the subscriber's loop are not impaired.

To 7

It has proved to be practical to record the measuring result optionally by means of an electron-beam oscilloscope with sufficient afterglow duration or by means of a pen recorder.

To 8

The following measurement accuracy is to be required:

for the range 200 Hz to 5000 Hz		10 to 25 microseconds
5 kHz to	12 kHz	3 to 10 microseconds
12 kHz to	120 kHz	~1 microsecond
120 kHz to	552 kHz	~0.1 microsecond

To 9

The tone signalling receivers in the voice channels can—during the measuring procedure—be prevented from responding and thus from releasing the connection to be measured by frequency blockings in the generator device which suppress the emission of the respective tone signalling frequency. It is also possible that by transmitting a frequency (blocking tone) in the voice-frequency trap the responding of the tone signalling receiver during simultaneous reception of the tone signalling frequency is prevented.

An objection to the use of frequency blocking devices is the fact that in (particularly international) switched connections so many tone signalling frequency ranges have to be suppressed that there will be no usable measurement result. There may indeed be the possibility of modulation products when blocking tones are used, but because of their small size they do not have any remarkable influence on the result of measurements on circuits using tone signalling receivers.

To 10

The sweep rate must be—independently of the frequency range chosen—adjustable in a wide range (~0.05 Hz to 10 Hz) in steps or continuously.

3. Meanwhile, Study Group IV requested Special Study Group A for further information on this subject.

The questions were as follows:

a) Is Special Study Group A in a position to confirm the values fixing the accuracy of the group delay measuring equipment, as specified in the annex to the questionnaire?

b) What should the accuracy be measuring ripples in group delay characteristics, such as those which may appear on equalized group of supergroup links?

Special Study Group A then formed the following conclusion on these questions:

Special Study Group A considered the request for information from Study Group IV concerning certain complementary specifications necessary to standardize measuring equipment to gauge the group delay distortion. As a result of this study, Special Study Group A is able to give the following reply:

1) Special Study Group A considers that there is an urgent need to standardize a group delay distortion measuring instrument.

2) The need to standardize a new instrument does not arise from the technical inadequacy of the equipment used at present, but from the fact that the existing types of instruments apply different standards, leading to operating incompatibility between the equipments supplied by different manufacturers.

3) Special Study Group A confirms the measurement accuracy values laid down in the annex to the questionnaire. This applies to all the frequency ranges indicated.

The accuracies required are necessary because they correspond to the group delay distortions generally met with in the various frequency ranges. However, Special Study Group A recognizes that these values are desirable for the measuring equipment itself, but that the actual accuracy may be affected by unfavourable line conditions, especially by noise on the circuit measured.

4) Special Study Group A is not in a position, for the time being, to fix the accuracy with which ripples in group delay characteristics should be measured, since the studies of the effect of ripple characteristics (amplitude and periodicity) on data transmission quality are not yet sufficiently advanced. This study will be pursued by Special Study Group A. The latter considers however, as a first approximation, that the information given in the annex to the questionnaire on the measurement accuracy for the various frequency ranges will provide sufficient accuracy for the measurement of ripple. The problem of ripple measurement should not therefore be an obstacle to the rapid standardization of a measuring instrument.

5) The problem of studying ripples could be solved with the aid of a measuring instrument using several modulation frequencies for each measurement frequency range. However, this solution would lead to an increase in the cost of measuring equipment and should be reserved for laboratory purposes.

6) Special Study Group A considers that the standardized equipment for measuring group delay distortion should be, first and foremost, an equipment for carrying out maintenance measurements, and that the problem of ripple measurement accuracy be considered a secondary matter, at least for the time being.

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.

(continuation of point U of the study programme 1964-1968)

(of concern to Study Group IV. Question to be studied in liaison with the LTG Joint Working Party)

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, *White Book*): G.101, G.123, G.124, G.125, G.132, G.133, G.141, G.151, G.152, G.153.

Note 3. — The study of characteristics of an impulsive-noise-measuring instrument for wideband data transmission is the subject of Question 7/C of Special Study Group C.

Note 4. — See Supplements Nos. 33 and 35 to 39 of this volume.

ANNEX

(to point U)

EXTRACT FROM THE REPORT BY SPECIAL STUDY GROUP A, OCTOBER 1968

The study of point U should be continued with reference to data transmission at modulation rates above 1200 bauds. For data transmission using the modems described in Recommendations V.21 and V.23, on the other hand, individual replies may be submitted:

a) With regard to impulsive noise on leased circuits, the adoption of Recommendation V.55 on the measuring instrument and of a recommendation specifying the limit for the number of impulsive noise counts (70 per hour) proposed in the annex to the Recommendation would constitute a reply to the question.

b) With regard to the attenuation/frequency characteristics and delay/frequency distortion, the Study Group noted the test results reported by the United Kingdom Administration (see Supplement No. 38).

The studies cannot be considered as completed as, on the one hand, Recommendations M.58 and M.102 are now being revised by Study Group IV and, on the other hand, Study Group IV has just asked that studies for the measurement of group delay distortion should be undertaken (Question 10/IV).

It should be noted that the information requested related to point-to-point leased circuits.

c) The Study Group noted that the C.C.I.R. was studying a question (Question 12/III) on distortion characteristics required for single-sideband and independent-sideband systems used for high-speed data transmission over h.f. radio circuits. The co-operation of Special Study Group A to the study of C.C.I.R. Question 12/III would be requested.

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

(continuation of point V of the study programme 1964-1968)

(of concern to Study Group IV, Question 8/IV, and Study Group XI, Question 8/XI)

Note 1. — See point 3 of the annex to Recommendation V.55.

Note 2. — The study will have to include data transmission with or without error control.

Note 3. — See Supplements Nos. 14, 34, 37 and 40 of this volume.

Question 1/A - point W* — Maintenance methods

(continuation of point W of the study programme 1964-1968, amended at Mar del Plata, 1968)

(study results to be reported to Study Group IV)

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

In particular, supplementary study of Recommendations V.50, V.51, V.52, V.53 and V.55 should be continued.

Note. — See Supplements Nos. 14, 34, 35, 37, 40 and 41 of this volume.

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

(continuation of point Y of the study programme 1964-1968, amended at Mar del Plata, 1968)

The following points should be considered:

a) Automatic calling equipment (ACE) for multiline operation.

Bearing in mind that the ACE is only required during the call establishment period and is idle during the information transfer period, the additional hardware for interfacing the ACE serves for only a small percentage of the time if it is used on a one ACE to one line basis.

The sharing of ACEs among a group of lines in both the telegraph and telephone cases should therefore be examined.

b) Simplified version of automatic calling on telephone networks.

Should a simplified version of automatic calling on telephone circuits be standardized?

If so, what would be:

1. the functional characteristics of the interchange circuits?

2. the interface procedure at the call originating station?

Note 1. — For this study, refer to Recommendation V.25 (Automatic calling and/or answering on the general switched telephone network).

Note 2. — See Supplement No. 42 of this volume.

Question 1/A - point Z* — Data transmission over 48-kHz and 240-kHz circuits

(continuation of point Z of the study programme 1964-1968, amended at Mar del Plata, 1968)

1. Study of points arising from the implementation of Recommendation V.35 (modems for 48 kilobits/s).

2. Further study of modems for the 48-kHz band including specification of characteristics and line signals for:

a) 48 kilobits/s transmission;

b) data signalling rates above 48 kilobits/s.

The study to take account of digital techniques (see Annex 1) and multi-condition modulation (see Annex 2) which give information on a 72-kilobits/s modem.

3. Study of characteristics to be standardized for data transmission over 240-kHz channels.

ANNEX 1

(to point Z)

Comments on Recommendation V.35

(Extract from the contribution by the Netherlands Administration; Mar del Plata, September 1968)

1. *General considerations*

In general the Netherlands Administration is in favour of recommending a line signal for 48 kilobits/s data transmission, as far as this may facilitate the exchange of data across the national frontiers. In establishing such recommendations, however, one should try to leave open as many possibilities of implementation as possible, at the sending side as well as at the receiving side, in order to be able to choose the most economical solution depending on the circumstances and on the use made of the transmission link. This point is extremely important, especially in view of the fast developing digital techniques.

In this light it seems to us that the wording of sections 4 and 5 of Recommendation V.35 had better be changed to the extent that only the line signal characteristics are described, without reference being made to a particular method of implementation. As such, the modulation technique should not be prescribed explicitly, as the same line signal can be realized with different methods. For example, with the synchronous mode of operation a line signal with the same characteristics as a vestigial sideband signal can be made by:

a.1) Vestigial sideband modulation with suppressed carrier.

b.1) Splitting the signal into two sequences, which are modulated in quadrature double sideband technique with suppressed carrier.

c.1) Using duobinary frequency modulation, combined with proper encoding (Bennett and Davey, *Data Transmission*, McGraw-Hill 1965, section 11.6).

These signals, no matter in which way they have been realized, can all be detected by the use of one of the following demodulation methods, combined with proper decoding:

a.2) Coherent demodulation with the (virtual) carrier frequency at the edge of the frequency band;

b.2) Coherent demodulation with a frequency at the centre of the band (quadrature demodulation, resulting in two sequences which have to be interleaved);

c.2) *Duobinary f.m.-detection*

The equivalence of the line signals for the cases a), b) and c) with the synchronous mode of operation disappears in the asynchronous mode of operation. However, asynchronous operation is still possible, with minor changes, in the combinations a.1)-a.2), b.1)-b.2) and c.1)-c.2). In the synchronous mode of operation especially the methods b) and c) offer extra possibilities compared with the vestigial sideband mode of operation: in the case of b.1) at the sending side and b.2) at the receiving side the use of the system as two separate 24 kbits/s channels is easily achieved; method c.2) at the receiving side is of interest for those cases where transmission quality gives rise to no problems, in which case a cheap duobinary f.m.-receiver can give quite a satisfactory performance.

The use of a certain method of modulation for 48 kilobits/s synchronous data transmission does not imply that the corresponding demodulation method has to be used. However, duobinary f.m.-detection is not consistent with the use of a carrier pilot nor is it consistent with a frequency gap around the carrier frequency as proposed in Recommendation V.35. In section 4 of this Recommendation it is stated that a pilot carrier will be necessary to permit homochronous demodulation. However, it is clear that the above-mentioned duobinary f.m.-demodulation does not require transmission of a pilot for the demodulation purpose. Another method of detection could be homochronous demodulation with the centre frequency of the signal (e.g. 84 kHz), which may be recovered in a similar way to the reference phase for a four-phase signal demodulator. This method has as an additional advantage that due to the position of the centre-frequency the influence of delay distortion is small compared with the vestigial sideband mode of operation where the virtual carrier lies at the edge of the group band.

The quadrature method of operation can have the same performance as a vestigial sideband system as far as signal-to-noise ratio is concerned (Bennett and Davey, section 11.1, table 11-1 at the bottom).

Still another method of operation is given in Bennett and Davey, section 13.2. In this section a carrier recovery method for a vestigial sideband system, permitting the transmission of the complete spectrum, is described. This system, using a carrier pilot in quadrature with the suppressed carrier, is given some preference over the method of carrier recovery with the aid of a frequency gap. The latter method also requires dc-restoration, especially when used for facsimile in the asynchronous mode.

In view of the above considerations the Netherlands Administration is in favour of recommending line signal characteristics, which are applicable in the duobinary f.m.-detection mode and the quadrature sending and receiving mode, as well as in the vestigial sideband mode of operation. As a consequence, the Netherlands Administration would like the recommendation of the frequency gap around the carrier frequency to be deleted, this frequency gap not being essential and even annoying for some modes of operation (especially for the duobinary f.m. mode of detection).

As homochronous demodulation in a pilot may be required, the Netherlands Administration suggests that a carrier pilot be recommended, as an optional feature, without at the same time a frequency gap around this pilot being recommended, as this gap is not essential and might give some difficulties in digital implementations of the modulator. As far as asynchronous operation of the system is concerned, it would be possible:

1) to recommend one of the given modes of operation for asynchronous use;

2) to make the mode of operation subject to bilateral agreement.

The Netherlands Administration is not in a position to judge which of the two possibilities seems to be the better, or even whether there should not be a separate recommendation for facsimile use. In the latter case, in our opinion, it is no longer a problem to be dealt with in Special Study Group A.

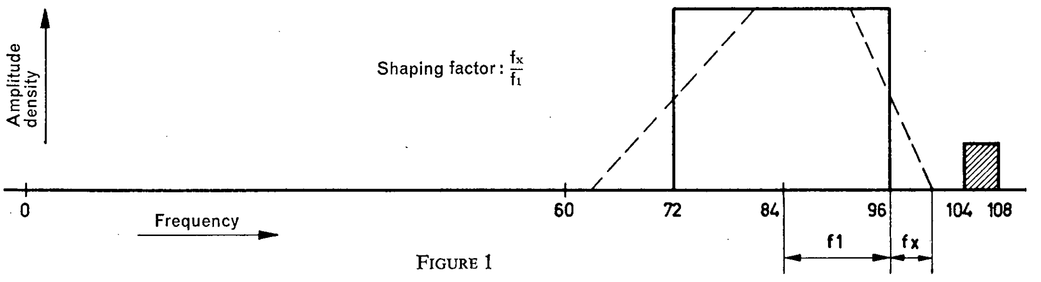


FIGURE 1

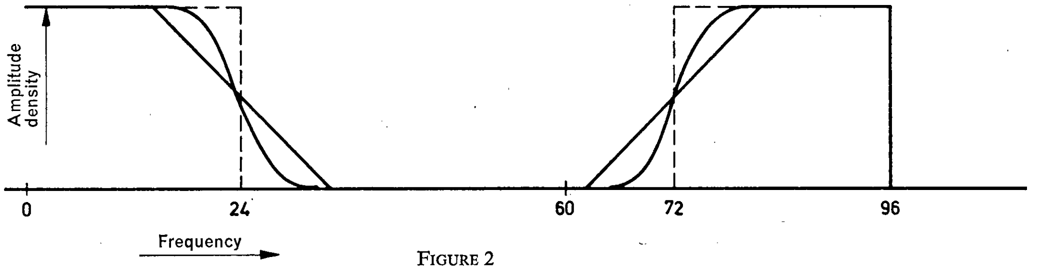


FIGURE 2

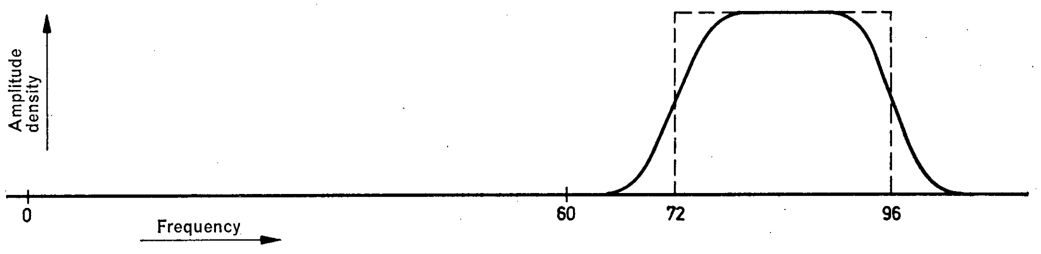


FIGURE 3

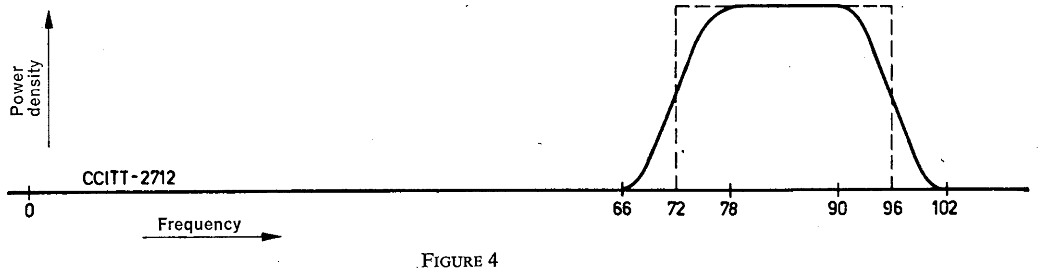


FIGURE 4

2. *Specific points*

2.1 *Carrier frequencies*

To avoid the exclusion of digital modulation techniques, it is necessary or at least advantageous to choose carrier frequencies equal to a multiple of 1/4 of the data signalling rate, the modem being used synchronously. With the preferred data signalling rate of 48 kilobits/s the (virtual) carrier frequency for the vestigial sideband mode should be 96 kHz (or 72 kHz at the lower edge of the band) rather than 100 kHz. The (virtual) carrier frequency at midband according to the quadrature mode will then be 84 kHz. Moreover, this choice results in a better spacing between the data-signal frequency band and the voice band 104-108 kHz. A shaping factor up to 0.66 is attainable in the proposed configuration of the frequency spectrum. See Figure 1.

2.2 *Signal spectrum*

In general, successive pulse signals can be detected without intersymbol interference when the amplitude spectrum of the pulses shows odd symmetry (on a linear scale) about the frequency equal to half the maximum signalling rate. With a 48-kilobits/s signal this frequency is 24 kHz for the baseband signal. Translated to the primary group in the vestigial sideband mode with a carrier frequency of 96 kHz, this frequency is 96 − 24 kHz = 72 kHz (see Figure 2). With an asymmetric sideband signal, the same rule of odd symmetry, but not about the carrier frequency, can ensure the absence of intersymbol interference due to low frequencies in the baseband signal (see Figure 3). The above-mentioned rules should lead to 6 dB-down points in the spectral density of the pulse signals at 72 kHz and 96 kHz. Analogous reasoning can be followed for the other modes of modulation, leading to the same results. These 6 dB-down points at 72 kHz and 96 kHz include:

i) pulse shaping and filtering at the sending side;

ii) filtering at the receiving side in the 60-104-kHz band and possibly some filtering after demodulation, but before detection has taken place.

To avoid intersymbol interference it does not matter whether the above-mentioned measures are all taken at the sending side, partly at the sending side and partly at the receiving side, or all at the receiving side. So one should not expect any consequence of this reasoning in the specification of the line signal characteristics. However, there is a connection between the foregoing and the signal-to-noise ratio. With regard to noise entering on the transmission path, which has essentially a flat frequency response, the optimum signal-to-noise ratio and hence the optimum security in the detection of the signal will be reached when the effects of the measures i) and ii) mentioned above are evenly divided. Consequently, this means that the spectral density of the signal at the sending side should have 3 dB points at 72 kHz and 96 kHz and odd symmetry in the *power* spectrum instead of the amplitude spectrum. When this characteristic is not met, signal power is spoiled, which results in signal-to-noise ratios worse than can be realized in the optimum situation.

Of course, when a gap should be used in the spectral density at a carrier frequency the optimum situation is intentionally abandoned as far as the low-frequency content of the signal is concerned. In view of the foregoing considerations the Netherlands Administration suggests that the shape of the spectral power density of the signal be prescribed in the above-mentioned sense.

ANNEX 2

(to point Z)

Use of wide-band circuits

(Contribution COM. Sp. A—No. 143 — U.S.S.R., October 1967)

General considerations

It is common practice to divide all data-transmission systems on the network of the Soviet Union into three groups according to the type of communication channel used and transmission rate:

1. Low-speed systems operating over telegraph channels with transmission rates more than 200-300 bauds.

2. Average-speed systems operating over telephone channels with transmission rates not more than 10 000 bauds.

3. High-speed systems operating over wide-band channels set up on the multi-channel links of modern carrier systems.

The design problem of high-speed data-transmission systems may be expediently divided into two parts:

1. Design of standard wide-band channels of different width on the basis of multi-channel links of modern carrier systems. It seems that this problem must be primarily studied by Study Group XV. It is also advisable to raise this question before this Study Group.

2. Design of standardized high-speed data-transmission equipment; this part of the problem relates to the research field of Special Study Group A.

Wide-band channel design

It is advisable to standardize:

— pre-group wide-band channel on the pre-group basis (three telephone channels),

— group wide-band channel on the group basis (12 channels),

— super-group wide-band channel on the super-group basis (60 channels),

— master-group wide-band channel on the master-group basis (300 channels).

In future a need to study data transmission over channels using wider bands and over standardized video channels may arise.

To set up a wide-band channel on the group link basis, for example, it is necessary to add to standard carrier equipment:

— channel translating equipment which is connected at the ends of the channel and which contains channel filters, frequency characteristic equalizers, suppression filters for pilots and amplifiers;

— transit equipment which is connected at intermediate points where the through connection of group links is provided; this equipment contains through-connection filters, frequency characteristic equalizers and suppression filters for line pilots.

Wide-band channels must be provided with means of frequency characteristic equalization. This is necessary because considerable phase distortions may cause inadmissible signal distortion which cannot be equalized at the channel end. Phase distortion should be equalized in each through-connection equipment.

Precision equalization devices can be provided in data-transmission equipment for final equalization of the wide-band channel together with the local line.

Group wide-band channel requirements

Requirements for group wide-band channel designed for data transmission can be formulated in the following way:

1. channel type: four-wire, non-switched;

2. nominal frequency band of 60 to 108 kHz;

3. nominal transmission test level (in power) must be equal to -4.5 Np at the input and -0.6 Np at the output;

4. data-transmission signal level at the input (at the test level point -4.5 Np) must not exceed -5.0 Np in average power and -4.5 Np in maximum power;

5. nominal value of input and output channel impedances must not exceed 150 ohms. The reflection coefficient relative to the nominal value must not exceed 15%;

6. irregularity of amplitude/frequency characteristic must not exceed 0.2 Np (±0.1 Np) in the frequency range of 65-103 kHz with channel up to 12 500 km (with the number of through connections within the group link up to 19 and within the super-group link up to 15) excluding the band 83.7 to 84.6 kHz;

7. irregularity of envelope delay/frequency characteristic in the range of 66-102 kHz must not exceed 30 microseconds with channel up to 12 500 km;

8. noise level at the point of test level −0.6 Np should not exceed −4.6 Np at the output of the channel having the length of 2500 km;

9. the level of selective noise introduced by carrier and pilot leaks at the output of the channel having the length of 2500 km should not exceed −5.6 Np at the point of test level −0.6 Np;

10. smooth changes of the test signal level in time at the output of the channel having the length of 2500 km should not exceed ±0.1 Np relative to the nominal level;

11. abrupt changes of signal level exceeding ±0.2 Np (but not more than ±0.7 Np) should not appear more than once within the period of 30 seconds in the channel having the length up to 12 500 km;

12. relative time period during which the impulse noise measured with the help of a device having the integration time of 10 microseconds exceeds 100 mV at the point −0.6 Np should not be above 5×10^{-5} within a one-hour period in the channel having the length up to 2500 km;

13. relative time during which the pilot level is reduced by 2.0 Np and more (within the period of more than 30 microseconds) should not be more than 8×10^{-6} within a one-hour period in the channel up to 12 500 km;

14. abrupt phase changes of the transmitted signal exceeding 30° should not appear more frequently than once within a two-hour period in the channel up to 12 500 km;

15. bit error rate in the speed range of 12-72 kilobauds should not exceed 5×10^{-5} in the channel up to 12 500 km;

16. in the first stage special selection and preparation of multi-channel links can be admitted for setting up wide-band channels designed for high-speed digital information transmission.

These requirements were formulated without taking into account the local lines. Perhaps they could be extended to comprise the whole channel including the local lines.
The requirements can be considered, naturally, only as provisional data to be used in raising this question before Study Group XV.

Reference pilots

It is not expedient to divide the wide-band channel into two parts because such a division causes frequency band losses and reduces total transmission speed. Using the whole frequency band of 60-108 kHz without division into sub-channels reduces the amount of data-transmission terminal equipment, makes it possible to obtain the highest data-transmission rate in a costly wide-band channel.
In order to eliminate the influence of the data-transmission signal on group reference pilot of about 84 kHz and the influence of this reference pilot on the data receiver, it is necessary to insert narrow-band rejection filters both at the beginning and at the end of the wide-band channel (in channel translating equipment).

The rejection filter causes adverse influence on data-transmission in two ways:

1. by cutting out a part of energy of a useful data signal;

2. by introducing into the surrounding medium a considerable phase distortion which cannot be corrected.

The first adverse factor is studied in the Annex to this contribution. This theoretical study shows that the presence of a rejection filter leads to the appearance of adverse impulses. The amplitude of these impulses is in most cases extremely small, therefore the influence of the impulses may not be taken into account. Carrier frequency and transmission rate must be chosen in such a way that main spectrum components of a signal do not fall in the stop-band of the rejection filter.

Experiments in data transmission over real group links were carried out. Mid-band frequency of the rejection filter was 84.14 kHz, the bandwidth at the level of 0.7 Np was approximately equal to 80 Hz and the attenuation at the mid-band frequency was about 3.5 Np. The data-signal carrier frequency was 84.00 kHz. Measurements were made by using frequency modulation, four-phase and eight-phase modulation methods. Transmission rates were 24-36 kilobauds, 48-60 kilobauds and 60-72 kilobauds, respectively. With the insertion of two or three rejection filters the impairment of stability to fluctuation noise (with the error rate of 10^{-5}) did not exceed 2 dB.

Therefore, in the near future with the use of modern modulation methods employing transmission rates of up to 72 kilobauds the influence of rejection filters may be neglected. There are no reasons to shift the group reference pilot.

In future, when it will be necessary to obtain even higher transmission rates and to use more perfect equalizers and more elaborate modulation methods, situations will possibly arise when the use of rejection filters in the middle of the passband will lead to considerable impairment of noise immunity. With this in mind, it is advisable to continue (in conjunction with Study Group XV) the study of the question concerning the shift of pilots and rejection filters beyond the passband of the wide-band channel or, at least, to the passband edges.

The second approach to the problem is possible, namely, cutting off group pilot and rejection filters when using the channel for data transmission. The possibility of such an approach can be determined when this question is studied jointly by Special Study Group A and Study Group XV.

Switched wide-band channels

In the Soviet Union high-speed data-transmission systems are supposed to be used in the near future only over leased circuits.

Transmission rates

The rule 600×2^n does not favour the creation of optimal systems. For example, for the channel of 48 kHz bandwidth the rates of 19 200, 38 400 and 76 800 bauds can be considered. The first two transmission rates do not ensure the use of all the channel possibilities, whereas the introduction of the third rate will result in considerable complication of the equipment.

The use of $600 \times \sqrt{2^n}$ rule adds intermediate rates of 27 072 and 54 144 bauds to the three rates mentioned above. This eases the situation but not to a great extent.

The adoption of the $600 \times m$ rule would mean, in fact, the refusal of the standardization of high rates. For example, 119 standard rates are obtained in the range of 12 000 to 84 000 bauds.

It is proposed to limit the number of standard rates by the new rule $12\,000 \times n$, adopting the $600 \times m$ rule as the basis. For example, the rates of 36 000, 48 000, 60 000 and 72 000 bauds obtained are very good for the 48-kHz channel.

Supervisory channel

The cost of a wide-band channel is, as a rule, high, therefore it is necessary to use the channel in the best way. It is proposed to use the whole frequency-band of the wide-band channel for data transmission without assigning a part of it (in the order of 4 kHz) for the supervisory channel. It is advisable to provide the possibility of establishing a supervisory speech connection instead of high-speed data transmission.

Such a solution is expedient at least for 12- and 48-kHz channels. In channels with wider bands, where the loss of 4 kHz will be negligible, the setting-up of a permanent supervisory channel might not reduce the possibilities of transmission.

If a customer urgently needs a permanent supervisory channel (for average-speed data transmission alternately with speech), it can be set up over a separate ordinary telephone channel.

Modulation method

When choosing a modulation method for a high-speed modem, it is necessary to take into account the considerably higher cost of a wide-band channel in comparison with that of a telephone channel. From the economical point of view it is advisable to use a more complicated data-transmission equipment in order to achieve the greatest possible rate.

Various modulation methods were tested on real channels:

— frequency modulation;

— amplitude modulation with a vestigial sideband;

— phase modulation with a vestigial sideband;

— four-phase differential modulation;

— eight-phase differential modulation.

It is proposed to use in data-transmission equipment the last method: eight-phase differential modulation. It ensures the highest transmission rate with normal error rate and good stability in operation. The 84 kilobaud rate was achieved. For the practical operation conditions, however, the lower rate of 72 kilobauds is recommended.

Description of eight-phase differential modulation modem

The block diagram of an eight-phase modulation modem transmitter is shown in Figure 1. The modem employs the ordinary digital principle of generating a phase-shift-keyed signal based on addition or inhibition of pulses in the process of frequency division in the frequency divider by eight. From a master oscillator the frequency of 672 kHz is fed to the divider input (672 kHz = 84 kHz × 8, where 84 kHz is a carrier frequency). From the divider output rectangular pulses are fed through the amplifier to the band-pass filter with the passband of 60-108 kHz. At the band-pass filter output the phase-keyed signal of 84-kHz carrier frequency is obtained.

From the transmitter input transmitted information is fed through the amplifier to the distributing circuit where binary signal elements are distributed among three sub-channels 1, 2 and 3. Data of three information signal elements are stored in triggers 1, 2 and 3. Then a coder generates three signals which control addition or inhibition of pulses in the divider by eight.

A " matching " circuit matches pulse trains of timing and carrier frequencies.

The output signal of the transmitter has eight possible phase shifts:

Sub-channels			Phase shift
1	2	3	
0	1	0	0
0	1	1	45°
0	0	1	90°
0	0	0	135°
1	0	0	180°
1	0	1	225°
1	1	1	270°
1	1	0	315°

The modulation code indicated above will give an error only in one sub-channel, if an adjacent phase position is received by mistake under the noise influence.

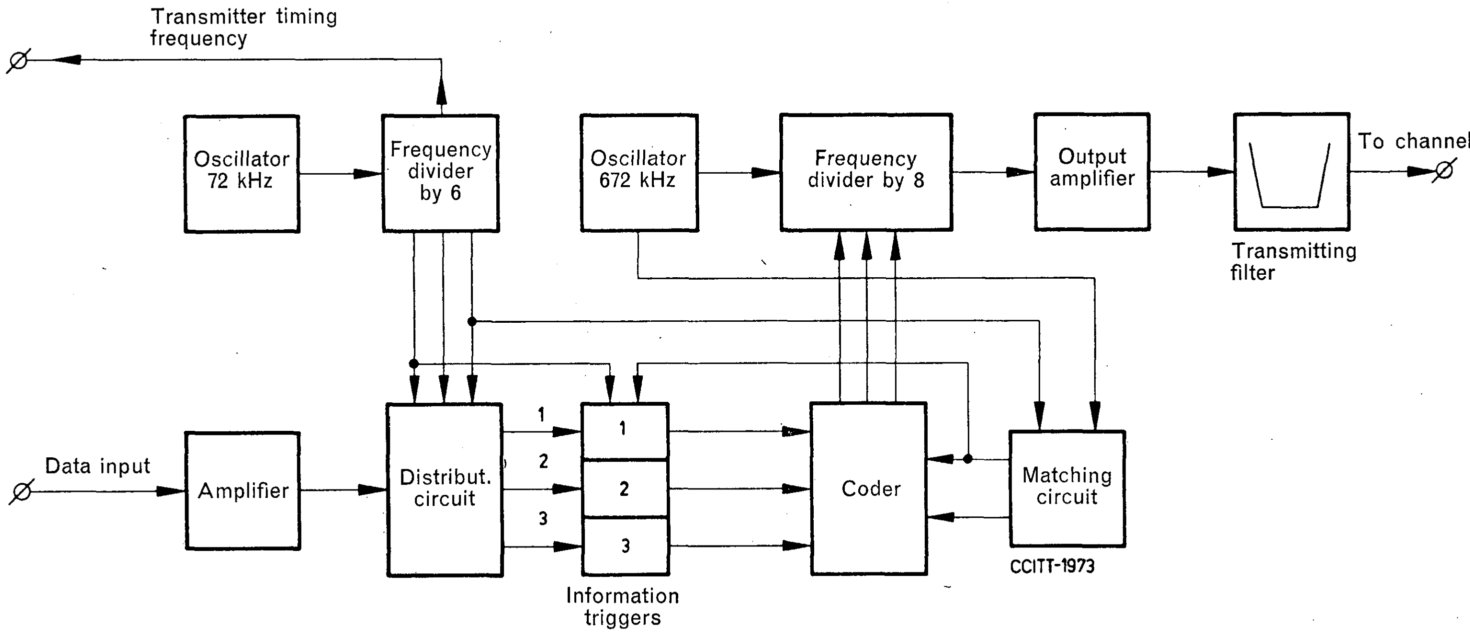


FIGURE 1. — **Transmitter block-diagram (72 kilobits/s)**

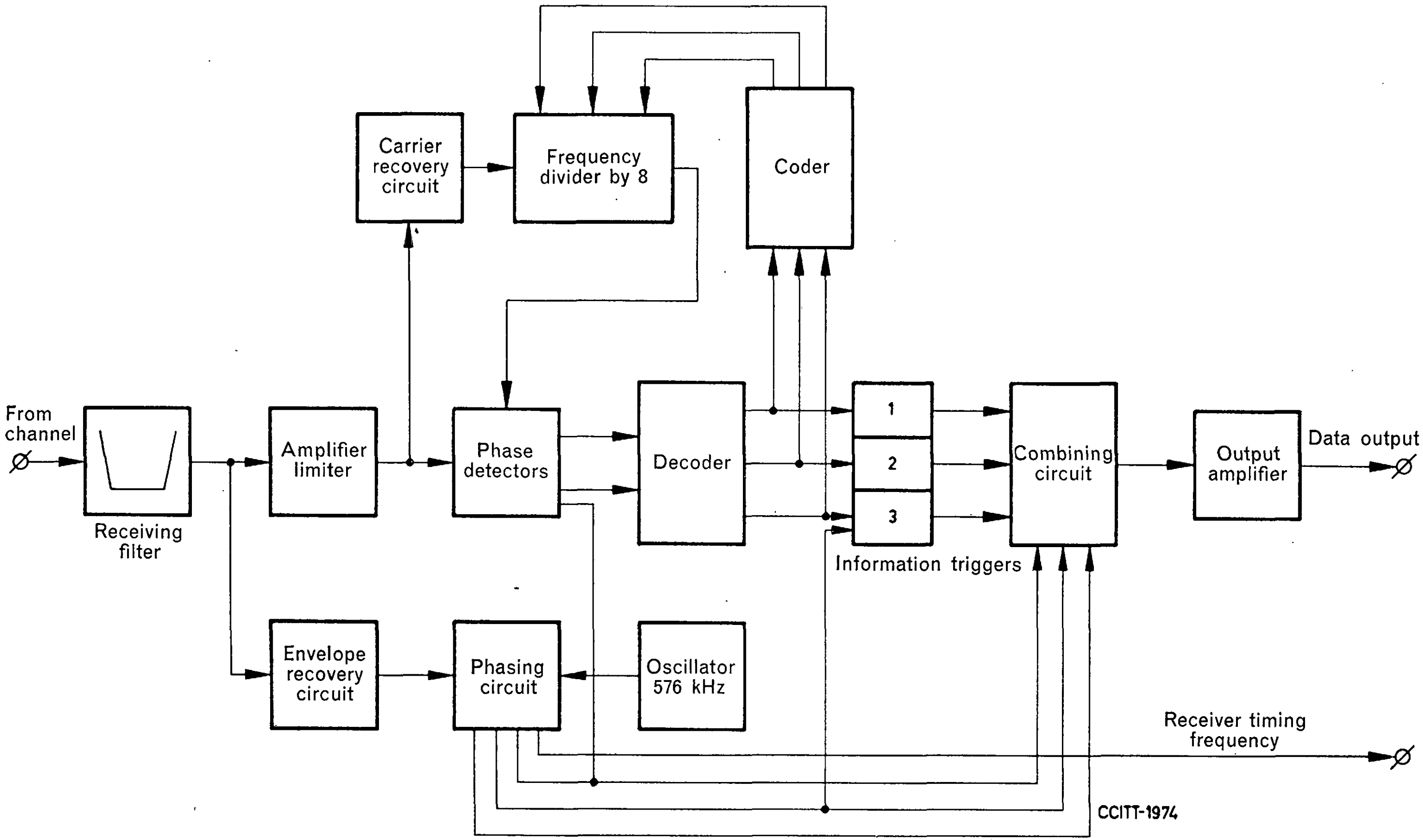


FIGURE 2. — **Receiver block-diagram (72 kilobits/s)**

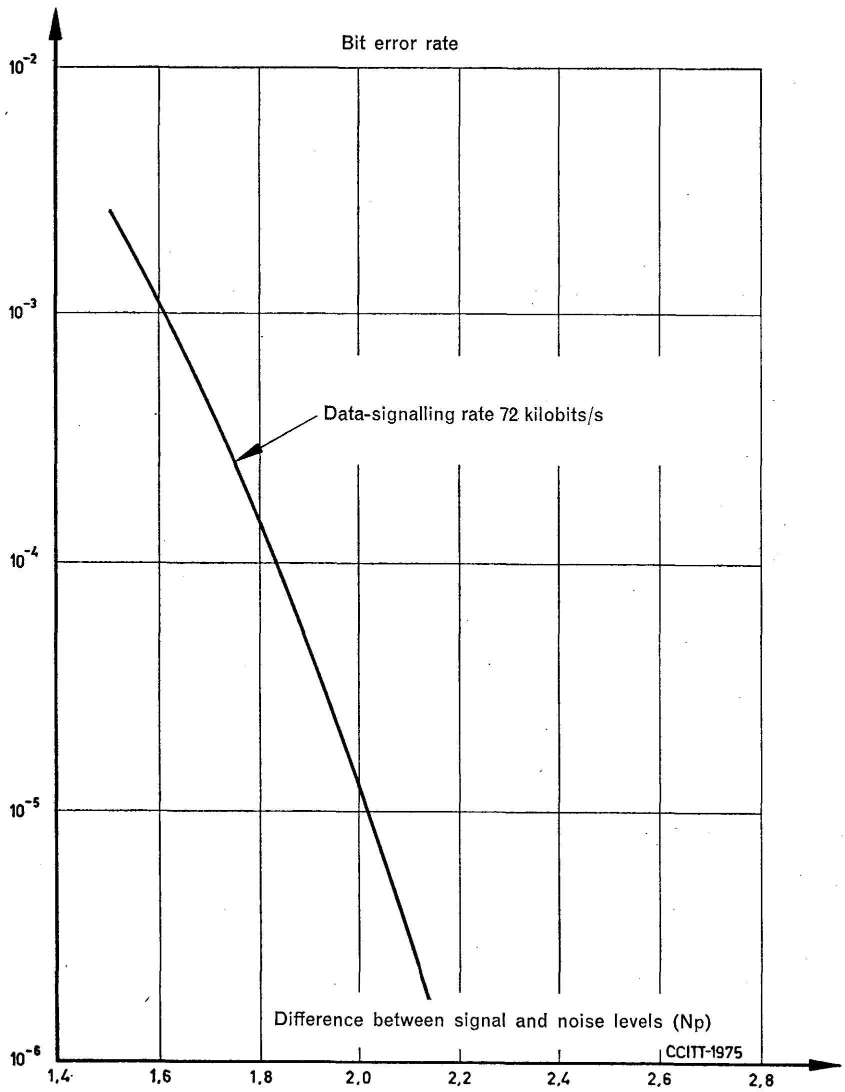


FIGURE 3. — **Interrelation of error rate and difference of signal and fluctuation noise levels**

The phase modulation rate is three times less than the operational transmission rate (in the case described it is 72 : 3 = 24 kilobauds).

The receiver block-diagram of the 8-phase modulation modem is shown in Figure 2. The received signal passes through the receiving band filter, then it is applied to the input amplifier limiter and to the envelope recovery circuit. As is known, the phase-shift-keyed signal envelope contains timing frequency components to which the local timing frequency is adjusted in the envelope recovery circuit.

The received signal and the local phase-shift-keyed signal with a " rotating " phase, obtained with the help of the divider by eight and coder, in the same way as in the transmitter, are fed to two phase detectors. The local signal repeats the phase positions of the preceding signal element received [1]. Thus, an original " differentially coherent method " is implemented in the receiver. The local signal, however, is free from noise.

Detected signals are applied to the decoder which recognizes signal elements in the three sub-channels. The use of the " rotating " phase method considerably simplified the decoder. The elements recorded in the information triggers are alternately fed to the output through the combining circuit.

Modem test results

From the tests carried out on the lines of 1500-3500 km the following results were obtained:

a) At the rate of 60 kilobauds bit error rate within a 10-minute period did not exceed 1×10^{-5} during 75% of the time and 1×10^{-4} during 98% of the time; total test time was 90 hours, average error rate was 1.8×10^{-5}.

b) At the rate of 72 kilobauds bit error rate within a 17-minute period did not exceed 1×10^{-5} during 60% of the time and 1×10^{-4} during 92% of the time; the total test time was 53 hours, average error rate was 2.5×10^{-5}.

Figure 3 shows interrelation of bit error rate and S/N ratio under the influence of fluctuation noise with the bandwidth of 48 kHz; the interrelation was measured in laboratory conditions.

ANNEX 3

(to point Z)

INFLUENCE OF REJECTION FILTERS ON PULSE TRANSMISSION OVER WIDE-BAND CIRCUITS

Let us first evaluate theoretically the influence of rejection filters on the transmission of unmodulated signals over a baseband system and then extend the results obtained on the transmission of modulated signals.

The baseband system

Suppose $A_0(\omega)$ is the spectrum of a pulse at the output of a baseband system having the linear phase response $\varphi_0(\omega) = \omega\tau_0$ in the absence of a rejection filter. Let the rejection filter with the midband frequency ω_r be connected in series with the baseband system. The amplitude response (amplitude/frequency characteristic) of this rejection filter is $A_r(\Omega)$ and the phase response is $\varphi_r(\Omega)$, where $\Omega = \omega - \omega_r$. Then the resulting amplitude response $A(\omega)$ and the resulting phase response $\varphi(\omega)$ are

$$A(\omega) = A_0(\omega)\, A_r(\omega - \omega_r); \qquad \varphi(\omega) = \omega\tau_0 + \varphi_r(\omega - \omega_r), \tag{1}$$

[1] The method applied is similar to the reception method described in the article by Kazuo Kawai and others. " A new carrier multiplex telegraph system using phase modulation " *(The Journal of the Institute of Electrical Communication Engineers of Japan,* 1965, Vol. 48, No. 8, pp. 1369-1377).

Assuming that for the pulse response (i.e. the response of the system to the Dirac's pulse) the stopband of the rejection filter is much narrower than the bandwidth of the channel, we have

$$P(t) = P_0(t) - A_0(\omega r)\, a(t_0) \cos \omega_r t_0, \tag{2}$$

where

$P_0(t)$ is the pulse response of the system without a rejection filter,

$$a(t_0) = \frac{2}{\pi} \left\{ \Omega_N \frac{\sin \Omega_N t_0}{\Omega_N t_0} - \int_0^{\Omega_N} A_r(\Omega) \cos [\Omega t + \varphi_r(\Omega)]\, d\Omega \right\}, \tag{3}$$

$\Omega_N = | \omega_r - \omega_N |$ is the frequency beyond which the influence of the rejection filter on the frequency response of the baseband system can be neglected;

$t_0 = t - \tau_0$.

Expression (3) shows that in the case of transmission of a single pulse with the spectrum $A_0(\omega)$, the rejection filter having the midband frequency ω_r causes spurious oscillations at the frequency ω_r. The amplitude of these oscillations depends on the shape of the spectrum of the pulse transmitted and on the relative position occupied by the rejection filter midband frequency in the channel (i.e. ω_r).

The envelope of these oscillations is defined only by the amplitude and phase responses of the rejection filter.

It can be shown [1] that the amplitude response and the phase response of the rejection filters used in the carrier transmission equipment can be approximated by the following expressions:

$$A_r(\Omega) = \frac{\tau\Omega}{\sqrt{1 + (\tau\Omega)^2}}; \quad \varphi_r(\Omega) = \text{arctg} \frac{1}{\tau\Omega} \tag{4}$$

where τ is a constant factor expressed in terms of the time and chosen so as to ensure that the characteristics mentioned above will correspond as nearly as possible to the actual rejection filter. For example, in the case of the 84.14 kHz rejection filter $\tau = 6.4 \times 10^{-3}$ s.

Substituting (4) in (3) and assuming the shape of the signal spectrum at the output of the baseband system to be cosine-squared [2], we have:

$$A_0(\omega) = \cos^2 \frac{\pi\omega}{2\omega_{max}}, \qquad f_{max} = \frac{\omega_{max}}{2\pi} > 36 \text{ kHz}$$

Thus for the 84.14 kHz rejection filter we obtain the maximum amplitude values of the interference voltage during transmission of a single pulse as shown in Table 1.

TABLE 1

$\frac{\omega_r}{\omega_{max}}$	0	0.25	0.33	0.5	0.75	1.0
The interference voltage amplitude/undistorted pulse amplitude ratio	0.0087	0.0074	0.0066	0.0044	0.0013	0

Table 1 shows that the influence of the rejection filter on transmission of a single pulse can be neglected since the amplitude of the interference voltage is less than 1 per cent of the pulse amplitude.

[1] B. S. Danilov: The influence of rejection filters on the pulse transmission over a telecommunication circuit *(Proceedings of the Central Research Communication Institute of the U.S.S.R. Ministry of Posts and Telecommunications,* 1963, No. 1*)*.

[2] The frequency band chosen is equivalent to the frequency spectrum of the modulated VSB signal transmitted over the group channel in the range of 60 kHz to 108 kHz at the modulation rate of 36 000 bauds.

The transmission of a random pulse train will result in a voltage being a sum of a great number of components. The ratio of the r.m.s. value of the interference voltage $\overline{U}$ to the undistorted pulse amplitude may be shown to be:

$$\overline{U} = \frac{1}{P_0(0)} \sqrt{\frac{1}{T} \int_{-\infty}^{\infty} (\Delta P)^2 \, dt}, \tag{5}$$

where $\Delta P = P\,(t_0) - P_0\,(t_0)$,

T is the time interval between the pulses transmitted.

For the particular case considered earlier, where $f_{max} = 36$ kHz and $\tau = 6.4 \times 10^{-3}$ s, the value of $\overline{U} = F \frac{(f_r)}{(f_{max})}$ found from (5) are shown in Table 2.

TABLE 2

$\frac{\omega_r}{\omega_{max}}$	0	0.25	0.33	0.5	0.75	1.0
$\overline{U}$	0.066	0.056	0.050	0.033	0.010	0

In the case where the transmission of bidirectional pulses shows a certain regularity, the maximum value of the interference voltage can be calculated by summing up the interfering actions of individual pulses.

For the particular case discussed and for certain periodical pulse combinations the estimated values of the actual voltages are shown in Table 3, where $\frac{\omega_r}{\omega_{max}} = 0.5$.

TABLE 3

Pulse train transmitted	Interference voltage	
	for double-current working	for single-current working
1. 1011010011000101 and further repeated	0.099	0.078
2. 1111111111111111	0.005	0.005
3. 1010101010101010	1.000	1.000
4. 1001001001001001	0.003	0.003
5. Random pulse train	0.003 (r.m.s. value)	

From the table one can see that in most cases the effect of the rejection filter during the pulse train transmission is rather small. For some combinations (e.g. 1111 and 1001001) almost complete mutual compensation of interference voltages takes place. In such cases the value of the interference voltage is much lower than the r.m.s. value of the noise.

We must also pay attention, however, to the case of transmission of " dots " (1010 signal). The values of the interference voltages in this case are summed up and the sum may have the same value as the wanted signal.

This has a straightforward physical meaning. In fact, during the transmission of " dots " the rejection filter midband frequency coincides with the first harmonic of the pulse train spectrum, this being the only harmonic for the given channel. In the cases 2 and 4, on the contrary, the rejection filter midband frequency is placed exactly between spectral lines of the pulse train transmitted. Case 1 is a close approximation to the random pulse transmission, therefore the interference voltage is of the same order of magnitude as the r.m.s. value of the voltage in the case of transmission of a random pulse combination.

Thus the ratio of the modulation rate to the channel bandwidth and the midband frequency of the rejection filter must be chosen in such a manner that the main spectrum signal components do not fall in the stopband of the rejection filter.

Transmission of modulated signals

The modulated signal in the phase-modulation systems is described in its most general form by:

$$P_{\omega}(t) = R(t) \cos(\omega_0 t - \varphi_0) + Q(t) \sin(\omega_0 t - \varphi_0), \tag{6}$$

where $R(t)$ is the in-phase component,

$Q(t)$ is the quadrature component,

ω_0 is the angular frequency,

φ_0 is the initial phase of the carrier.

a) With the vestigial-sideband transmission of modulated signals and a synchronous detector in the receiver, the latter responds to the in-phase component only. Therefore, the output signal $P(t)$ of the receiver will be:

$$P_{\text{out}}(t) = R(t) = P(t),$$

where $P(t) = P_0(t_0) - A_0(\Omega_r)\, a(t_0) \cos \Omega_r t_0$ is the output signal of the baseband system which is described by (2)

Ω_r is the difference between the rejection filter midband frequency and the carrier frequency.

Thus in the PM/VSB system the effect of the rejection filter is similar to its effect in the baseband system.

b) With the PM/DSB system the rejection filter midband frequency is generally placed to the left or to the right of the carrier frequency, thus affecting the signal spectrum symmetry. From (2) and (6) the envelope of the in-phase component is

$$R(t) = 2 P_0(t_0) - A_0(\Omega_r)\, a(t_0) \cos \Omega_r t_0.$$

The quadrature component of the voltage appears as well.

In the case of synchronous demodulation, we have:

$$P_{\text{out}}(t) = 2 P_0(t_0) - A_0(\Omega_r)\, a(t_0) \cos \Omega_r t_0,$$

i.e. the effect is half of that in the baseband system.

With the four-phase modulation system, in each of the two quadrature sub-channels the interference voltage comprises two components:

a) a component due to the influence of the rejection filter on transmission of signals over the given sub-channel which is fully identical to the influence of that in PM/DSB system;

b) a component due to the appearance of the quadrature component of the interference voltage during the transmission of signals over another sub-channel.

Supposing the amplitude of the latter component is almost identical to that of the former component, we may conclude that the influence of the rejection filter in the four-phase modulated system is twice as large as in the PM/DSB system and approximately equal to that in the PM/VSB system.

Question 1/A - point AA* — Acoustic coupling of data terminals to the telephone instrument

(continuation of point AA of the study programme 1964-1968)

a) Is there a definite need for the use of acoustic coupling?

b) If so, can means of acoustic coupling be defined which ensure at once:

1. a reliable transmission of the data;

2. a reliable protection of the telephone network from signals which cannot be tolerated, for instance because of their nature, or their level?

Acoustic coupling of data terminals offers a number of attractive features to the user, and has found some application. However, it poses certain technical problems which would have to be studied and solved before this mode of operation could become generally acceptable.

ANNEX

(to point AA)

POINTS PROPOSED TO STUDY TO COME TO THE STANDARDIZATION OF ACOUSTIC COUPLING WITH TELEPHONE SETS

(Contribution COM Sp. A—No. 202, I.S.O., June 1968)

1. *Scope*

Acoustic coupling of transportable data communication equipment with a telephone set which is connected to the public telephone networks, provides the facility to transmit from any place where a telephone set is available. In the near future several kinds of ambulant persons will need transportable data communication equipment to transmit data from nearly every place to a data-processing centre.

Such a transportable data set is of great importance when the collected information has to be processed within a certain time limit or when the ambulant person needs a quick response.

2. *Categories of application*

Two main categories of application can be distinguished:

2.1 *Conversational*

For example a little portable keyboard with the same layout as the keyboard of a touch tone telephone can acoustically be coupled with a normal telephone set. This portable keyboard itself generates the audio frequencies. On the other end of the line there can be a computer with audio response capability.

This type of device can be used by, e.g.:

— a salesman to ask whether a certain article is in stock;

— a salesman to enter a rush order;

— a repairman to report the end of his job and to be informed about his next instruction;

— an insurance agent to inquire for policy conditions.

For this category a low data signalling rate is sufficient. Parallel transmission can be used.

As we may expect that the touch tone telephone sets will not be available for general public use in Europe within the next decade, such a simple keyboard with acoustic coupling might overcome this shortage of technical possibilities for data transmission.

2.2 *Batch*

In this case data are collected and recorded during the day.

The recorded information will be transmitted as a batch to the central location where it will be processed.

The application can be a salesman who has recorded his orders received during the day. In the evening he has his portable transmitter available in his hotel-room to transmit the batch of data to the central location. Also for this type of application it is essential that acoustic coupling is available.

3. *Data signalling rates*

The following data signalling rates have to be considered:

— for the conversational type: up to 20 characters per second, with parallel transmission;

— for the batch type : up to 40 characters per second, with parallel transmission; up to 75 characters per second, with serial transmission.

4. *Conversational type*

Subjects which have to be considered to come to a standardization for the conversational type operating with parallel transmission are:

4.1 Has the code set to be restricted to numerical information only or is there also a need for an alphanumeric code set?

4.2 What type of function codes, such as: " end of transmission ", " operator intervention required ", " cancel ", etc., are needed?

4.3 Do we only consider the manual entry of data or has also the use of a simple card or badge reader to be considered?

This influences the maximum data signalling rate needed.

4.4 Which frequencies will be used?

The standardized touch tone frequencies or parallel modem frequencies?

4.5 What shall be the duration of the transmitted tones?

4.6 Is there a preference for a synchronous or asynchronous technique?

4.7 Is there a need for a non-simultaneous backward channel for tones, voice or data?

4.8 The power level on the line at the end of the transmitting telephone set.

4.9 What shall be the requirements of the acoustic coupling with respect to varieties of telephone handsets in relation to:

— the different sizes;
— the deviation from the mean power level by the coupling itself;
— the deviation of the characteristics of different microphones;
— the deviation of the characteristics of one microphone with time?

4.10 What coupling method, in respect to minimize the introduction of ambient radiated and conducted noise, has to be recommended?

4.11 What is the influence of the harmonics produced by the microphone?

4.12 Measuring method for the acoustic power level.

4.13 Can the parallel modem as described in C.C.I.T.T. Recommendation V.30 with the exception of the choice of frequencies, be used at the receiving end?

5. *Batch type, parallel transmission*

Subjects which have to be considered to come to a standardization for the batch type operating with parallel transmission:

5.1 Has the code set to be restricted to numerical information only or is there also a need for an alphanumeric code set?

5.2 What type of function codes, such as: " end of transmission ", " operator intervention required ", " cancel ", etc., are needed?

5.3 Which frequencies will be used?

The standardized touch tone frequencies or parallel modem frequencies:

5.4 What shall be the duration of the transmitted tones?

5.5 Is there a preference for a synchronous or asynchronous technique?

5.6 Is there a need for a non-simultaneous backward channel for tones, voice or data?

5.7 The power level on the line at the end of the transmitting telephone set?

5.8 What shall be the requirements of the acoustic coupling method with respect to varieties of telephone handsets in relation to:
— the different sizes;
— the deviation from the mean power level by the coupling itself;
— the deviation of the characteristics of different microphones with time;
— the deviation of the characteristics of one microphone with time?

5.9. What coupling method, in respect to minimize the introduction of ambient radiated and conducted noise, has to be recommended?

5.10 What is the influence of the harmonics produced by the microphone?

5.11 Measuring method for the acoustic power level.

5.12 Can the parallel modem as described in C.C.I.T.T. Recommendation V.30 with the exception of the choice of frequencies, be used at the receiving end?

6. *Batch type, serial transmission*

Subjects which have to be considered to come to a standardization for the batch type operating with serial transmission:

6.1 Is there a preference for a synchronous or asynchronous technique?

6.2 Is there a need for a non-simultaneous backward channel for tones, voice or data?

6.3 The power level on the line at the end of the transmitting telephone set.

6.4 What shall be the requirements of the acoustic coupling method with respect to varieties of telephone handsets in relation to:
— the different sizes;
— the deviation from the mean power level by the coupling itself;
— the deviation of the characteristics of different microphones;
— the deviation of the characteristics of one microphone with time?

6.5. What coupling method in respect to minimize the introduction of ambient radiated and conducted noise has to be recommended?

6.6 What is the influence of the harmonics produced by the microphone?

6.7 Measuring method for the acoustic power level.

6.8 Can the serial modem, 600/1200 baud, as recommended by C.C.I.T.T. be used at the receiving end?

Question 1/A - point AB* — **Use of digital transmission (or pulse-code modulation)**

(continuation of point AB of the study programme 1964-1968)

(to be studied first by Special Study Group D — Questions 11/D and 7/D)

Question 1/A - point AC* — Use of circuits established by means of satellite

(continuation of point AC of the study programme 1964-1968, amended at Mar del Plata, 1968)

Satellite links may possess properties different from other links with regard to noise distribution, available bandwidth, and propagation delay. In view of these different properties, it is recommended that the following factors be studied:

a) What general characteristics for error detection and correction should be standardized to enable efficient use to be made of data transmission over circuits involving satellite links (see Recommendation V.41)?

b) What methods should be standardized for the efficient establishment of data connections, especially in the automatic mode over circuits composed, in part, of a satellite link?

Question 1/A - point AD* — Comparative tests of modems

(Mar del Plata, 1968)

To facilitate the work of administrations and recognized private operating agencies in making a comparison of modems offered by different manufacturers, should a standard test method for different types of modems be established?

If so, what characteristics of the test devices should be standardized?

Note. — These tests should be made in the laboratory under a specific operating condition, the modem being connected to line simulators of particular characteristics.

The test assembly should include various items of equipment capable of reproducing in the laboratory the main parameters affecting the performance of a data-transmission system. These include:

1) a white-noise generator;
2) an impulsive noise generator;
3) a frequency shift simulator;
4) a phase variation generator;
5) line simulators for reproducing the amplitude and group-delay characteristics of circuits such as:
   a) carrier circuits,
   b) coil-loaded lines,
   c) local subscriber lines.

The data generator and receiver for recording the error rate of a transmission system should employ the 511-bit pseudo-random system specified in Recommendation V.52.

Question 1/A - point AE* — Continuation of the compilation of the list of interchange circuits in general (see Recommendation V.24)

(Mar del Plata, 1968)

1. This will include the following items which will be studied under the appropriate study points:

1.1 Review of electrical specifications to improve the performance relative to cross-talk between interchange circuits.

1.2 The new electrical characteristics and interchange circuits for the outstation modem used for parallel transmission.

1.3 The interchange circuits for the instation modem used for parallel transmission.

1.4 Inclusion of electrical specification for interchange circuits operating at greater than 20 000 bits per second.

1.5 A new section to define the use and inter-relationships of the interchange circuits provided in Recommendation V.24.

This will enable Recommendation V.24 to be limited to a list of circuits used in modems and of the relevant electrical characteristics.

2. It should also cover the review of all problems that arise in implementing the existing recommendations.

Note. — See Supplement No. 43 of this volume.

ANNEX
(to point AE)

Further points to be studied with the revised Recommendation V.24

(Mr. Van Egmond, Rapporteur on Recommendation V.24—Extract from COM Sp. A—No. 199, June 1968)

To make Recommendation V.24 a real shopping list, more changes to Recommendation V.24 have to be accepted.

Actually, the V.24 new version shall consist of two shopping lists:

1. A range of interchange circuit definitions, usefully divided into series for specific applications,

e.g. — circuits, normally used for and associated with data transfer . . . . (100-series)
— circuits, specifically for automatic calling . . . . (200-series)
— circuits, specifically for intermediate equipment . . . . (300-series)
— circuits, specifically for speech . . . . (400-series)

2. A collection of electrical characteristics for:

— circuits, operating with a common return . . . . (V-1)
— circuits, each consisting of a twin lead connection . . . . (V-2)
— circuits, operating on coaxial connections . . . . (V-3)

In any type of practical equipment a selection will be made from these two collections. As an example:

Essential interchange circuits for equipment X

Circuit	*Electrical characteristics*
101	V-1
102	V-1
103	V-3
140	V-1
402	V-2

Sections I and II contain proposals concerning this matter.

I. New interchange circuits to be added to Recommendation V.24

I–1 *Proposed new circuit, required for parallel data transmission (instation)*

Circuit ... — Voice answering input

Direction: TO data communication equipment

Speech signals, originated by the data terminal equipment, are transferred on these circuits to the data communication equipment.

I–2 *Proposed new circuits, required for parallel data transmission (outstation)*

Circuit 301 — Protective ground

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 302 — Data common return

This conductor establishes the common reference potential for all 300-series data interchange circuits. Within the data communication equipment, this circuit shall be brought to one point, and it shall be possible to connect this point to circuit 301 by means of a wire strap inside the equipment. This wire strap can be connected or removed at installation, as may be required to meet applicable regulations or to minimize the introduction of noise into electronic circuitry.

Circuit 303 — Transmitted data

Direction: TO data communication equipment

Data, originated by the data terminal equipment are passed on this circuit to the (local) data communication equipment for transmission to one or more remote data stations.

Circuit 304 — Activate transmitter

Direction: TO data communication equipment.

Signals on these circuits control the data channel transmitter. The ON condition causes the data communication equipment to activate the transmitter. The OFF condition causes the data communication equipment to disable the transmitter.

Circuit 305 — Activate receiver

Direction: TO data communication equipment.

Signals on these circuits control the data channel receiver. The ON condition causes the data communication equipment to activate the receiver. The OFF condition causes the data communication equipment to disable the receiver.

Circuit 306 — Received backward signal

Direction: FROM data communication equipment.

Signals originated by the data communication equipment in response to line signals received on the backward channel, are transferred on these circuits to the data terminal equipment.

Circuit 307 — Voice answering output

Direction: FROM data communication equipment.

Speech signals, received by the data communication equipment, are transferred on these circuits to the data terminal equipment.

Circuit 308 — Data set connected

Direction: FROM data communication equipment.

I–3 *Notes from the Rapporteur*

Referring to point 2 (Recommendation V.24 as shopping list) of this report, it would be more useful to have a separate series of circuits to be used for voice answering in- and outputs. Additionally, the Rapporteur proposes to delete circuit 301, circuit 302 and circuit 303 (as proposed in I–2), and to insert circuits 304, 305, 306 and 308 into the 100-series interchange circuits, and to have a new 300-series, applicable to voice operation.

Proposals: Circuit 135 — Activate transmitter
Circuit 136 — Activate receiver
Circuit 137 — Received backward signal
Circuit 138 — Data set connected

with the same definitions as proposed in I–2.

Note. — If circuit 138 (308) is equivalent with circuit 107, the Data Set Connected-Circuit should be deleted and, *300-series specifically for speech:*

Circuit 301 — Voice answering input
Circuit 302 — Voice answering output

with the same definitions as proposed in I–1 and 1–2.

In V.24 a new section V-2 has to be inserted for the electrical specifications for twin lead circuits and if required another section for the 300-series, when the characteristics of the 300-series circuits differ from those specified for twin lead connections.

The essential circuits for the out- and instation interface will then be:

Instation interface

Circuit	*Electrical characteristics*
101, 102, 104 (12 or 8), 107, 108.1, 108.2, 109, 110, 125, 130, 131	V-1
301	V-2 or 3

Outstation interface

Circuit	*Electrical characteristics*
101, 102, 103 (9 or 6)	V-1 (or whichever is required)
135, 136, 137, 138	V-2
302	V-2 or 3

In this case paragraph I–4 of the V.24 scope and the heading of Section V of the V.24 need to be amended, and the definition of Signal Ground or Common Return (102) will have to be re-worded.

II. Proposals of amendments to Recommendation V.24 exceeding the terms of reference

II–1 *Deletion of paragraph I–2*

When the definition of data communication equipment has been added to the list of definitions in *Blue Book*, Volume VIII, the illustration of general layout in V.24, section II, would be sufficient.

II–2 *Deletion of paragraphs I–3 a, b and c*

These points are redundant and may be proper to modem recommendations.

II–3 *Deletion of last two paragraphs of circuits 103 and 118*

The data terminal equipment is responsible for the transfer of data on these circuits. It is evident that, when one of the four circuits is OFF, proper transmission may be impaired. The data terminal equipment knows the conditions on these four circuits and will take the appropriate decisions. Therefore, these paragraphs are redundant.

II–4 *Circuit 105*

CONTRIBUTION OF THE C.C.I.T.T.-U.S.A. STUDY GROUP ON DATA TRANSMISSION (MARCH 1968)

Section III–1, circuit 105 (Request-to-send)

Suggested re-wording:

" Signals on this circuit are used to condition the local data set to transmit. The ON condition is maintained on this circuit whenever the data-processing terminal equipment has information ready for transmission or being transmitted to the data channel.

A transition from OFF to ON instructs the data communication equipment to enter the transmit mode as rapidly as possible. A sustained ON condition maintains the data communication equipment in transmit mode.

A transition from ON to OFF instructs the data communication equipment to complete the transmission of all data, which was previously transferred across the interface point on interchange circuit 103 (transmitted data), and then assume a non-transmit mode. On a simplex (U.S. half duplex) circuit where circuit 105 controls the direction of transmission of a local data set, a receive mode is assumed.

When circuit 105 is turned OFF, the data communication equipment responds by turning OFF circuit 106 (ready for sending) when it is prepared again to respond to a subsequent ON condition of circuit CA.

It is permitted to turn ON circuit 105 at any time when circuit 106 is in the OFF condition, regardless of the condition of any other interchange circuit. The condition of circuit 105 may not be changed from OFF to ON if circuit 106 (ready for sending) is ON."

Note. — A non-transmit mode does not necessarily imply that all line signals have been removed from the communication channel.

" For operation with clamping of circuit 104 (received data) and circuit 109 (data channel received line signal detector) refer to section III."

The choice of words to avoid the reference to turning ON or transmitting a " line signal " is intentional. Although the modems built in accordance with C.C.I.T.T. Recommendations V.21 and V.23 transmit an identifiable line signal and the Recommendations for these data sets might justifiably dictate that the line signal be turned ON not only " as soon as possible " but actually within a specified time interval. It is quite conceivable that some future data sets might not involve the turning on of a line signal to enter a transmit mode, for example, those using the transmission of a modified digital " baseband " signal or p.c.m. The general definition of the circuit in Recommendation V.24 should not presume to specify *how* a data set gets into the transmit mode. Rather it should suffice to say that it *does* go into transmit mode and leave the mechanism whereby it accomplishes this to the individual data set recommendation.

It is agreed that the data communication equipment should get out of the transmit mode as soon as reasonably possible after circuit 105 is turned OFF; however, it must protect the data-processing terminal equipment against loss of data. It has been the experience with some data sets that they must complete the transmission of available data, remove the line signal, reset internal timing circuits, etc. before they turn OFF circuit 106 (ready for sending). If circuit 105 is again turned ON before circuit 106 is turned OFF, the data communication equipment may be in the process of performing these internal " Housekeeping " tasks and may not be able to accommodate data presented to it by the data-processing terminal equipment. As a result, data are lost. The responsibility for preventing this lies in two places. As already stated, when the data-processing terminal equipment turns OFF circuit 105, it should under no circumstances turn it back ON until the data communication equipment finishes housekeeping and turns OFF circuit 106. On the other side of the contract, the data communication equipment can now lock out the data terminal equipment by holding circuit 106 in the ON condition. It should therefore not turn this circuit OFF until it has reached an internal state in which it can legitimately respond to a new ON condition on circuit 105 and provide adequate service. If these *two* conditions are observed, no data will be lost. In the case of the V.21 and V.23 data sets, the time involved is negligible and the OFF condition on circuit 106 can follow the OFF condition of circuit 105 in a matter of milliseconds; however, when more sophisticated data sets using two to four level encoders (e.g., four-phase data sets) are used or if error correction is introduced and redundancy is added by data communication equipment it is conceivable that circuit 106 may remain ON for several seconds after circuit 105 is turned OFF by the data-processing terminal equipment. These comments are for future consideration.

On half duplex circuits (U.S. duplex communication channels with half-duplex data-processing terminal equipment) the turning OFF of circuit 105 does not necessarily mean that the line signal transmitted by the associated transmitting signal converter will be turned off. When circuit 105 is turned OFF, some data sets continue to transmit autonomously to:

— maintain a constant energy level on the circuit for companders, AGC circuits, etc.;

— transmit training signals to keep " clocks " at both data stations synchronized;

— transmit training signals to keep automatic equalizers adjusted;

— hold data channel received line signal detectors at remote data station in the ON condition to minimize turn-around time;

— maintain synchronization between " scramblers " used for transparency purposes.

Thus, if a data set does transmit a line signal, the turning OFF of circuit 105 (request-to-send) will cause the data set to eventually assume a non-transmit mode in the sense that it will not accept data from the data-processing terminal equipment, but it will not necessarily cause the data set to take its line signal off the communication channel.

Note from the Rapporteur

The problem is to find a circuit 105 definition which covers all the possible and required applications. There are two solutions to this problem:

a) to refer to the modem recommendation for a proper definition of the transmit mode and the non-transmit mode;

b) to define more circuits which cover two or three types of application.

II–5 *Deletion of last sentence of 2nd paragraph of circuits 105 and 119*

" The ON condition must be maintained ...... (data)." This sentence is redundant. If the DTE wants data to be transmitted, of course it should have conditioned the DCE to transmit, i.e. circuit 105 or 119 ON.

II–6 *Deletion of last sentence of 2nd and 3rd paragraphs of circuit 107*

" Establishment of a .... turned ON " and " The OFF condition .... (calling indicator) ".

Both parts are redundant and may be better added to modem recommendations.

II–7 *Circuits 108.1 and 108.2*

Deletion of " The OFF condition .... (calling indicator) ", as proposed in II–6 above.

It would be useful to define one circuit 108, which covers both existing circuits 108.1/108.2. Such as:

Circuit 108 — Data terminal ready

Direction: TO data communication equipment.

Signals on this circuit control switching of the signal-conversion or similar equipment to or from the line.

The ON condition, indicating that the data terminal equipment is ready to operate, prepares the data communication equipment to connect the signal-conversion or similar equipment to the line and maintains this connection after it has been established by other means.

When the data communication equipment is conditioned for a) automatic answering of calls, b) automatic originating of calls, c) manual originating or answering of calls, connection to the communication channel occurs only in response to a combination of an ON condition on circuit 108 and a) the calling signal, b) a signal from the automatic calling equipment, c) a signal from the local operator.

A wiring option shall be provided in the data communication equipment in such a way that, using this option, the ON condition on circuit 108 causes the establishment of line connection without the help of other means.

(The OFF condition is identical with the 108.1/108.2 definition.)

However, the Rapporteur strongly recommends taking advantage of the new shopping list V.24 and defining one circuit 108 with the 108.2 definition, and a new circuit, e.g. circuit 138, with the 108.1 definition. This would simplify recommendations, because it prevents these recommendations from confusing notes.

II–8 *Circuit 112*

PROPOSAL MADE BY THE FEDERAL REPUBLIC OF GERMANY

" *Circuit 112:*

This circuit would be better called " Data signalling rate indicator ".

" *Circuit 112, second paragraph:*

Delete " are used to select one " and replace by " indicate which ". Delete " in the data processing ... of rates in use ". After " asynchronous data set " *insert* " is in use ".

" *Circuit 112, third paragraph:*

Delete " selects " twice and replace each time by " indicates ". "

The Rapporteur agrees with this proposal because the DCE can only indicate. For a proper selection a decision has to be made.

II–9 *Timing circuits 113, 114, 115, 128.*

PROPOSALS MADE BY THE FEDERAL REPUBLIC OF GERMANY

" *Circuit 115, last paragraph:*

Delete " Timing information on ... data communication equipment " since the first sentence of this paragraph does not appear to be reasonable and it is not sure that the timing information after interruptions is still correct. Moreover, it may be favourable for the data terminal equipment to receive timing information at all times."

" *Circuit 128, last paragraph:*

We suggest formulating the text in the way similar to the one proposed for circuit 114 (page 10). " The condition on this circuit shall be nominally ON and OFF for equal periods of time. The data communication equipment shall present a data signal on circuit 104 (received 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 circuit 128."

The Rapporteur agrees with the deletion of the last paragraph of circuit 115. In addition, the Rapporteur recommends the deletion of this paragraph from circuit 115 as well as from the circuits 113, 114 and 128. When timing information should be provided is a matter proper to modem recommendations.

The Rapporteur also supports the amendment of the circuit 128 definition (last paragraph).

II–10 *Circuit 120*

CONTRIBUTION FROM C.C.I.T.T.–U.S.A. STUDY GROUP ON DATA TRANSMISSION

" *Blue Book*, Volume III, item 15 of definitions for data transmission defines a " backward channel " as one in which the direction of transmission is always opposite to that in which information is being transferred. There are apparently no constraints in Recommendation V.24 which enforce this requirement. If this is desirable, a statement might be added to the definition of circuit 120 (transmit backward channel line signal) to the effect that the ON condition of circuit 105 (request-to-send) disables circuit 120 and clamps it in the OFF condition within the data set. Without such an interlock, the data communication equipment is at liberty to turn on *both* the data channel transmitting signal converter and the backward channel transmitting signal converter in the same data set. This is submitted for your consideration."

The Rapporteur only agrees with this interlock on circuit 120, when the backward channel will always be used in the opposite direction, but this would be restrictive for special use of the backward channel. In addition, the DTE is responsible for the control of the transmit functions provided within the DCE.

When both transmitters are used at the same time for some reason, would there be a degradation of the operation of the data channel or the telecommunication circuit? Even when a backward channel is used alternatively, the power should be 3 dB less than with operation without a backward channel.

II–11 *Circuits 124 and 129*

The C.C.I.T.T.–U.S.A. Study Group on data transmission as well as the Rapporteur suggest deleting these circuits from the V.24 because their functions are covered by the clamping option.

In the case that these circuits will not be deleted, the C.C.I.T.T.–U.S.A. Study Group proposes:

" In the event that these interchange circuits are not deleted from Recommendation V.24, the terminators for these circuits should be classified as " fail safe " so that they will interpret an open circuit condition (i.e. no driver provided by the data terminal equipment) as an OFF condition."

However, such a statement is proper to modem recommendations (Rapporteur).

II–12 *Circuits 126 and 127*

The Rapporteur proposes to delete these circuits and to adopt a new circuit 126:

Circuit 126 — Frequency selector

Direction: TO data communication equipment.

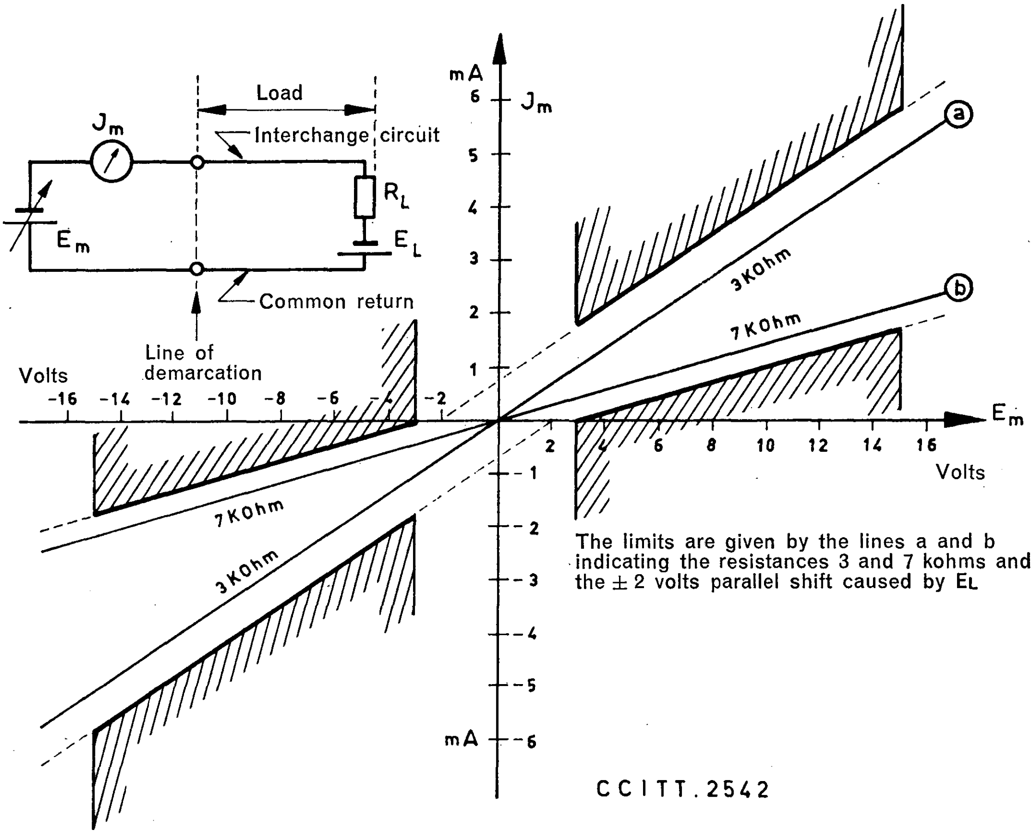
Signals on this circuit are used to select the required transmit and receive frequency within the data communication equipment.

The ON condition selects the higher transmit frequency and the lower receive frequency.

The OFF condition selects the lower transmit frequency and the higher receive frequency.

II–13 *Tolerance scheme for measurement of load resistance R_L*

The Federal Republic of Germany proposes to insert the tolerance scheme, which appears on the next page, in section V.



Such a scheme may be useful, but section V-2 together with Figure V-1 defines R_L properly and clearly.

II–14 *Circuit names and numbers*

The following circuit names are proposed:

Circuit

101	Protective ground *	
102	Common return *	
103	Transmitted data	
104	Received data	
105	Send request *	(Request to send)
106	Data channel ready *	
107	Data set ready	
108	Data terminal ready	(old 108.2)

* Circuits with an amended name.

Circuit

109	Signal detector *	
110	Quality detector *	
111	Rate selector *	
112	Rate indicator *	
113	Transmitter timing (DTE) *	
114	Transmitter timing (DCE) *	
115	Receiver timing (DCE) *	
116	Stand-by selector *	
117	Stand-by indicator	
118	Transmitted backward data *	
119	Received backward data *	
120	Backward send request *	(Backward request to send)
121	Backward channel ready	
122	Backward signal detector *	
123	Backward quality detector *	
125	Calling indicator	
126	Frequency selector *	
128	Receiver timing (DTE) *	
130	Transmit backward signal *	
131	Received character timing	
132	Non-data *	
133	Ready to accept *	
134	Received data present	
135	Active transmitter	
136	Active receiver	
137	Received backward tone	
138	Data set connected	(may not be required)
139	Connect data set *	(old 108.1)

When circuits 124 and 129 will not be deleted, the following names are proposed:

Circuit

124	Receiver clamp *
129	Backward receiver clamp *

Question 1/A - point AF* — Planning limits for the characteristics of telephone-type circuits and wideband circuits used for data transmission

(Mar del Plata, 1968)

What telegraphic distortion and error-rate limits, if any, should be recommended for the planning of telephone-type circuits used for data transmission?

Question 1/A - point AG* — Transmission constraints on wideband data system

(Mar del Plata, 1968)

In wide-band data systems, what constraints should be placed on the distribution of data signal (inband and outband) energy with respect to its concentration within a given frequency band and the permissible duration of such concentration?

* Circuits with an amended name.

1. The problems associated with interference to other services is more complex and potentially more serious with wide-band than with voiceband data transmission. This is because the total allowable power is greater in the wide-band than in the voiceband facility (viz., about 10 dB higher on 48 kHz than on a voice channel). It is desirable for this energy to be uniformly distributed over the entire frequency spectrum.

2. Difficulties and interference are likely if, as a result of the data terminal sending a repeated pattern of bits (repeated character or an idle condition), the total signal power is concentrated in a single frequency or a few frequencies for any extended period of time—viz., 1 second or more.

3. This problem is minimized in Recommendation V.35 by use of an encoder/decoder (i.e. a scrambler) which ensures a random data signal. However, these constraints need defining to aid in future work on standardizing new modems and in cases where non-standardized modems are permitted.

4. Limits should be set on the maximum permissible concentration of signal energy (power) within a portion of the total bandwidth (viz., within any 1 kHz of bandwidth) when this energy is integrated over some specified interval (viz., 1 second). It may be necessary to specify different limits for integration over several different time intervals.

Question 1/A - point AH — Modems for transmission of medical analogue data

(Mar del Plata, 1968)

In some countries, use is made of the general telephone network for the transmission of ECG (electrocardiogram) analogue data directly from a patient in his home or a doctor's office to a major medical centre where specialists are available. The transmitting modem may be electrically connected to the subscriber line or may be acoustically coupled to the telephone. A bandwidth of about 100 Hz is used.

Requirements for similar international ECG transmissions are likely to develop.

Additionally, experimental work is under way on systems for the simultaneous transmission of three channels of ECG data from a patient directly to a computer. The medical community, in co-operation with the communication and computing experts, are devising automated diagnostic procedures which hold great promises for mankind.

What are the requirements for the transmission of medical analogue data and what are the characteristics of modems to meet these needs?

Question 1/A - point AI* — Transmission of data over intercontinental telephone-type circuits

(Mar del Plata, 1968)

Should use be made of a special signal to distinguish data calls which may involve equipment different from that of normal speech connection?

Question 1/A - point AJ — Future integrated networks for data transmission

(Mar del Plata, 1968)

Considering

1. the rapid expansion of data-type services all over the world;

2. the indicated need for data-type services at high data signalling rates up to 50 kilobits/s and above;

3. the indicated need for very fast set-up time in a switched network for data-transmission;

4. the indicated need for flexibility in the selection of data signalling rates for particular services;

5. the indicated need for auxiliary controls on a data connection, such as combined data-voice, data-telegraph;

6. the indicated need for interconnection of facilities which carry data either in a switched network or a leased service;

7. the indicated need for transmission at low error rates;

8. the indicated need for transparency in the transmission medium with respect to the format and codes of the data;

9. the possibility of a switched network using " store-and-forward " principles;

10. the possibility of an integrated network using time-division principles in the facilities for transmission, switching and controls,

it is proposed to study the following:

1. the use of telegraph and telephone-type switched networks for combined operation with data services;

2. the characteristics of a separate switched network for data-type services;

3. the use for data of a future network for general telecommunication services (data, telephone, telegraph, etc.) employing time-division techniques on an integrated basis for transmission, switching, and controls.

Note. — For this study, refer to the Annex to point H.

Note 2. — For the purpose of this study, the word " data " may be assumed to include facsimile telegraphy where appropriate.

Note 3. — It is to be understood that in framing the requirements of this new network for data transmission, Special Study Group A will need to join the study of other interested Study Groups.

Note 4. — See Supplements Nos. 44 and 45 of this volume.

Comments

1. The study of point 1 of this question has been carried on in many respects by Special Study Group A for years.

2. Point 2 of the question could be considered under point H as far as telegraph-type network is concerned.

3. Point 3 of the question is to be studied by Special Study Group D (Question 11/D).

SUPPLEMENTS TO RECOMMENDATIONS AND QUESTIONS CONCERNING DATA TRANSMISSION

Contributions received during 1964-1968 which are published on account of their special interest

Supplement No.	Source	Title	Study point to be concerned with
1	Working Party on "New Alphabet"	Supplementary points to Alphabet No. 5	C
2	I.S.O.	Graphical representation for the control characters of the CCITT/ISO 7-bit coded character set	C
3	I.S.O.	Code extension procedures for the CCITT/ISO 7-bit code	C
4	I.S.O.	Transparent mode control procedures for data communication systems	C
5	Chile Telephone Co.	Error control in the telegraph network	I
6	I.S.O.	Use of longitudinal parity to detect errors in information messages	I, R
7	K.D.D.	A system for high-speed digital transmission over a telephone circuit	J
8	Tele-signal Corporation	Combined operation of a 1200-2400-baud data transmission channel and several 50-75-baud telegraph channels in a single voice-band	J
9	United Kingdom	Tests for data transmission systems on telephone-type circuits	L
10	I.B.M. (W.T.E.C.)	Data-transmission tests on a multipoint network (200, 600, 1200 and 1800 bits/s)	L
11	Chile Telephone Co.	Test measurements	L
12	Italy	Line tests over Italian telephone network	L
13	Italy	Italian line tests—Line breaking analysis	L
14	Federal Republic of Germany	Study of the general switched telephone network with a view to its suitability for data transmission	L, S+T ,V, W
15	United Kingdom	Tests for data transmission systems on telephone-type circuits	L
16	United Kingdom	Complement to standardization of the modem at 600-1200 bauds	M

Supplement No.	Source	Title	Study point to be concerned with
17	Italy	Complement to standardization of the modem at 600-1200 bauds	M
18	France	Draft specifications for a standard clock to be included in the 600-1200-baud modem referred to in Rec. V.23	M
19	United Kingdom	A method of modulation and demodulation for a four-phase line signal	O
20	United Kingdom	Coding methods for parallel transmission	Q
21	Federal Republic of Germany	Proposals of coding for parallel transmission	Q
22	Chile Telephone Co.	Error control in the telephone network	R
23	AEG-Telefunken	Signal quality detector	R
24	AEG-Telefunken	Data protection on telephone lines	R
25	United Kingdom	Cyclic code polynomial	R
26	AEG-Telefunken	Comparison of error protection methods for data transmission on telephone channels	R
27	AEG-Telefunken	Suggestions for 2 codes with 63 binary digits per block	R
28	Federal Republic of Germany	Measurement of phase distortion between subscribers	S+T
29	Federal Republic of Germany	Subscriber-to-subscriber loss	S+T
30	Netherlands	Subscriber-to-subscriber loss in international telephone connection	S+T
31	Study Group XVI	Characteristics of world-wide connections with regard to data transmission	S+T
32	United Kingdom	Measurement of phase distortion and transmission loss between subscribers	S+T
33	U.S.S.R.	Methods and results of measurements of envelope delay/frequency characteristics of telephone channels	S+T, U
34	A.T.T.	Future study of characteristics of subscriber lines used for data transmission over the telephone network	S+T, U, V
35	Siemens & Halske	Telegraph distortion caused by attenuation distortion and envelope delay distortion	S+T, U, W
36	United Kingdom	Maintenance methods	U, W
37	A.T.T.	Use of impulse noise counters	U, V, W

Supplement No.	Source	Title	Study point to be concerned with
38	United Kingdom	Specification of frequency/loss characteristics, phase distortion and impulse noise limits for circuits leased for data transmission, measuring techniques to check these specifications	U
39	France	Specification of an ordinary telephone circuit for data transmission	U
40	Australia	Measurements of impulsive noise at Sydney international switching centre	V, W
41	U.S.S.R.	Study of short interruptions in telephone channels designed for data transmission	W
42	Siemens A.G.	Automatic originating and answering of calls in the telephone network	Y
43	I.S.O.	Assignment of connector pin numbers for interchange circuits between data terminal equipment and data communication equipment where Rec. V.24 applies	AE
44	Chile Telephone Co.	Use of digital transmission	H, AJ
45	Netherlands	Some principal considerations concerning the future development of a switched network for data transmission	H, AJ

Supplement No. 1

JOINT WORKING PARTY ON NEW ALPHABET. — *(Extract from the report to the IVth Plenary Assembly of the C.C.I.T.T. AP IV/35 as amended—October 1968)*

SUPPLEMENTARY POINTS TO ALPHABET NO. 5

1. *Principles for the study of the question*

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 telephone network or the telex network.

This alphabet for data transmission will not 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 3rd Plenary Assembly in June 1964. The 3rd Plenary Assembly approved the method hitherto adopted and confirmed that the study should be continued on the same lines.

2. *I.S.O./C.C.I.T.T. 7-unit code*

2.1 *General*

At the initial stage of the study period 1964-1968, there were some differences between the I.S.O. draft and the C.C.I.T.T. draft concerning the allotment and use of the code combination and the definition of the control characters in the alphabet.

In order to seek to reach a compromise between these two drafts and to draw up a common I.S.O./C.C.I.T.T. 7-unit alphabet for data transmission and telegraphy, the two organizations met jointly as mentioned before and reached an agreement.

Slight editorial modifications to the agreed I.S.O./C.C.I.T.T. draft were subsequently made by the I.S.O. as well as by the C.C.I.T.T. Joint Working Party without any change of substance and this alphabet should be known as International Alphabet No. 5 for the following reason:

— No. 3 is in practice used for the 7 (4+3) units alphabet in the ARQ systems described in C.C.I.T.T. Recommendation S.13;

— No. 4 is allotted to the alphabet derived from Alphabet No. 2 used in time-division multiplex systems described in Recommendation R.44.

Note by the Secretariat. — The I.S.O. Council decided in December 1967 to accept the I.S.O.\ C.C.I.T.T. 7-unit code as an I.S.O. Recommendation. The new I.S.O. Recommendation has been registered under the following reference:

ISO\R 646-1967 — 6- and 7-bit coded character sets for information processing interchange.

2.2 *Table of Alphabet No. 5* (see Recommendation V.3).

2.3 *Current status on national choices of Alphabet No. 5*

2.3.1 *Positions reserved for national use*

Alphabet No. 5 gives allotments of combinations used exclusively for national purposes (sign ③ in the alphabet table). The allotment of these combinations is a national matter and the following table indicates the data recognized as valid at the present time:

Positions / Countries		4/0 ((@) *)	5/11 (([) *)	5/12	5/13 ((]) *)	7/11	7/12	7/13
F.R. of Germany	1	@	[	\	]	{	|	}
	2	§	Ä	Ö	Ü	ä	ö	ü
Belgium		@	[	\	]	{	¦	}
U.S.A.		@	[	\	]	{	¦	}
France	1	à	0	ç	§	é	ù	è
	2		[	\	]			
Italy	1	@	[	\ (°)	]	{	¦	}
	2	§	i	°	ò	é	ù	è
Japan		@	[	¥	]	{	|	}
Netherlands	1	@	[	\	]	{	|	}
	2			IJ			ij	
United Kingdom	1			\				
	2	@	[	·10	]	{	|	}
	3			$\frac{1}{2}$				
Denmark Finland Norway Sweden			Ä	Ö	Å	ä	ö	å
New Zealand		@	[		]			

C.C.I.T.T. 2847

* Preferred signs for international use.

Note. — This table has been supplemented by the data obtained from the I.S.O.

2.3.2 *Positions reserved for optional international use*

For certain combinations (marked with a ④ or a ⑦ used internationally there are a number of options. The options known at the present time are as follows:

Positions / Countries		2/3 £ (#)	2/4 $ (¤)	5/14 ∧	6/0 \	7/14 — (~)
F.R. of Germany	1	#		∧	\	—
	2	£				β
Belgium		#		∧	\	~
U.S.A.		#	$	∧	\	~
France		£		∧	\	~
Italy				∧	\ à	~
Japan	1	#		∧	\	—
	2	£			#	
Netherlands				∧	\	~
United Kingdom	1 2 3	£		∧	\	—
New Zealand				∧	\	—

C.C.I.T.T. 2848

Note. — This table has been supplemented by the data obtained from the I.S.O.

2.4 *Graphical representation of all function characters*

A study is being made by the I.S.O. either for all the functions, or for part of them. (See Supplement No. 2.)

For this purpose, the signs already standardized by the C.C.I.T.T. for Alphabet No. 2 (Recommendations S.4 and S.5) have been reported to the I.S.O.

The results of the I.S.O. study will be transmitted to the C.C.I.T.T.

2.5 *Definitions of functions*

Some of the definitions agreed on at the joint I.S.O./C.C.I.T.T. meeting are as yet only expressed in general terms and more precise definitions for users will be necessary for the accurate application of the alphabet. These definitions might be the subject of other recommendations.

The problem of the method of using the relevant functions has been and continues to be the subject of important studies by the I.S.O. In particular, according to information supplied by the I.S.O. the special functions for press requirements are being duly considered. The C.C.I.T.T. wishes to be informed of the results of these studies, even if they are only partial.

3. *Character structure of Alphabet No. 5* (see Recommendation V.4).

4. *Printing of a sign to indicate an error detected by the error detection system* (see Recommendation V.40).

5. *Implementation on punched tape*

Note. — It is not the role of the C.C.I.T.T. to lay down the detailed physical specifications of equipment. Thus, standards for the punched tape of the 5-unit code have never formed the subject of a C.C.I.T.T. Recommendation. Nevertheless, the Joint Working Party expressed the view that the foregoing amended extract from the I.S.O. Recommendation should appear as such in a forthcoming edition of the books of C.C.I.T.T. Recommendations. It is suggested that it might be annexed, for example, to the Recommendation on the 7-unit alphabet.

5.1 *Object and scope*

The following rules apply to the implementation of character sets on punched tape. The specifications relate more especially to the implementation of standard sets of coded characters of SEVEN information bits on 25.4 mm (1 inch) wide tape.

5.2 *References*

5.2.1 *Coded character sets*

7-unit coded character sets form the subject of I.S.O. Recommendation R.646 and C.C.I.T.T. Recommendation V.3.

The characters in this set contain 7 bits. To allow protection against errors in the tape punching, an extra bit called a " parity bit " is added to each character, making EIGHT bits to be punched in the tape.

5.2.2 *Punched tape*

The tape to be used (of paper or other material) must correspond to the provisions of Recommendation ISO/R ... [1] as regards the physical characteristics of 25.4 mm or 1 inch wide punched tape.

[1] At present at the stage of I.S.O. draft proposal.

The tape contains, parallel to the edges, a track of small holes constituting a feed hole track. On either side of this track are 3 and 5 information tracks respectively, capable of receiving holes. The positioning and dimension of the holes are specified in Recommendation ISO/R ... [1]

5.3 *Specifications*

5.3.1 *Lay-out of the tracks*

5.3.1.1. *Feed hole track*

The punched tape should have feed holes, as specified in 5.2.2 above.

5.3.1.2 *Reference edge*

The reference edge is that which has 3 data tracks between it and the feed hole track.

5.3.1.3 *Numbering of the information tracks*

The information tracks are numbered consecutively from 1 to 8 starting from the reference edge. Thus the feed hole track is between tracks 3 and 4.

5.3.2 *Implementation of coded characters*

5.3.2.1 Each position where a hole can be punched is assigned to the implementation of a bit. The absence of a hole represents binary NOUGHT. The presence of a hole represents binary ONE.

5.3.2.2 Each transverse line of holes can contain the implementation of only one character with its parity check bit.

5.3.2.3 Bits b_1 to b_7 of a character from the 7-bit alphabet table are assigned respectively to tracks 1 to 7.

5.3.3 *Parity bit*

A bit called the parity bit should be added to each coded character. This bit will be placed after the significant bits, i.e. in track 8.

This bit should be chosen so that the number of binary ONES in the same transverse line is *even*.

Note. — In certain instances, the transmission and the translation of this parity bit are optional.

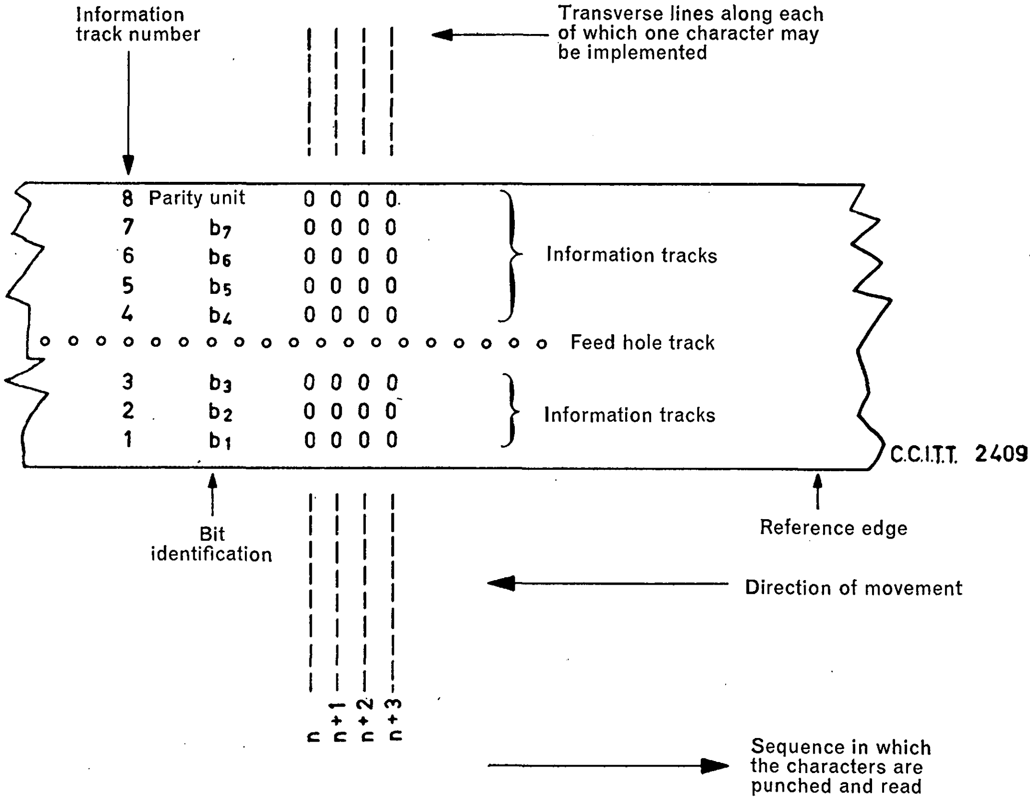
5.3.4 *Character sequence and direction of movement of tape*

The direction of movement of the punched tape is opposite to the character sequence.

[1] At present at the stage of I.S.O. draft proposal.

5.4 *Figure*

The following figure indicates the method of implementation specified above.



ISO/TC 97/SC 2. — *(Contribution COM Sp. A—No. 215—August 1968)*

GRAPHICAL REPRESENTATIONS FOR THE CONTROL CHARACTERS OF THE I.S.O. 7-BIT CODED CHARACTER SET

1. *Scope*

This I.S.O. Recommendation [1] specifies graphical representations for the control characters of the I.S.O. 7-bit coded character set, which is defined in Recommendation ISO/R 646-1967 " 6- and 7-bit coded character sets for information processing interchange ".

The control characters of the ISO 7-bit coded character set referred to in this I.S.O. Recommendation are the 32 characters coded in column 0 and in column 1 and the characters coded in positions 2/0 and 7/15 of the 7-bit set table.

Two modes of representation are given in this I.S.O. Recommendation:

— a set of specific symbols where a single character is required for the graphical representation of each of these normally non-printing characters;

— a set of two-lettered abbreviations, derived from the abbreviations used in the I.S.O. 7-bit set table.

2. *Field of application*

These graphical representations are intended for such applications as monitoring or diagnostic printers or display devices.

The precise font design for the symbols is not part of this I.S.O. Recommendation.

There is no necessity to implement all symbols.

[1] At present at the stage of I.S.O. draft proposal.

3. *Representations*

Position in the 7-bit set table	Set table name	2-lettered abbreviation	Symbol	Position in the 7-bit set table	Set table name	2-lettered abbreviation	Symbol
0/0	NUL	NL*		1/1	DC_1	D1	
0/1	(TC_1) SOH	SH		1/2	DC_2	D2	
0/2	(TC_2) STX	SX		1/3	DC_3	D3	
0/3	(TC_3) ETX	EX		1/4	DC_4	D4	
0/4	(TC_4) EOT	ET		1/5	(TC_8) NAK	NK	
0/5	(TC_5) ENQ	EQ		1/6	(TC_9) SYN	SY	
0/6	(TC_6) ACK	AK		1/7	(TC_{10}) ETB	EB	
0/7	BEL	BL		1/8	CAN	CN	
0/8	FE_0 (BS)	BS		1/9	EM	EM	
0/9	FE_1 (HT)	HT		1/10	SUB	SB	
0/10	FE_2 (LF)	LF		1/11	ESC	ES	
0/11	FE_3 (VT)	VT		1/12	IS_4 (FS)	FS	
0/12	FE_4 (FF)	FF		1/13	IS_3 (GS)	GS	
0/13	FE_5 (CR)	CR		1/14	IS_2 (RS)	RS	
0/14	SO	SO		1/15	IS_1 (US)	US	
0/15	SI	SI		2/0	SP	SP	
1/0	(TC_7) DLE	DL		7/15	DEL	DE	

CCITT 2383 A

* This proposal from the German Member Body cannot be retained, because of the confusion with " new line " NL.

ISO/TC 97/SC 2. — *(Contribution GM/ALP—No. 25—May 1968)*

CODE EXTENSION PROCEDURES FOR THE C.C.I.T.T./I.S.O. 7-BIT CODE

1. *General*

The I.S.O. 7-bit code (I.S.O. Recommendation R 646) [1] provides coded representations for a set of graphic and control characters having general utility in information interchange. In some applications, it may be desirable to augment the standard repertory of characters with additional graphic symbols or control functions.

The code includes several special characters intended to facilitate the representation of such additional symbols or functions, a process known as *code extension* [2].

The term CODE EXTENSION describes methods of using characters such as shift-out, shift-in, and/or escape, combined if necessary with other characters of a data-processing code, to represent additional characters which are not directly represented in that code. Characters which are included in the code by the exercise of national options or by substitution are excluded from this definition. It is an essential characteristic of this concept that additional characters so obtained are still encoded with the same number of bits as the basic code.

Although the basic nature of code extension—providing for encoding of information beyond the standard—limits the degree to which it may be standardized, there are advantages to adherence to certain standard rules of procedure. These advantages include: minimized risk of conflict between systems required to interoperate, and the possibility of including advance provision for code extension in the design of general purpose data handling systems.

2. *Scope*

This working paper specifies a set of procedures for the representation of graphic symbols and control functions not directly represented in the I.S.O. 7-bit code by characters of the I.S.O. 7-bit code.

3. *Characters SO, SI and ESC*

This working paper further defines the use of the I.S.O. 7-bit code characters:

SO shift out
SI shift in
ESC escape

The use of the I.S.O. 7-bit character DLE which is used to extend communications control functions will be defined in another Recommendation.

Notes by the C.C.I.T.T. Secretariat

[1] International Alphabet No. 5.

[2] In contrast with the term " code extension ", the term " code expansion " is used by the I.S.O. to define a method of obtaining a larger (8-bit) code table from the I.S.O. 7-bit code.

The converse process, passing from the 8-bit code to the 7-bit representation is called " code contraction ".

4. *Graphic set extension*

The characters SO and SI are used exclusively for extension of the I.S.O. 7-bit code graphic set. Inversely, extension of the I.S.O. graphic set is obtained by the use of SO and SI.

SO indicates that the 95 graphics of the I.S.O. 7-bit code are to be replaced by an " alternate " set of at most 95 graphics [1] not all of which need be different from I.S.O. 7-bit code graphics.

The 32 control characters and the character DEL (delete) of the I.S.O. 7-bit code are not affected by the Shift Out or Shift In operation.

If more than one alternate set is required to co-exist in a given system, sequences using ESC may be assigned to identify which one of the defined alternate sets SO calls for.

SI indicates that the standard graphics of the I.S.O. 7-bit code are restored to the code table.

5. *Extension of control functions*

With the exception of additional communication control functions the character ESC is used only for the addition of control functions. Inversely, non-communication control functions are obtained only by the use of ESC.

An additional control function or altered meanings to a group of existing control functions may be introduced by an escape sequence. The ESC character is the first character of an escape sequence which may also contain any number of *intermediate* characters and is always terminated by a *final* character.

Intermediate characters are the 16 characters of column 2 of the I.S.O. 7-bit code table, i.e. space and certain special graphics.

All other characters of the I.S.O. 7-bit code are *final* characters, except that the following characters must not be used intentionally to terminate an escape sequence.

Designation	Name	Code Table Column/Row
NUL	Null	0/0
SOH	Start of heading	0/1
STX	Start of text	0/2
ETX	End of text	0/3
EOT	End of transmission	0/4
ENQ	Enquiry	0/5
ACK	Acknowledge	0/6
DLE	Data link escape	1/0
NAK	Negative acknowledge	1/5
SYN	Synchronous idle	1/6
ETB	End of transmission block	1/7
CAN	Cancel	1/8
SUB	Substitute	1/10
ESC	Escape	1/11
DEL	Delete	7/15

[1] Not all 95 graphics of the alternate set need be assigned.

6. *Qualifications*

The need for standardizing assignment of specific escape sequences to specific functions and for the assignment of standard alternate graphic sets is under study.

A first attempt at defining such classes of escape sequences follows.

Further considerations on possible structuring of escape sequences

1. The intention of this document is to pursue the work on the standardization of ESC sequences. Provision is made for variable ESC sequences by dividing the code table into intermediate and final characters:

Intermediate character	(I):	the 16 characters of column 2 of the table;
Final characters	(F):	the 79 characters of columns from 3 to 7 of the table;
	(f):	the remaining 18 characters of columns 1 and 2 of the table.

The 18 control characters referred as (f) are permitted as final characters by the above-mentioned working paper, but they are probably not to be recommended for wide use due to the interference they can have with the hardware of equipment, particularly for the simplest devices. In the following of this document, when a final character is mentioned, it is intended to be one of the 79 graphics previously referred as (F).

2. *Requirements for extension of the 7-bit code by ESC sequences*

2.1 *Sequences which define*

control characters of an expanded set;
control characters independent of an expanded set.

2.2 *Sequences which define a command calling into force alternate sets of*

controls;
graphics;
controls and graphics.

2.3 *Sequences for private use*

3. *Examples*

Some tentative examples are considered here and submitted for examination and further evaluation. The intention is to structure the ESC sequences in such a way that similar ESC sequences should relate to similar control functions. A grouping of controls in this way is considered to be more important than assigning two character ESC sequences to frequently used controls only. The grouping takes also into consideration the relationship with a proposed structure for a family of 8-bit codes.

3.1 *Two character sequences*

They take the form:
ESC (F)

The 79 final characters may be subdivided into three groups as specified in the following and shown in the figure:

		←(F)→		
0 1	2	3	4 5	6 7
Controls	(I)	(A) - - - - - - (D)	(E)	(A) - - - - - - (D)

(E): 32 characters of columns 4 and 5 for additional controls to be used in columns 08 and 09 of the proposed family of 8-bit codes;

(A): a chosen number of characters from columns 3, 6 and 7 for additional agreeable controls;

(D): the remaining characters of columns 3, 6 and 7 for "do it yourself" controls.

3.2 *Three character sequences*

They take the form:

ESC (I) (F)

It is suggested that these sequences be used to create groups of families of similar controls or to nominate different sets as specified in the following tentative examples:

ESC ! (F): to designate a maximum of 79 device controls

ESC " (F): to designate a maximum of 79 format controls

ESC # (F): to designate a maximum of 79 groups of 32 additional controls for the proposed family of 8-bit codes

ESC $ (F): to nominate a maximum of 79 different control sets (columns 0 and 1 of the I.S.O. 7-bit table)

ESC % (F): to nominate a maximum of 79 different shift out sets (columns 0 to 7 of the I.S.O. 7-bit table)

ESC & (F): to nominate a maximum of 79 different transparent sets (columns 0 to 7 of the I.S.O. 7-bit table)

ESC ' (F): to nominate a maximum of 79 different sets with a lower or higher number of bits than the I.S.O. 7-bit code

ESC * (F): to nominate a maximum of 79 different national choices of the I.S.O. 7-bit code.

3.3 *Four character sequences*

They take the form:

ESC (I) (I) (F)

and are reserved for private use.

ISO/TC 97/SC 6. — *(Extract from Contribution COM Sp. A—No. 207—June 1968)*

DRAFT I.S.O. PROPOSAL ON TRANSPARENT MODE CONTROL PROCEDURES FOR DATA COMMUNICATION SYSTEMS

1. *Introduction and scope*

There exist various reasons for using transparent mode. In basic mode (non-transparent) the originator is constrained to the use of 118 characters in data. Some form of transparency therefore is required for other conditions, such as:

1) Grouping of basic mode messages to be transmitted as a block on intermediate links within a multi-link transmission system. In this case, some but not all of the transmission control characters might be within the transmission block, hence there is no requirement for full transparency (i.e., to 128 codes). This case is, therefore, called "format-independent" mode of operation.

2) Using the 7-bit ISO code, where all 128 possible code combinations can appear in data, is considered to be full "transparent" mode. It can occur where the information is encrypted or where binary data are put in 7-bit groups with an appended parity bit. This type of data could have the same error checking system as applicable in basic mode, but requires the addition of the "double DLE" technique in transmission.

3) In using all 8 bits, 256 possible code combinations can occur in the actual data being transmitted. This type of transmission will occur when binary data are broken up into 8-bit groupings, instead of 7-bit groups with a parity bit. It is also useful if it is desirable to transmit 8-bit codes instead of the 7-bit ISO code, as would be the case for EBCDIC code or "double numeric" data. In all these cases, the I.S.O. transmission control characters may appear in the data being transmitted. This type of transmission would require both the "double DLE" operation of (2) above and a block checking scheme different from the longitudinal parity check as presented in Supplement No. 6.

2. *Rules for each of the types of transparency*

2.1 *Seven-bit format independent blocking mode*

The character sequence DLE ... starts the format independent block
The character sequence DLE ETB (or optionally DLE ETX) ends the block
DLE and SYN are not allowed in the data, but all other control characters could occur in data

The error-checking technique is the same as that used in basic mode.

2.2 *Seven-bit text transparent mode*

DLE STX starts a transparent text or a transparent block
DLE ETX ends a transparent text
DLE ETB ends a transparent block
The "double DLE" technique is required

The error-checking technique is the same as that used in basic mode.

2.3 *Eight-bit text transparent mode*

DLE STX starts a transparent text
DLE ETX ends a transparent text [1]
DLE ETB ends a transparent block [1]
The "double DLE" scheme is required

A cyclic type of check (block check sequence BCS) is needed in lieu of character and longitudinal parity check. The composition of the BCS is subject to further study and will be specified at a later date.

3. *Rules for "double DLE" technique*

If the data characters or binary information transmitted in seven- or eight-bit text transparent mode include a bit pattern that could be interpreted as DLE by the slave station, the following procedure must be followed:

1) The master station inserts an additional DLE contiguous to the existing DLE in the data stream.

2) When the slave station detects the first DLE in the transparent message, it examines the following character to take one of the following actions:

a) if the following character is DLE, one of the pair is discarded and the other is accepted as data;
b) if the following character is ETX, ETB or SYN the appropriate rules in section 2.2 and 2.3 above apply;
c) if the sequence is DLE DLE ETX, DLE DLE ETB, or DLE DLE SYN, the sequence shall be interpreted as independent characters DLE and ETX, or ETB, or SYN;
d) if any other character follows DLE, it is to be considered as an error.

Supplement No. 5

CHILE TELEPHONE COMPANY. — *(Extract from Contribution COM Sp. A—No. 50—November 1965)*

ERROR CONTROL IN THE TELEGRAPH NETWORK

It would seem necessary that at least one of the methods to be standardized for error control of data transmitted over the telex network should be suitable for use on connections including or requiring:

a) ARQ equipment,
b) regenerative repeaters, etc., or
c) simultaneous transmission in both directions.

[1] No ending DLE—extension is required when predetermined fixed-length blocks are transmitted.

Of the three methods of error control envisaged in Recommendation V.10, that employing the principle of block retransmission is best suited for general application.

For the automatic transmission of 5-track paper tape, reader to punch, storage in the error-control equipment can be envisaged as being either electronic or the paper tape itself (back-stepping readers and punches). Further, the receiving station arrangements could be such as to provide a clean tape or simply to overpunch those blocks containing errors. Whatever the arrangements adopted, it would be very desirable that the error-control procedures should permit satisfactory interworking of such equipments.

The characteristics of a block retransmission error-control equipment which it is considered would require standardization, if interworking of equipments of different origin is to be ensured, are:

1. block size
2. block identification
3. block check code
4. block repetition procedure
5. precautions against block errors
6. initiation and termination of protected transmission.

The proposals which follow are based on experience with a system using back-stepping paper-tape readers and punches and providing an output tape having faulty blocks overpunched. However, the proposals would prove equally suitable for interworking with and between systems using electronic storage.

Repetition procedure

The receiving station acknowledges correct receipt of a block by returning the sequence character of the next block, i.e. correct receipt of a block Y is acknowledged by returning character I.

The receiving station indicates that a block has been incorrectly received by returning the sequence character of that block. The sending station repeats this same character to acknowledge receipt. Actual retransmission of the repeated block is initiated by the receiving station sending a second time the sequence character of the block to be repeated.

Precautions against block errors

A block is received by the receiving equipment only if preceded by the sequence character next in sequence to that of the last block correctly received. The only acknowledgement signal returned by a receiving station is the sequence character of the next block to be received. This acknowledgement signal is repeated at intervals of approximately 1.5 second until the requested block is actually received.

The sending station prepares to send next the block indicated by the acknowledgement or repetition request signal; actual transmission of a block, unless it is the next in sequence, begins only when its sequence character has been received twice in succession.

Initiating and terminating protected transmission

Protected transmission is started by sending "the signal for transfer to data" specified in Recommendation V.10. The revision to normal teleprinter working is effected by sending the sequence XXXX (or *////*), i.e. signal 24 of International Alphabet No. 2.

If either station fails to receive a sequence character, either as a block-indication character or an acknowledgement signal, over an interval of 10 seconds, it reverts to teleprinter operation and

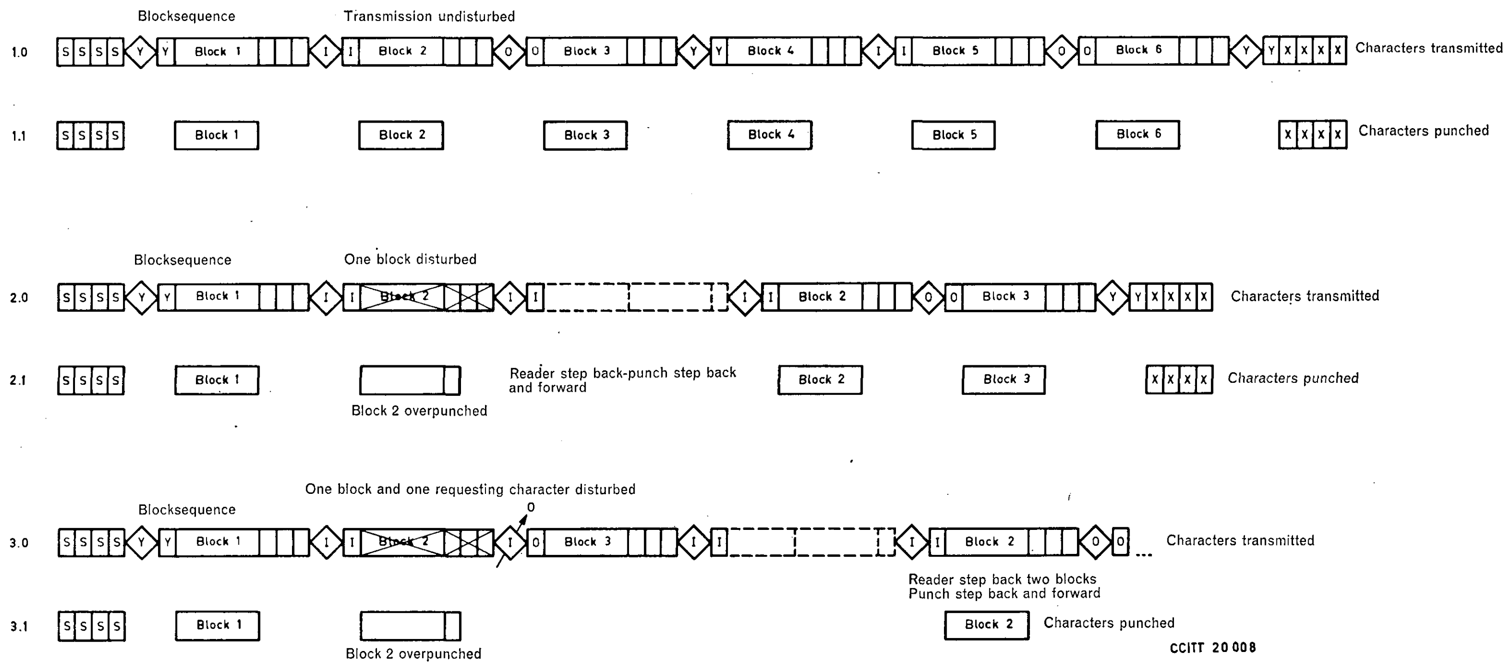


FIGURE 1. — **Error detection and correction**

initiates an operator-recall signal. The sending station similarly reverts to teleprinter operation if it receives two successive sequence character signals referring to a block which is no longer accessible for retransmission.

The correction of errors is automatic, and the degree of error control exerted is powerful. Erroneous blocks will be detected if they contain three or fewer bits in error or if the errors do not extend over more than 12 consecutive bits. Reduction factors for different numbers of check bits are indicated in Figure 2 of Supplement No. 22.

The error-control equipment, with associated tape reader and tape punch, is connected between the teleprinter and the line terminal unit of a telex installation. Characters are transmitted in 5-unit start/stop code (stop elements 1½ units duration) at 50 bauds. Operation at 75, 100 and 200 bauds is also possible. Figure 1 illustrates the various line procedures of error control.

Block size

Alternatively 48 or 112 information characters (5 bits), excluding block identification and checking characters (see below).

Block identification

A block is identified by one of the characters Y, I or O used in a cyclic sequence. This sequence character is the first of a block, immediately preceding the information characters.

Check code

Three check characters added at the end of each block are derived from the transmitted information by a 12-stage shift register with feedback over module 2 adders, as in Figure 2. The check characters are derived from the state of the register after transmission of the last information character as follows:

	Elements				
	1	*2*	*3*	*4*	*5*
Check character 1	12	11	10	9	12
Check character 2	8	7	6	5	8
Check character 3	4	3	2	1	4

The polarity of the first four elements of each check character is determined by the 12-stage shift register. The polarity of the fifth element of each check character is a reversal of the polarity of the first element. This systematic reversal ensures that all check characters are printable.

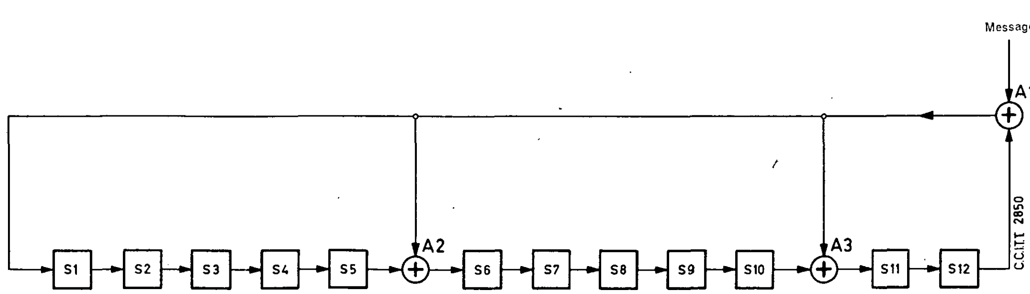


FIGURE 2. — **Feed-back shift-register**

ISO/TC 97/SC 6. — *(Extract from Contribution COM Sp. A—No. 207—June 1968)*

LONGITUDINAL PARITY TO DETECT ERRORS IN INFORMATION MESSAGES

(Draft I.S.O. Recommendation No. 1732)

Introduction

In data communication systems the information formats and the redundancy in the data to be transmitted differ widely from one application to another. It is therefore clear that a number of classes of error protection systems may be required.

This I.S.O. Recommendation defines one method of error detection which satisfies a wide range of applications. It consists of accompanying the data block or text by one checking character (in addition to character parity) and it is often referred to as the "longitudinal parity method".

1. *Scope*

This I.S.O. Recommendation specifies a method of detecting errors in information messages by attaching one block check character to the transmitted information block (or text) and checking this character when it is received. The method of correcting errors when they are detected is specified in (reference to be inserted in final version [1]).

The method is applicable to systems which use the I.S.O. 7-bit coded character set, which is the subject of Recommendation ISO/R 646 and the basic mode of implementing this 7-bit code in data communication systems [2].

The rules for generating the character parity bits [3] are that the character parity sense shall be odd in synchronous systems and even in asynchronous systems.

2. *Rules for generating the longitudinal parity block check character*

2.1 *Block check character*

2.1.1 The block check character shall be composed of 7 bits plus a parity bit.

2.1.2 Each of the first 7 bits of the block check character shall be the modulo 2 binary sum of every element in the same bit 1 to bit 7 column of the successive characters of the transmitted block.

2.1.3 The longitudinal parity of each column of the block, including the block check character, shall be even.

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[1] Being prepared. Presently at the draft proposal stage.

[2] Being prepared. Presently Draft I.S.O. Recommendation No. 1745.

[3] Being prepared. Presently Draft I.S.O. Recommendation No. 1734.

2.1.4 The sense of the parity bit of the block check character shall be the same as for the information characters (odd for synchronous transmissions, even for asynchronous transmission).

2.2 *Summation*

2.2.1 The summation to obtain the block check character shall be started by the first appearance of either SOH (Start of heading) or STX (Start of text).

2.2.2 The starting character shall not be included in the summation.

2.2.3 If an STX character appears after the summation has been started by SOH, then the STX character shall be included in the summation as if it were a text character.

2.3 With the exception of SYN (Synchronous idle), all the characters which are transmitted after the start of the block check summation shall be included in the summation, including the *ETB* (End of transmitted block) or ETX (End of text) control character which signals that the next following character is the block check character.

2.4 No character, SYN or otherwise, shall be inserted between the ETB or ETX character and the block check character.

Supplement No. 7

KOKUSAI DENSHIN DENWA. — *(Contribut.on COM Sp. A—No. 127—October 1967)*

A SYSTEM FOR HIGH-SPEED DIGITAL TRANSMISSION OVER A TELEPHONE CIRCUIT

General

A phase-modulation digital transmission system is scheduled to be put in practical operation over the transpacific submarine cable. This system named Rectiplex provides various combinations of data channels and telegraph channels by using a telephone type channel. It consists of a modem and some intermediate equipments. The former is a modem of multicarrier 8-phase modulation for transmission up to 5184 bits per second over a telephone type channel of 3 kHz bandwidth. The modem provides 54 synchronous 96-baud channels ($96 \times 54 = 5184$). The latter provides data channels and telegraph channels of various modulation rates such as 4800, 2400, 1200, 600, 200, 75, 50 bauds and subdivided 50 bauds by combining or dividing those 96-baud channels. These equipments also perform conversion of telegraph modulation between synchronous or asynchronous modulation of data channels and telegraph channels which are not necessarily in synchronism with the modem. Thus the total signalling rate of 5184 bits per second can be used for various combinations of data channels and telegraph channels. For example:

1) 4800 B$\times$1+50 B$\times$4
2) 1200 B$\times$4+50 B$\times$4
3) 1200 B$\times$1+600 B$\times$2+200 B$\times$4+50 B$\times$34
4) 1200 B$\times$1+75 B$\times$8+50 B$\times$70
5) 75 B$\times$20+50 B$\times$60+½(50 B)$\times$18+¼(50 B)$\times$36
6) 50 B$\times$108

Transmission tests were carried out by using submarine cable circuits between Tokyo/D.D.D. –San Francisco/I.T.T.W.C.[1] and R.C.A.C.[2] and between Tokyo/K.D.D. – Hongkong/C. & W.[3], in February 1966 and in July 1967 respectively. Both tests resulted satisfactorily in an average bit-error rate of 1.2×10^{-6} on 1200-baud channels in the Tokyo–Hongkong test, and in average character-error rates around 1 to 3×10^{-5} on 50-baud channels in both the tests. Rectiplex system is scheduled to inaugurate its operation on K.D.D.–R.C.A.C. circuit and K.D.D.–I.T.T.W.C. circuit in 1968.

Designing considerations and specifications of the system and the test results are outlined hereafter.

Considerations on system design

In considerations on design of Rectiplex system, the most important is use of multi-carrier system in the modem and use of stuffing technique in the intermediate equipments.

1. *Multi-carrier system*

Channel filtering in a general multi-carrier system is achieved by using band-pass filters. In the case of a synchronous phase-modulation system, however, a multi-carrier system without loss of bandwidth is made possible by using integrators for channel filtering instead of band-pass filters, as widely known.

Merits of a multi-carrier system over a single-carrier system are well known, as follows:

1) As modulation rate is low, it has stronger immunity to abrupt disturbances such as impulsive noises on the transmission line.

2) Strict equalization of amplitude and delay characteristics of the transmission line are not necessarily required.

Thanks to employment of the multi-carrier system, Rectiplex system shows satisfactory performance on an existing telephone type channel in spite of high-speed transmission.

In the case of Rectiplex system unique demodulators are used to materialize the method of channel filtering by integrators. In this demodulator, a reference carrier for phase detection is generated by a voltage-controlled oscillator. Phase error of the reference carrier is automatically corrected by the function of a feed-back loop comprising the oscillator, the phase detectors, the integrators, a phase-error detector and back to the oscillator. The phase-error detector makes a voltage proportionate to phase error of the reference carrier by using output voltages of the integrators and demodulated output signals.

2. *Stuffing technique*

In order to connect data circuits and telegraph circuits to Rectiplex modem, conversion of telegraph modulation is necessary because these circuits are not usually synchronized with the modem. For such purposes the function called "stuffing" is generally required.

Three different methods of stuffing are employed in Rectiplex system; one for data circuits, one for 50-baud and 75-baud telegraph circuits and the other for subdivided telegraph circuits.

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[1] I.T.T. World Communications Inc.

[2] R.C.A. Communications, Inc.

[3] Cable and Wireless Ltd.

Specifications

1. *Modem*

1) Carrier frequencies

Information carriers (in Hz)

1T 540	2T 660	3T 780	4T 900	5T 1020	6T 1140	7T 1260	8T 1380	9T 1500
10T 1620	11T 1740	12T 1860	13T 1980	14T 2100	15T 2220	16T 2340	17T 2460	18T 2580

Synchronizing carrier: 2820 Hz

2) Modulation

Type: Differential 4-phase or 8-phase modulation

Modulation scheme for 4-phase:

Polarities of signals		Phase advance in reference to the preceding element (degrees)
Channel 1	Channel 2	
1	1	0
1	0	90
0	0	180
0	1	270

Modulation scheme for 8-phase:

Polarities of signals			Phase advance in reference to the preceding element (degrees)
Channel 1	Channel 2	Channel 3	
1	1	1	0
1	1	0	45
1	0	0	90
1	0	1	135
0	0	1	180
0	0	0	225
0	1	0	270
0	1	1	315

Synchronizing carrier: phase inversion at 96 bauds.

Synchronism in modulation: modulation elements of all carriers coincide with each other.

3) Channel

Modulation rate: synchronous 96 bauds with accuracy better than 1×10^{-6}.

Number of channels: 36 (4-phase) to 54 (8-phase).

Type: duplex channels.

4) Automatic frequency correction

Type: heterodyne method.

Pilot: upper side-band of the synchronizing carrier ($2820+48 = 2868$ Hz).

Bandwidth for the correcting signal: $\pm$ 45 Hz.

Delay of correction: delay is given to information carriers in order to compensate delay of correcting signal due to its extraction.

5) Delay equalization: equalizers for delay characteristics of the transmission line are internally equipped.

2. *Intermediate equipment for data circuits*

1) Type of data signal: signal with equal element length regardless of start-stop or synchronous type

2) 96-baud channels: 13 channels of 96 bauds are used for one 1200-baud channel. For different modulation rates of data circuits, proportionately different numbers of 96-baud channels are required.

3) Stuffing: every 156 elements, a frame, of the synchronous signal to the modem includes 5 or 7 idle elements.

4) Arrangements of elements in a frame:

(Frame A) I_{12} Z I_{13} I_{12} A I_{13} I_{12} A I_{13} I_{12} A I_{13} I_{12} A I_{13} I_{12} A I_{12} A

(Frame D) I_{12} A I_{13} I_{12} Z I_{13} I_{12} Z I_{13} I_{12} Z I_{13} I_{12} Z I_{13} I_{13} I_{13}

where: I_{12}: 12 information elements

I_{13}: 13 information elements

Z : an idle element of Z polarity

A : an idle element of A polarity

5) Frame phasing: automatic.

6) Tolerance for error in modulation rate: $\pm$ 0.66%.

3. *Intermediate equipment for 75-baud telegraph circuits*

1) Type of telegraph signal: 75-baud start-stop signal of 5-unit code.

2) Character rate: 617-1/7 characters per minute or lower.

3) Stuffing: an idle element of Z polarity is occasionally added after the stop element of character.

4) Multiplex

Method: four channels of 75 bauds are combined into three channels of 96 bauds by time-division multiplex

Designation of channels: channels A, B, C and D

Arrangement of elements:

96-baud channel 1: AAADAAADAAAD . . . .

96-baud channel 2: DBBBDBBBDBBB . . . .

96-baud channel 3: CDCCCDCCCDCC . . . .

where A, B, C and D show elements of channels A, B, C and D respectively.

5) Synchronization: synchronized by timing pulses from Rectiplex modem.

6) Phasing

Type: automatic or manual phasing.

Phasing channel: one channel A for whole 75-baud channels in the system.

Phasing signal: reversals at 72 bauds.

External phasing: phasing by phasing pulses from intermediate equipment for subdivided circuits as available. Phasing channel of 75 bauds is unnecessary.

4. *Intermediate equipment for 50-baud telegraph circuits*

1) Type of telegraph signal: 50-baud start-stop signal of 5-unit code.

2) Character rate: 411-3/7 characters per minute or lower.

3) Stuffing: an idle element of Z polarity is occasionally added after the stop element of character.

4) Multiplex

Method: two-channel multiplex by element interleaving.

Designation of channels: channels A and B.

5) Polarity inversion: polarities of 3rd, 4th and 5th elements of telegraph signal on channel B are inverted on the aggregate signal for preserving privacy in case of failure of phase correction.

6) Synchronization: synchronized by timing pulses from Rectiplex modem.

7) Phasing

Type: automatic or manual phasing.

Phasing channel: one channel B for whole 50-baud channels in the system.

Phasing signal: reversals at 48 bauds.

External phasing: phasing by phasing pulses from intermediate equipment for subdivided circuits is available. Phasing channel of 50 bauds is unnecessary.

8) Transmission of telex signals: types A, B and C standardized in Recommendations U.1 and U.11 of the C.C.I.T.T.

5. *Intermediate equipment for subdivided telegraph circuits*

1) Type of telegraph signal: 50-baud start-stop signal of 5-unit code.

2) Character rate: 205-2/7 characters per minute or lower on a half-speed channel and 102-6/7 characters per minute or lower on a quarter-speed channel.

3) Stuffing: an idle character of 7 successive Z elements is occasionally inserted.

4) Multiplex

Type: eight-channel time-division multiplex.

Designation of channels: channels A1, A2, A3, A4, B1, B2, B3 and B4. In case of half speed, A3, A4, B3 and B4 are annexed to A1, A2, B1 and B2 respectively.

Multiplexing method: character interlacing for multiplex of channels A1, A2, A3 and A4, and for multiplex of channels B1, B2, B3 and B4; element interleaving for multiplex of A-channel group and B-channel group.

5) Polarity inversion: Polarities of elements of telegraph signals are inverted in accordance with the inversion pattern particular to each quarter-speed channel, and further polarity inversion is given to whole B-channel group. The inversion pattern is as follows:

Channel \ Element	1	2	3	4	5
A1/B1	N	N	N	N	N
A2/B2	I	I	N	I	I
A3/B3	N	N	I	I	I
A4/B4	I	I	N	N	N

where N: normal polarity
I : polarity inversion

For a half-speed channel, the inversion pattern of the channel with smaller suffix is applied to both channels.

6) Synchronization: synchronized by timing pulses from Rectiplex modem.

7) Phasing

Type: automatic or manual phasing.

Phasing channel: one B4 channel and another B4 channel as a spare phasing channel if possible for whole system.

Phasing signal: ZZZZAAZ at the aggregate signal.

Test results

1. *Tokyo/K.D.D.–San Francisco/I.T.T.W.C. and R.C.A.C. test*

Date: February 1966.

Transmission line used: Tokyo – San Francisco via T.P.C. cable and Hawaii cable.

Modulation: four-phase and eight-phase modulation.

Character error rate on 50-baud telegraph channels:

	Total hours	Character error rate
TOK→SF (4-phase)	117 (58×2)	0.56×10^{-5}
SF→TOK (4-phase)	119 (59×2)	0.59×10^{-5}
TOK→SF (8-phase)	117 (58×2)	0.99×10^{-5}
SF→TOK (8-phase)	110 (55×2)	1.26×10^{-5}

Each error rate is an average of two channels.

2. *Tokyo/K.D.D.–Hongkong / C. & W. test*

Date: July 1967.

Transmission line used: Tokyo–Hongkong via T.P.C. cable and SEACOM cable.

Modulation: eight-phase modulation.

Bit error rate on 1200-baud channels:

	Total hours	Bit error rate
TOK→HKG HKG→TOK	130 120	1.2×10^{-6} 1.2×10^{-6}

Character error rate on 50-baud channels:

	Total hours	Character error rate
HKG→TOK→HKG (loop back at 50-baud circuit in Tokyo)	136	4.19×10^{-5}
HKG→TOK	136	2.66×10^{-5}

Supplement No. 8

TELE-SIGNAL CORPORATION (U.S.A.). — *(Contribution COM Sp. A—No. 182—March 1968)*

COMBINED OPERATION OF A 1200-2400-BAUD DATA TRANSMISSION CHANNEL AND SEVERAL 50-75-BAUD TELEGRAPH CHANNELS IN A SINGLE VOICE-BAND

1. For economic reasons it is desirable to combine on leased cable circuits a number of telegraph channels along with a 1200-2400-baud data channel.

2. The standardized 600-1200-baud modem according to Recommendation V.23 may be combined with a number of lower-speed frequency modulated telegraph channels specified under Recommendations R.35 and R.36.

3. A preferred method of operation is to allocate five 50-75-baud channels (1-5) below and five 50-75-baud channels (18-22) above the spectrum occupied by the 600-1200-baud modem centred on 1700 Hz as shown in Figure 1 b.

Assuming the bandpass filters for the 600-1200-baud modem are made with adequate attenuation characteristics, there will be little interference between the channels.

This has been proven by thorough testing and also by practical operation on long-haul cable circuits.

4. Improvements in bandpass filter production allow the nominal 50-baud channels to be operated at 75 bauds with only a negligible increase of distortion of about 2%.

5. By using multi-level phase modulation, speeds higher than 1200 bauds are possible in the available spectrum centred on 1700 Hz.

For example (Figure 1 a), with eight-phase modulation it is possible to operate synchronously at 2000 bauds along with twelve 50-75-baud channels on the same voice-channel. Synchronous operation at 2400 bauds is possible along with ten 50-75-baud channels. Such systems have been tested with satisfactory results (Figure 1 c).

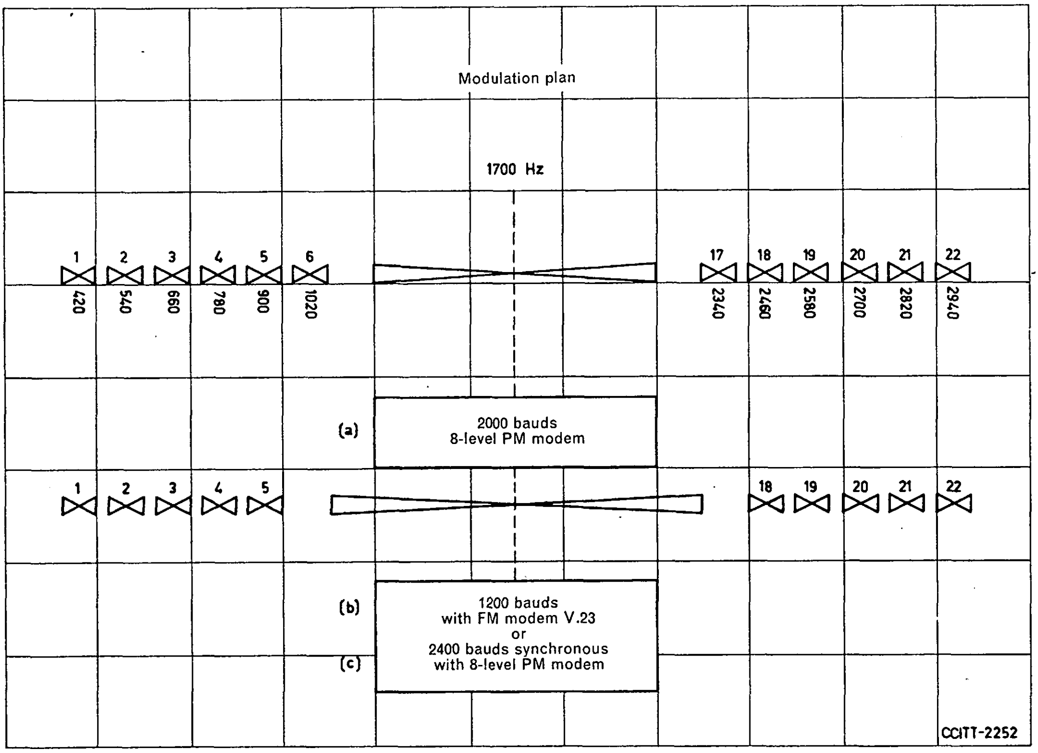


FIGURE 1. — **Combined operation of FMVFT 50-75-baud and 1200, 2000, or 2400 bauds on one voice-channel**

Supplement No. 9

UNITED KINGDOM. — *(Extract from contribution COM Sp. A—No. 27—October 1965)*

TESTS FOR DATA TRANSMISSION SYSTEMS ON TELEPHONE-TYPE CIRCUITS

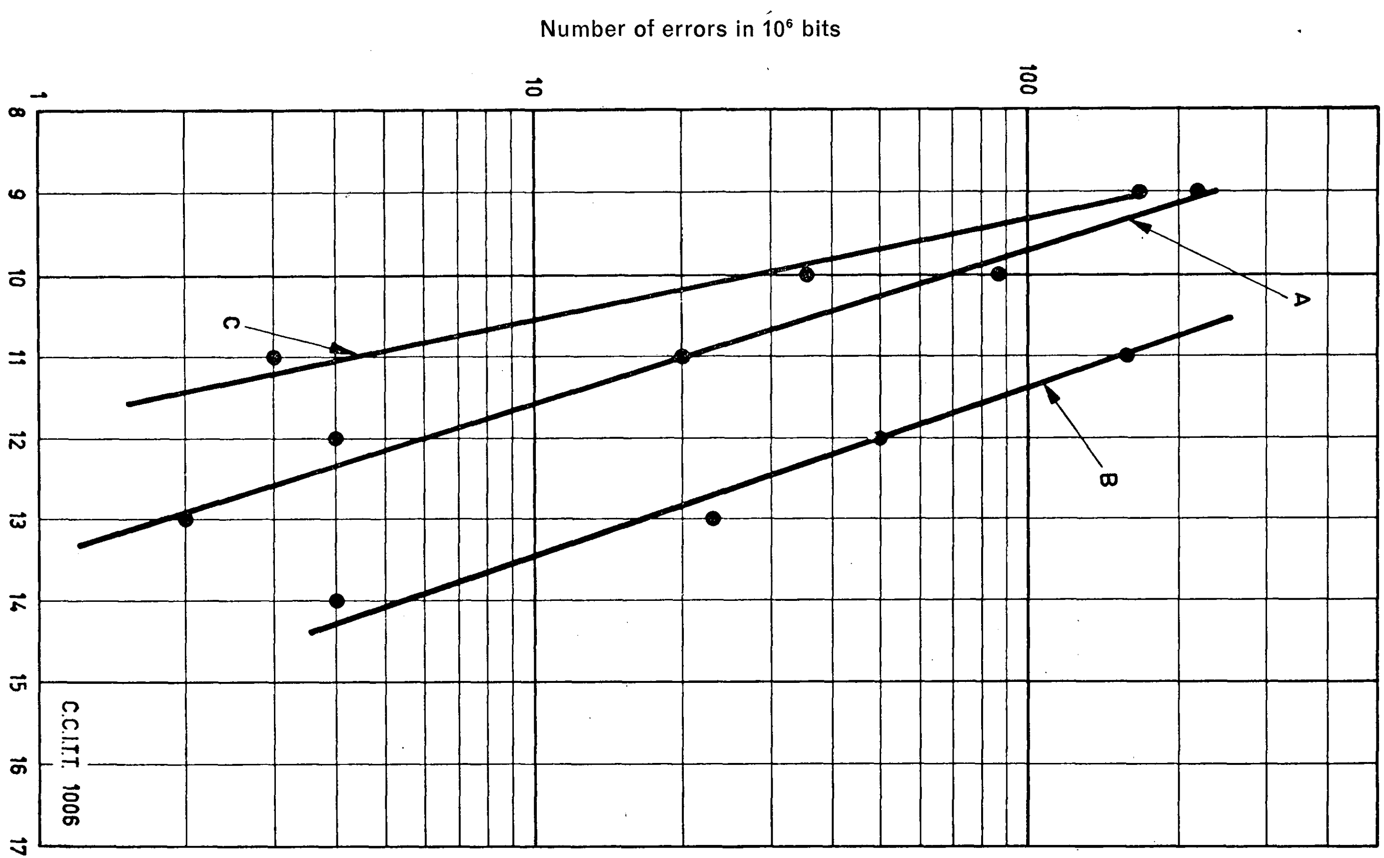
Since January 1965 the United Kingdom Administration has had available for hire by customers modems for the transmission of data over telephone-type circuits and the public telephone switching network. The modem complies with Recommendation V.23 and is known in the United Kingdom as Datel modem No. 1A. The modem has been subjected to laboratory tests to determine its performance as measured in terms of bit errors in the face of known signal impairment introduced by adding uniform spectrum random noise limited to the band 175 to 3300 Hz to the line signals after they have been transmitted through artificial line networks simulating:

a) Three telephone carrier channels connected in tandem, or

b) 100 miles of standard loaded cable (20/88/1.136).

Tests have been made at signalling rates of 75 bits/s, 600 bits/s and 1200 bits/s. Typical results at 1200 bits/s are shown in Figure 1.

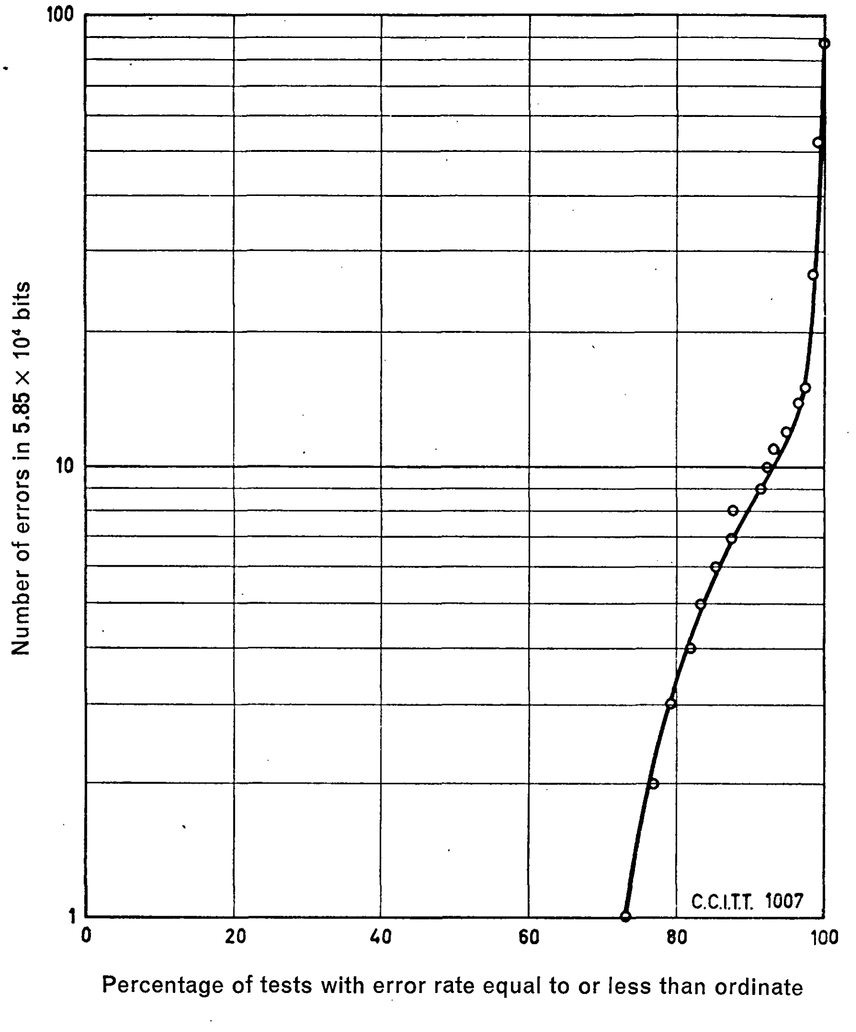
Further tests have been made over connections set up over the public telephone switched network at signalling rates of 75, 600, 1000 and 1200 bits per second and the results of these tests are summarized in Figures 2, 3, 4 and 5. These tests have been made using a tester of the type described in the United Kingdom reply to point W.



A = Back to back
B = Three telephone carrier channels
C = 100 miles standard loaded cable (20/88/1-126)

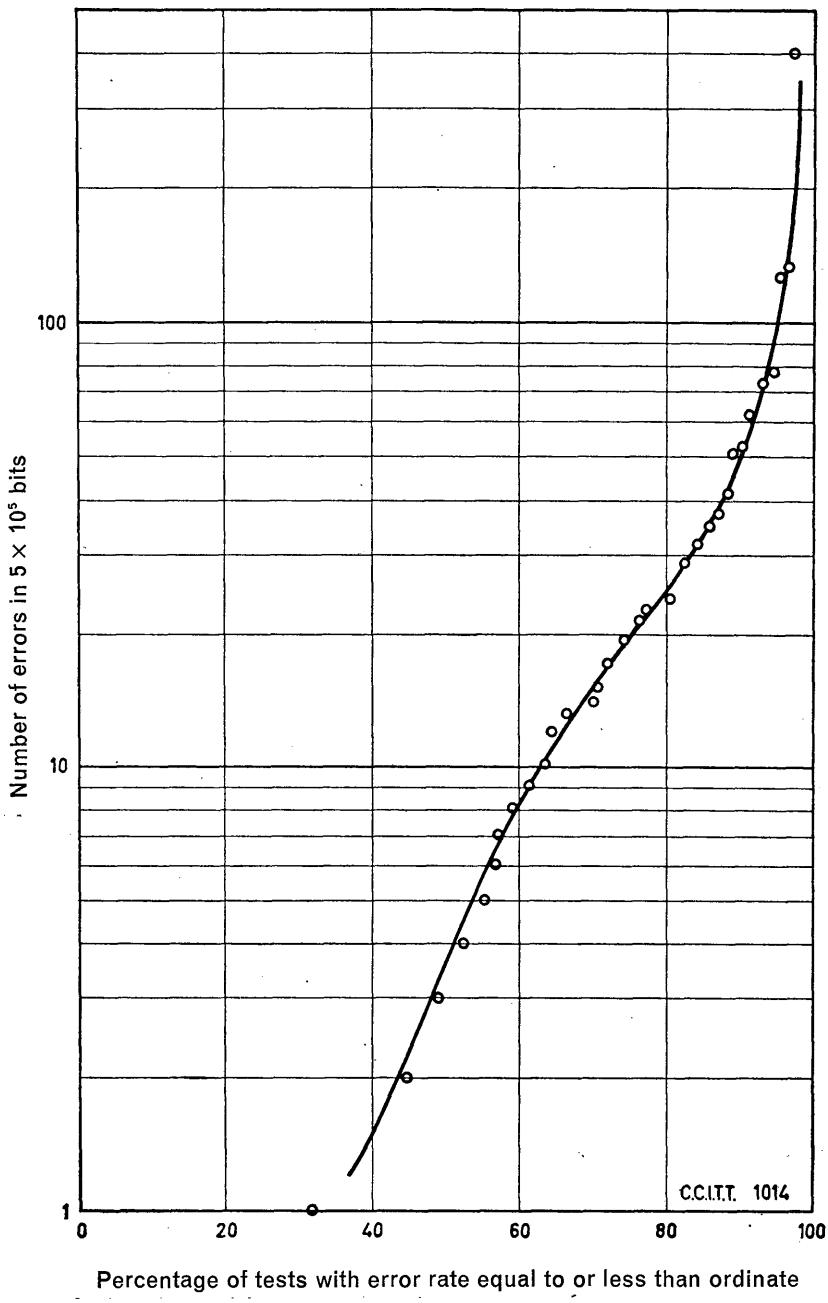
Data sig. rate — 1200 bits per second mixed signal
Modem setting — A2 (1200 bits per second)
Noise bandwidth limited to 175-3300 Hz (3 dB points)

FIGURE 1. — **Errors versus signal/uniform spectrum random noise ratio for Datel modem No. 1A connected via various line simulators**



Number of connections tested: 49
Number of tests: 117

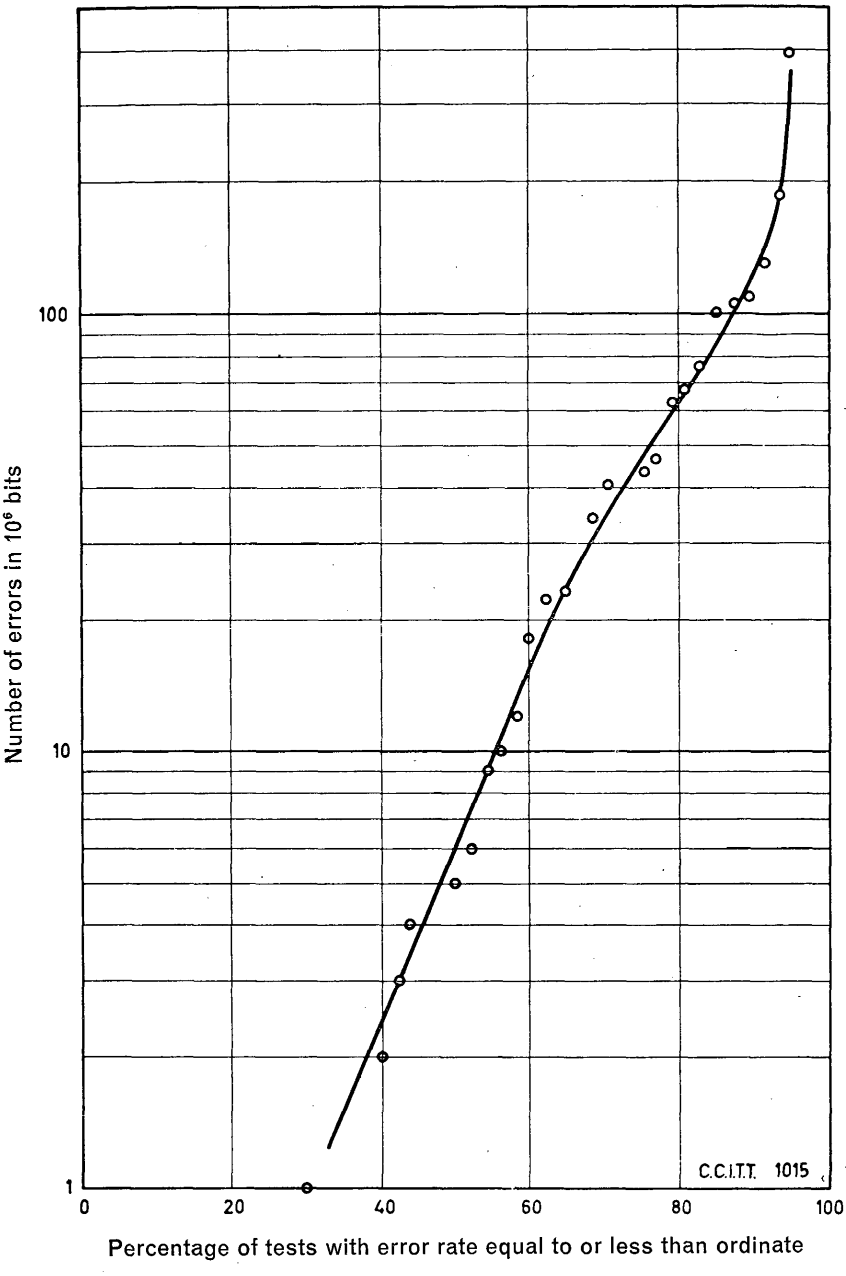
FIGURE 2. — **Datel modem No. 1A error rate at 75 bits per second over public telephone network**



Number of connections tested: 66 [1]
Number of tests: 97
Modem setting — A1 mode (600 bits per second)

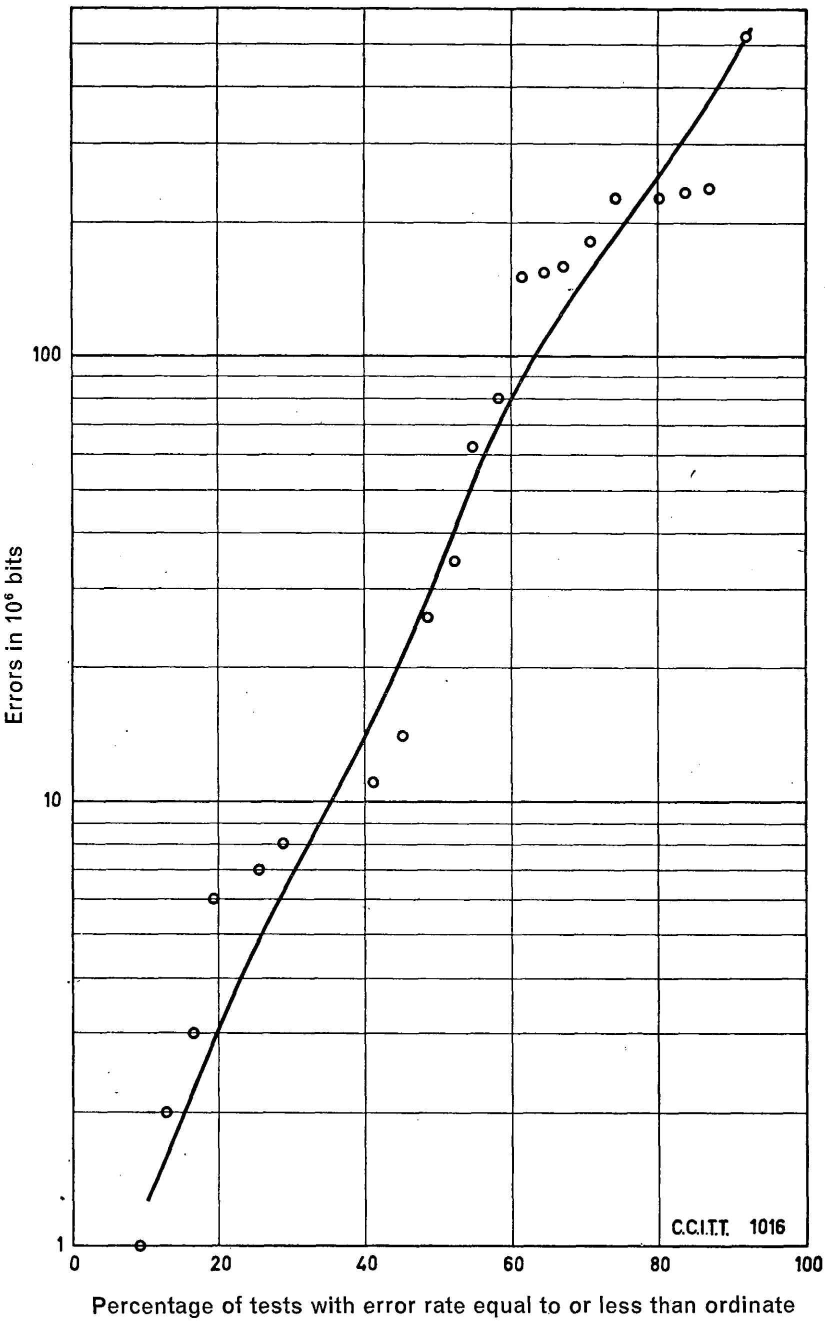
[1] One connection would not work at 600 bits per second.

FIGURE 3. — **Datel modem No. 1A error rate at 600 bits per second over public telephone network**



Number of connections tested: 28
Number of tests: 48
Modem setting — A2 mode (1200 bits per second)

FIGURE 4. — **Datel modem No. 1A error rate at 1000 bits per second over public telephone network**



Number of connections tested: 31 [1]
Number of tests: 31
Modem setting — A2 mode (1200 bits per second)

[1] Two connections would not work at 1200 bits per second. One connection would not work at 1200 or 600 bits per second.

FIGURE 5. — **Datel modem No. 1A error rate at 1200 bits per second over public telephone network**

I.B.M. WORLD TRADE EUROPE CORPORATION. — *(Contribution COM Sp. A—No. 33—October 1965)*

DATA-TRANSMISSION TESTS ON A MULTIPOINT NETWORK (200, 600, 1200 AND 1800 BITS PER SECOND)

Introduction

This contribution describes data-transmission tests which were carried out in France on a multipoint network of leased lines. The results obtained provide information on the possibilities of operating with data signalling rates of 200, 600, 1200 and 1800 bits per second.

Contribution COM Sp. A—No. 64 (1960-1964 period) described similar tests for data signalling rates of 2000 bits per second (Supplement No. 37 to the *Blue Book*, Volume VIII).

Organization of tests

The tests were conducted from 25 May to 27 June 1964 by the I.B.M. France Company in co-operation with the French Postal and Telecommunication Administration.

The I.B.M. 3976 and 3977 Modems were used as well as the portable I.B.M. 3998, Model 1 (see brief description of equipments in Annex II).

The network was of the multipoint type, made up of leased lines (see diagram described below).

The main circuit was established on a four-wire basis. The terminations in Paris and Nice were successively two-wire and four-wire. Local drops were made at Marseille and Toulon. Two drops were made at Le Pontet, one towards Avignon and the other towards Montpellier; the latter was connected to local pairs to simulate a distant subscriber station.

Multipoint network

A multipoint network generally consists of a main line and of drops made by the P.T.T. Administration at its technical centres (telephone exchanges, LGD (long-distance centres), etc.).

With this type of network, it is possible to set up both "centralized" and "non-centralized" data-transmission systems:

— *centralized multipoint system*

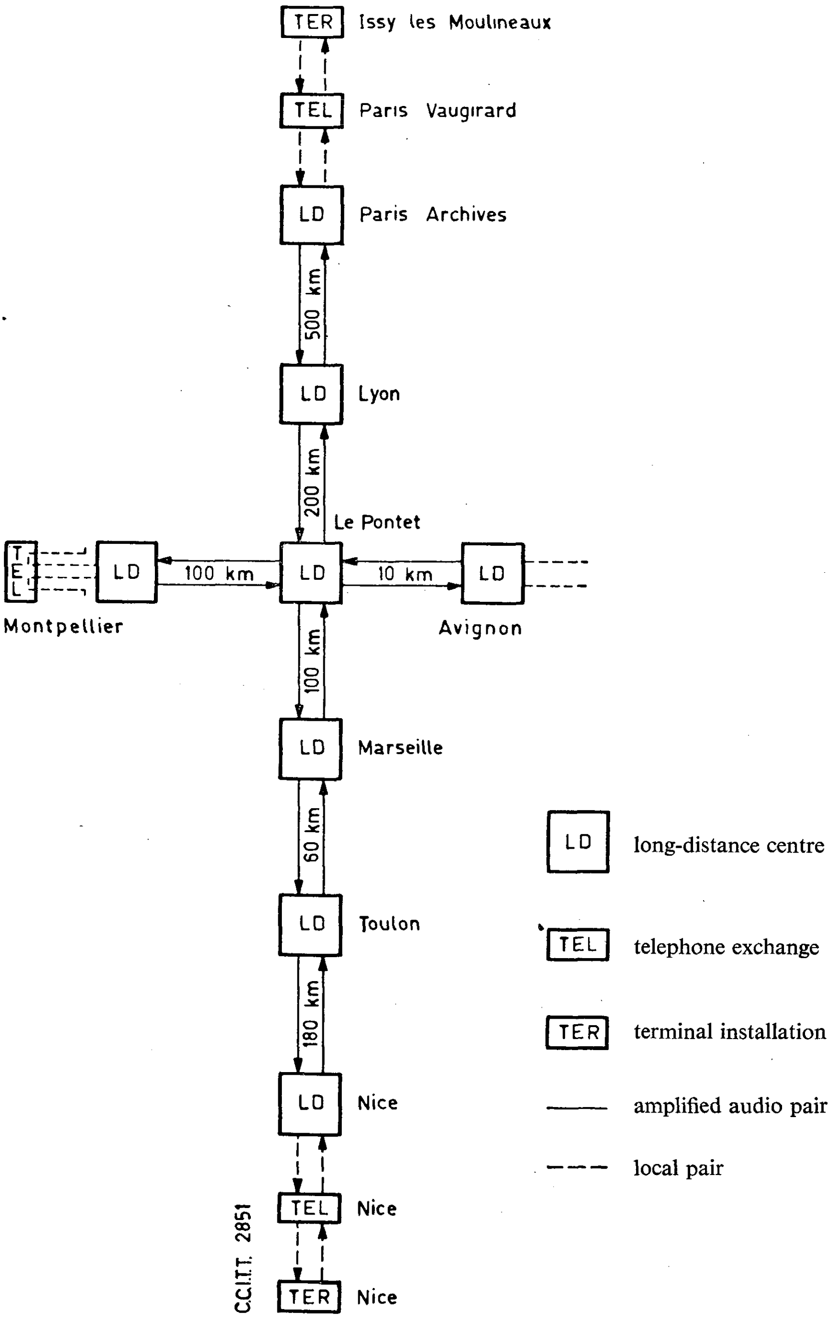
all terminal equipments receive the signals transmitted by a central equipment but only the latter receives the signals transmitted by all the other equipments;

— *non-centralized multipoint system*

all terminal equipments can function as central equipment. All the signals transmitted by one equipment are received by all the other equipments.

Test objectives

1. To measure the electrical characteristics of the network.
2. To carry out transmission tests without drops, so that comparisons can be made.
3. To carry out transmission tests on the multipoint network at 200 bits per second with two-wire and four-wire terminations in Paris and Nice.


TER Issy les Moulineaux
TEL Paris Vaugirard
LD Paris Archives
500 km
LD Lyon
200 km
Le Pontet
TEL
LD
100 km
LD
10 km
LD
Montpellier
Avignon
100 km
LD Marseille
60 km
LD Toulon
180 km
LD Nice
TEL Nice
TER Nice
LD long-distance centre
TEL telephone exchange
TER terminal installation
amplified audio pair
local pair
C.C.I.T.T. 2851

4. To carry out the same tests at 600, 1200 and 1800 bits per second.

5. To analyse outages and anomalies.

Conclusions

The tests showed that it is possible to transmit data over a multipoint network and that, in some cases (very extensive networks), it is preferable to use four-wire terminations, even with relatively low data signalling rates (200 bits per second).

The performance of the four-wire terminated network was distinctly better than that of the two-wire terminated network. Circuit adjustments are less critical and certain difficulties resulting from echo phenomena are eliminated.

Test results

1. *Electrical characteristics of the network*

Two patterns were used to determine the electrical characteristics and to carry out the transmission tests described below:

a) Issy-les-Moulineaux:	two-wire termination
Le Pontet-Montpellier:	drop loaded at 600 ohms at Montpellier LGD
Marseille:	local drop loaded at 600 ohms
Toulon:	local drop loaded at 600 ohms
Nice:	two-wire termination

Note. — Measurements were made in Paris, Toulon and Nice.

b) Issy-les-Moulineaux:	four-wire termination
Le Pontet-Montpellier:	four-wire drop
Le Pontet-Avignon:	four-wire drop
Montpellier-Subscriber:	two local pairs looped for representing a distant subscriber station
Marseille:	local drop loaded at 600 ohms
Toulon:	local drop loaded at 600 ohms
Nice:	four-wire termination

Note. — Measurements were made in Paris, Montpellier, Avignon, Toulon and Nice.

Annex I shows diagram of a drop.

Drops were loaded at 600 ohms when the test equipments were not connected.

2. *Transmission tests without drop*

These tests were carried out on a Paris–Nice circuit with four-wire terminations.

a) *Level of circuit noise*

Data signalling rate (bits/s)	Frequency (Hz)	Circuit noise (dBm)		
		Lower band (Hz)	Band used (Hz)	Upper band (Hz)
200	1080 } 1750	$f < 700$ – 31.8	$700 \leqslant f \leqslant 2150$ – 43.6	$f > 2150$ – 50
600	1500	$f < 1200$ – 32.3	$1200 \leqslant f \leqslant 1800$ – 51.8	$f > 1800$ – 53
1200	1500	$f < 900$ – 31.8	$900 \leqslant f \leqslant 2100$ – 43.6	$f > 2100$ – 50
1200	1800	$f < 1200$ – 31.8	$1200 \leqslant f \leqslant 2400$ – 45.3	$f > 2400$ – 53

b) *Receive level* (600 and 1200 bits/s)

Transmitting station	Receiving station	Receive level dBm
Nice	Paris	– 5 to – 6
Paris	Nice	– 14

c) *Error rate*

Transmitting station	Receiving station	Data signalling rate (bits/s)	Error rate	
			bit	message (511 bits)
Paris	Nice	600	$7.4 \cdot 10^{-7}$	$2 \cdot 10^{-4}$
Paris	Nice	1200	$2.0 \cdot 10^{-5}$	$3 \cdot 10^{-3}$
Nice	Paris	600	0	0
Nice	Paris	1200	0	0

3. *Transmission tests at 200 bits per second with drop*

3.1 *Two-wire termination* (25 May to 4 June 1964)

a) *Level of circuit noise*

Place	Time	$f < 700$ Hz (dBm)	700 Hz $< f <$ 2150 Hz (dBm)	$f > 2150$ Hz (dBm)
Paris	11.30	— 49	— 60	— 65
	16.30	— 37	— 56	— 62
Toulon	11.30	— 38 to — 39	— 44 to — 48	— 45 to — 47
	16.30	— 39.5 to — 41.5	— 44 to — 47	— 45 to — 48
Nice	16.30	— 35	— 44	— 43
	8.00	— 35.5	— 44	— 43

Noise impulses were observed in Toulon in the upper band: impulses at approximately 5 kHz (20 to 30 Hz, 120 mV peak to peak).

b) *Receive level*

Transmitting station	Receiving station	Receive level (dBm)
Paris	Nice	— 22 to — 24
Paris	Toulon	— 7 to — 7.5
Nice	Paris	— 25 to — 26.5
Toulon	Paris	— 20.5 to — 22

Sending level was ajdusted at — 6 dBm.

c) *Error rate*

Transmitting station	Receiving station	Frequency (Hz)	Error rate	
			bit	message (511 bits)
Paris	Toulon	1750	0	0
Paris	Nice	1750	$1.4 \cdot 10^{-7}$	$7.1 \cdot 10^{-5}$
Nice	Paris	1080	$9.6 \cdot 10^{-7}$	$5.0 \cdot 10^{-4}$
Toulon	Paris	1080	$1.1 \cdot 10^{-6}$	$2.1 \cdot 10^{-4}$

Transmission time was such that, at 200 bits per second, more than 7×10^6 bits were transmitted.

3.2 *Four-wire termination* (5 June to 27 June 1964)

a) *Level of circuit noise*

Identical to that of the two-wire termination.

b) *Receive level*

Transmitting station	Receiving station	Frequency (Hz)	Receive level (dBm)
Paris	Nice	1750	– 11 to – 15
Paris	Avignon	1750	– 9.2 to – 10.5
Paris	Montpellier	1750	– 11.5 to – 13
Nice	Paris	1080	– 10 to – 11.5
Avignon	Paris	1080	– 8 to – 9.5
Montpellier	Paris	1080	– 8

Send level was adjusted at – 6 dBm.

c) *Error rate*

Transmitting station	Receiving station	Frequency (Hz)	Error rate	
			bit	message (511 bits)
Paris	Nice	1750	0	0
Paris	Avignon	1750	0	0
Paris	Montpellier	1750	0	0
Nice	Paris	1080	0	0
Avignon	Paris	1080	$5.7 \cdot 10^{-7}$	$1.5 \cdot 10^{-4}$
Montpellier	Paris	1080	$1.0 \cdot 10^{-6}$	$1.5 \cdot 10^{-4}$

Transmission time was such that, at 200 bits per second, more than 7×10^6 bits were transmitted.

4. *Transmission tests at 600, 1200 and 1800 bits per second*

4.1 *Two-wire termination* (25 May to 4 June 1964)

a) *Level of circuit noise*

Place	Carrier frequency (Hz)	Data signalling rate (bits/s)	Time	Noise level (dBm)		
				Lower band	Band used	Upper band
Paris	1500	600	11.30 16.30	$f < 1200$ Hz – 49 – 37	$1200 < f < 1800$ – 63 – 62	$f > 1800$ Hz – 65 – 62
	1500	1200	11.30 16.30	$f < 900$ – 49 – 37	$900 < f < 2100$ – 60 – 56	$f > 2100$ – 65 – 62
	1800	1200	11.30 16.30	$f < 1200$ – 49 – 37	$1200 < f < 2400$ – 63 – 62	$f > 2400$ – 65 – 62
Toulon	1500	600	11.30 16.30	$f < 1200$ – 38.5 – 40	$1200 < f < 1800$ – 45 to – 50 – 49	$f > 1800$ – 46 – 43 to – 47
	1500	1200	11.20 16.30	$f < 900$ – 38.5 – 40.5	$900 < f < 2100$ – 44 to – 49 – 44 to – 48	$f > 2100$ – 46 – 45 to – 48
	1800	1200	11.30 16.30	$f < 1200$ – 38.5 – 40.5	$1200 < f < 2400$ – 46 to – 50 – 46 to – 49	$f > 2400$ – 45 to – 47 – 45 to – 48.5

Noise impulses were observed at Toulon in the upper band: impulses at approximately 5 kHz (20 to 30 Hz, 120 mV peak to peak).

b) *Receive level*

Transmitting station	Receiving station	Data signalling rate (bits/sec.)	Carrier frequency (Hz)	Receive level (dBm)
Paris	Nice	600 1200	1500	– 23
Paris	Toulon	600 1200	1500	– 9
Nice	Paris	600 1200	1500	– 27
Toulon	Paris	600 1200	1500	– 23

Send level, in all cases, was adjusted at –6 dBm.

c) *Echo level on the line*

The echo caused by imperfect balancing of the terminations used in Paris and in Nice was measured as follows:

— modulated signal transmitted in Nice, received in Paris, echo received in Toulon;

— modulated signal transmitted in Toulon, received in Paris, echo received in Nice.

Transmitting station	Carrier frequency (Hz)	Data signalling rate (bits/s)	Echo level received (dBm)	
			in Nice	in Toulon
Nice	1500	600		— 22
Toulon	1500	600	— 35	
Nice	1500	1200		— 22
Toulon	1500	1200	— 36	
Nice	1800	1200		— 22
Toulon	1800	1200	< — 40	

d) *Error rate*

Transmitting station	Receiving station	Data signalling rate (bits/s)	Error rate	
			bit	message (511 bits)
Paris	Nice	600	0	0
Paris	Toulon	600	0	0
Nice	Paris	600	$1.1 \cdot 10^{-5}$	$2.0 \cdot 10^{-4}$
Toulon	Paris	600	$5.8 \cdot 10^{-7}$	$1.5 \cdot 10^{-4}$

Transmission time was such that, at 600 bits per second, more than 2×10^7 bits were transmitted.

4.2 *Four-wire termination* (5 June to 27 June 1964)

a) *Level of circuit noise*

Identical to that of the two-wire termination.

b) *Receive level*

Transmitting station	Receiving station	Data signalling rate (bits/s)	Receive level (dBm)
Paris	Nice	600	− 15.5 to − 17
		1200	− 14.5 to − 17
	Toulon	600	− 11 to − 12
		1200	
	Avignon	600	− 10.5 to − 11.5
		1200	− 12
	Montpellier	600	− 9.1 to − 9.3
		1200	− 13 to − 15
Nice	Paris	600	− 10.5 to − 11
		1200	− 9 to − 12
Toulon		600	− 11.5 to − 13.5
		1200	
Avignon		600	− 9 to − 12.5
		1200	− 11.5 to − 12
Montpellier		600	− 8 to − 12
		1200	− 11.5

Send level was adjusted at −6 dBm; the carrier frequency was 1500 Hz.

c) *Error rate*

Transmitting station	Receiving station	Data signalling rate (bits/s)	Error rate	
			bit	message (511 bits)
Paris	Nice	600	$6.3 \cdot 10^{-6}$	$1.5 \cdot 10^{-3}$
Paris	Nice	1200	$3.3 \cdot 10^{-5}$	$3.6 \cdot 10^{-3}$
Paris	Toulon	1200	$3.4 \cdot 10^{-7}$	$2.2 \cdot 10^{-5}$
Paris	Avignon	600	0	0
Paris	Avignon	1200	0	0
Paris	Montpellier	600	0	0
Paris	Montpellier	1200	0	0
Nice	Paris	600	$8.1 \cdot 10^{-7}$	$5.9 \cdot 10^{-5}$
Nice	Paris	1200	$4.4 \cdot 10^{-7}$	$1.1 \cdot 10^{-4}$
Toulon	Paris	1200	$2.2 \cdot 10^{-5}$	$1.4 \cdot 10^{-4}$
Avignon	Paris	600	0	0
Avignon	Paris	1200	$4.6 \cdot 10^{-7}$	$7.1 \cdot 10^{-5}$
Montpellier	Paris	600	0	0
Montpellier	Paris	1200	$3.1 \cdot 10^{-7}$	$6.6 \cdot 10^{-5}$

The number of bits transmitted was greater than:

$2 \cdot 10^7$ bits at 600 bits per second

$4 \cdot 10^7$ bits at 1200 bits per second

d) *Transmission tests at 1800 bits per second*

After the tests at 600 and 1200 bits per second with four-wire termination, tests were made at 1800 bits per second.

An equalizer was added at Nice and, using the multipoint pattern, 10^8 bits were transmitted from Paris.

The error rate was as follows:

" bit " error rate: $6.4 \cdot 10^{-6}$

" message " error rate: $1.2 \cdot 10^{-3}$

5. *Outages and anomalies*

5.1 *Outages*

Test conditions were such that only outages lasting longer than three minutes could be recorded.

Date	Beginning of outage	Length of outage (h)	Remarks
12.6.64	9.45	0.5	Outage on Paris → Nice and Nice → Paris
	13.00	2	Outage on Paris → Nice and Nice → Paris
17.6.64	12.45	2	Outage on Avignon → Paris and Paris → Avignon
23.6.64	10.45	1	Outage on Montpellier → Paris

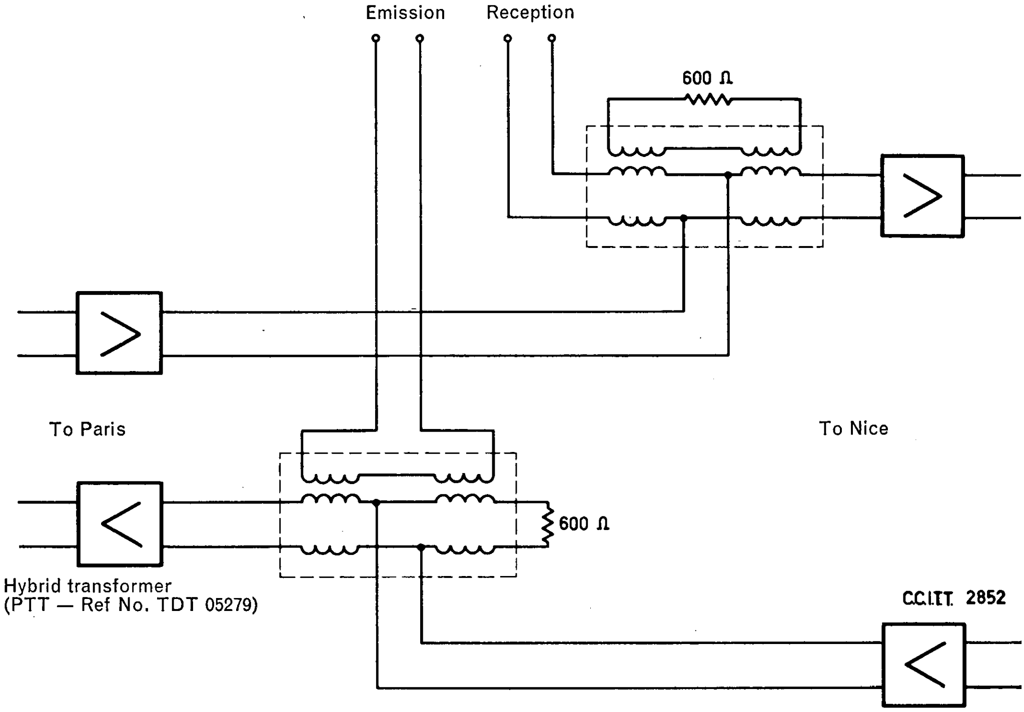
5.2 *Anomalies*

Two anomalies were observed during the test month.

Date	Time	Anomaly
16.6.64	8.00 to 10.00	Variation in receive level of 10 dB
27.6.64	8.00 to 9.00	Receive level unusually low, became normal again after a burst of errors

ANNEX 1

Four-wire drops



ANNEX II

Test equipment

The tests were performed with three I.B.M. equipments:

a) *I.B.M. Modem 3976*

Data signalling rate	200 bits per second, duplex in accordance with C.C.I.T.T. Recommendation V.21 channel No. 1 mean frequency 1080 Hz channel No. 2 mean frequency 1750 Hz
Modulation:	two-condition frequency, non-coherent, condition coding
Transmitter:	send level adjustable between 0 and – 10 dBm at the input of the subscriber's line
Receiver:	receive level variable between 0 and – 40 dBm.

b) *I.B.M. Modem 3977*

Data signalling rate	600 to 2400 bits per second (equalizing possible above 1200 bits per second)
Modulation:	two-phase, transition or condition coding, non-coherent
Transmitter:	carrier frequency, 1500 Hz or 1800 Hz, phase-modulated, send level adjustable between 0 and – 10 dBm at the input of the subscriber's line
Receiver:	receive level variable between 0 and – 40 dBm. The phase indetermination (π), when data are condition coded, is cleared by sending series of "1" for approximately 50 ms.

c) *I.B.M. Teletest 3998*

Transmission:	a generator supplies adjacent blocks of 511 binary digits constituting a message. These blocks are generated by a looped shift register with nine positions.
Reception:	The blocks received are compared with identical blocks generated locally and suitably synchronized.
Errors:	bits and messages are counted. Send and receive equipments include a quartz-stabilized clock.

Supplement No. 11

CHILE TELEPHONE COMPANY. — *(Extract from Contribution COM Sp. A—No. 50—November 1965)*

TEST MEASUREMENTS

a) *Parallel operation*

A series of tests has been carried out to determine the error rate and error distributions when data are transmitted in parallel form over leased telephone connections. These connections included both carrier and loaded circuits. The test equipment comprised existing frequency-shift-modulated

V.F. telegraph equipment, which was designed to transmit 75 eight-bit characters per second.

The output from the error-recording equipment permitted subsequent analysis by computer. The analyses included investigations into the effect of time of day, the burst lengths, the interaction between channels, and a comparison between serial and parallel characters.

The tests comprised the transmission of 2×10^9 bits, during which 8×10^3 errors were recorded. The character error rate was approximately 1.34 in 10^5. It was found that the outside channels had a much higher error rate than the centre channels (an increase of 50%).

The overall bit error rates for the serial and parallel modes of transmission were found to be rather similar, but the error patterns were very different, as can be seen from the following table relating to test transmissions including 8056 errors.

Number of errors per character	Number of faulty parallel characters	Number of faulty serial characters
1	1297	2578
2	548	689
3	476	394
4	450	301
5	261	215
6	126	84
7	42	17
8	10	2
	3210	4280

In spite of the much larger number of character errors with serial transmission, there were more undetected errors with the parallel mode if a single parity bit only had been used for detection. Serial characters showed a tendency for errors to occur in adjacent bits.

With parallel characters some channels were more prone to errors than others. It was found that for some of the channels in the parallel mode about 33% of the errors were not accompanied by other errors in the same character; for other channels this figure was less than 10%.

Theoretically it would seem possible to counteract the different error liabilities of the parallel channels by some element transposition at the two terminations. However, if any processing is to be employed, it seems preferable to use the serial mode, which is more flexible and more easily protected by error-detection and error-correction arrangements.

b) *2000 bits/second phase-modulated operation*

During 1964 a series of tests were carried out on four-phase modems operating at 2000 bits/second. Operation over leased carrier-type circuits caused no difficulty, but operation over loaded circuits proved difficult as the phase distortion and attenuation were too severe and upset the receiver modem synchronization.

The tests involved the transmission of about 1.4×10^9 bits in the form of 2048 bit blocks. About 9000 errors were recorded, giving an overall bit error rate of 6.5 in 10^6. The block error rate was found to be 1 in 1900. The results showed the usual variation with time of day and the usual nature of the errors, 60% of the errors being present in bursts of less than 20 bits.

Supplement No. 12

ITALIAN ADMINISTRATION. — *(Extract from Contribution COM Sp. A—No. 53—November 1965)*

LINE TESTS OVER ITALIAN TELEPHONE NETWORK

Some extensive line tests were made on the Italian public telephone network since year 1961 for over 2000 hours of effective transmission in collaboration among the Italian P.T. Administration, the Italian telephonic operating agencies and the I.B.M. Italia.

The goal of these tests was the examination of all types of circuits and telephonic equipment existing in Italy and some tests were made expressly for testing purpose with a low transmitting level.

The tests were made at 1200 bauds with an IBM two phases IBM laboratory model modem or with the frequency shift ATT modem model 3A and at 2000 bits/s with the four-phase ATT modem model 201A.

This paper presents a summary of the results aiming to assist for a first approach of design for a data transmission link into the Italian telephonic network.

All tests are divided into three classes:

1. Tests that were recorded only with reading every 15-20 minutes by an operator of two counters, one for erroneous bits, the other for erroneous messages. All these tests were made at 1200 bauds.

2. Tests at 1200 bauds recorded bit by bit into a magnetic tape and then elaborated by a computer.

These tests were made normally with IBM two-phase modem (a few tests with ATT modem model 3A) over:

a) point-to-point line or loop,

b) only automatic telephone exchange equipment;

c) switched line (direct distance dialling or operator distance dialling)

3. Tests at 2000 bits/s recorded bit by bit as indicated in number 2.

These tests were made with ATT modem 201A over:

a) one point-to-point line, frequency carrier system,

b) three looped lines coaxial cable system.

All tested lines are drawn into Figure 1.

All tests are listed into the tables with the number of the received bits and messages, the numbers of the erroneous bits and messages, the error rate in bits and messages and the transmission time in minutes.

Some notes indicate the differences among the tests over the same line.

Table 1 summarizes all results that are detailed into the following tables.

. . . .

FIGURE 1. — **Tests on Italian telephone public network**

TABLE 1. — *Summary of all results*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Tests 2000 bits/s									
Point to point	5 909 680 000	11 581 279	210 225	27 097	3.55 10^{-5}	2.94 10^{-3}	49 097	511	
Tests 1200 bits/s									
Data from counter	276 030 000	503 257	51 607	2 560	1.86 10^{-4}	5.08 10^{-3}	3 846	511	
Automatic telephone exchange	1 443 175 500	1 499 237	491 939	37 717	3.41 10^{-4}	2.52 10^{-2}	22 508	960	
Direct distance dialling	467 280 760	483 915	92 308	8 051	1.97 10^{-4}	1.67 10^{-2}	7 458	960	
Operator distance dialling	103 200 000	202 000	6 185	465	5.99 10^{-5}	2.3 10^{-3}	3 033	511	
Operator distance dialling	341 470 960	355 634	229 191	18 681	6.61 10^{-4}	5.25 10^{-2}	5 383	960	
Radio links	621 446 400	647 248	38 722	9 426	6.25 10^{-5}	1.45 10^{-2}	9 712	960	
Voice-frequency cable	329 415 880	345 343	26 041	6 583	7.91 10^{-5}	1.9 10^{-2}	5 180	960	
Frequency carrier	57 084 400	160 211	2 836	489	4.96 10^{-5}	3.05 10^{-3}	1 158	400	
Frequency carrier	187 332 880	197 148	4 980	1 719	2.66 10^{-5}	8.72 10^{-3}	3 015	960	
Coaxial cable	419 204 240	483 689	33 192	4 689	7.92 10^{-5}	9.6 10^{-3}	6 258	960	
Open-wire line	23 610 240	24 594	2	2	8.47 10^{-8}	8.13 10^{-5}	369	960	
Total messages 400 bits	57 084 400	160 211	2 836	489	4.96 10^{-5}	3.05 10^{-3}	1 158		
Total messages 511 bits	6 288 910 000	12 286 536	268 017	30 122	4.26 10^{-5}	2.45 10^{-3}	55 976		
Total messages 960 bits	3 732 937 060	4 036 808	916 375	86 868	2.45 10^{-4}	2.15 10^{-2}	59 883		
Total	10 088 931 460		1 187 228		1.18 10^{-4}		117 017		

TABLE 2. — *Test at 2000 bits/s (class 3)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Roma–GE–MI–Roma	2 562 000 000	5 013 000	81 995	16 120	3.2 10^{-5}	3.22 10^{-3}	21 350	511	
Roma-Pescara-AN-MI-Roma	1 763 000 000	3 450 309	49 596	3 614	2.81 10^{-5}	1.05 10^{-3}	14 690	511	
Roma–Palermo–Roma	1 360 000 000	2 676 830	75 390	6 944	5.54 10^{-5}	2.59 10^{-3}	11 333	511	
Milano–Cremona–Parma	224 680 000	441 140	3 244	419	3.1 10^{-5}	9.4 10^{-4}	1 873	511	
Total	5 909 680 000	11 581 279	210 225	27 097	3.55 10^{-5}	2.34 10^{-3}	49 246		

TABLE 3. — *Tests with only data from counter (class 1)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Genova–Napoli test No. 1	8 280 000	16 194	11 653	226	1.41 10^{-3}	1.4 10^{-2}	110	511	
Giunzione–Roma test No. 1	144 320 000	282 300	0	0	0	0	2 030	511	
Genova–Napoli test No. 2	6 480 000	12 720	7 813	409	1.21 10^{-3}	3.21 10^{-2}	95	511	
Genova–Piombino	23 100 000	45 400	11 520	547	5 10^{-4}	1.2 10^{-2}	321	511	
Genova–Taranto	24 300 000	47 350	15 606	978	6.44 10^{-4}	2.06 10^{-2}	330	511	
Milano–Arluno	21 910 000	42 700	4 982	382	2.1 10^{-4}	0.89 10^{-2}	290	511	
Giunzione–Roma test No. 2	47 640 000	56 593	33	18	0.7 10^{-6}	3.18 10^{-4}	670	511	
Total	276 030 000	503 257	51 607	2 560	1.86 10^{-4}	5.08 10^{-3}	3 846		

TABLE 4. — *Automatic telephone exchange in urban area (class 2 b)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Model A	160 296 000	162 975	2 303	693	1.43 10^{-5}	4.25 10^{-3}	2 447	960	
Model B	200 512 420	208 967	14 871	1 054	7.41 10^{-5}	5.04 10^{-3}	3 137	960	
Model C	195 217 920	203 352	4 803	1 035	2.46 10^{-5}	5.08 10^{-3}	3 053	960	
Model D	86 110 560	89 661	1 480	377	1.71 10^{-5}	4.20 10^{-3}	1 347	960	
Model E	218 098 080	227 220	220 472	19 411	1.01 10^{-3}	0.85 10^{-1}	3 411	960	
Model F	121 097 280	126 143	55 527	3 447	4.58 10^{-4}	2.73 10^{-2}	1 894	960	
Model G	251 784 320	262 067	8 357	524	3.32 10^{-5}	2 10^{-3}	3 934	960	
Model H	210 058 920	218 852	184 126	11 176	8.76 10^{-4}	5.1 10^{-2}	3 285	960	
Total	1 443 175 500	1 499 237	491 939	37 717	3.41 10^{-4}	2.52 10^{-2}	22 508		

Note. — Model A = Common control systems with x-bar switch
Model B * = Common control systems with two-motions switch
Model C * = Common control systems with two-motions switch
Model D = Common control system with x-bar switch
Model E = Common control systems with one-motion switch
Model F ** = Step by step systems with two-motions switch
Model G = Step by step systems with motorselector
Model H ** = Step by step systems with two-motions switch

* of different types
** of different manufacturers

TABLE 5. — *Direct distance dialling (class 2 c)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Brescia–Milano	76 383 360	79 576	29 640	2 015	3.88 10^{-4}	2.53 10^{-2}	1 194	960	
Mestre–Vicenza	4 029 120	4 197	42	3	1.04 10^{-5}	7.15 10^{-4}	63	960	1
Mestre–Vicenza	23 598 720	24 582	975	111	4.15 10^{-5}	4.52 10^{-3}	370	960	1
Rimini–BO–Milano	15 016 320	15 642	3 119	400	2.08 10^{-4}	2.55 10^{-2}	234	960	
Rimini–BO–Milano	2 669 760	2 781	5 291	478	1.98 10^{-3}	1.72 10^{-1}	41	960	3
Rimini–BO–Milano	79 763 520	83 087	8 183	1 174	1.03 10^{-4}	1.41 10^{-2}	1 247	960	4
Rimini–Milano	24 091 200	25 095	2 043	328	8.48 10^{-5}	1.3 10^{-2}	376	960	
Genova–Livorno	75 866 880	79 028	10 568	1 137	1.4 10^{-4}	1.29 10^{-2}	1 186	960	
Napoli–Sora	3 576 800	3 830	289	252	2.15 10^{-4}	6.58 10^{-2}	57	960	2
Napoli–Sora	19 650 240	20 469	1 307	448	6.65 10^{-5}	2.19 10^{-2}	307	960	2
Napoli–Sora	2 776 560	2 861	29	6	1.06 10^{-5}	2.1 10^{-3}	42	960	2
Napoli–Sora	3 549 120	3 697	4	4	1.1 10^{-6}	1.2 10^{-3}	55	960	2
Napoli–Sora	17 079 360	17 791	487	66	2.85 10^{-5}	3.71 10^{-3}	267	960	2-3
Napoli–Sora	23 587 200	24 570	3 731	112	1.58 10^{-4}	4.56 10^{-3}	368	960	2-4
Napoli–Sora	2 508 080	2 673	581	54	2.26 10^{-5}	7.7 10^{-4}	40	960	2
Napoli–Sora	17 460 480	18 188	239	14	1.37 10^{-5}	3.09 10^{-3}	273	960	2
Colonna–Eur–(Roma)	32 526 720	33 882	15 010	351	4.61 10^{-4}	1.92 10^{-3}	508	960	
Colonna–Eur–(Roma)	39 287 360	41 966	10 770	1 098	2.74 10^{-4}	2.62 10^{-2}	630	960	4
Total	467 280 760	483 915	92 308	8 051	1.97 10^{-4}	1.67 10^{-2}	7 458		

Table 6. — *Operator distance dialling (class 2 c)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Napoli–Sora	15 240 960	15 876	444	49	2.91 10^{-5}	3.09 10^{-3}	267	960	
Macerata–Roma	109 000 000	113 418	93 935	9 671	8.6 10^{-4}	8.52 10^{-2}	1 702	960	
Macerata–Roma	3 830 000	3 997	2 394	198	6.25 10^{-4}	4.96 10^{-2}	60	960	2
Milano–Taormina	96 200 000	100 219	35 805	2 914	3.72 10^{-4}	2.9 10^{-2}	1 504	960	
Mantova–Verona–Roma	117 200 000	122 124	96 613	6 849	7.3 10^{-4}	5.6 10^{-2}	1 850	960	
Genova–Mestre	103 200 000	202 000	6 185	465	5.99 10^{-5}	2.32 10^{-3}	3 033	511	
Total messages 960 bits	341 470 960	355 634	229 191	18 681	6.61 10^{-4}	5.25 10^{-2}	5 383		
Total messages 511 bits	103 200 000	202 000	6 185	465	5.99 10^{-5}	2.3 10^{-3}	3 033		
Total	444 670 960		235 376		5.27 10^{-4}		8 416		

TABLE 7. — *Radio links (class 2 a)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Mestre–Cortina–Mestre	44 133 120	45 972	735	88	1.6 10^{-5}	1.9 10^{-3}	690	960	
Mestre–Cortina–Mestre	5 402 880	5 628	0	0	0	0	84	960	2
Mestre–Cervignano–Mestre	27 208 320	28 342	20 528	6 980	7.5 10^{-4}	2.46 10^{-1}	425	960	
Mestre–Cervignano–Mestre	5 442 080	5 648	20	9	3.6 10^{-6}	1.59 10^{-3}	84	960	3
Mestre–Udine–Cervignano–Mestre	8 898 240	9 269	97	46	1.09 10^{-5}	4.96 10^{-3}	139	960	
Mestre–Udine–Cervignano–Mestre	1 820 160	1 896	23	10	1.26 10^{-5}	5.27 10^{-3}	28	960	3
Mestre–Vicenza–Mestre	15 598 400	16 040	46	13	2.99 10^{-6}	8.1 10^{-4}	240	960	
Bologna–Porretta–Bologna	31 768 320	33 092	168	7	5.23 10^{-6}	2.12 10^{-4}	496	960	
Bologna–Ferrara–Bologna	39 563 520	41 212	75	13	1.9 10^{-6}	3.15 10^{-4}	618	960	
Rimini–Bologna–Milano	3 860 160	4 021	2	1	5.18 10^{-7}	2.49 10^{-4}	60	960	
Roma–Firenze–Roma	3 996 160	181 246	1 595	198	0.917 10^{-5}	1.09 10^{-3}	2 721	960	
Roma–Pescara–Roma	263 755 040	274 882	15 433	2 061	5.87 10^{-5}	7.5 10^{-3}	4 127	960	
Total	621 446 400	647 248	38 722	9 426	6.23 10^{-5}	1.45 10^{-2}	9 712		

TABLE 8. — *Voice-frequency cable (class 2 a)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Torino–Rivarolo–Torino	30 980 320	32 992	99	4	3.2 10^{-6}	1.24 10^{-4}	499	960	
Torino–Lanzo–Torino	15 732 840	16 379	0	0	0	0	245	960	
Torino–Lanzo–Torino	12 206 080	11 673	4	4	3.28 10^{-7}	3.43 10^{-4}	175	960	2
Rete distribuzione Roma	30 406 080	31 673	6	1	1.97 10^{-7}	3.16 10^{-5}	475	960	
Roma–Milano–Roma	41 077 440	42 789	514	66	1.26 10^{-5}	1.68 10^{-3}	642	960	
Roma–Civitavecchia	82 500 000	85 865	80	9	9.7 10^{-7}	1.05 10^{-4}	1 289	960	
Roma–Firenze–Roma	50 186 880	52 278	18 595	4 486	3.7 10^{-4}	8.58 10^{-2}	785	960	
Roma–Firenze–Roma	45 702 720	47 607	4 524	1 683	1.09 10^{-4}	3.53 10^{-2}	714	960	1
Roma–Firenze–Roma	23 123 520	24 087	2 719	330	1.17 10^{-4}	1.37 10^{-2}	356	960	1
Total	329 415 880	345 343	26 041	6 583	7.91 10^{-5}	1.9 10^{-2}	5 180		

TABLE 9. — *Frequency carrier (class 2 a)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Torino–Vercelli–Torino	23 454 720	24 452	0	0	0	0	367	960	
Torino–Vigone–Torino	31 450 560	32 761	3 240	1 299	1.03 10^{-4}	3.97 10^{-2}	491	960	5
Torino–Vigone–Torino	4 034 880	4 203	0	0	0	0	63	960	6
Bologna–Molinella–Bologna	15 625 920	16 277	63	22	4.03 10^{-6}	1.35 10^{-3}	244	960	2
Bologna–Molinella–Bologna	5 553 600	5 785	0	0	0	0	86	960	2
Bologna–Ancona–Bologna	22 537 920	25 477	4	2	1.77 10^{-7}	9.2 10^{-5}	382	960	2
Bologna–Ancona–Bologna	8 009 280	8 343	0	0	0	0	125	960	2
Roma–Bologna–Roma	46 692 480	48 638	450	31	9.63 10^{-6}	6.37 10^{-4}	790	960	
Roma–Ostia–Roma	29 867 520	31 112	1 223	365	4.1 10^{-5}	1.17 10^{-2}	467	960	
Roma–Bologna–Roma	25 867 200	64 668	2 645	322	1.02 10^{-4}	4.98 10^{-3}	467	400	6
Roma–Bologna–Roma	31 217 200	95 543	191	167	6.12 10^{-6}	1.75 10^{-3}	691	400	5
Total messages 960 bits	187 332 880	197 148	4 980	1 719	2.66 10^{-5}	8.72 10^{-3}	3 015		
Total messages 400 bits	57 084 400	160 211	2 836	489	4.96 10^{-5}	3.05 10^{-3}	1 158		
Total	244 417 280		7 816		3.19 10^{-5}		4 173		

TABLE 10. — *Coaxial cable (class 2 a) (4 MHz)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Roma–Milano–Roma	213 812 240	222 794	6 384	2 763	2.59 10^{-5}	1.23 10^{-2}	3 344	960	1
Roma–Milano–Roma	205 365 120	213 992	1 198	68	5.84 10^{-6}	3.2 10^{-4}	3 210	960	1
Roma–Milano–Roma	45 026 880	46 903	25 610	1 858	5.7 10^{-4}	3.98 10^{-2}	704	960	1
Total	419 204 240	483 689	33 192	4 689	7.92 10^{-5}	9.6 10^{-3}	6 258		

TABLE 11. — *Open-wire line (class 2 a)*

Description	Received		Error		Error rate		Time minutes	Mess. length	Notes
	Bits	Messages	Bits	Messages	Bits	Messages			
Bologna–Parma–Bologna	23 610 240	24 594	2	2	8.47 10^{-8}	8.13 10^{-5}	369	960	
Total	23 610 240	24 594	2	2	8.47 10^{-8}	8.13 10^{-5}	369		

ITALIAN ADMINISTRATION. — *(Contribution COM Sp. A—No. 54—November 1965)*

ITALIAN LINE TESTS — LINE BREAKING ANALYSIS

Some extensive line tests were made in Italy on the three loops:

a) Roma - Milano - Roma (tirrenic coaxial cable)

b) Roma - Pescara - Milano - Pescara - Roma (radio links Roma - Pescara - and Adriatic coaxial cable Pescara - Milano).

c) Roma - Palermo - Roma (coaxial cable).

with a channel demo-modulation at the far end, respectively Milano and Palermo.

These tests were programmed to examine not only the error rate but also the effective availability of the line, with continuous record bit by bit 24 hours every day of each error and line interruption.

The loop a) was examined during three weeks and the other two b) and c) during two weeks.

For the line tests were used the A.T. & T. modem 201A four-phase modulation at 2000 bits/s and the I.B.M. teletest for data generation and comparison.

For saving personnel and duty time the teletest outputs were recorded through a special laboratory equipment directly into magnetic tape in digital form.

The teletest gives for each erroneous bit an output with level "1" and for each line interruption bit an output with a twofold level.

The signal coming from the telephone line goes to the modem through the teletest.

The teletest examines the level and if it is less than 45 dB for an 8-bits time or more, the condition "line breaking" (L.B.) is recorded bit by bit for the whole time of line interruption.

This contribution presents some statistical results coming from these tests.

All the magnetic tapes recorded during the test are examined with a computer and 8 bits (teletest delay) are added at every recorded line breaking.

The L.B. are divided in groups of thirty minutes for time occurrence and in groups for lengths.

The line breaking more than 100 000 bits length are tabulated with indication of the hour and minute of occurrence, the starting bits into the message and lengths in bits.

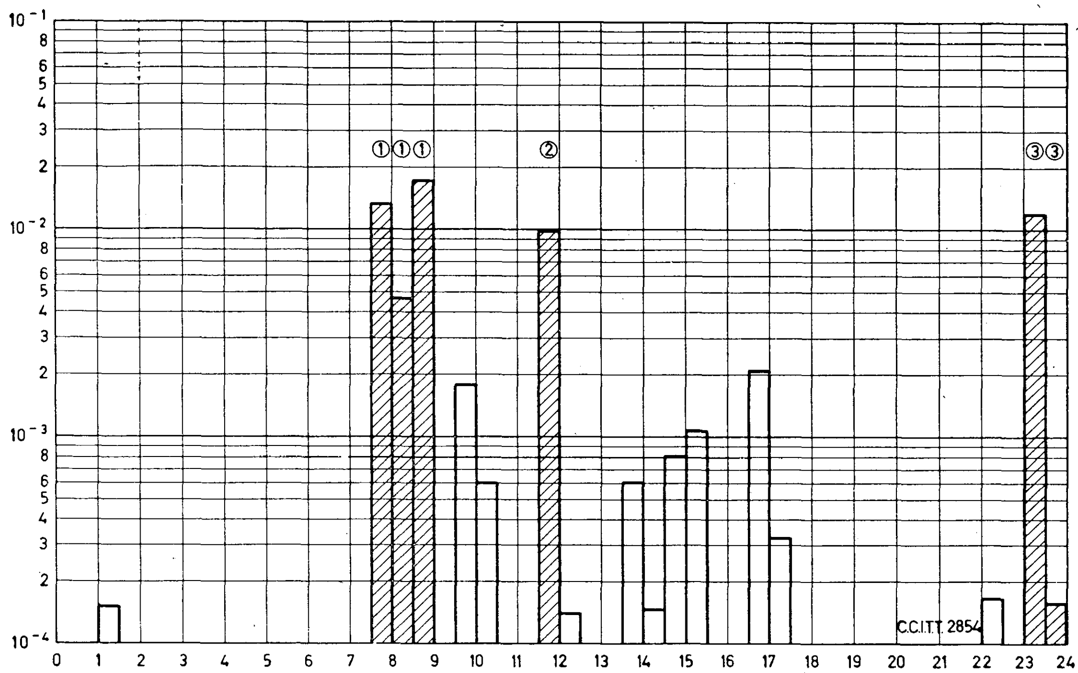
This table permits an easy research of the causes into the journal of the central office.

The three histograms Figures 1, 2, 3 respectively for the loops a, b, c present for every half-hour the ratio between the L.B. in bits and the bits effectively transmitted during the whole test in the half-hour considered.

The hatched drawn zones in the histograms indicate identified causes, the white zones not identified causes.

The causes are attached to the figures.

The number of the L.B. of the three tests divided for lengths are summarized in Table 1, for L.B. up to 300 ms and in Table 2 for interruption more than 300 ms length.



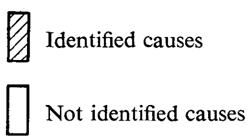


FIGURE 1. — **Roma-Milano-Roma. L.B. rate histogram in half an hour**

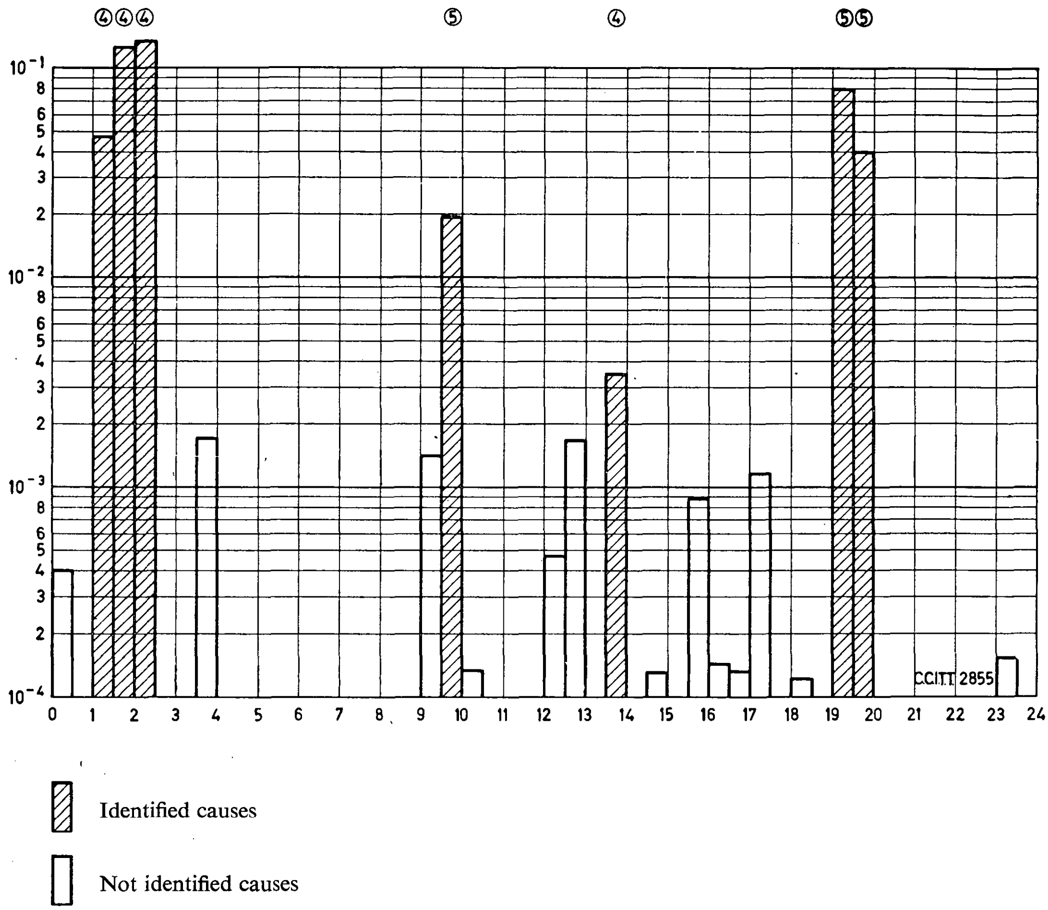


FIGURE 2. — **Roma-Pescara-Milano-Roma. L.B. rate histogram in half an hour**

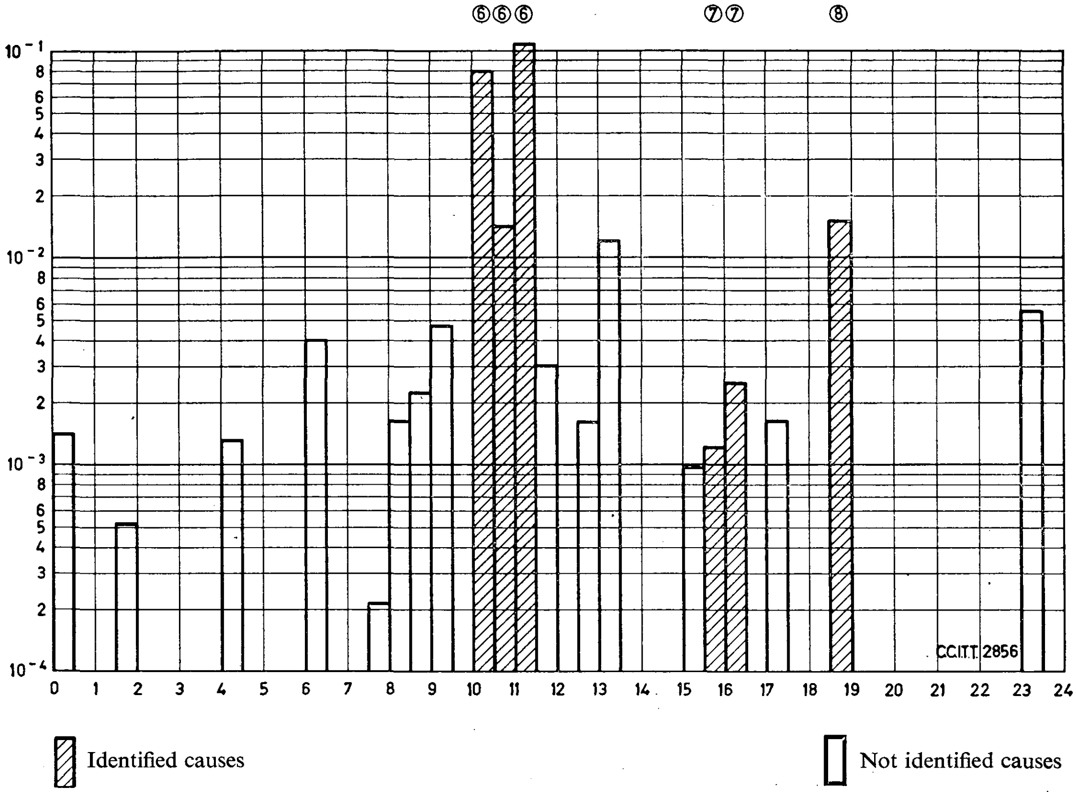


FIGURE 3. — **Roma-Palermo-Roma. L.B. rate histogram in half an hour**

TABLE 1

Distribution by length in bits for L.B. length up to 300 ms (1 bit = 0.5 ms)

L.B. length		Test A	Test B	Test C
From	To			
1	10	99	2	36
11	20	5141	36	536
21	30	3634	35	503
31	40	1364	26	375
41	50	579	22	172
51	60	282	20	132
61	70	176	7	93
71	80	113	16	117
81	90	68	10	67
91	100	55	16	91
101	200	215	52	357
201	300	102	42	142
301	400	46	24	63
401	500	45	12	53
501	600	32	16	26

TABLE 2

Distribution by length in bits for L.B. length more than 300 ms (1 bit = 0.5 ms)

L.B. length		Test A	Test B	Test C
From	To			
601	700	31	15	29
701	800	15	10	23
801	900	19	6	20
901	1 000	16	8	12
1 001	2 000	94	25	82
2 001	3 000	23	13	37
3 001	4 000	13	6	15
4 001	5 000	7	6	6
5 001	6 000	12	1	5
6 001	7 000	5	2	4
7 001	8 000	10	3	3
8 001	9 000	1		7
9 001	10 000	1	1	4
10 001	15 000	3	4	9
15 001	20 000	4	2	7
20 001	25 000	7	2	8
25 001	30 000	3	1	1
30 001	35 000	1	4	3
35 001	40 000	3	2	11
40 001	45 000	1	1	2
45 001	50 000		2	1
50 001	55 000	1		3
55 001	60 000			3
60 001	65 000			2
65 001	70 000	1		1
70 001	75 000	2		
75 001	80 000			1
80 001	85 000		1	1
85 001	90 000			1
90 001	95 000	1		
95 001	100 000		1	1
More than	100 000	5	17	14

Identified causes

1. Troubles on coaxial Pisa - Genova, switch on radio-link.
2. Return on coaxial system.
3. Trouble on channel equipment in Milano.
4. Maintenance on radio-link power supply.
5. Trouble on radio-link.
6. Defective loop in Palermo.
7. Troubles on supergroup amplifier in Roma.
8. Main power off. Delayed insertion of emergency electric generator.

FEDERAL REPUBLIC OF GERMANY. — *(Contribution COM Sp. A—No. 95—February 1967)*

STUDY OF THE GENERAL SWITCHED TELEPHONE NETWORK WITH A VIEW TO ITS SUITABILITY FOR DATA TRANSMISSION

1. *Test programme*

During the period from March to May 1966 the German Administration made extensive measurements in the general switched telephone network of the Deutsche Bundespost, so as to obtain a survey of the characteristics of telephone connections being of interest to data transmission.

These measurements supplement the tests carried out in 1963 and 1964, the results of which are dealt with in Documents COM Sp. A No. 17 and No. 18 (see Supplements No. 28 and No. 29 respectively).

It is pointed out that in order to provide as real conditions as possible, all measurements were carried out from subscriber station to subscriber station. In 69 places distributed all over the Federal Republic of Germany a Deutsche Bundespost measuring van—especially equipped for the purpose of these tests—was connected with subscriber stations of the general switched telephone network. Three central test points were established, viz:

Test point A: Darmstadt (Fernmeldetechnisches Zentralamt)

Test point B: München (Siemens AG)

Test point C: Bad Godesberg

One or several switched connections were always set up between the measuring van and the three central test points.

The following characteristics of the connections were measured:

a) overall loss;

b) attenuation distortion [1];

c) group delay distortion [1];

d) relative amplitude of the listener echo;

e) bit and block error rates [2] for data transmission at 1200 bits/s (modem V.23);

f) isochronous distortion [2] for data transmission at 1200 bits/s (modem V.23);

g) impulsive noise rate.

The characteristics enumerated in a), b), c), d) and g) were determined on connections set up to all central test points, whereas e) and f) could only be measured on connections between the measuring van and test point A, since there were no more than two test equipments available.

[1] Attenuation and group delay distortions were measured by means of the sweep-frequency delay measuring set of Messrs. Siemens (Rel 3K 220) and photographed.

[2] Bit and block error rates as well as isochronous distortion were measured by means of the error rate analyser, type 1004 or the error distribution analyser, type 1005, of Standard Radio & Telegraph, Bromma, Sweden.

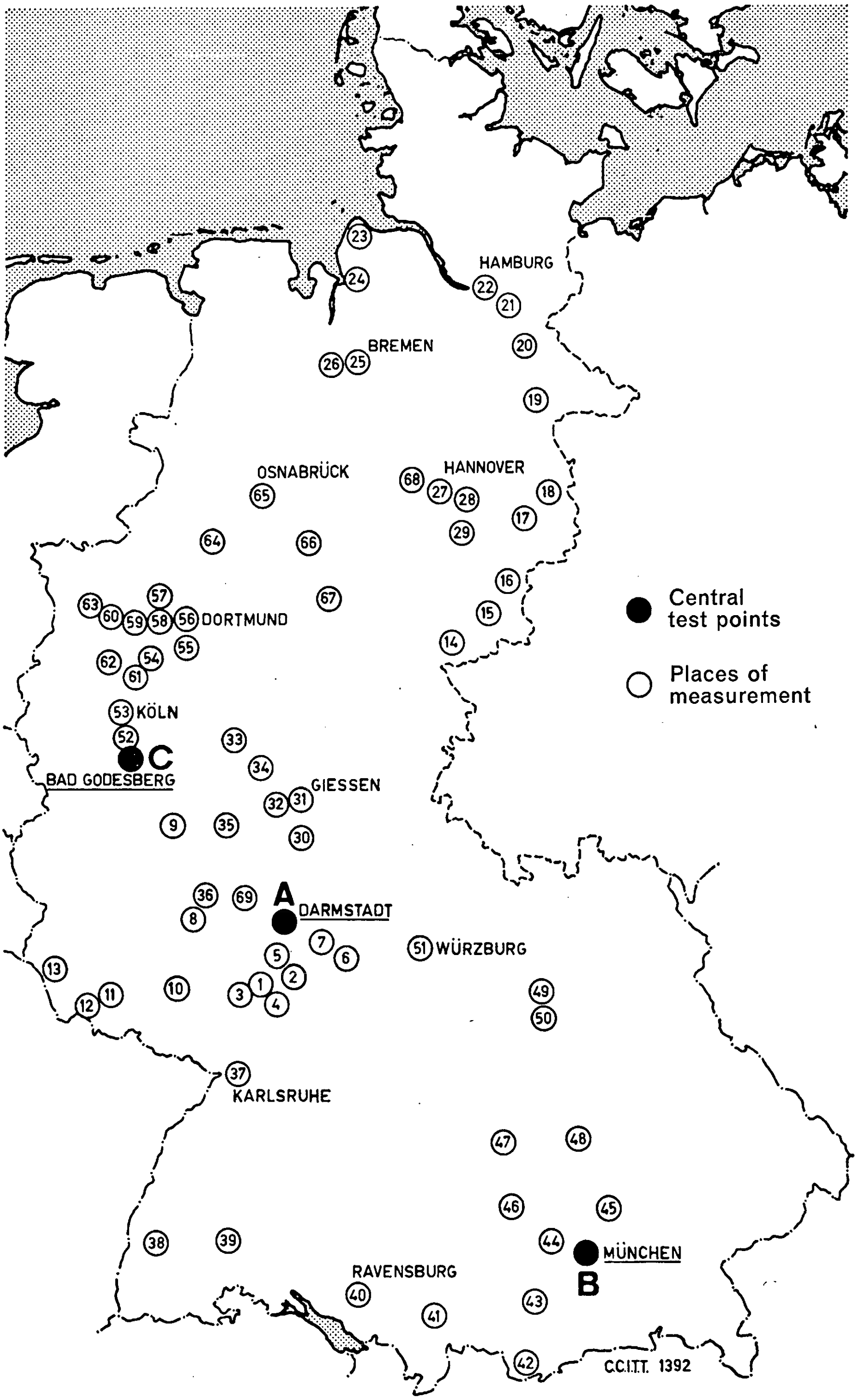


FIGURE 1. — Survey of the places of measurement and the three central test points for measurements in the German telephone network

The 69 places where the measuring van had been set up (Figure 1) were so selected that, amongst others, the geographical distribution and the composition of the connections to the central test points complied as closely as possible with the distribution of the data sources and sinks to be expected in future. The location of the telephone stations with regard to their exchanges should also be so that the distribution of the lengths of the subscribers' lines (curve 1 of Figure 2) is not more favourable than that of all telephone subscribers' lines (curve 2 of Figure 2—situation some ten years ago).

All data transmission tests were made on workdays during the period from 0800 to 1700 hours. More than 70% of the transmission tests took place from 0830 to 1200 hours or 1430 to 1700 hours, i.e. during periods of heavy telephone traffic.

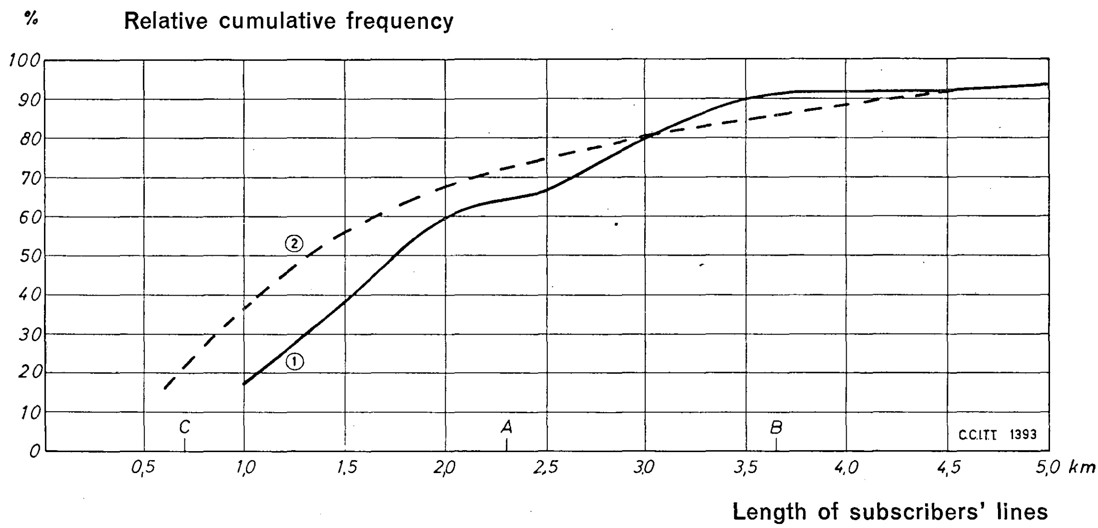


FIGURE 2. — **Length of subscribers' lines**

2. *Test results*

2.1 *Overall loss*

The overall loss of the subscriber-to-subscriber lines terminated with 600 ohms was measured at 1000 Hz. Since attenuation distortion measurements (cf. item 2.2) were performed as well, the overall loss can also be determined at other frequencies.

Figures 3, 4, 5, 6, and 7 show the cumulative frequency distribution of the overall loss a_r at 800 Hz, 1000 Hz, 1700 Hz, 2500 Hz, and 2800 Hz.

The curves designated A, B and C apply to the connections established between the measuring van and the relevant test points A, B, and C. Curve D refers to the total number of connections.

For comparison, curve E is plotted in Figures 3 and 5. It is taken from Document COM Sp. A No. 18 and represents the cumulative frequency distribution of the overall loss measured three years ago on connections between some 200 telephone stations all over the Federal Republic of Germany and Darmstadt (test point A).

The values of the overall loss not exceeded in 50% and 95% respectively of the cases are listed in Table 1. From Table 1 and Figures 3 and 5 it can be seen that the values measured in 1966 (curve A) are some 4 to 5 dB better than those of curve E measured in 1963. This is thought to be mainly due to the improvements of the network achieved during the last years. For instance,

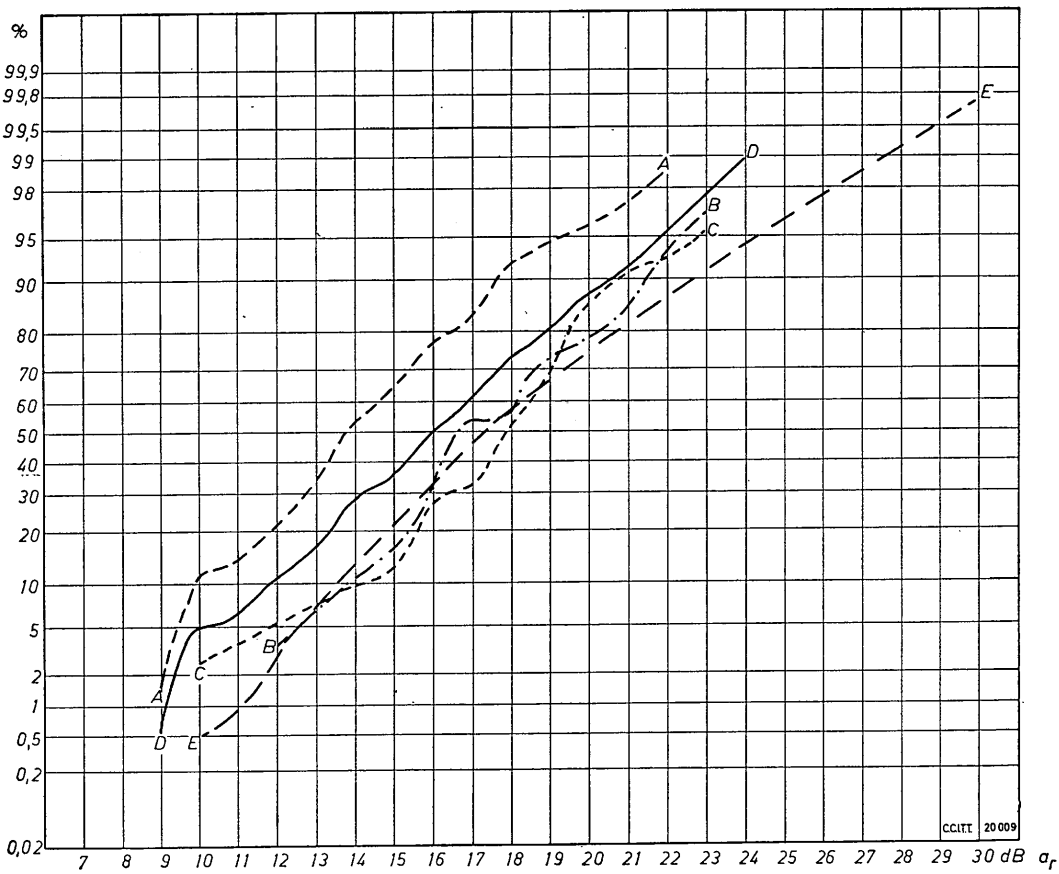


FIGURE 3. — Relative cumulative frequency distribution of overall loss at 800 Hz

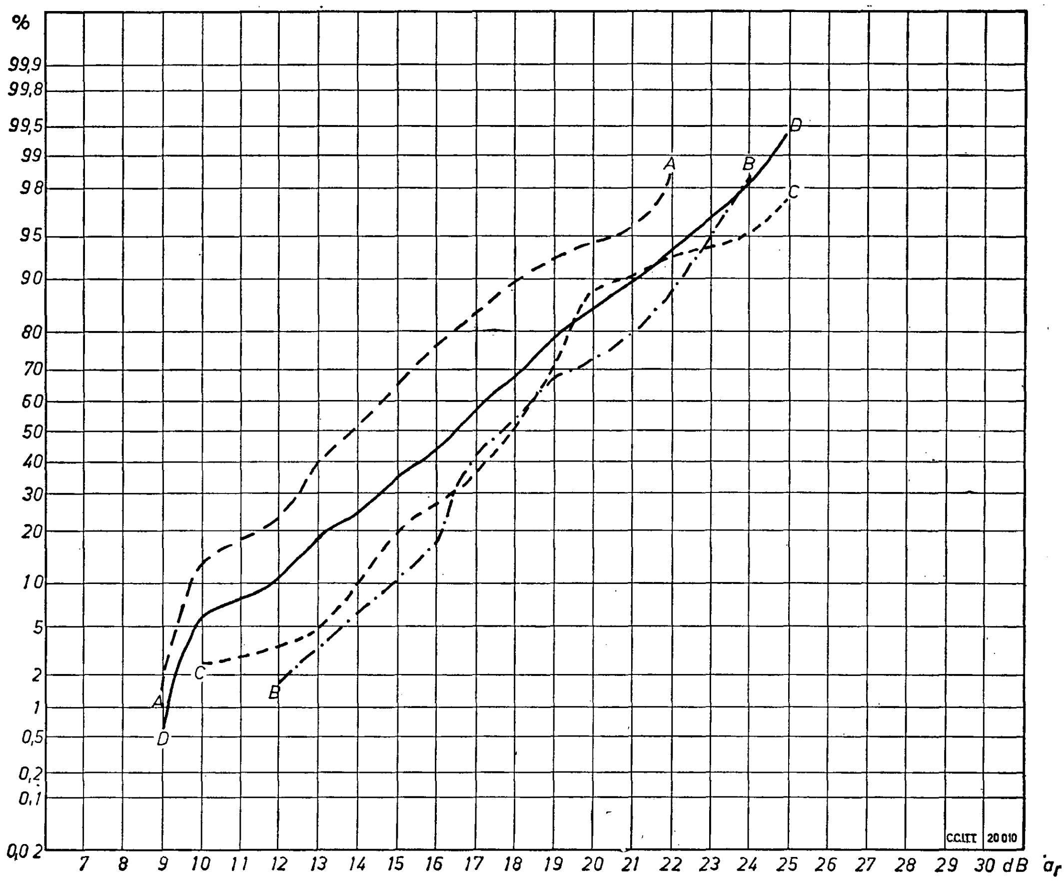


FIGURE 4. — Relative cumulative frequency distribution of overall loss at 1000 Hz

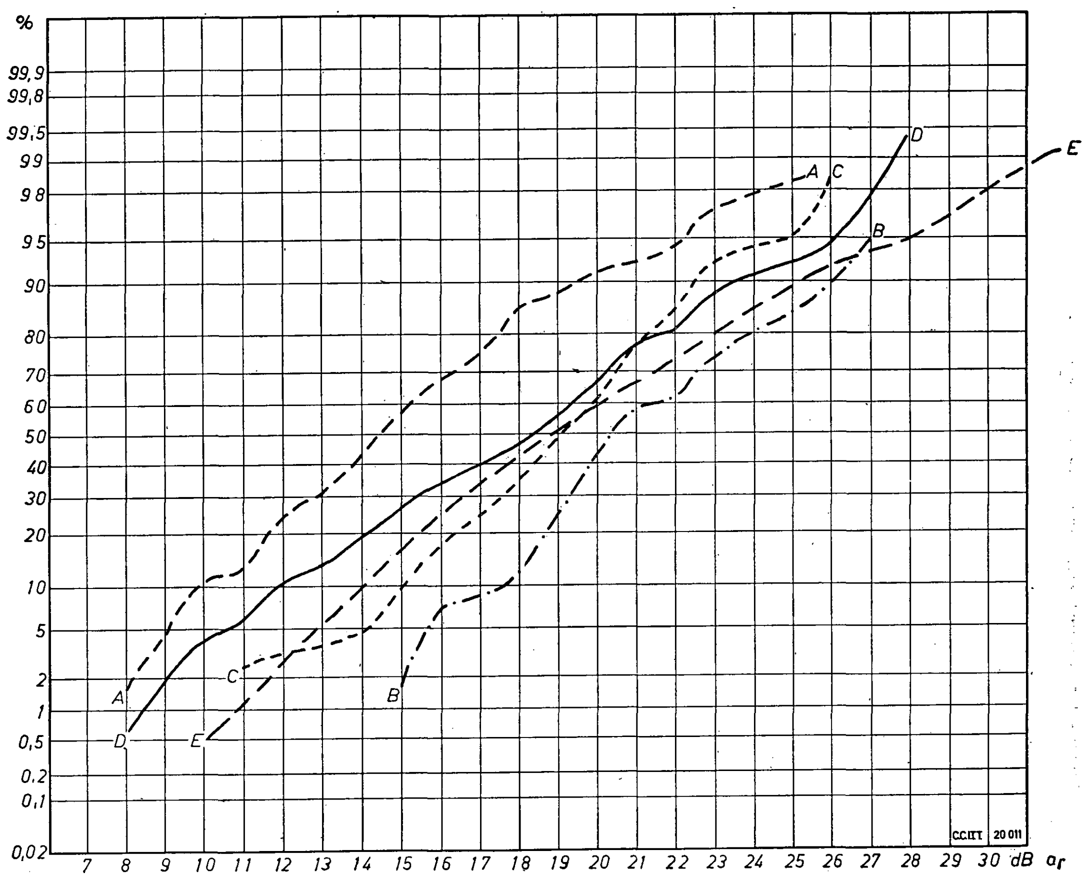


FIGURE 5. — Relative cumulative frequency distribution of overall loss at 1700 Hz

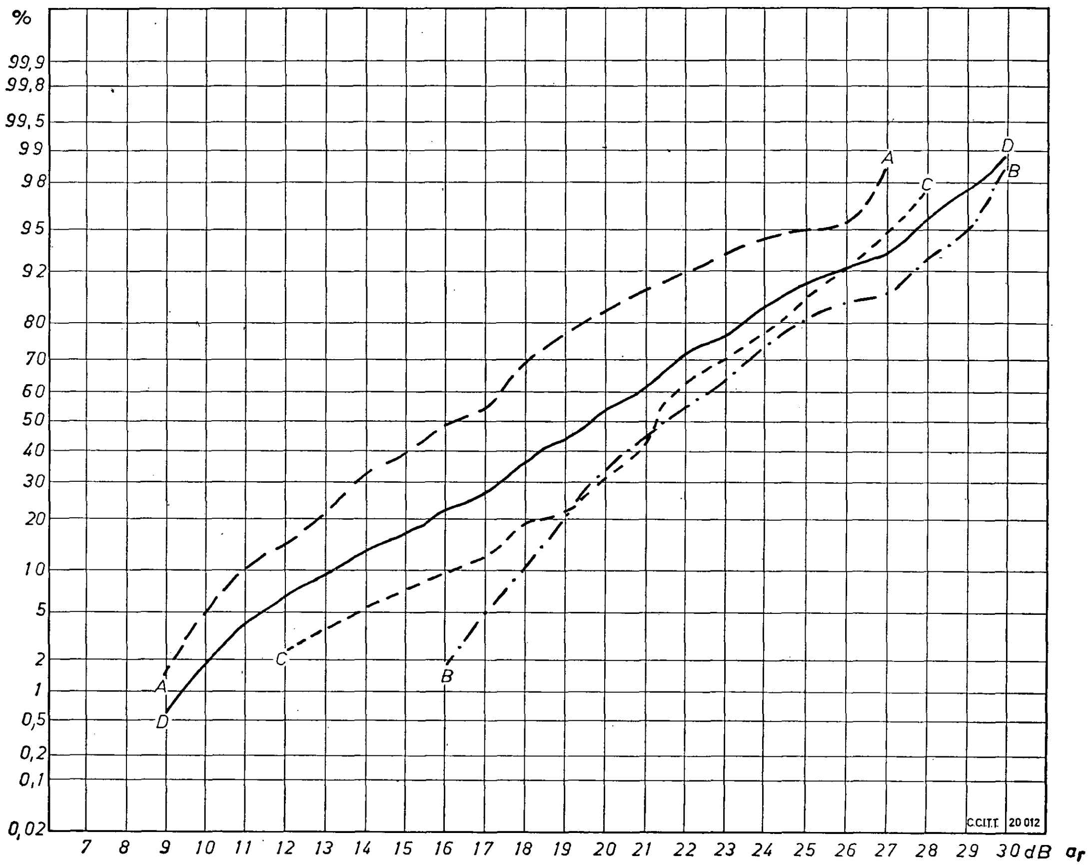


FIGURE 6. — **Relative cumulative frequency distribution of overall loss at 2500 Hz**

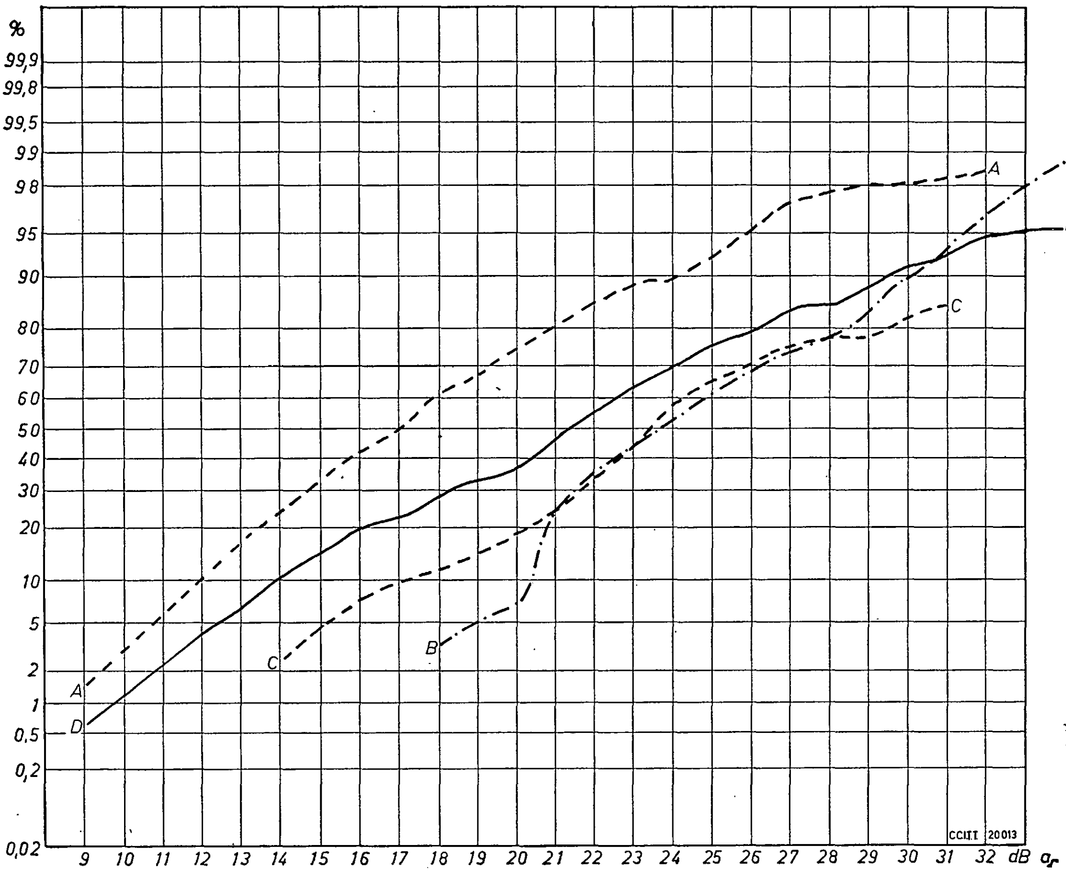


FIGURE 7. — Relative cumulative frequency distribution of overall loss at 2800 Hz

a great number of loaded audio circuits were replaced by carrier links. On the link Darmstadt—Frankfurt, over which nearly all calls to Darmstadt are routed, this resulted in an average decrease of attenuation by 1-2 dB. Although at the time of these measurements the average length of subscribers' lines was still greater than that of all telephone subscribers' lines (see Figure 2) it might yet be slightly below the figure obtained in 1963.

TABLE 1

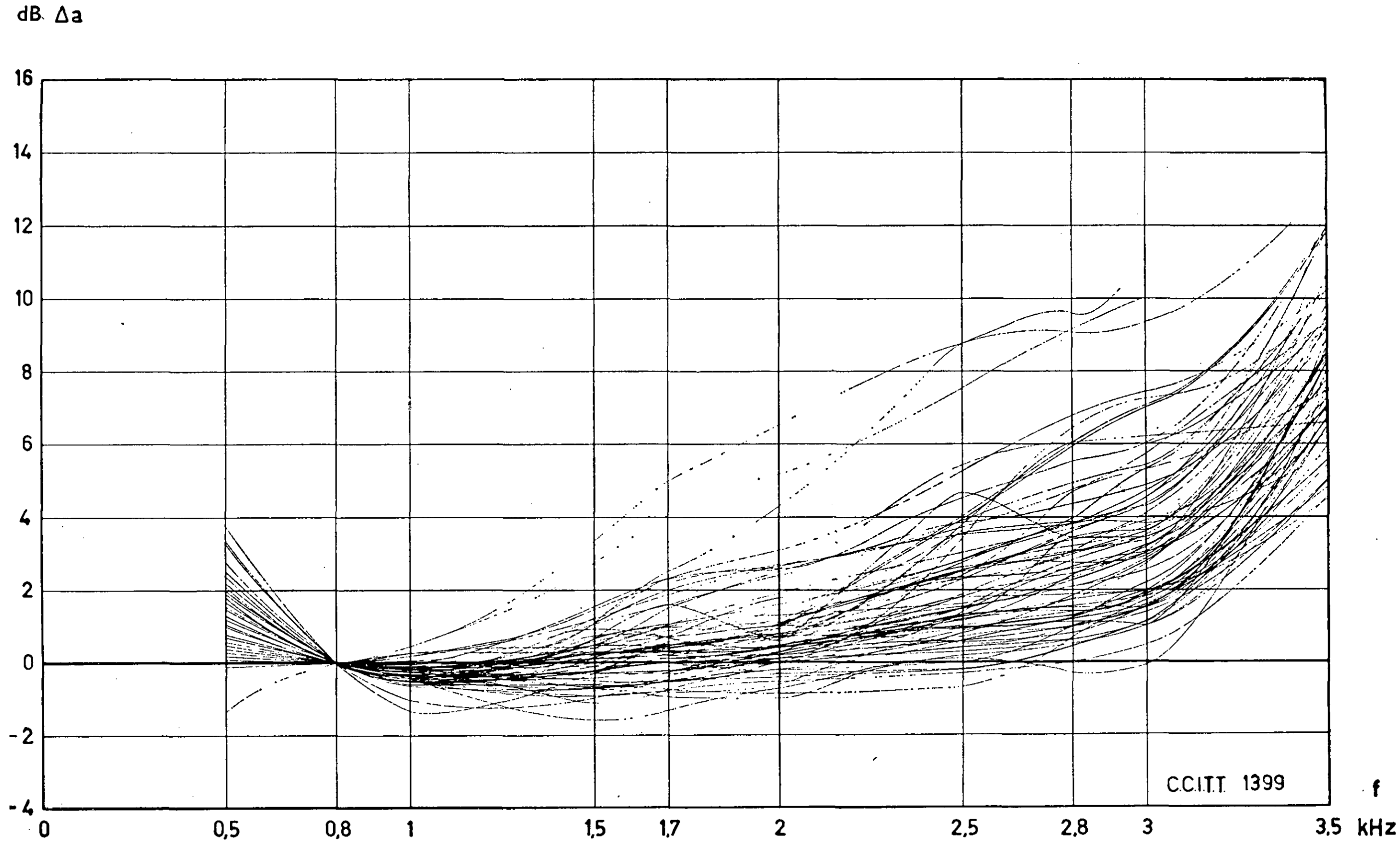
a_r in dB for cumulative frequency %	A	B	C	D	E	
50 95	13.9 19.6	16.7 22.5	18.0 23.0	16.1 21.9	17.5 24.5	$f =$ 800 Hz
50 95	13.9 20.6	17.7 23.0	17.9 23.9	16.5 22.4	— —	$f =$ 1000 Hz
50 95	14.5 22.2	20.4 27.0	19.1 24.8	18.4 26.1	18.8 28.0	$f =$ 1700 Hz
50 95	16.2 25.3	21.5 29.1	21.3 27.1	19.7 27.8	— —	$f =$ 2500 Hz
50 95	17.0 25.9	21.4 31.5	23.4 33.0	23.6 —	— —	$f =$ 2800 Hz

2.2 *Attenuation distortion*

Figures 8, 9 and 10 show the spread of the attenuation distortion measured at several frequencies and referred to 800 Hz. It can be seen that the majority of the collective consists of carrier circuits. There are only some connections established over older medium and heavily loaded lines. The difference in the slopes of the sets of curves is to be attributed to the different lengths of the audio cable links between the central test points and the associated carrier repeater stations. The length of the subscribers' lines of the test points is also shown in Figure 2. The curves of Figures 8, 9, and 10 do not indicate the fine structure (fluctuations due to echoes at the ends of the four-wire circuit). It was, however, evaluated when determining the relative amplitude of the listener echoes (see item 2.4).

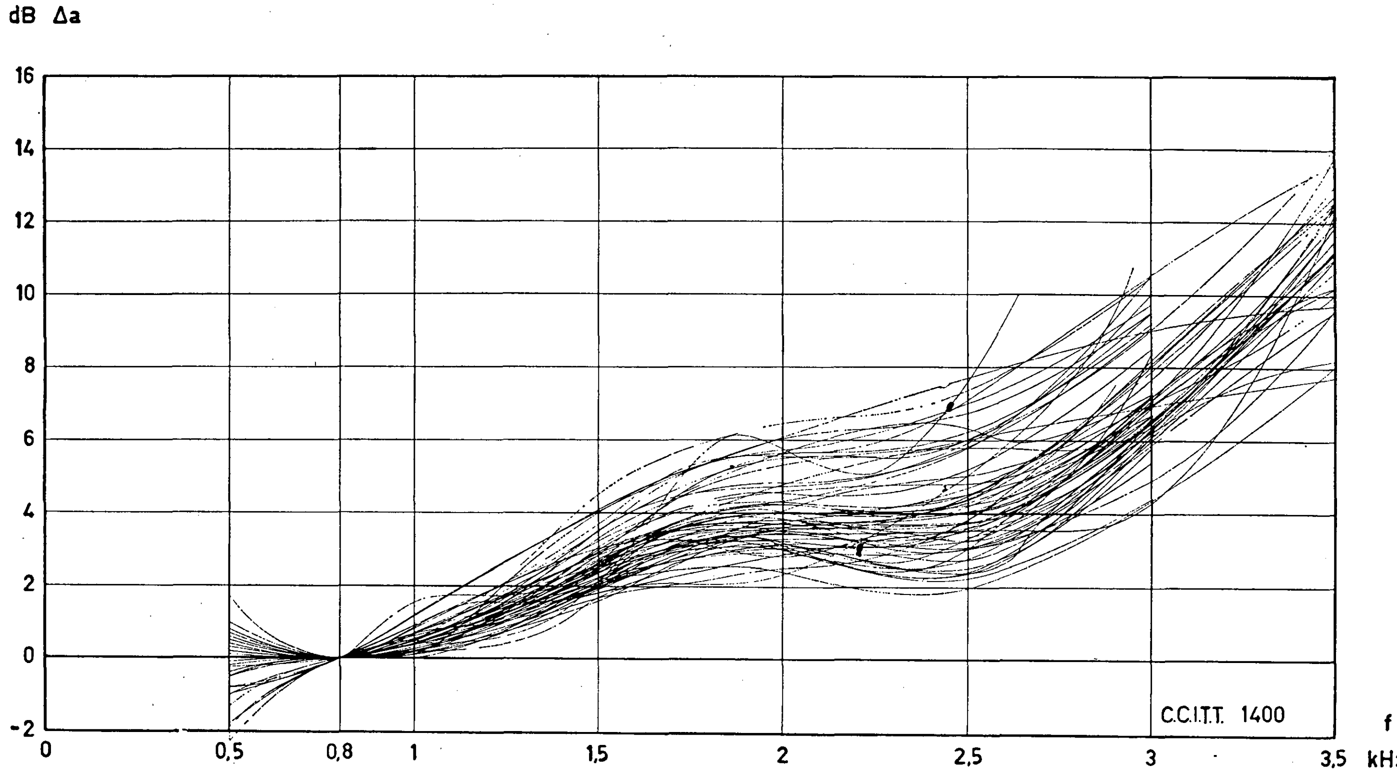
The attenuation distortion curves of Figure 9 show a characteristic hump which can be explained by the point of reflection between a loaded terminal exchange line and the comparatively long unloaded subscriber's line to the central test point B (München, Siemens AG).

The cumulative frequency distribution of the attenuation distortion Δ *a* between the frequencies 800 Hz and 2500 Hz as well as 800 Hz and 2800 Hz is plotted in Figures 11 and 12.



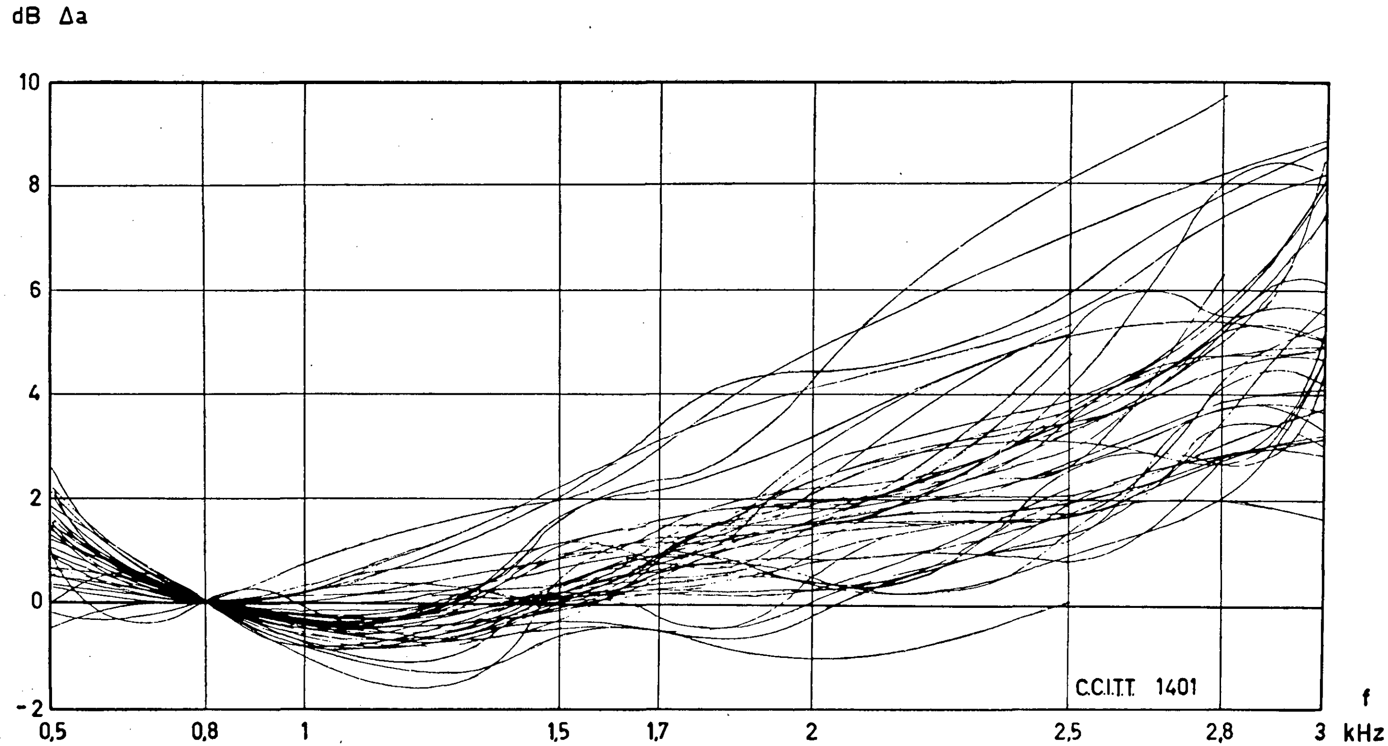
Central test point: A

FIGURE 8. — **Attenuation distortion in the German telephone network referred to the value at 800 Hz**



Central test point: B

FIGURE 9. — **Attenuation distortion in the German telephone network referred to the value at 800 Hz**



Central test point: C

FIGURE 10. — **Attenuation distortion in the German telephone network referred to the value at 800 Hz**

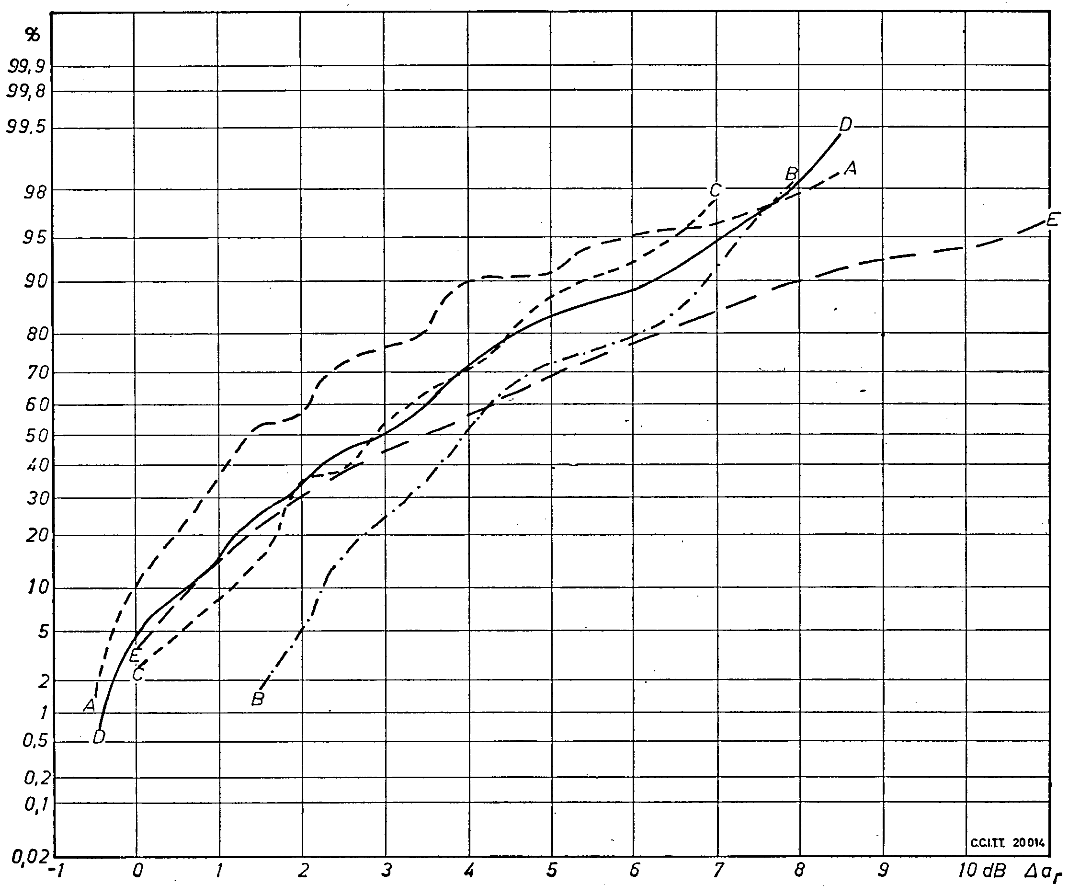


FIGURE 11. — Relative cumulative frequency distribution of Δa_r between 800 Hz and 2500 Hz

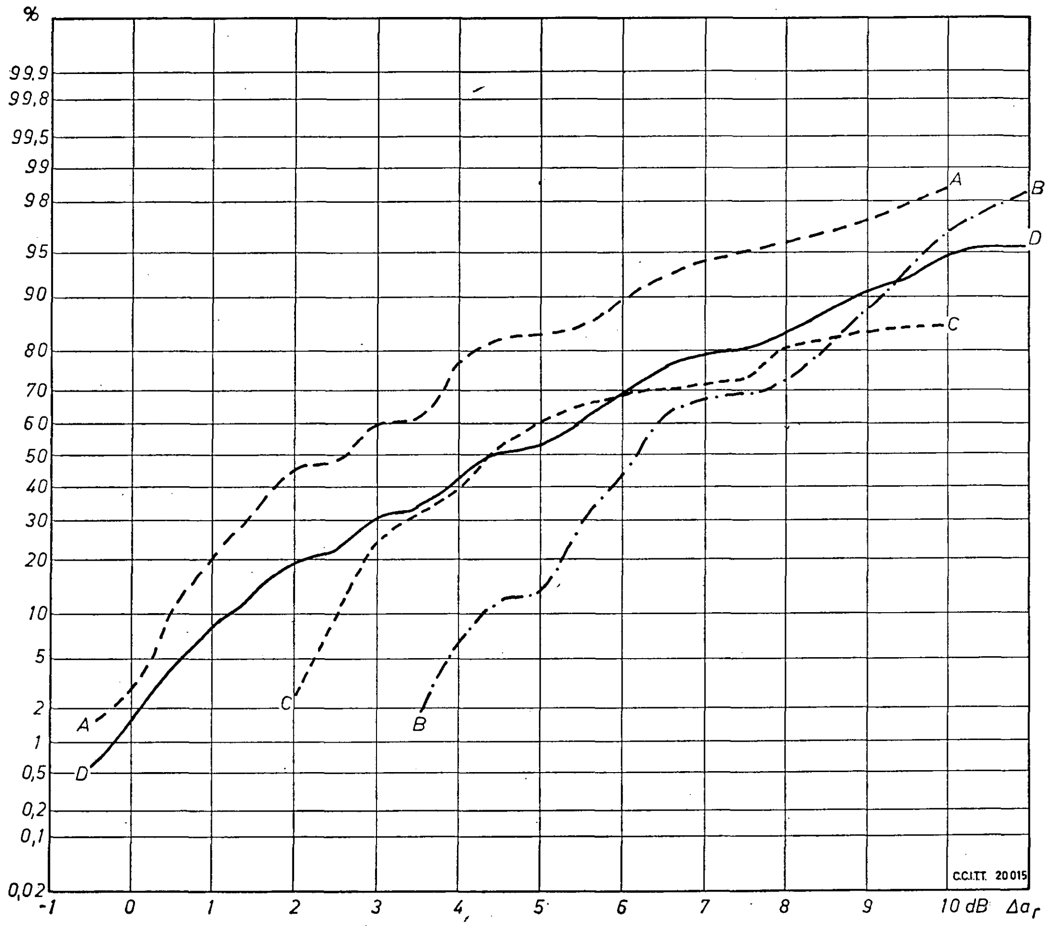


FIGURE 12. — Relative cumulative frequency distribution of Δa_r between 800 Hz and 2800 Hz

The designation of the curves (A, B, C, D, E) is described in item 2.1. The attenuation distortion values not exceeded in 50% and 95% respectively of the cases are listed in Table 2.

TABLE 2

a_r in dB for cumulative frequency %	A	B	C	D	E	
50 95	1.4 5.9	4.0 7.3	2.8 6.5	3.0 7.1	3.5 10.4	800 Hz . . . 2500 Hz
50 95	2.6 7.5	6.2 9.7	4.4 —	4.4 10.4	— —	800 Hz . . . 2800 Hz

Here, the values for A are also better than those for E, a fact which might again be due to the circumstances mentioned in 2.1 above.

2.3 *Group delay distortion*

Figures 13, 14 and 15 give the dispersion range of the group delay distortion referred to a value at 2000 Hz. These curves, too, do not indicate the fluctuations in the fine structure caused by echoes. When comparing the tolerance range for delay distortions, reproduced in Supplement No. 28, with the dispersion ranges of Figures 13, 14 and 15, it will be found that the frequency band is only slightly exceeded at the edges.

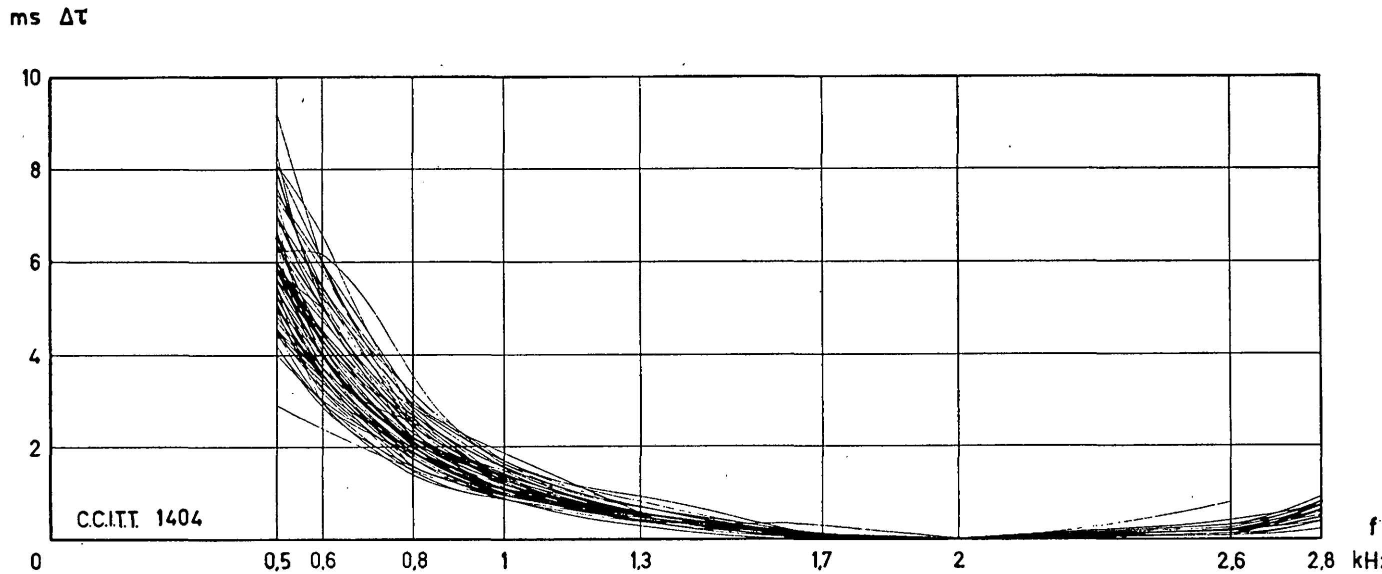
As can be seen from Figures 13, 14 and 15 all delay curves have similar characteristics. A more detailed study revealed that approximation of all delay curves can be effected quite well by multiples of the delay distortion of a single average carrier channel after previous subtraction of the delay distortion of two local line transformers, which are, amongst others, inserted as isolation transformers in the two-wire section.

Figure 16 shows the cumulative frequency distribution of the delay distortion. The abscissa indicates how often the delay distortion of an average carrier channel was contained in the delay distortion of the connection. At the lower right of Figure 16 the delay distortion of a reference average carrier channel is plotted as curve 1 and that of two local line transformers as curve 2.

From Figure 16 it can be seen that the connections with test point B comprise, on the average, one carrier channel less than the connections to A and C. This is to be attributed to the fact that contrary to A and C, test point B is connected to a terminal exchange located at the same place as the central exchange.

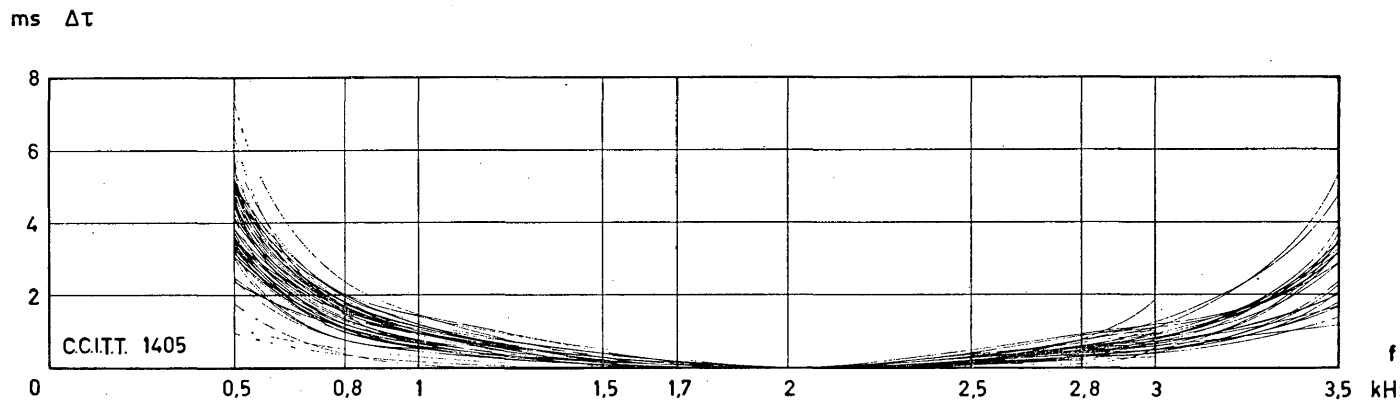
2.4 *Echoes*

The reflections, occurring at both ends of the four-wire section of a telephone connection, give rise to the so-called listener echoes. These listener echoes add up with different phase with respect to the wanted signal as a function of frequency so that the attenuation (and delay) curve of such a connection shows some fluctuations. The "amplitudes" of these fluctuations are the higher the lower the return loss; the frequency spacing between the peaks of these fluctuations is the shorter the longer the echo delay.



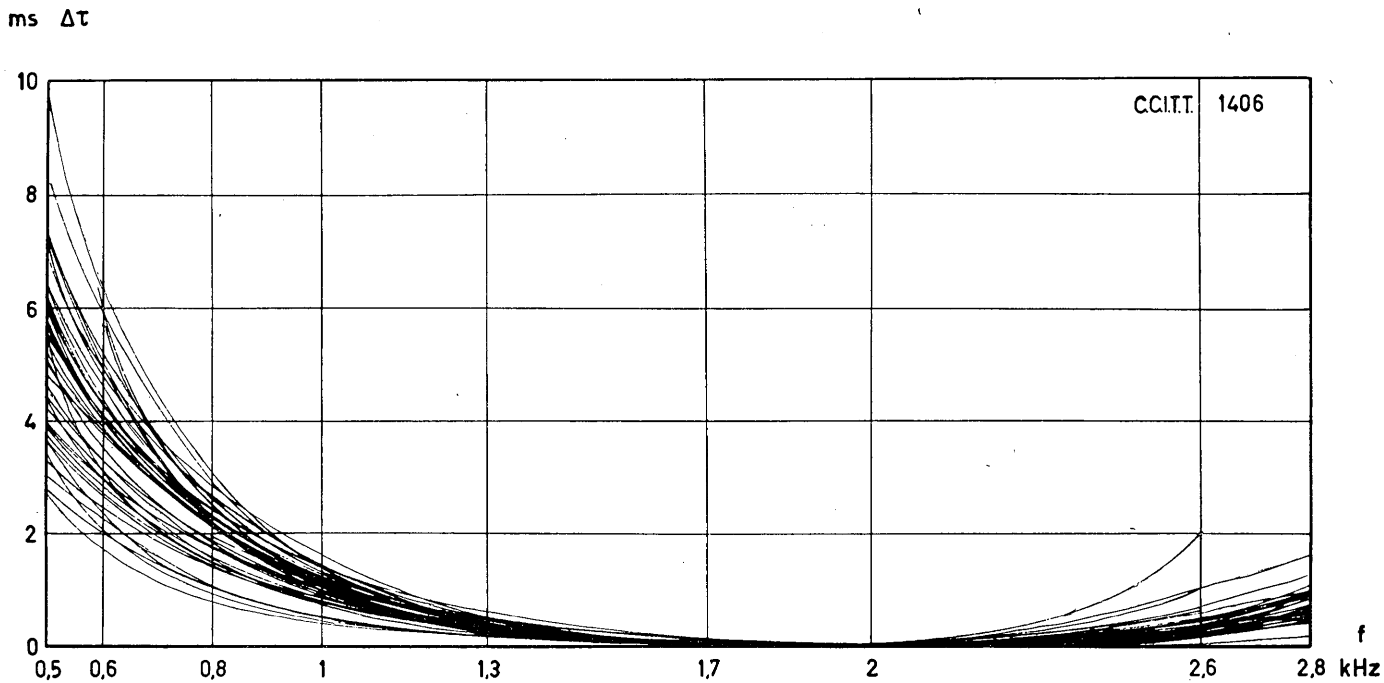
Central test point: A

FIGURE 13. — **Delay distortion in the German telephone network referred to the value at 2000 Hz**



Central test point: B

FIGURE 14. — **Delay distortion in the German telephone network referred to the value at 2000 Hz**



Central test point: C

FIGURE 15. — **Delay distortion in the German telephone network referred to the value at 2000 Hz**

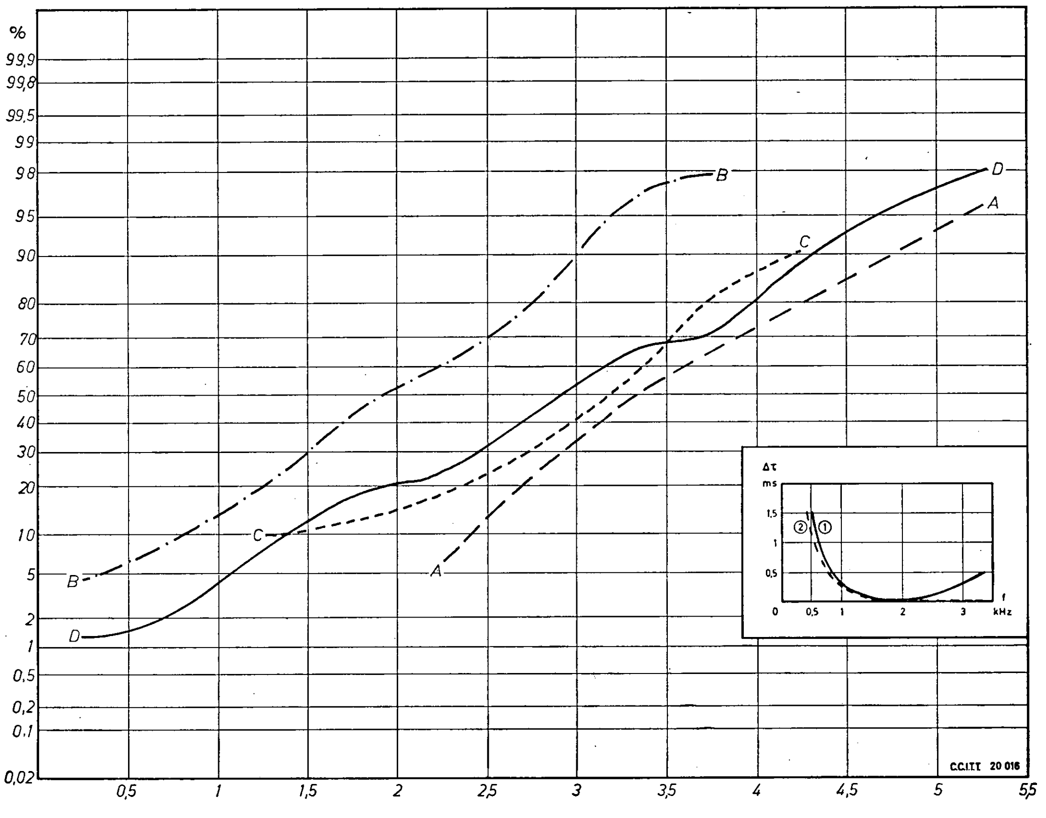


FIGURE 16. — Multiples of delay distortion of one carrier channel

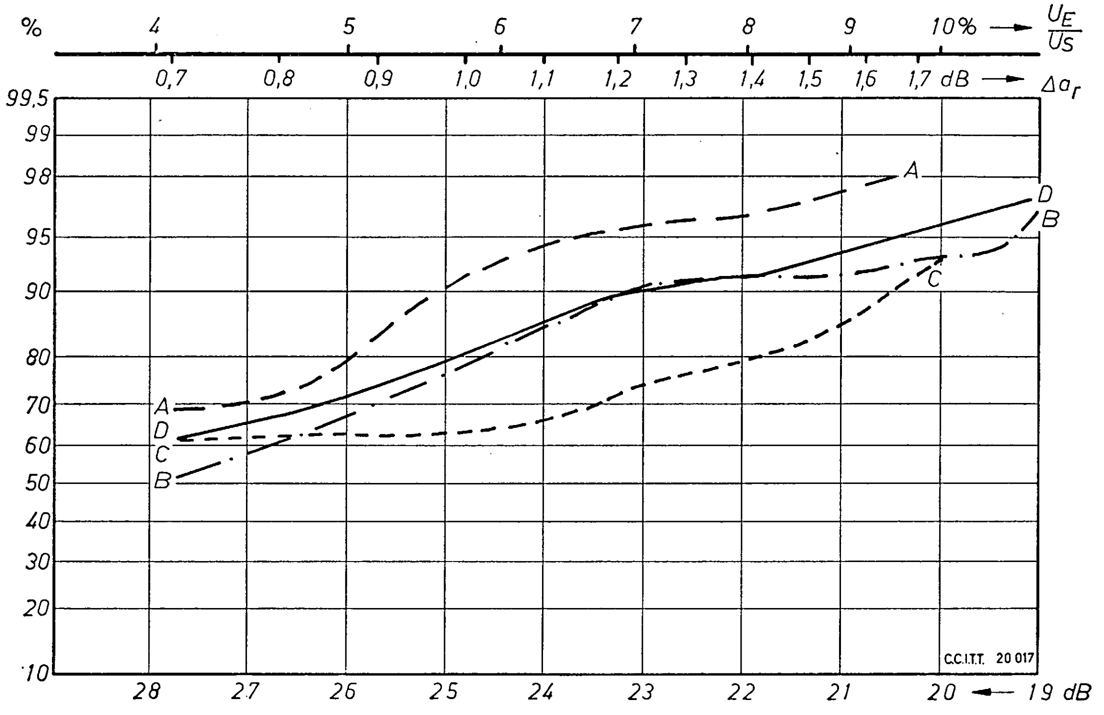


FIGURE 17. — Relative cumulative frequency distribution of the signal-to-listener echo ratio

The minimum signal-to-listener echo ratio of each connection was determined with the help of these fluctuations of the photographed attenuation curves.

The relative cumulative frequency distribution of the signal-to-listener echo ratio is plotted in Figure 17. It reveals that for some 70% of all connections (curve D), the ratio is equal to or greater than 26 dB, i.e. the echo voltage was lower than or equal to 5% of the wanted signal voltage. For about 5% of all connections the ratio was less than or equal to 20 dB. The differences between the curves A, B and C are due to the different composition of the two-wire sections connecting the three central test points with the long-distance network. When analysing these curves it must, however, be borne in mind that the greatest echo occurring in the entire frequency range from 500 Hz to 3400 Hz—i.e. not only the frequency band of the data and backward channel respectively which is of interest to data transmission—was recorded for each connection. Figure 18 shows the cumulative frequency distribution of the frequencies at which the fluctuations of the attenuation curve and thus the echoes were most pronounced. For connections to test point B the peak values of the echoes were experienced at 1.8 kHz. Even for connections to test point C they were still within the range of the data channel of the V.23 modem, whereas the echoes on the connections with test point A were at their highest levels within the frequency range above the data channel.

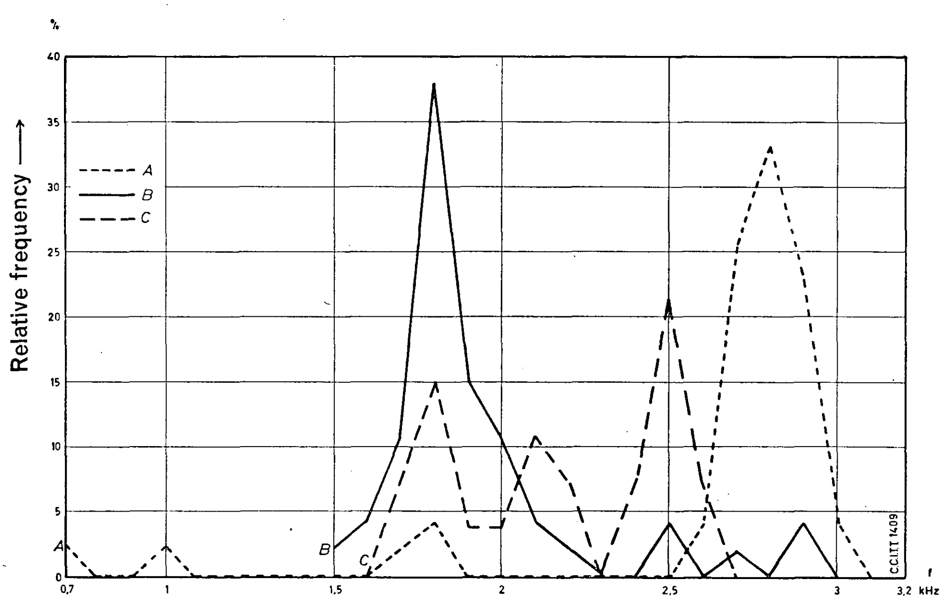


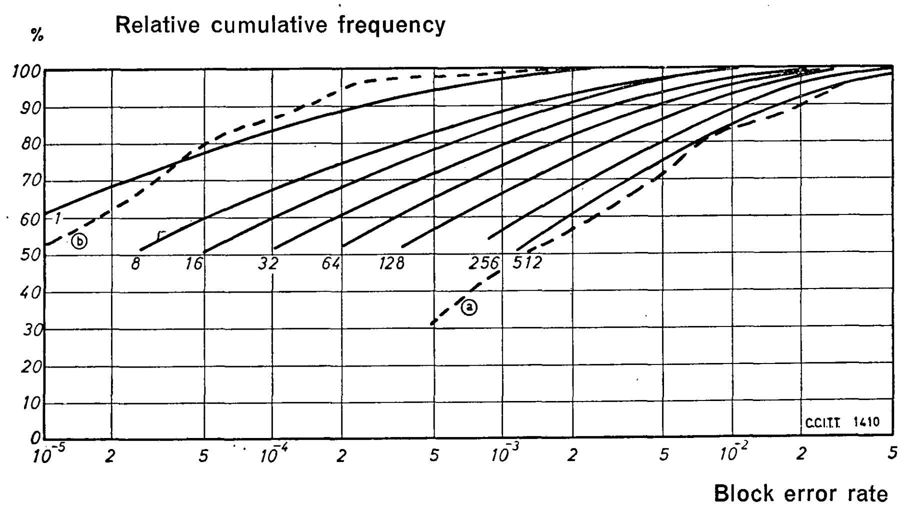
FIGURE 18. — **Relative frequency distribution of the frequencies at which each attenuation curve shows the most pronounced fluctuations**

2.5 *Bit and block error rates for data transmission at 1200 bits/s (Modem V.23)*

These measurements were made only between test point A and the measuring van. Since the previous measurements revealed that the quality of the final selectors of the last exchange in the direction of the establishment of a call has a great influence on the error rate, all connections were set up in the direction from test point A to the measuring van. As a rule, three connections were established with each test point. Three hundred blocks with 1024 bits each (quasi-random programme), i.e. 307 200 bits, were transmitted on each connection. Since the data signalling rate was 1200 bits/s, each transmission took 256 s.

At the receiving end the incoming data were checked bit by bit, and the bit errors were recorded. At the same time the block error rate was determined for the following block lengths: 8; 16; 32; 64; 128; 256 and 512 bits.

The cumulative frequency distribution of the bit and block error rates respectively for the various block lengths is plotted in Figure 19. The bit and block error rates respectively which were not exceeded for 50%, 80% and 95% of all connections, are listed in Table 3 for several block lengths.



1, 8, 16, 32, 64, 128, 256, 512 = bits block length

FIGURE 19

TABLE 3

Block error rate for cumulative frequency %	Block length 1	8	128	512 bits
50	$3.8 \cdot 10^{-6}$	$2.4 \cdot 10^{-5}$	$3.1 \cdot 10^{-4}$	$1.1 \cdot 10^{-3}$
80	$6.6 \cdot 10^{-5}$	$3.8 \cdot 10^{-4}$	$2.8 \cdot 10^{-3}$	$6.8 \cdot 10^{-3}$
95	$6 \cdot 10^{-4}$	$2.7 \cdot 10^{-3}$	$1.8 \cdot 10^{-2}$	$3 \cdot 10^{-2}$

These results indicate a comparatively low error rate on most connections. On the few connections with a relatively high bit error rate (95% value in Table 3), the fact that the errors occurred in the form of bursts proved to be of advantage with regard to the block error rate.

For instance, at a bit error rate of $6 \cdot 10^{-4}$ and uniform distribution of errors the block error rate of blocks of 512 bits would then be approximately 0.31. Because of the concentration of errors in the form of bursts, the block error rate decreased approximately by the factor 10 to $3 \cdot 10^{-2}$. That means that only for 5% of all switched connections established for data transmission the block error rate (blocks with 512 bits) was equal to or higher than $3 \cdot 10^{-2}$.

As already mentioned above, previous measurements revealed that the error rate is considerably influenced by the quality of the final selector, i.e. that in general the majority of errors is caused by impulsive noise arising in the final selectors. Therefore, the connections were divided into two groups. The first group encompassed all connections having two-motion selectors as final selectors, whereas the second group included those with EMD (precious metal uniselector motor) switches. Here, the differences became evident. The error rate of connections with EMD final selectors was, on the average, better by the factor 10 than connections using two-motion selectors as final selectors. Details may be learnt from Figures 20, 21, 22, and 23. Curve ① applies always to connections set up by means of two-motion final selectors and curves ③ to EMD final selectors. Curves ② (which are also plotted in Figure 19) refer to the total number of tested connections.

2.6 *Isochronous distortion*

The isochronous distortion was measured on all connections established for the data transmission tests. These measurements were made by means of the SRC error rate analyser mentioned in 1 above. For the purpose of these tests data signals were transmitted synchronously at a speed of 1200 bits/s. The blocks had a length of 1024 bits. At the receiving end those bits could be counted whose distortion was equal to or higher than the degree of distortion to which the measuring instrument was set. The degree of distortion can be adjusted in steps of 5%. The result of a distortion measurement was indicated as the range at the lower end of which the counter responded rather frequently and at the upper end of which it responded not at all or only every now and then (because of impulsive noise). The measuring time was about 1½ minutes (i.e. approximately 100 000 bits).

In Figure 24 the bit error rate is plotted for all tested connections as a function of the isochronous distortion. The distortion levels were entered within the relevant 5% range. Each plotted point represents one connection. Connections, on which no errors occurred during data transmission are reproduced by points below the dashed line plotted at $3.25 \cdot 10^{-6}$. (Since $300 \times 1024 = 307\,200$ bits were transmitted during each test, it can only be said that for an error-free transmission the error rate was $< 3.25 \times 10^{-6}$).

Figure 24 shows that as far as data transmission over the general telephone network is concerned, there is no correlation between isochronous distortion and bit error rate. This result was to be expected since isochronous distortion is mainly caused by attenuation and delay distortions and the basic noise of the connection, whereas errors are chiefly to be attributed to impulsive noise; which, in turn, depends almost exclusively on the selector quality. Since, on the other hand, impulsive noise has an amplitude, which is generally in the order of magnitude of the signal voltage, it does not matter very much whether the data signal had already been adversely affected by linear distortion on the line or not.

2.7 *Impulsive noise rate*

In order to be able to analyse the noise voltages occurring on the switched connections from various points of view, the noise voltages measured on 171 connections were recorded on magnetic tape for approximately 4½ minutes.

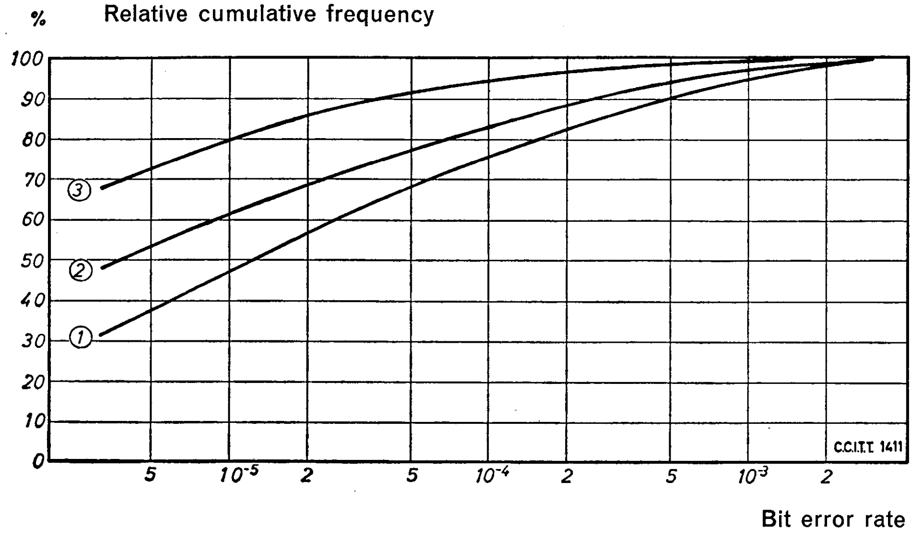
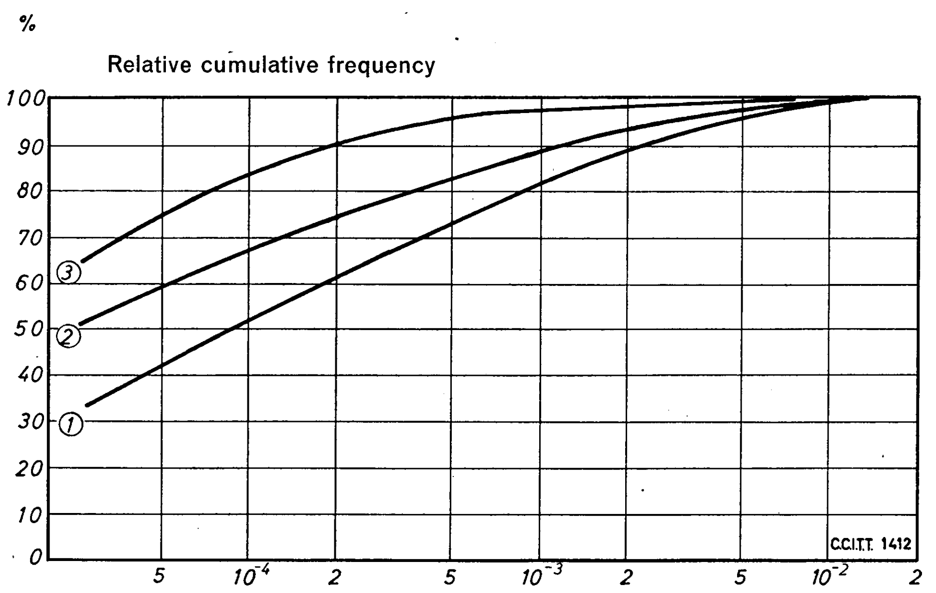
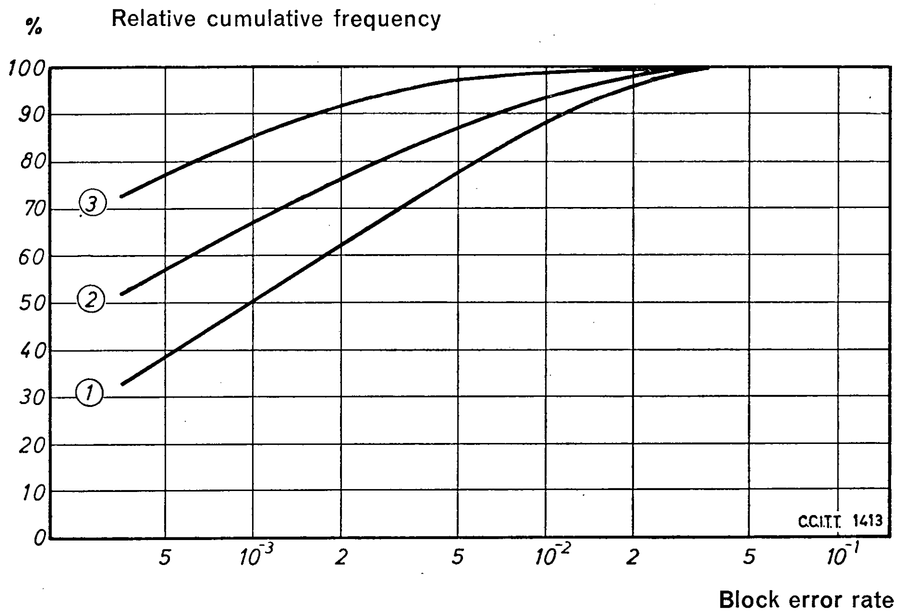


FIGURE 20



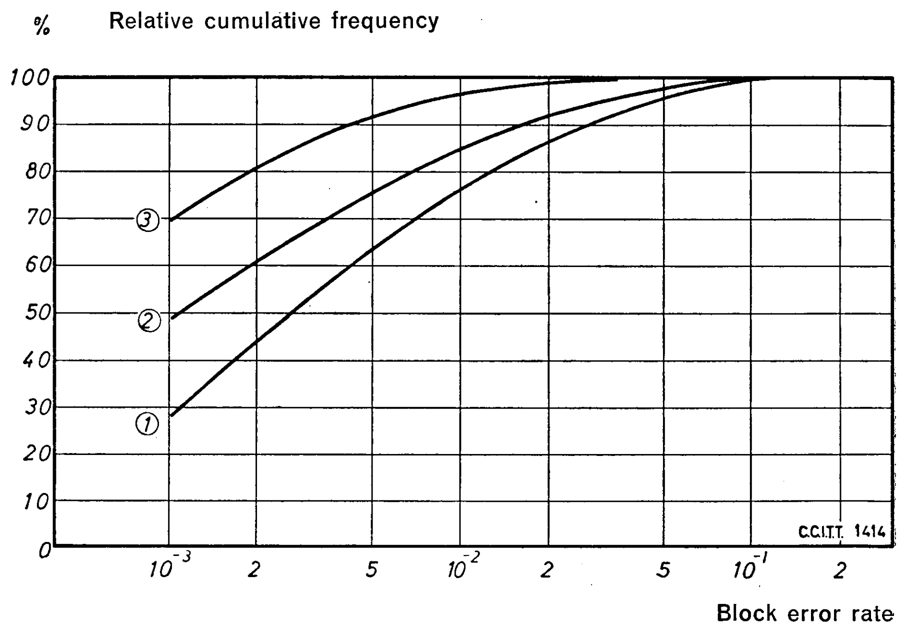
Block length: 8 bits

FIGURE 21



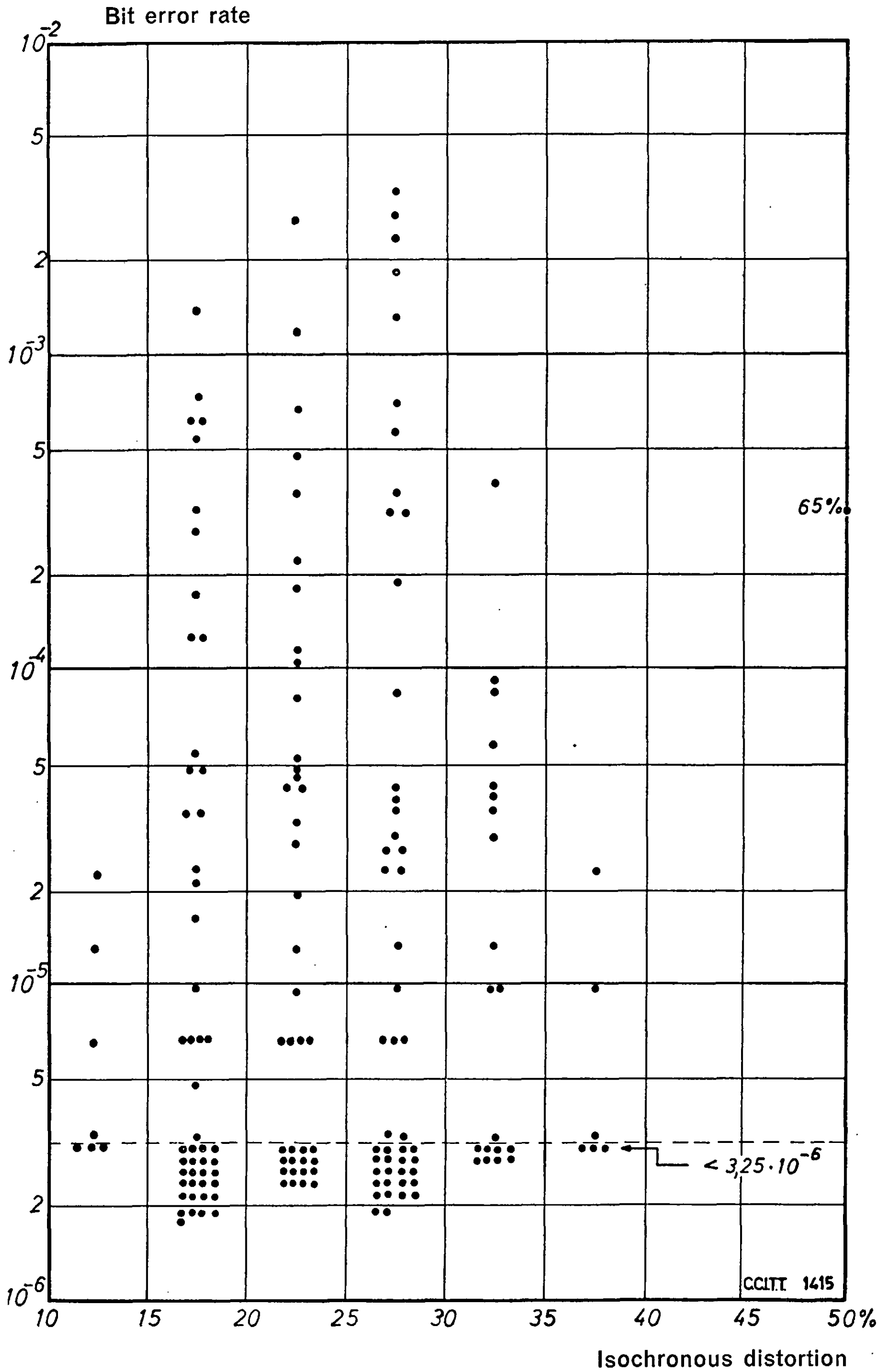
Block length: 128 bits

FIGURE 22



Block length: 512 bits

FIGURE 23



Bit error rate
10⁻²
5
2
10⁻³
5
2
10⁻⁴
5
2
10⁻⁵
5
2
10⁻⁶
65%
< 3,25·10⁻⁶
CCITT 1415
10
15
20
25
30
35
40
45
50%
Isochronous distortion

FIGURE 24

Later on, the tape records were analysed by means of a pulse measuring instrument (instrument constructed in the laboratories of Siemens AG), the electrical characteristics of which comply with those specified in Recommendation V.55.

For this analysis the data signal was transmitted over an artificial line having an attenuation and delay distortion corresponding to that of 3 carrier channels connected in tandem. At the end of this artificial line the recorded noise was injected via a hybrid so that the data-signal-to-noise ratio corresponded to the conditions existing when recording the noise. The number of bit errors was measured in this way. At the same time the impulsive noise was recorded at various operating thresholds. On this occasion it was found that the number of recorded noise pulses is quite close to the number of bit errors, provided that the operating threshold of the pulse noise measuring instrument is equal to the level of the data signal.

The cumulative frequency distribution of the number of bit errors per noise pulse was plotted in Figure 25. For this purpose only such noise pulses were recorded the amplitude of which reached or exceeded that of the data signal.

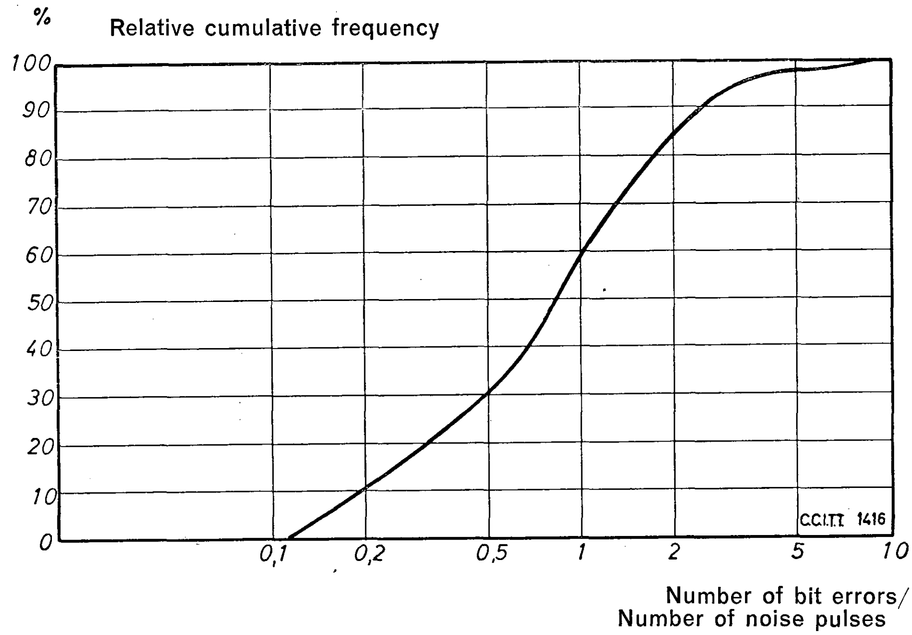
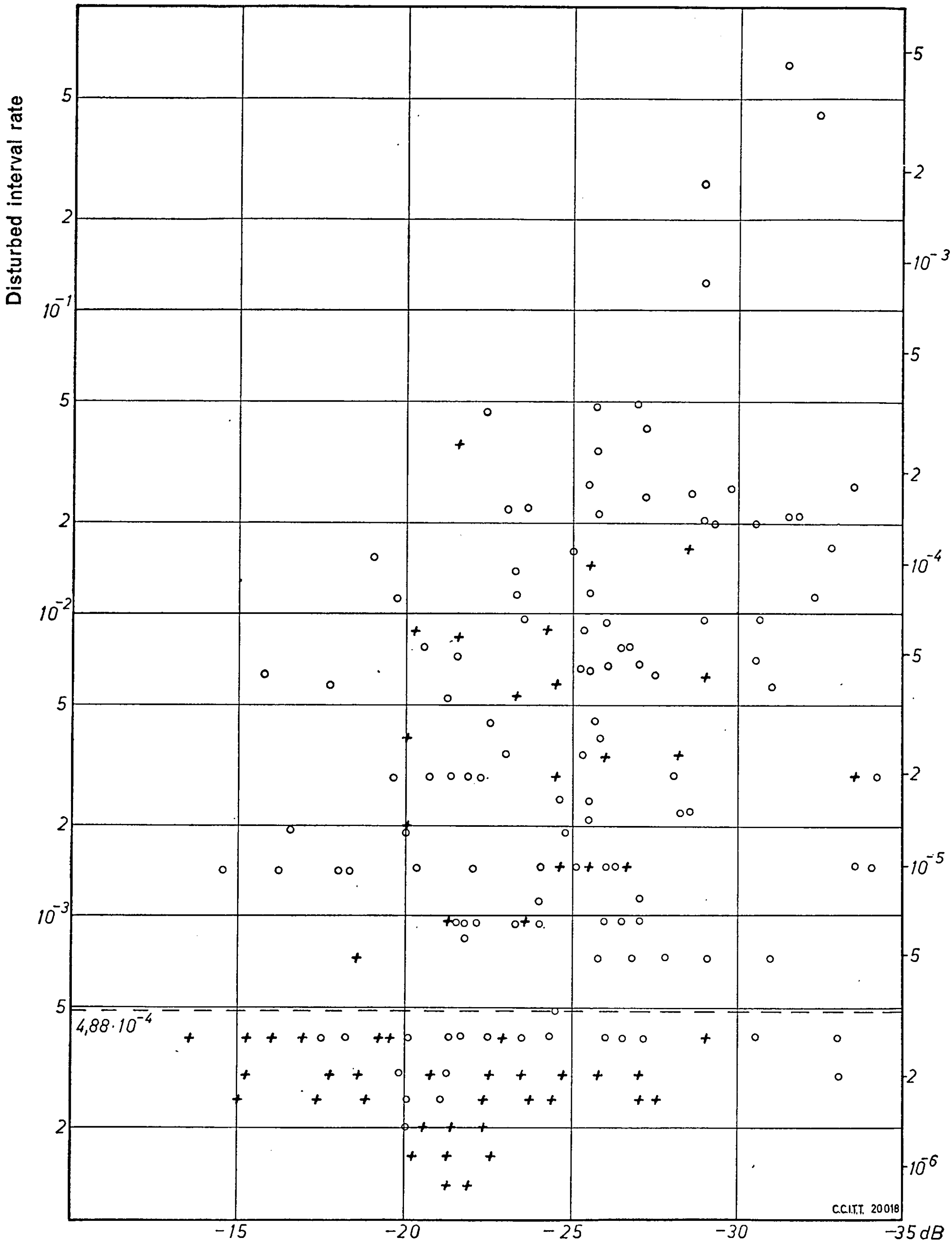


FIGURE 25

Figure 25 shows that about 60% of the cases fall into the range of 0.1 ... 1 bit error per noise pulse and about 40% in the range of 1 ... 10 bit errors per noise pulse. The average value is about 0.8 bit errors per noise pulse. Approximately 75% of all cases are in the range of $1/3$ ... 3 bit errors per noise pulse.

At first, the result that, on the average, only such impulsive noise gives rise to errors the amplitude of which is equal to or higher than that of the data signal seems to be surprising. It can mainly be explained by the facts that the bandwidth of the pulse noise measuring instrument is considerably wider than that of the data channel of the V.23 modem and the point of concentration of the pulse noise energy is outside the data channel but within the passband of the pulse noise measuring instrument.

Figure 26 shows the "disturbed interval rate" versus the receiving signal level. The "disturbed interval rate" is defined as the ratio



O Two-motion selectors + Emd-selectors

FIGURE 26

$$\frac{\text{number of disturbed 125 ms-intervals}}{\text{number of 125 ms-intervals during the test period}}$$

(The dead time of the pulse noise measuring instrument is 125 ms.) Before every recording the operating threshold of the pulse noise measuring instrument was set to the value of the receiving signal level of the connection concerned. (Operating threshold = transmitting level of the data signal minus overall loss of the connection at 1700 Hz.) The circles indicate connections with two-motion final selectors and the crosses connections with EMD final selectors. The connections for which no impulsive noise was recorded are plotted below the dashed line at $4.88 \cdot 10^{-4}$.

At first, it is to be expected that the cumulative frequency distribution of the disturbed interval rate is similar to the cumulative frequency distribution of the block error rate for one block length of 125 ms $\cdot$ 1200 bits/s = 150 bits.

In Figure 19 the disturbed interval rate is plotted as dotted curve ⓐ. It is apparent that it is only inferior by the factor 3 to the block error rate for 150 bit-blocks. However, when converting disturbed interval rate into the bit error rate by forming the ratio of the number of noise pulses to the number of bits transmitted during the test period at a speed of 1200 bits/s, we obtained the dashed curve ⓑ of Figure 19. This curve agrees quite well with the cumulative frequency curve for the bit error rate. In this connection it should be mentioned that curves ⓐ and ⓑ were determined from all recorded tapes so that they include also connections between the measuring van and the central test points B and C.

Summing up, one result of the impulsive noise measurements is that the pulse noise measuring instrument complying with document COM Sp. A No. 66 allows a rough estimate to be made of the bit error rate to be expected on a connection of the general telephone network when transmitting data at a speed of 1200 bits/s (V.23 modem). For this purpose the operating threshold is to be set to the value which the level of the data signal will have at the respective point of the connection.

Reference:

M. Williams, The characteristics of telephone circuits in relation to data transmission, *P.O.E.E.J.*, October 1966.

Supplement No. 15

UNITED KINGDOM. — *(Contribution COM Sp. A—No. 129—October 1967)*

TESTS FOR DATA TRANSMISSION SYSTEMS ON TELEPHONE-TYPE CIRCUITS

The following statistics were collected on a looped leased line over a period of three weeks. The transmission was synchronous at 1200 bauds and used a repetitive 4092-bit pseudo-random pattern.

Line and modems

The line composition was:

Two-wire unloaded local cable	—	1.3 miles
Four-wire loaded cable 20/88/1.136	—	9.6 miles

Telephone carrier channel	—	133.0 miles
Four-wire loaded cable 20/88/1.136	—	81.0 miles
Telephone carrier channel	—	150.0 miles
Four-wire loaded cable 20/88/1.136	—	9.6 miles
Two-wire unloaded local cable	—	1.3 miles

This gave the following overall loss and group delay characteristics:

Frequency (Hz)	Gain (dB)	Group delay distortion (ms) (relative to the minimum delay)
800	– 9	0.32
1000	– 10	0.09
1600	– 12	0.15
2000	– 16	0.25
2400	– 18	0.99
2600	– 19	2.00

The group delay curve had ripples which did not exceed 0.5 mS peak-to-peak, these being caused by echoes. The terminating modems complied with Recommendation V.23 and no backward channel was used. The signal level was – 10 dBm at the circuit zero reference point. The peak individual distortion at 1200 bauds was 17%.

The receiving equipment was fitted with a commercial synchronizer having a margin less than 49%. The results may therefore be pessimistic compared with those which would have been obtained had the synchronizer been similar to the one proposed for the C.C.I.T.T. data test instrument.

Error analysis

The total number of data bits transmitted was 309 917 819. At certain times, the "carrier failed" condition occurred. Since it is meaningless to describe individual error bits under these circumstances, all bits received during a period of carrier failure longer than 16 bits were classed as unusable. Where the carrier was restored for a period of not more than 48 bits between two occurrences of carrier failure, the intervening bits were also classed as unusable, as it was found that those bits usually bore no relation to the transmitted bits. Therefore it can be seen that errors in such "unusable" bits are of less consequence than errors which occur without warning. The total number of unusable bits was 878 421 representing a total time of = 10 min, 12 s.

For engineering reasons, a total of 326 bits which occurred in the region of error bursts could not be recorded by the equipment. An attempt was therefore made to extrapolate some of the results to compensate for these unrecorded data.

Results

TABLE 1

	Recorded		Extrapolated
Total number of error bits	1 576		1 654
Bit error rate — 1 bit in	196 648		187 375
Number of bits changed from 0 → 1	981		1 030
Number of bits changed from 1 → 0	595		624
Percentage of corrupted bits 0 → 1		62.2%	
Percentage of corrupted bits 1 → 0		37.8%	

The distribution of error groups of different sizes was:

TABLE 2

	Recorded	Extrapolated
Number of 1 bit errors	584	612 occurrences
Number of 2 bit errors	204	214 occurrences
Number of 3 bit errors	73	76 occurrences
Number of 4 bit errors	43	45 occurrences
Number of 5 bit errors	16	16 occurrences
Number of 6 bit errors	9	9 occurrences
Number of 7 bit errors	5	5 occurrences
Number of 8 bit errors	3	3 occurrences
Total number of error groups	937	983
The average number of bits in an error group = 1.68		

The distribution of sequences of uncorrupted bits was:

TABLE 3

Sequences between	1	and	10 bits long	741 occurrences
Sequences between	11	and	100 bits long	77 occurrences
Sequences between	101	and	1 000 bits long	16 occurrences
Sequences between	1 001	and	10 000 bits long	31 occurrences
Sequences between	10 001	and	100 000 bits long	22 occurrences
Sequences between	100 001	and	1 000 000 bits long	44 occurrences
Sequences between	1 000 001	and	10 000 000 bits long	82 occurrences

A sequence of uncorrupted bits may be terminated at either end by either a corrupted bit or the beginning or end of a test period.

The variation of bit error rate over different days of the week is:

TABLE 4

Monday	1 bit in	1 647 283
Tuesday	1 bit in	1 272 290
Wednesday	1 bit in	55 770
Thursday	1 bit in	3 006 290
Friday	1 bit in	647 140

For Wednesdays only, the variation of bit error rate over the times of day is:

TABLE 5

9.00 hrs	10.00 hrs	1 bit in	75 343
10.00 hrs	11.00 hrs	1 bit in	14 368
11.00 hrs	12.00 hrs	1 bit in	46 014
12.00 hrs	13.00 hrs	1 bit in	1 277 811
13.00 hrs	14.00 hrs	1 bit in	12 876 768
14.00 hrs	15.00 hrs	1 bit in	79 529
15.00 hrs	16.00 hrs	Indeterminate—better than 1 bit in	6 425 000
16.00 hrs	17.00 hrs	1 bit in	10 261
17.00 hrs	17.30 hrs	Indeterminate—better than 1 bit in	2 140 000

Comments. — These line characteristics are such that performance in the face of uniform spectrum random noise would be worse for binary 0 than binary 1.

Table 1 shows such a characteristic but it does not follow that the noise spectrum over the data band was therefore uniform.

Table 2 gives the distribution of various numbers of consecutive errors. The proportion of double errors compared to single errors may be artificially high if any errors are due to lack of adequate margin in the synchronizer.

Table 3 stresses the importance of error burst detection in any errord-etecting system. The uncorrupted sequences of 1 to 10 bits are clearly bursts. The large number of sequences 11 to 100 bits illustrates the wisdom of choosing a cyclic code polynomial having double burst detection capabilities (see COM Sp. A—No. 61, page 100, paragraph 3.3).

Although considerable efforts have been made to trace the reason for the poor performance on Wednesdays, none has been found. It would therefore be unwise to draw any conclusions from Table 4 except that considerable variations in error rate from day to day may be experienced.

The results expressed in Table 5 are similar in form to those quoted elsewhere and a general conclusion can be drawn that use of a data link in traffic off-peak periods will yield better results than in the busy period.

Supplement No. 16

UNITED KINGDOM. — *(Extract from Contribution COM Sp. A—No. 27—October 1965)*

COMPLEMENT TO STANDARDIZATION OF THE MODEM AT 600/1200 BAUDS

a) *Automatic Selection between rates of 600 and 1200 bauds*

The usual reasons for changing from the 1200 to the 600 baud transmission rate would be that the quality of the line was so low that data cannot be conveyed at the higher rate. Alternatively it may be transmitted more accurately or with a higher transfer rate using 600 bauds than using 1200 bauds owing to the larger number of repetitions required at 1200 bauds. The particular amount of line degradation which would warrant such a change varies according to its characteristics and the particular error control system in use. It would seem therefore that the data signalling rate should be selected by the error control terminal and not by the modem unless the former is part of the modem. This control can be effected over Interchange circuit 11.

Automatic rate reduction from 1200 to 600 bauds could be simply achieved by the error control apparatus by using a timing device in the terminal equipment at each end of the link. If unsatisfactory conditions at 1200 bauds were detected continuously for some selected time, the lower receiving rate could then be selected automatically by the terminal equipment. The 600-baud rate would then be used for the remainder of the data call.

. . . .

c) *Equalization in the modem of phase distortion on calls*

In the United Kingdom modem—Datel modem No. 1A—group delay correction has been applied permanently to the send and receive filters. The group delay/frequency characteristic of the transmitting terminal is such that the differential group delay at any frequency relative to any other frequency in the range 1100 Hz to 2300 Hz is less than 0.2 ms, in the range 950 Hz to 1100 Hz is less than 0.6 ms, and in the range 2300 Hz to 2500 Hz is less than 0.5 ms. At the receiving terminal the equalization is such that a data connection has an improved performance on circuits containing up to 100 miles of standard loaded cable (20/88/1.136) compared with the back-to-back performance. With this arrangement additional compromise group delay correction on practical connections over the United Kingdom switched network has so far produced only marginal improvement.

Supplement No. 17

ITALY. — *(Extract from Contribution COM Sp. A—No. 42—October 1965)*

COMPLEMENT TO STANDARDIZATION OF THE MODEM AT 600-1200 BAUDS

a) *Automatic selection between the rates of 600 and 1200 bauds*

It is not felt necessary to complement the standard 600-1200 modem with additional functions for the automatic selection of the modulation rate, since all decisions concerning the rate may be taken automatically within the data processing equipment and communicated to the modem, via the interface circuit No. 11. This may be done as follows.

As soon as the data terminals are connected to the line, they must find themselves in one fixed rate condition, say 600: the modems are also held in the 600 rate condition through interface circuit 11.

If the terminals are to be switched to the 1200 rate condition, the transmitter sends at 600 bauds a special character, or sequence, thereafter it switches to the new condition, forcing also also the modem to switch.

The receiving data terminal, upon receiving the special character or sequence, also switches to the 1200-rate condition and forces the modem to do so through the interface circuit 11.

b) *Equalization in the modem of phase distortion in connection*

It is not felt necessary to equalize, from case to case, phase distortion in the 600-1200-baud modem. There may well be a fixed phase equalization within the modem, but it seems not convenient to ask the operator to make a selection among two or more phase equalizations, according to the particular connection.

Moreover, the possibility of selecting between the two rates, 600 or 1200, seems to be sufficient to cover the great majority of possible connections.

Automatic equalization, on the other hand, with its rather sophisticated circuitry, seems to fit more advanced types of modems.

Supplement No. 18

FRANCE. — *(Contribution COM Sp. A—No. 184—December 1967)*

DRAFT SPECIFICATIONS FOR A STANDARD CLOCK TO BE INCLUDED IN THE 600-1200-BAUD MODEM REFERRED TO IN RECOMMENDATION V.23

1. *Introduction*

The 600-1200-baud modem specified in Recommendation V.23 can be used for synchronous as well as asynchronous operation.

At a time when the I.S.O. is specifying procedures for the use of data transmission terminal equipment, it is a matter of urgency to draw up specifications for a built-in modem clock to make sure that there is no incompatibility between the operation of this clock and the new procedures.

The French P.T.T. Administration has decided to make provision for the possibility of including a built-in clock in the standardized modem it is at present designing. In preparing specifications for this clock, it has had to make a number of assumptions on the nature of the signals to be transmitted. The specifications that have been established correspond to the transmission of isochronous signals in a code of eight units, including a parity unit, and in particular in the I.S.O./C.C.I.T.T. code. The same specifications are, however, applicable to the transmission of codes having more than eight units but in this case the permissible distortion or modulation rate limits have to be re-examined.

The following draft is a contribution to this work of standardization:

2. *Characteristics of the signals sent by the clock*

2.1 *Circuit 114, time base for signal elements at the transmitter (Source: data transmission equipment)*

2.1.1 *Modulation rate*

The modulation rates in the data transmission channel shall be 600 or 1200 bauds depending on the indications given by circuit 111 (data signalling rate selector).

2.1.2 *Tolerance on the frequency*

The frequency of the time base signals shall not deviate by more than 0.01 % from the assigned value.

2.1.3 *Symmetry*

The half-cycle t of the time-base signals (OFF or ON condition) shall be such that

at 600 bauds	$666 < t < 1000$ µs
at 1200 bauds	$333 < t < 500$ µs

This corresponds to a service cycle (ratio between the OFF or ON condition and the full cycle of (50 ± 10) %.

2.1.4 *Distortion*

The degree of isochronous distortion in the ON to OFF transitions of the time-base signal shall not exceed 1 % at the modulation rate of the data signals.

2.1.5 *Note*

The time-base signals shall be kept in the OFF condition when circuit 105 (Request to send) is also in this condition.

2.2 *Circuit 115, time base for signal elements at the receiver (Source: data transmission equipment)*

2.2.1 *Modulation rate*

The modulation rates in the data transmission channel shall be 600 or 1200 bauds depending on the indication given by circuit 111 (data signalling rate selector).

2.2.2 *Permissible modulation rates*

The clock shall be synchronized for modulation rates of

600 bauds $\pm$ 0.1 %
1200 bauds $\pm$ 0.1 %

provided that the signals received do not contain any bursts of more than 15 consecutive bits without transition.

Note. — For a given time-base signal distortion, the permissible deviation of the modulation rate is inversely proportional to the number of consecutive bits without transition.

2.2.3 *Symmetry*

The half-cycle t of the time-base signals (OFF or ON condition) shall be such that

at 600 bauds	$666 < t < 1000$ µs
at 1200 bauds	$333 < t < 500$ µs

This corresponds to a service cycle (ratio between the OFF or ON condition and the full cycle) of (50 ± 10) %.

2.2.4 *Distortion*

The degree of isochronous distortion of the ON to OFF transitions is 15 %.

In circuits set up on a permanent basis in which the synchronizing time is not a factor to be considered, this degree of isochronous distortion can be reduced to 7.5%. The figures given in brackets below are proposed for connections of this type.

2.2.5 *Position of* ON *to* OFF *transitions with respect to the transitions of the signal*

If the clock is fed with undistorted signals, the ON to OFF transitions shall not deviate by more than ± 7.5% (or ± 3.75%) from the middle of each signal element on circuit 104.

2.2.6 *Synchronizing time*

If the clock is fed with undistorted signals produced by a pseudo-random 511-bit generator, the clock shall fulfil the conditions specified in 2.2.5 before 8 (or 16) transitions of the input signal. These conditions correspond to 2 (or 4) SYNCH characters of the I.S.O./C.C.I.T.T. 7-bit code.

2.2.7 *Note 1*

The time-base signals for signal elements at the receiver may be sent only if the following two conditions are met:

— circuit 105 (request to send) is in the OFF position

— circuit 109 (data carrier detector) is in the ON position.

Moreover, in alternate working after each transition of circuits 105 and 109 from the OFF to the ON position, the time-base signals for the signal elements at the receiver may be sent only if their phase is considered correct.

2.2.8 *Note 2*

The signals of circuit 104 (data reception) may not be regenerated by receiving time base signals. This is necessary, for example, if they are to be used to evaluate transmission quality.

3. *Example of work*

In constructing a clock to be built into the modem, use can be made of a quartz oscillator set at a frequency of $(2n - 1)\ F$, where F is the nominal modulation rate, followed by division stages for which there may be three division rates: $2n - 2$, $2n - 1$, and $2n$.

When a modem is operated on an alternate-working basis, the same clock can be used for transmission and reception. When circuit 105 (Request to send) is in the ON position, the value for the division rate is $2n - 1$ and the output signals are supplied over circuit 114. When circuit 105 is in the OFF position, the division rate is made variable by a device which compares the respective positions of the clock and the transitions of the signals received over circuit 104. The signals thus obtained are transmitted only over circuit 115 if circuit 109 is in the ON position and if, in addition, a pre-determined phase relationship has been established between the clock and the transitions of the signals received each time circuit 105 passes from the ON to the OFF position.

The French Administration has provisionally established two division rates corresponding to $n = 4$ and $n = 5$, i.e. 15 and 31.

On this basis, the characteristics of the signals sent by the clock are as follows:

	Characteristics of time-base signals	$n = 4$	$n = 4$
2.1.2	Tolerance on the frequencies	0.01%	0.01%
2.1.3	Symmetry at the transmitter	< 7%	< 4%
2.1.4	Degree of isochronous distortion at the transmitter	< 1%	< 1%
2.2.2	Permissible modulation rate, 600 or 1200 for signals not containing bursts of more than 15 consecutive bits without transition and for a degree of isochronous distortion as specified in 2.2.4	± 0.1 %	± 0.05%
2.2.3	Symmetry at the receiver	< 7%	< 4%
2.2.4	Degree of isochronous distortion at the receiver	13.33%	6.66%
2.2.5	Position of transitions	± 6.66%	± 3.33%
2.2.6	Maximum synchronizing time (number of transitions)	8	16

Supplement No. 19

UNITED KINGDOM. — *(Mar del Plata—September 1968)*

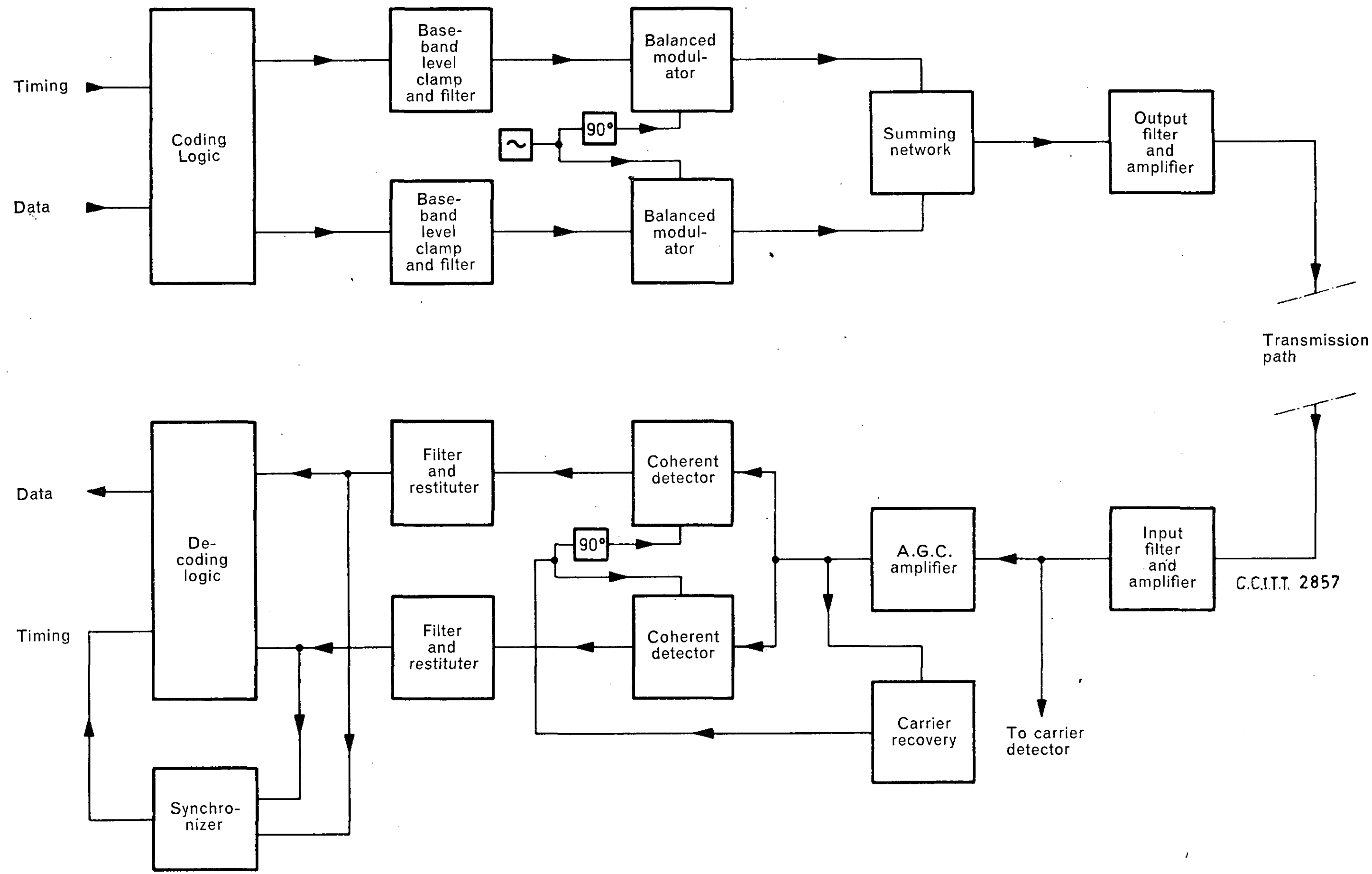
A METHOD OF MODULATION AND DEMODULATION FOR A FOUR-PHASE LINE SIGNAL

The British Post Office Datel modem No. 7 transmits data in the code conforming to Alternative A of Recommendation V.26. Referring to the figure attached, it can be seen that the serial data stream is divided, retimed and coded (see the coding rules mentioned below) into two streams (*P* and *Q*), each of which is filtered and then passed through a balanced modulator. The carrier waves feeding these modulators have a 90° phase separation, and thus the modulator outputs can be directly combined to produce the four-phase signal. A common filter is included to remove higher-order modulation products.

Before demodulation, the carrier frequency is recovered from the signal (see figure). The standard amplitude signal is frequency quadrupled by a pair of tandem full-wave rectifiers, filtered by a single tuned circuit and reduced to a square wave. This is divided by four in such a way that two carrier frequency outputs at 90° phase separation are provided.

Demodulation is the inverse of modulation. The standard amplitude signal is split into two by a pair of coherent detectors, one being fed by the leading carrier and the other by the lagging carrier. The resulting baseband signals are restituted and these transitions fed to the bit synchronizer. The synchronizer generates the inspection instants and the regenerated signals are decoded and reassembled to form a serial bit stream.

The synchronizer is equipped with a rapid start-up facility whereby the synchronizing steps following the initial detection of the carrier are large. Once acceptable element synchronism has been achieved, the synchronizer steps are reduced to ½% of an element.



British Post Office 2400 bits/s modem

The coding rules are:

To produce 0° phase shift, states P and Q remain unaltered.

To produce -180° phase shift, states P and Q are both altered.

To produce -90° phase shift, only P is altered when P was the same as Q, only Q is altered when P was different from Q.

To produce -270° phase shift, only Q is altered when P was the same as Q, only P is altered when P was different from Q.

This coding has a limitation in that, if the receiver clock has an accuracy of $\pm R\%$ and the synchronizer timing can be altered by $\pm T\%$ of a signal element per significant instant, then the number of consecutive binary "0"s occurring at any point within the data streams should be limited to $\left(\frac{2T}{R + 0.01}\right)$ if the modem conforms to Recommendation V.26. Suggested values are $R = 0.01$, $T = 0.5$ giving a limitation of 50 binary "0"s.

Supplement No. 20

UNITED KINGDOM. — *(Extract from Contribution COM Sp. A—No. 132—October 1967)*

CODING METHODS FOR PARALLEL TRANSMISSION

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6. *Arrangements particular to the ($3 \times {}^1/_4$) system*

6.1 Normally, an inter-character separator is used consisting of the combination A1, B1, C1. A modulation rate of 40 bauds therefore yields a transfer rate of 20 characters per second with 63-data combination available.

6.2 The outstation incorporates a keyboard for variable data. In addition to keys for digits 1 to 9 and zero, the keyboard has 5 keys marked with geometric symbols to which the customer may assign meanings of his own choice. The data combination is transmitted for the entire period of key depression. A further key controls the transmission of an "end-of-block" control combination for a limited period before the group oscillators are silenced.

6.3 The card reader reads any standard 80-column punched tabulator card having nominal dimensions $7^3/_8'' \times 3\frac{1}{4}''$ (approximately 18.7 cm $\times$ 8.25 cm) when punched in accordance with any of the codes specified in British Standard 3174: 1959. This type of card requires up to 39 different combinations. Blank columns are indistinguishable from the inter-character separator.

6.4 The basic 16 characters are provided as a subset of the expanded 64 and thus no coding change between the basic and expanded systems is required: both are provided by a $3 \times$ (1 out of 4) selection, as set out in Table 2. This is regarded as important since it is usually necessary to include transmissions from a keyboard in conjunction with transmission from a card reader catering for more than 16 combinations.

TABLE 2

Card system alphabet

Group A frequency	Group B frequency	Card or keyboard meaning with group C frequency as below:			
		$f1$	$f2$	$f3$	$f4$
$f4$	$f4$	1		J	A
$f4$	$f3$	2	S	K	B
$f4$	$f2$	3	T	L	C
$f4$	$f1$	4	U	M	D
$f3$	$f4$	5	V	N	E
$f3$	$f3$	6	W	O	F
$f3$	$f2$	7	X	P	G
$f3$	$f1$	8	Y	Q	H
$f2$	$f4$	9	Z	R	I
$f2$	$f3$			End of block	
$f2$	$f2$				
$f2$	$f1$				
$f1$	$f4$				
$f1$	$f3$				
$f1$	$f2$				
$f1$	$f1$	Inter-character separator	Zero		

6.5 An interface can be provided which is used to modulate the group oscillators directly and thus provide for the use of all 64 frequency combinations. This is a contact interface in 3 × (1 out of 4) form having 3 wires to each group and a common return. Frequency f_0 is assumed where no wire within a group is connected to common return.

When the inter-character separator is omitted the character transfer rate may be doubled.

6.6 The instation is fundamentally a modem only because of the variety of data-processing equipments required by different users. The output is presented in parallel form by 12 interchange wires, each corresponding to a line frequency.

A data-checking facility is available depending on the redundancy in the 3 × (1 out of 4) code and persistence checks of both the data combination and its preceding separator. Only the characters which are adjudged correct are then presented on the 12 wires. Two further wires indicate the character timing and checking failures respectively. This data-checking facility is not available when the inter-character separator is not used.

7. *Arrangements particular to the tape system*

7.1 Integral with the modem, a paper tape reader is provided which is capable of reading at 20 characters per second all normal tapes having from 5 to 8 tracks, or cards of similar punching. Quaternary groups A and C are used to convey the data with each character being sent in two sequential halves. The higher frequency (1480 Hz) of the timing channel is transmitted simultaneously with the first half of a character (tracks 1-4).

7.2 The tracks are paired thus: 1 and 2, 3 and 4, 5 and 6, 7 and 8. Each pair is considered as a binary number, the higher track having the higher significance, and a hole being interpreted as binary 1. The frequencies $f0$ to $f3$ within a group corresponds to this binary number. Group A is modulated by track pairs 3 and 4, 7 and 8.

7.3 When the instation modem contains no data-checking facility the interface includes eight wires for the data (one for each line frequency in groups A and C) and a ninth wire indicates the state of the timing channel.

7.4 When a data-checking unit is fitted at the instation, not only are errors detected, but also the first half of a character is decoded and stored until the second half is received and similarly treated. The whole character is then presented on the eight wires referred to in para. 7.3 in parallel form corresponding to an assumed 8-track input. Two further wires indicate checking failures and character timing respectively.

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Supplement No. 21

FEDERAL REPUBLIC OF GERMANY. — *(Extract from Contribution COM Sp. A— No. 167—December 1967)*

PROPOSALS OF CODING FOR PARALLEL TRANSMISSION

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5. *Proposals for coding*

The following prerequisites are taken as a basis:

a) Both numeric and alphanumeric data (sets of 16 or 64 combinations) should be transmitted using the same frequency allocation scheme.

b) Outstations operating on numeric code and outstations operating on alphanumeric code should be capable of inter-operating with a common instation.

c) Transmission should proceed using already established alphabets or subsets thereof.

5.1 *Basic considerations*

Each character consists of two parts. Their durations may have any, including different (arhythmic mode), values above 25 ms. The first part is a combination of frequencies characteristic for the character to be transmitted. The second part is the rest combination, which serves to derive the character timing. The characteristic and the rest combination consist in the numerical system of one frequency out of the groups A and C and in the alphanumerical system of one frequency out of the groups A, B and C.

5.2 *Alphabet for numerical and alphanumerical parallel transmission*

In Figure 1 a proposal is given for alphabets for numerical (2 × 1 out of 4) and alphanumerical (3 × 1 out of 4) parallel transmission. The two are compatible:

Bits $b_7\ b_6\ b_5\ b_4\ b_3\ b_2\ b_1$		Row \ Column			$f_{B/3}$	$f_{B/4}$			$f_{B/1}$	$f_{B/2}$
					2 2)	3			6	7
$f_{C/1}$	$f_{A/1}$	0				0			2)	p
	$f_{A/2}$	1				1			a	q
	$f_{A/3}$	2				2			b	r
	$f_{A/4}$	3				3			c	s
$f_{C/2}$	$f_{A/1}$	4				4			d	t
	$f_{A/2}$	5				5			e	u
	$f_{A/3}$	6				6			f	v
	$f_{A/4}$	7				7			g	w
$f_{C/3}$	$f_{A/1}$	8				8			h	x
	$f_{A/2}$	9				9			i	y
	$f_{A/3}$	10				1)			j	z
	$f_{A/4}$	11				1)			k	2)
$f_{C/4}$	$f_{A/1}$	12				EOT			l	2)
	$f_{A/2}$	13				CAN			m	2)
	$f_{A/3}$	14				1)			n	2)
	$f_{A/4}$	15				RC			o	2)

CCITT. 2241

RC = rest combination.

Alphabetical system: one frequency out of groups A, B and C.
Numerical system : one frequency out of groups A and C; only column 3 is used.

[1] May be used for graphical symbols or device controls in the numerical system.
[2] May be used for graphical symbols or device controls in the alphanumerical system.

FIGURE 1. — **Alphabet for numerical (2 × (1 out of 4)) and alphanumerical (3 × (1 out of 4)) parallel transmission on the basis of C.C.I.T.T. alphabet No. 5**

One gets the numerical alphabet if in the alphanumerical case group B is simply omitted. Hence, an alphanumerical receiving station can be switched over to a numerical one very easily by clamping receive data lead B/4 permanently in ON condition.

The proposed alphabet offers the advantage of being readily convertible into columns 2, 3, 6 and 7 of C.C.I.T.T. alphabet No. 5. Only the signs EOT, CAN and ? will then change their positions. The assignment of the frequencies to the binary digits of C.C.I.T.T. alphabet No. 5 can be seen in Figure 1.

5.3 *Transmission of a 256-character set*

With the proposed system also transmission of a 256-character set is possible. In this case two frequency combinations of the groups A and C are transmitted sequentially. To distinguish the first from the second combination, one of the two middle frequencies of group B (*f* B/2 or *f* B/3) will be added.

Supplement No. 22

CHILE TELEPHONE COMPANY. — *(Extract from Contribution COM Sp. A—No. 50—November 1965)*

ERROR CONTROL IN THE TELEPHONE NETWORK

1. *Introduction*

1.1 *Undetected error rate*

Users are likely to be more interested in the interval between undetected errors than in whether such errors appear in a character or in a block of one size or another.

For a specified period of transmission at any number of bits/ second, the number of characters or blocks transmitted will be inversely proportional to their sizes. As a consequence, the percentages of characters or blocks which may contain undetected errors will not be equal for different block lengths. This point is illustrated in Table 1 (see 2.5).

1.2 *Error detection methods*

In Supplement 40 of Volume VIII of the *Blue Book* information was given on the distribution of error bursts for data transmission at 200 bits/second on a telephone channel. Further examination confirms that error detection by message feedback does not offer a satisfactory form of operation.

Adequate protection from undetected errors can be obtained either by a character-by-character system using a signal quality detector and parity checking or by a block system with a cyclic checking code.

For applications where the messages are long or where substantial amounts of data have to be transmitted, the block-repetition system provides extremely powerful error control. Table 1 illustrates the expected resultant quality of data transmission for different speeds of operation over switched and leased connections in terms meaningful to the user. Further, the efficiency of the system as regards effective transmission time, taking account of redundancy requirements and repetition arrangements, is high.

On the other hand, where the traffic to be transmitted consists of short messages the adaptability of the character-by-character control system more than offsets the relatively high percentage redundancy resulting from the use of check bits (one or two) for each character. When used in conjunction with a signal quality detector, very adequate error control can be exerted, particularly at the lower data-signalling rates where the effect of the signal quality detector is greater. The terminal equipment requirements are rather less than for a block-retransmission system.

For these reasons, it is considered that Special Study Group A should pursue standardization of error-control arrangements of both categories.

1.3 *Loop-propagation delay*

Both block-retransmission and character-by-character error-control systems depend on the retransmission of faulty characters or blocks; thus loop-propagation time, which varies with the length of the connection, is an important factor. Provision for abnormally long propagation times imposes a cost penalty except at the lowest signalling rates. It is proposed that the normal data-transmission equipment should cater for loop-propagation times of approximately 125 ms. This would suffice for most international connections at 600 bits/second over surface transmission media and for many connections at 1200 bits/second. Modified equipments could cater for longer propagation loops (e.g. satellite connections) and also permit transmission at 2400 bits/second.

2. *Block-retransmission system for switched circuits*

2.1 *Block size*

The size of the block does not need to exceed 250 bits in order to cater for a loop-propagation time of 125 ms. For messages of 1 minute or more, blocks of 250 bits lead to more efficient line usage than blocks of 100 or 500 bits. With blocs of 250 bits and messages exceeding 1 minute, the effective transmission time for data is greater than 90%.

When traffic includes a considerable proportion of shorter messages, there are advantages in a character-by-character form of operation.

For the modified equipment applicable to very long connections it is suggested that a larger block with between 500 and 600 bits should be used as described later.

2.2 *Amount of data transmission traffic*

It is assumed that switched connections will chiefly be used when the traffic is light and that, in general, the traffic will increase with the bits per second. In Table 1 intervals between undetected errors are shown for a range of usage figures.

2.3 *Block error rates*

Figure 1 is based on measurements carried out over a series of switched and leased circuits. It provides a relationship between bit and block error rates for blocks of different size.

2.4 *Cyclic-code reduction factor*

Measurements and simulations have shown that a cyclic code for block error detection is economical, efficient, and preferable to "row and column" or "3 co-ordinate" arrangements. Figure 2 shows how the number of check bits in a cyclic detection code affects the ratio of the number of erroneous blocks to the number of blocks with undetected errors.

The freedom to select the number of check bits for the cyclic code represents a flexibility not available with many other checking arrangements.

2.5 *Intervals between undetected errors*

Assuming an element error rate of 1 in 10^5 for leased circuits, and 1 in 10^4 for switched circuits, Table 1 shows the intervals between errors for a range of data traffic with blocks of 250 or 500 bits and modulation rates of 600, 1200 and 2400 bits per second.

TABLE 1

Intervals between undetected errors (Block-retransmission system)

1. Data-signalling rate (bits/s)	600	600	1200	1200	2400	2400
2. Traffic (erlang-hours/ month)	50	100	50	125	125	150
3. Block size (bits)	250	250	250	250	250	500
4. Blocks per month	4.3×10^5	8.6×10^5	8.6×10^5	2.2×10^6	4.3×10^6	2.2×10^6
5. Switched or leased connection	S	L	S	L	L	L
6. Block error rate (see Figure 1)	1.3 in 10^2	3 in 10^4	1.3 in 10^2	3 in 10^4	3 in 10^4	6 in 10^4
7. Erroneous blocks per month	5600	258	11 200	660	1290	1560
8. Intervals between undetected errors (months)						
12 check bits [a]	9	194	4.5	76	38.8	32
14 check bits [a]	45	970	22.5	379	194	160
16 check bits [a]	178	3880	90	1516	776	640

[a] The corresponding error-reduction factors (shown in Figure 2) are:
12 bit — 5 $\times 10^4$
14 bit — 2.5×10^5
16 bit — 1 $\times 10^6$

It is only practical to treat bit and block error rates on the basis of a long-term average for a large sample of different connections. Is is probably unavoidable that some connections will constantly give a poorer service than others. It is not logical that all equipments should be made more costly to cope with a few difficult applications. The following comments indicate a number of ways in which undetected errors can be reduced without adding to the expense of all equipments.

2.6 *Number of check bits*

The interval times in Table 1 suggest that 14 check bits would be suitable. However, it is a well-known characteristic of error distribution that many errors occur in bursts; furthermore, a data system is liable to undetected errors only when noise or some interruption causes several errors per block. During such periods there are three other system safeguards which have an influence on the overall performance.

When a large number of errors are present in a block, a considerable percentage of the blocks with undetectable errors will be rejected due to the serial number failing to check. More than 33% of the blocks with undetectable errors will be rejected because they follow immediately after a block for which a retransmission has been indicated. In the case of blocks disturbed by many errors, it has been found by measurement that the backward supervisory channel is also interrupted and that retransmission is often caused regardless of any error detection at the received station. With these three additional safeguards it is thought that 12 check bits are sufficient to provide an adequate interval for the majority of users.

Where a user requires even better protection it is possible to include auxiliary checking within the data.

2.7 *Very long propagation times*

Some users may need to transmit data over circuits with long propagation times. It is proposed that a block length of approximately 500-600 bits should be recognized for these cases. It is proposed that for this larger block size 16 check elements should be provided, but that the equipments should be designed so that they are capable of working with the smaller block size using 12 check elements.

2.8 *Cyclic-code check*

A simple form of checking includes the use of a shift register at each station, the correctness of each block being checked by the condition of the shift register at the receiving station. It is necessary to check that this acceptance is not being given owing to some faulty condition in the terminal equipment resulting in the shift register remaining permanently in the "accept" position.

2.9 *Generator polynomial for cyclic-code*

A series of simulations has been carried out to compare the different feedback combinations. Some codes are several times better than others; they are usually those which give protection against four errors in bursts whose length just exceeds the number of check bits.

Simulation programmes are being employed to verify the conclusions reached as to the desirable characteristics of the generator polynomials.

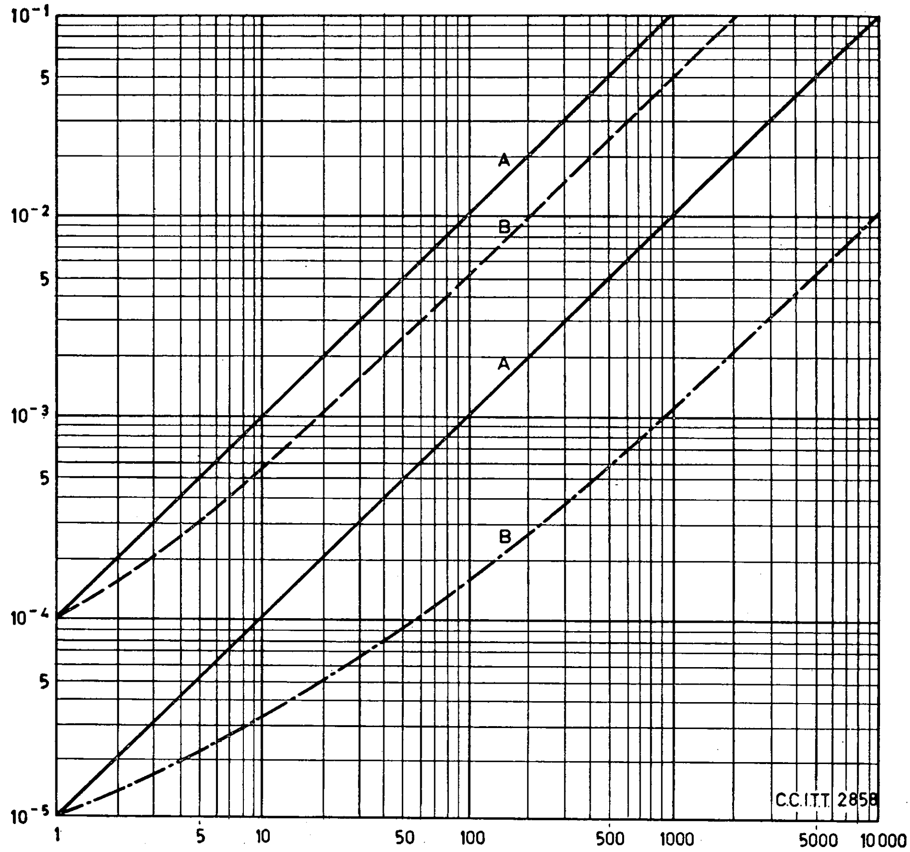
2.10 *Decision feedback signal*

The presence of the accept signal on the backward supervisory channel should be taken as the instruction for the sending station to continue transmitting new blocks in sequence. Any interruption in the reception of the answer signal should initiate the retransmission cycle.

The loop-propagation delay will vary with different connections. Nevertheless, it is essential there should be no uncertainty on the part of the sending station about the identity of blocks to which repeat signals refer. It is proposed that the receiving station should apply the repeat signal immediately after the completion of a block which does not check correctly and that this signal should continue for the length of half a block.

2.11 *Block omission or duplication*

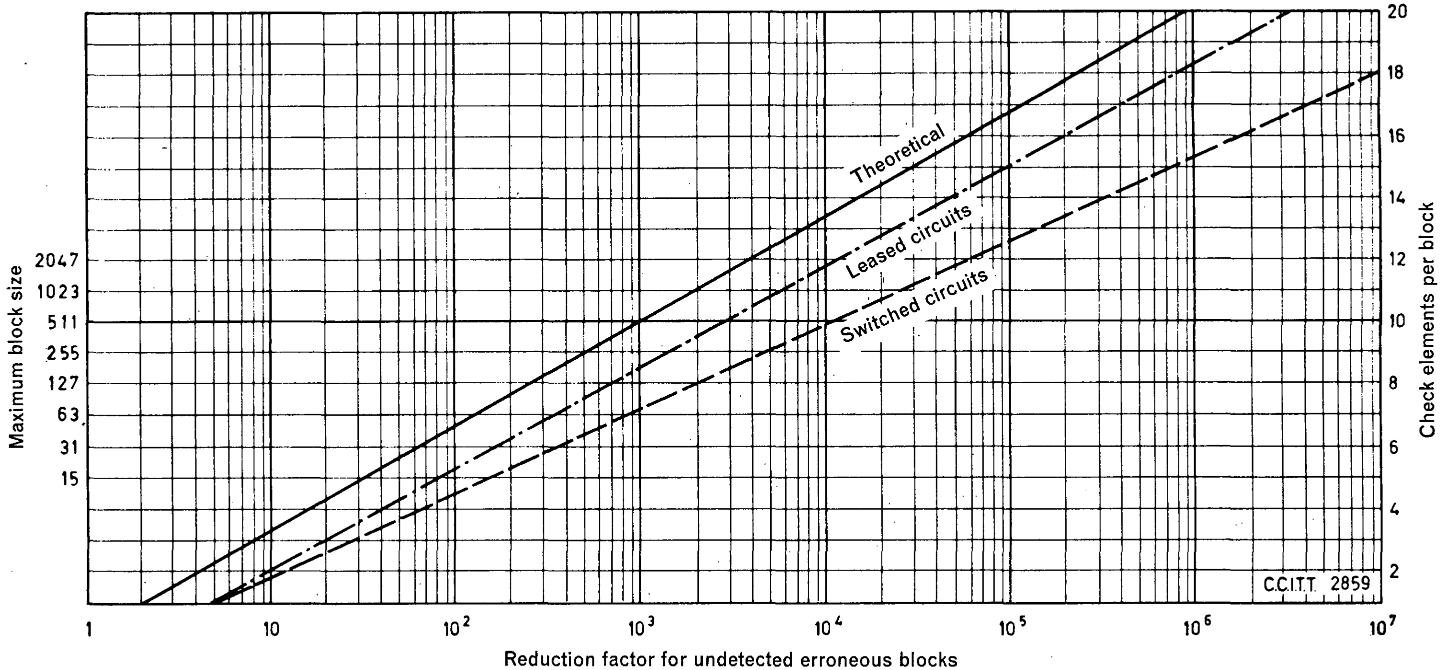
As interference to the backward supervisory channel may result in the sending station transmitting blocks which the receiving station is not expecting, message disturbance should be safeguarded against by employing a small cycle of, say, four sequence numbers used cyclically to



- - - - - = switched circuits
——.—— = leased circuits

A = random distribution
B = measured distribution

FIGURE 1. — **Block error rate**



Curves based on generator polynomials

$X^6 + X^5 + X^3 + X^2 + X + 1$

$X^{12} + X^{10} + X^5 + X^3 + X + 1$

$X^{16} + X^{11} + X^9 + X^7 + X^4 + 1$

FIGURE 2. — **Improvement factor of cyclic codes**

prefix each new block. The sequence-number bits should be checked with the remainder of the data in the block.

2.12 *General arrangement of block-retransmission system*

An outline of a block system appears in Supplement No. 61 of Volume VIII of the *Blue Book*.

Such a block system is suitable for use with the modem defined in Recommendation V.23 and would seem to be appropriate both for existing networks and for possible p.c.m.-type networks of the future.

It is known that such systems are operating in several national networks. Field experience over a period exceeding one year confirms the undetected-error expectancy which has been predicted by analysis of line-measurement programmes.

3. *Character-by-character system*

3.1 *Performance*

Laboratory tests involving a line simulator indicate that at 200 bits/second a satisfactory interval between undetected errors can be achieved with one parity bit per character and a signal quality detector. Table 2 illustrates the error-reduction factor needed to provide error-free intervals of three months for given amounts of data. For 600 and 1200 bits/second it is considered that two parity bits per character are required. Error-reduction factors have been established by a series of laboratory and field tests.

TABLE 2

Intervals between undetected errors (Character-by-character system)

Data-signalling rate (bits/s)	200	600	1200
Traffic (erlang-hours/month)	25	50	50
Character size (bits)	8	9	9
Characters per month	2.2×10^6	1.2×10^7	2.4×10^7
Switched or leased connection	S	S	S
Character error rate (see Figure 1)	5 in 10^4	5 in 10^4	5 in 10^4
Erroneous characters per month	1100	6000	12 000
Reduction factor needed to provide free intervals of three months	3.3×10^3	1.8×10^4	3.6×10^4

As regards the number of repetition cycles, the laboratory tests have indicated that the action of the signal quality detector involved approximately 66% more repetitions than were necessary to correct errors. In the field tests this increase was over 100%. The number of repetition cycles is not proportional to the character error rate because a number of erroneous characters can occur within one repetition cycle. Despite the action of the signal quality detector, the transmission time involved by repetition cycles is relatively small. At 600 bits/second the effective transmission time for data is approximately 70% even when there are two redundant parity bits per character.

The proposed characteristics of the signal quality detector are included in the Annex. Other characteristics of the system appear at the end of Supplement No. 61 (*Blue Book*, Volume VIII).

4. *Error control on leased telephone circuits*

4.1 *Off-line applications*

For "off-line" applications the requirements are not dissimilar to those for the switched case.

The amount of transmission and the message lengths would probably be greater, but these two factors would not make it desirable to increase the block lengths. It is likely that 1200 or 2400 bits/second would be transmitted, although the higher rate would restrict the permissible loop-propagation time. The additional safeguards described for switched connections are also applicable, and it is, therefore, reasonable to use the same block sizes and number of check bits for both cases.

4.2 *On-line applications*

Study Group XI is examining the advisability of using data transmission on a signalling channel common to a large number of speech circuits. Advice is sought concerning the use of signal quality detectors.

The common signalling channel will be a four-wire circuit with a continuous stream of signal units in each direction; the signal units are likely to have some fixed number of bits between 16 and 36, to which will be added check bits generated as a cyclic code. The signalling circuit will be a typical 3-kHz speech circuit carrying 1200 or 2400 bits/second. Equalization will be used if necessary.

The basic requirements for the C.C.I.T.T. signalling system No. 6 include rapid signal transfer and a high degree of dependability. The need for speed makes it necessary to include check bits per signal unit to avoid awaiting the end of the block. The dependability requirements will demand a standby signalling channel, and it is possible that duplicate operation will be used so that the received signal units can be compared in turn. Only a small number of check bits will normally be needed with duplicate operation, but it is necessary to make provision for the periods when only one of the two signalling circuits is operating correctly. It is questioned whether at such times a signal quality detector can be relied on to avoid the acceptance of erroneous signal units which would be too numerous if only a few check elements were used. In normal duplicate operation the signal quality detectors would be disabled to prevent too-frequent requests for retransmission. It is believed that the periods when only one signalling circuit is operating would be relatively short and that during such periods an excessive number of requests for retransmission could be accepted.

ANNEX

Signal quality detector

It is proposed that the response of the signal quality detector should not exceed 2 ms in the case of an instantaneous increase of level by 3 dB or decrease of level by 6 dB.

Supplement No. 23

A.E.G.-TELEFUNKEN. — *(Contribution COM Sp. A—No. 139—October 1967)*

SIGNAL QUALITY DETECTOR

Signal quality detectors provide a low-cost and effective means for error detection in data transmission over telephone channels. Their use has already been proposed to the COM Sp. A of the C.C.I.T.T. in several contributions, and the present contribution will support these proposals. Based on the results of our investigations, we feel it is advisable to include in the modems, as per Recommendations V.21 and V.23, provisions for signal quality detection.

The objective of the modem Recommendations V.21 and V.23 was a concept enabling the design of comparatively uncomplicated and hence low-cost modems. For this reason, not only the principle of binary frequency modulation has been adopted, but also have all functional units that are not necessarily needed in the modem (e.g. the timing generator) been assigned to the data-processing terminal equipment. This disposition affects, however, the supervision of the received signal quality by disturbance detectors, since in various types of these latter, the bit timing is required for the analysing process if—as defined by Recommendation V.24 (page 50 of Volume VIII of the *Blue Book*)—not only a disturbance criterion must be transmitted over the interchange circuit 10 (signal quality detector), but already the decision of whether a binary digit is faulty or not. In order to obviate these difficulties, we therefore would suggest that the definition be formulated anew to read as follows:

Circuit 10: Data signal quality detector,
Direction: From data communication equipment.

A signal on that circuit is used to indicate whether certain characteristics of the received signal are within defined tolerance limits. The duration and position of the signals on that line provide a measure for the quality of the data signals received, and may be a measure for the probability of the occurrence of transmission faults.

ON condition: The characteristics lie within the tolerance limits
OFF condition: The characteristics lie outside the tolerance limits.

Quality supervision of the received signals can be made prior to and/or after demodulation, and may be either a continuous or sampled check. For quality supervision various criteria such as amplitude, frequency, phase, duration energy, correlation function may be used. The efficiency of a quality supervision, but also the complexity of the circuitry—increase with the number of analysed signal parameters. A low-cost and rather efficient quality supervision is obtained with the aid of the amplitude tolerance detector which evaluates the amplitudes of the demodulated signals with the aid of thresholds (Figure 1). This type of detector which is recommended for quality supervision provides a compromise between system complexity and quality.

Design proposal

The signal at the output of the discriminator $U(t)$ is compared with four thresholds (S_1; S_2; S_3; S_4) as shown in Figure 1 a. If the voltage $U(t)$ is within the hatched areas, the modem sends negative polarity over the interchange circuit 10 (Recommendation V.24), otherwise it will send positive polarity (Figure 1 b). This signal is sampled outside the modem in the error detection and correction unit in the bit rhythm, and a fault alarm will occur if during the time

$$t_i \pm \frac{\Delta t}{2}, \; U+ < 0,$$

or, respectively, $\quad |U| < S_3 = S_4$

or $\quad |U| > S_1 = S_2$

Efficiency of the amplitude tolerance detector

Definition: $R = \dfrac{\text{Quantity of undetected erroneous bits}}{\text{Total quantity of erroneous bits}}$

$r = \dfrac{\text{Quantity of bits indicated as disturbed, and correctly interpreted by the receiver}}{\text{Total quantity of bits transmitted}}$

$R_B = \dfrac{\text{Quantity of erroneous data blocks for which no disturbance indication was given}}{\text{Total quantity of erroneous data blocks}}$

$r_B = \dfrac{\text{Quantity of data blocks indicated as disturbed, that are, however, without error}}{\text{Total quantity of data blocks transmitted}}$

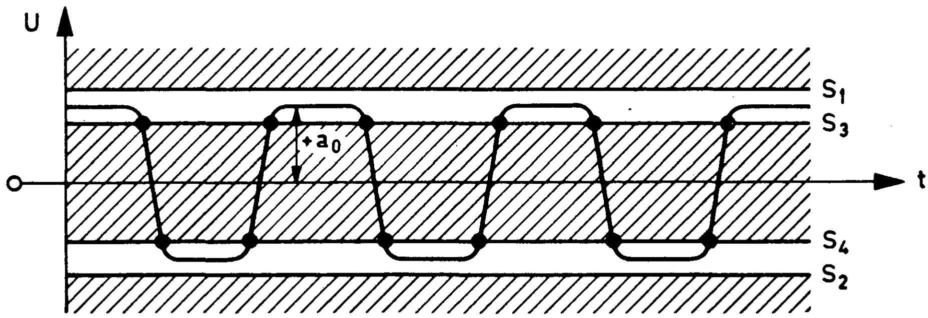


FIGURE 1 a

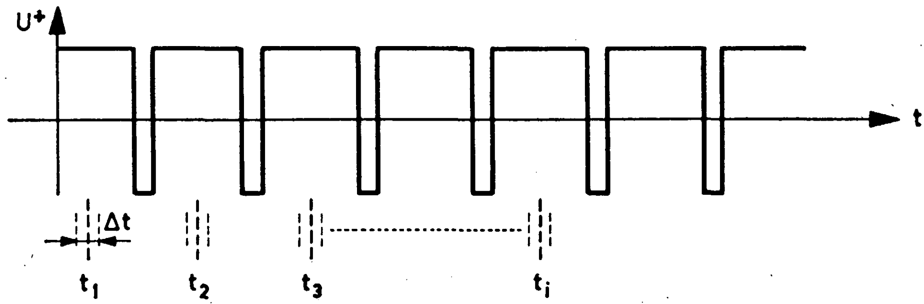


FIGURE 1 b

R and R_B, respectively, are the reduction factors for the probability of non-recognizable errors; r_B corresponds to the redundancy caused by the quality supervision.

R is a function of the threshold margin $S+$ from the nominal amplitude of the steady-state signal $\pm a_0$; for symmetrical threshold margins

$$S+ = \frac{S_1}{a_0} - 1 = 1 - \frac{S_3}{a_0} = \frac{S_2}{a_0} - 1 = 1 - \frac{S^4}{a_0}$$

The functions $R = f(r)$ and $R = f(S+)$ have been measured under the disturbance conditions occurring in telephone channels, and are shown in Figures 2 a and 2 b.

The quality supervision is analysed practically always for whole data blocks.

This increases the effectiveness of the quality supervision considerably. Figures 3 a and 3 b show the measured functions

$$R_B = R_B(n_B; S+) \quad \text{and}$$
$$r_B = r_B(n_B; S+),$$

where $n_B =$ number of binary digits per data block.

From the figures, it is for instance apparent that for

$S+ = \pm 0.2$ and
$n_B = 64$ binary digits, a reduction factor of
$R_B = 9 \cdot 10^{-4}$ is reached at a redundancy of
$r_B = 6 \cdot 10^{-3}$.

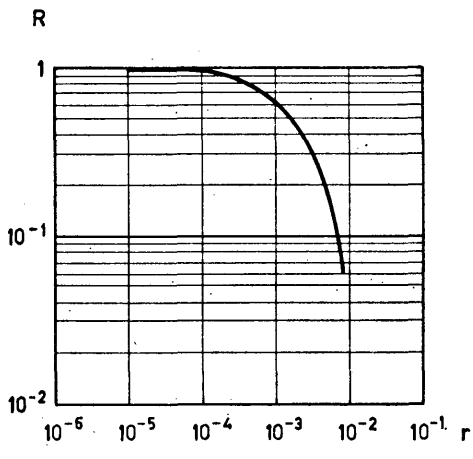


FIGURE 2 a

Reduction factor of the bit error probability as a function of redundancy.

Amplitude detector arranged subsequent to demodulation.

Binary frequency modulation 1200 bits/s.
Receive level $a_0 = -30$ dBm.

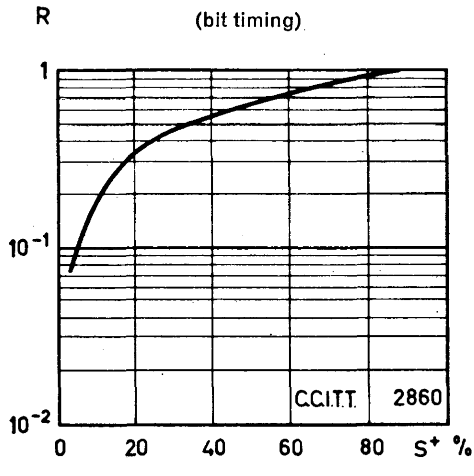


FIGURE 2 b

Reduction factor of the bit error probability as a function of the threshold margin.

Amplitude detector arranged subsequent to demodulation.

Binary frequency modulation 1200 bits/s.
Receive level $a_0 = -30$ dBm.

With the aid of redundant codes, an about equal reduction factor is reached when using 10 binary test digits.

For an efficient quality supervision, the transient behaviour of the voltage U at the discriminator output is of great significance. In case of strong overshooting or great asymmetry of the amplitudes with respect to the zero axis, the signal U needs special treatment to allow comparison with the thresholds.

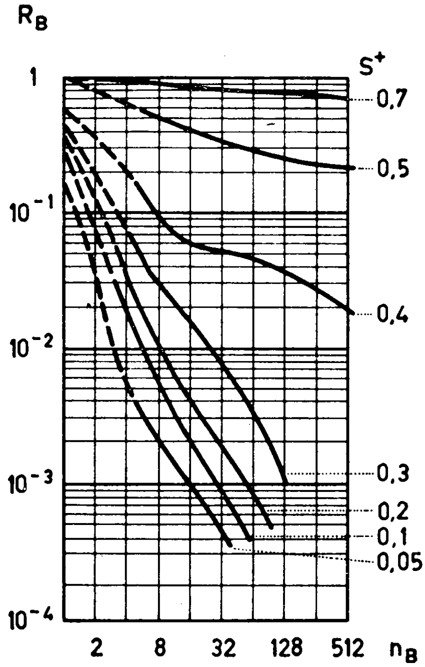


FIGURE 3 a

Reduction factor of the block error probability as a function of the block length with the threshold margin being constant.

Amplitude detector arranged subsequent to demodulation.

Binary frequency modulation 1200 bits/s.
Receive level a_0 = − 30 dBm.

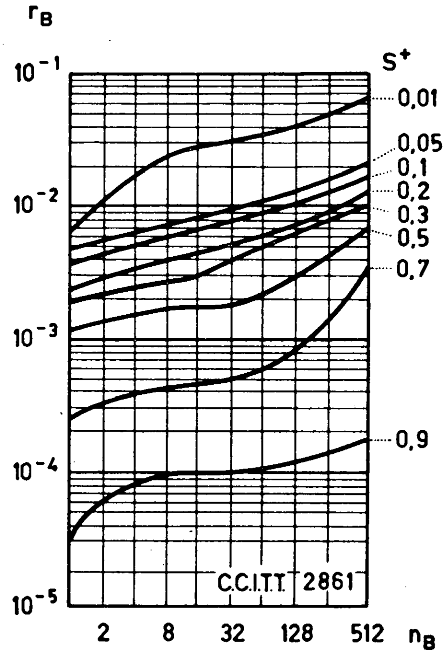


FIGURE 3 b

Block redundancy as a function of the block length with the threshold margin being constant.

Amplitude detector arranged subsequent to demodulation.

Binary frequency modulation 1200 bits/s.
Receive level a_0 = − 30 dBm.

Supplement No. 24

A.E.G.-TELEFUNKEN. — *(Contribution COM Sp. A—No. 142—October 1967)*

DATA PROTECTION ON TELEPHONE LINES

1. *General*

Protection against errors in data transmission via telephone lines is generally ensured by redundant coding of information. On telephone lines, however, faults occur frequently which result in whole bursts of errors. The effectiveness of a redundant code against error bursts decreases with the length of the error bursts. The amount of redundancy needed to provide the requisite high degree of protection against undetected errors may therefore become relatively high. But there are other facilities for the detection of these errors, namely the data signal quality detectors. The greater the bursts, the better is the effectiveness of these data signal quality detectors. Therefore, a combination of both procedures will furnish more favourable results with a smaller amount of redundancy than each procedure employed by itself.

2. *Example of calculation for a block length of 32 bits*

The improvement in the transmission protection is denoted by the reduction factor. If the two procedures are independent of each other, this reduction factor can be equated as a first approximation to the product of the distinct reduction factors.

$$R = R_C \cdot R_Q \qquad \begin{array}{l} C = \text{code} \\ Q = \text{qualité} \end{array}$$

In [1] it is shown that under practical circumstances R is even smaller than the product of the distinct reduction factors that means $R < R_C \cdot R_Q$.

If proper detectors are chosen, the reduction R_Q will easily attain the value $R_Q \leqslant 10^{-3}$ where less than 1 % redundancy is caused by unnecessary repetition. (see the contribution of Telefunken about quality detectors, Annex 1 to this document). This value is rather favourable.

In the first approximation, for the reduction factor caused by good redundant codes there is an upper bound:

$$R_C \leqslant \frac{1}{2^m}$$

where m is the number of the added checking steps derived from the information. Hence, for a reduction by the factor

$$R_C \approx 10^{-2}$$

approximately 6 bits are required as check information.

According to measurements, the mean error density in the faulty blocks is approximately:

$$d_m = 0.2$$

such that, on the average, an error-affected block contains

$$n_f = 0.2 \cdot n$$

wrong bits.

Hence, after error detection is accomplished, the residual bit error probability is:

$$P_{\text{rsd bit}} \leqslant P_{\text{bloc}} \cdot R_C \cdot R_Q \cdot d_m$$

For a value of

$$P_{\text{rsd bit}} = 10^{-8}$$

with a block error probability.

$P_{\text{block}} \approx 5 \cdot 10^{-3}$, according to measurements (Figure 1), one obtains the requisite reduction factor of the code

$$R_C \leqslant \frac{P_{\text{rsd bit}}}{P_{\text{block}} \cdot d_m \cdot R_Q} = \frac{10^{-8}}{5\ 10^{-3}\ 0.2\ 10^{-3}}$$

$$R_C \approx 10^{-2}$$

Thus, the number of check bits is obtained

$$R_C \leqslant \frac{1}{2m} \qquad m = 6$$

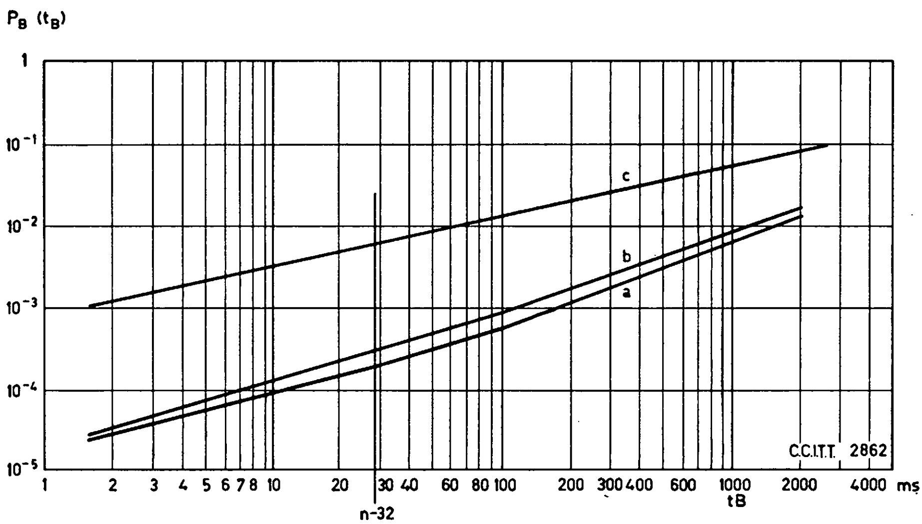
The redundancy r_C is a function of the block length n

$$r_C = \frac{m}{n}$$

Given a block length of

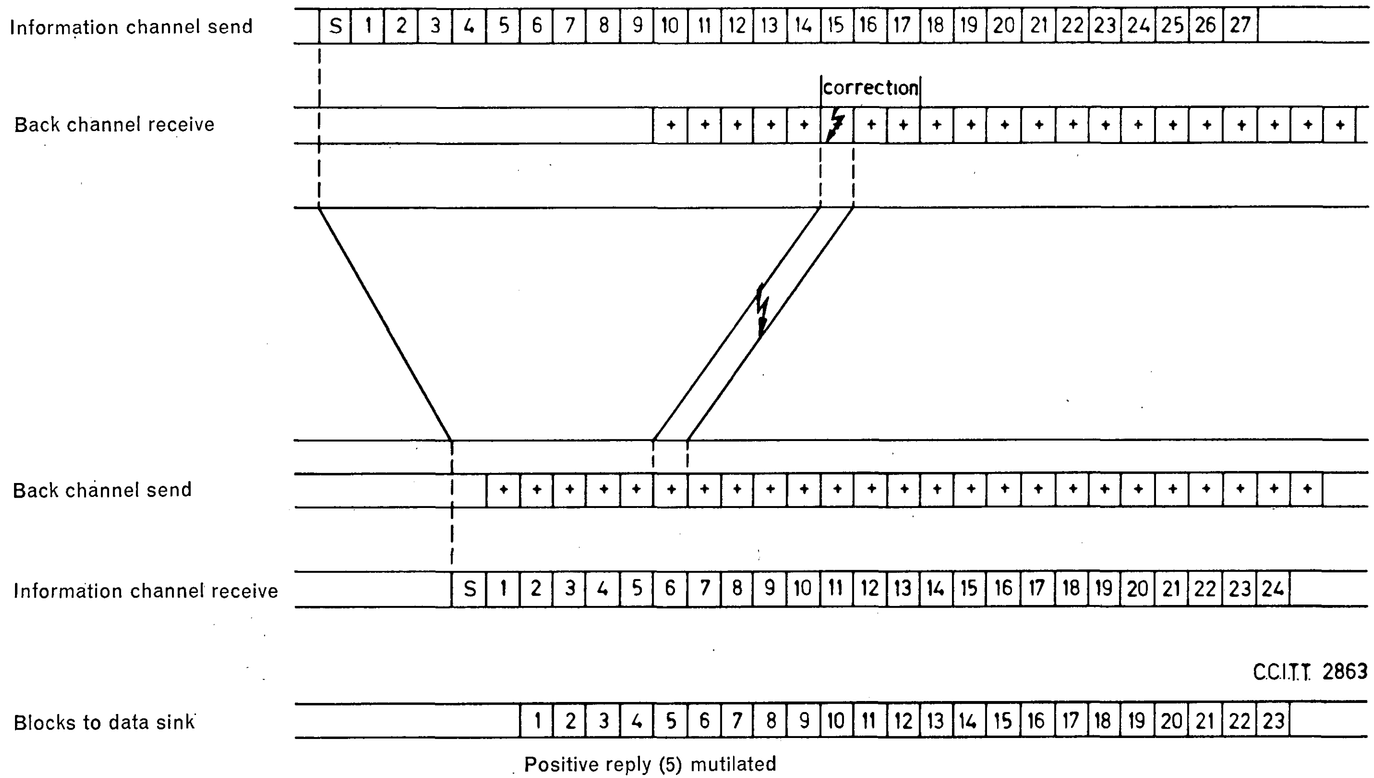
$$n = 32 \text{ bits}, r_C \approx 19\%.$$

Hence, a redundancy of $r = 20\%$ is required for the total reduction factor $R \leqslant 10^{-5}$.

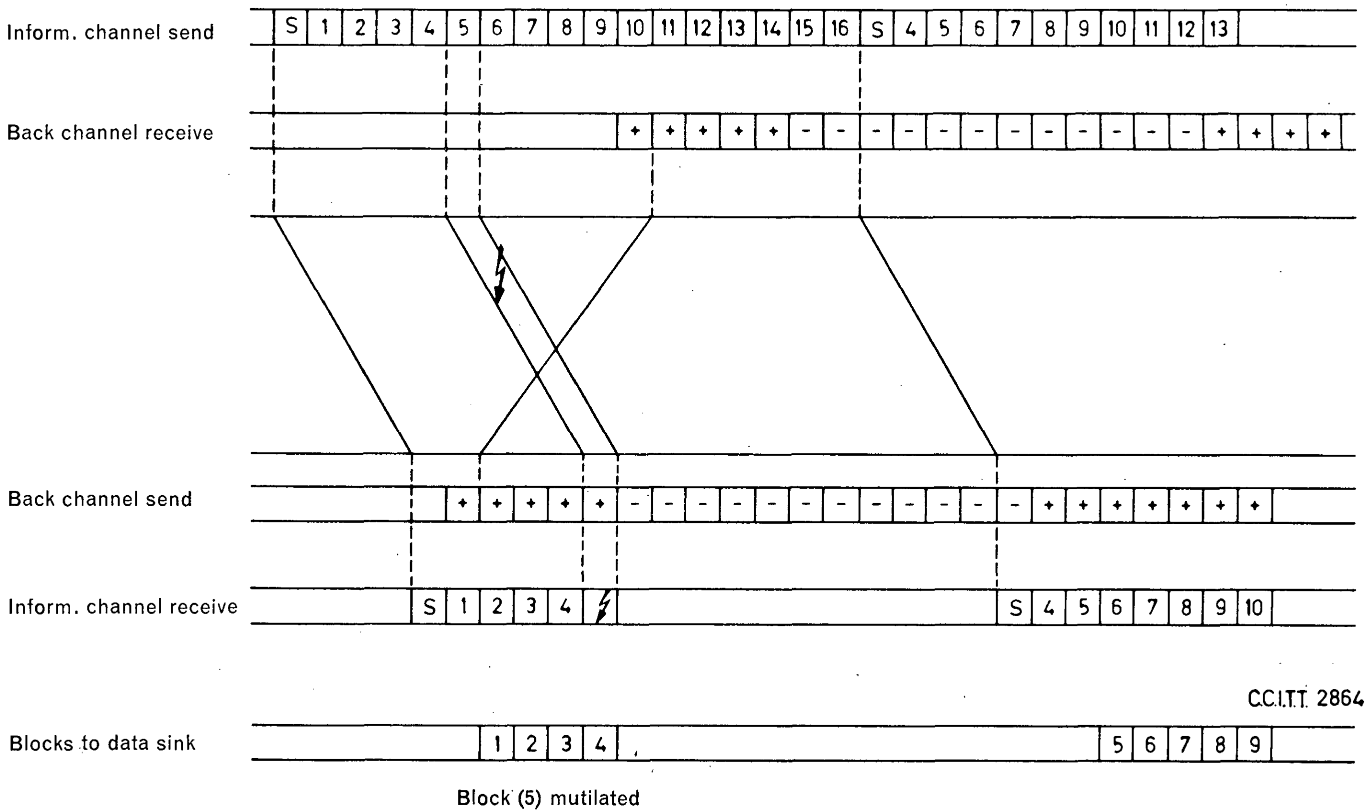


$P_B = P_{\text{block}}$ = Block error rate
t_B = Block length in ms (data-signalling rate = 1200 bits/s)
Transmission loss of the connection at 1300 Hz: 17.5 dB
Transmitted signal level: a) −5.5 dBm b) −8.5 dBm c) −16.5 dBm
Received signal level: a) −23 dBm b) −26 dBm c) −34 dBm

FIGURE 1. — **Block error rate measured on a heavily disturbed telephone connection in the general switched network of DBP**



Error control procedure



Error control procedure

3. *Procedure of transmission and error correction by means of repetition*

3.1 *The backward channel*

With a data-signalling rate of 1200 bits/s, one obtains the required minimum data-signalling rate on the backward channel

$$fB > \frac{1}{n} \cdot fI \qquad \begin{array}{l} fI = \text{data-signalling rate on information channel} \\ fB = \text{data-signalling rate on backward channel} \end{array}$$

Hence, with a block length of $n = 32$ bits, one obtains

$$fB > \frac{1200}{32} \text{ bps} = 37.5 \text{ bits/secund}$$

This requirement is met with the backward channel of the normal FM-modem.

3.2 *Transmission of Information*

After establishment of the connection and synchronization of the blocks, the information is continuously transmitted block by block. The emitted blocks are stored and counted in the transmitting station A.

After a certain period of time, the first reply message will arrive resetting by one the transmitter block counter. Each incoming reply message has the same effect. Thus, the counter indicates the number of blocks not acknowledged yet by a reply message. Upon arrival of a negative reply, station A is capable of telling precisely how many blocks must be repeated. This is, then, the minimum possible amount of repetition bits. Protection against falsified reply message is enhanced by applying a reply message correlation. A single negative reply which is followed by a positive reply does not cause a repetition and is counted as a positive reply. This correlation can be extended over several reply messages. It is made feasible by having the receiver transmitting negative replies only upon detection of an error.

3.3 *The receiver*

Station B (receiver) is provided with 2 stores of the length of one block. Due to the shortness of blocks, the amount of expenditure is very low. By turns, one of these stores is being filled with incoming information while the other is delivering its contents to the data sink.

3.4 *Error detection*

The error detection analyses the indications of the signal quality detector and of the redundant decoding. Upon detection of an error, each succeeding block is acknowledged by a negative reply. No block is stored any more. The last correct block is preserved. Station B will wait for a sync-block announcing the repetition. Upon arrival of the sync-block, the information is checked. The receiver compares the blocks of information with the correct block in the store received last. In the case of coincidence, the correct junction will be easily found, and the next block is handled as described above. Coincidence ensures that errors in the backward channel which occur immediately before a negative reply, and which mutilate positive replies, do not cause correct information to be repeated to the data sink.

4. *A brief summary of the merits of the system reads as follows:*

Low storage requirements due to shortness of blocks. The amount of repeated information is matched to the loop delay time and to the properties of the backward channel. Effective protection against blocking caused by heavy line fault. High-grade exploitation of the lines. A detailed description of the system is given in [2].

REFERENCES

[1] H. OHNSORGE, W. WAGNER: Zur Kombination von Störungsdetektoren und redundanten Codes für die Fehlererkennung; *AEÜ* 24 (1967), Volume 9.

[2] U. HALLER, H. OHNSORGE: Laufzeitgesteuertes Datenübertragungssystem, *Telefunkenzeitung* 40 (1967), Volume 1/2, p. 76.

Supplement No. 25

UNITED KINGDOM. — *(Extract from Contribution COM Sp. A—No. 153—October 1967)*

CYCLIC CODE POLYNOMIAL

. . . .

1. *Comparison of cyclic polynomials*

A number of suitable polynomials have been suggested. In order to see if the performance of any one polynomial is markedly superior to all others a computer simulation has been carried out.

The polynomials included in this exercise were:

a) $X^{16}+X^{12}+X^5+1$
b) $X^{16}+X^{15}+X^{10}+X^6+X^5+1$
c) $X^{16}+X^{15}+X^2+1$
d) $X^{16}+X^{13}+X^{11}+X^5+X^2+1$

Some properties of these polynomials are 1) a Hamming distance of 4, 2) error detection of any one error burst not exceeding 16 bits in length, 3) detection of all odd numbers of errors and 4) detection of any two-error bursts, which for polynomials a) and c) must not exceed a length of 2 bits each, for b) must not exceed lengths of 5 and 6 each, and for d) must not exceed lengths of 5 and 7 each. However, for a given block size and a given number of check bits there is a limited degree of protection available and thus if protection against certain types of error distribution is strengthened, protection against other types of error distribution will be weakened.

The computer simulation used the error statistics quoted in the *Blue Book*, Volume VIII, Supplement No. 22. Although these statistics are admitted to be insufficient for an exhaustive comparison of the polynomials, nevertheless they are judged to be sufficient to demonstrate the marked superiority of any one polynomial.

The statistics, together with a pseudo-random process, yielded 222 034 blocks containing an even number of errors equal to or greater than four bits. These blocks represented approximately 15% of all erroneous blocks, and assuming that this is a typical percentage and that the long-term block error rate is 1%, the test is equivalent to approximately one year of data transmission at 1200 bits/second. The residual block error rate for each polynomial tested during the "year" is as follows:

TABLE 2

Polynomial	Residual block error rate
$X^{16}+X^{12}+X^5+1$	one in 9.1 million
$X^{16}+X^{15}+X^{10}+X^6+X^5+1$	one in 6.9 million
$X^{16}+X^{15}+X^2+1$	one in 5.6 million
$X^{16}+X^{13}+X^{11}+X^5+X^2+1$	one in 3.8 million

TABLE 3

Analysis of 16-bit cyclic error detection codes

Number of blocks processed = 222 034

Size of block = 240+16 check bits = 256 bits

Polynomial		$X^{16}+X^{12}+X^5+1$	$X^{16}+X^{15}+X^{10}+X^6+X^5+1$	$X^{16}+X^{15}+X^2+1$	$X^{16}+X^{13}+X^{11}+X^5+X^2+1$
Errors per block	Number of blocks containing these errors	Number of undetected errors			
4	96 112	12	16	19	28
6	53 603	1	3	6	5
8	30 898	2	0	0	4
10	17 796	0	1	1	1
12	10 212	1	1		
14	5 845				
16	3 291				
18	1 909				
20	1 038				
22	555				
24	348				
26	195				
28	105				
30	55				
32	36				
34	15				
36	9				
38	6				
40	3				
42	3				
44	0				
46	0				
48	0				
Total number of blocks	222 034				

The results of the test, given in Table 3, demonstrate that there were differences in performance between the polynomials and since $X^{16} + X^{12} + X^5 + 1$ is simple and appears to be slightly superior it is therefore suggested. However, should the study group feel that longer double burst error protection is of paramount importance then the U. K. Administration would suggest $X^{16} + X^{15} + X^{10} + X^6 + X^5 + 1$.

2. *Definition of the coding process*

The following definition of cycle coding is offered, since it is important that the coding process be agreed otherwise compatibility may not be ensured.

The service bits and information bits, taken in conjunction, constitute a message polynomial which is then divided in a module two manner by the generating polynomial. The check bits correspond to the remainder polynomial found at the completion of this division process. In both the message and the check bits the higher-order terms of the polynomial precede the lower-order terms.

Supplement No. 26

A.E.G.-TELEFUNKEN. — *(Contribution COM Sp. A—No. 165—December 1967)*

COMPARISON OF ERROR PROTECTION METHODS FOR DATA TRANSMISSION ON TELEPHONE CHANNELS

The following three system principles have become known for error protection apparatus with decision feed-back and correction through repetition of disturbed data blocks:

1) semi-duplex systems;

2) storage-change systems;

3) delay time controlled systems.

If the transmission safety is demanded to begin with, the necessary apparatus expenditure and the transmission rate are decisive in the evaluation of the systems. It will be shown in the Appendix that, based on measurements and theoretical considerations, data transmission with error correction by the principle of "delay time controlled systems" offers the highest effective transmission rate (compare Figures 9.1 and 9.2). Furthermore, it will be shown that especially with noisy transmission channels the principle of "delay time controlled systems" is to be favoured in comparison with other principles (see Figure 10). Delay time controlled systems work with short block length, while the necessary block length of storage change systems is limited towards small values by the maximum loop delay time to be expected of principal and return channel. If for a delay time controlled system the block length n is chosen, and for a storage change system the block length $n_w = L \cdot n$, the effective transmission rates can be taken from

Figure 11 as a function of L for channels with disturbance bursts (telephone channels), with mean error probability P_s as a parameter. It will be recognized that especially at times with high error frequency the "delay time controlled system" is substantially superior to the storage change system for instance with $n = 64$ and $L = 5$—i.e. with a block length of $n_w = 5 \cdot 64 = 320$ binary digits per block for the storage change system—the effective transmission rate for the storage change system falls nearly to zero when the error frequency rises to 1%. The storage change system is practically blocked through permanent repetition under these circumstances, while the "delay time controlled system" still operates with a transmission rate of approx. 30%. The principle of "delay time controlled systems" is described in Supplement No. 24. More detailed descriptions of the various system principles are found in the literature quoted in the Appendix. The system principles are schematically presented by means of the storage organizations in Figure 7 of the Appendix. The minimum storage expenditure for the storage change system amounts to $4\ n_w = 4 \cdot 1\ 4 \cdot L \cdot n$, for the delay time controlled system to $4\ n + L \cdot n$. With $n = 64$, $n_w = 512$, i.e. $L = 8$, the delay time controlled system requires only about ⅓ of the storage expense necessary for the storage change system.

For the calculations in the Appendix the combined employment of a redundant code and a signal quality detector for error recognition are presupposed. The properties of the signal quality detector considered have been measured and presented in Figures 4 and 5 in the Appendix. The respective signal quality detector is described in Supplement No. 23.

Appendix

For fast binary data transmission the telephone network is advertised today by the German Federal Post Office. The fee for telephone channels is relatively high, therefore the question of channel economy is of interest. For data transmission with error correction a number of procedures are known: in the following it will be shown which channel economy may be attained with the various methods. Since redundance is necessary for data safeguarding, the mean speed V_m of the data fed into the sink must always be smaller than the transmission speed V_k of the channel. As a yardstick for the channel utilization we choose the normalized effective transmission rate

$$V_{\text{eff}} = \frac{V_m}{V_k} \tag{1}$$

1. *Error correction through redundant codes*

If each binary digit transmitted is being falsified with the same probability p (statistically independent errors), then the code with the largest minimum distance is optimum for the error correction—as is known [1]. If a code is to correct e errors per code word, the Hamming relation [2] delivers the minimum necessary redundance for a given block length n

$$r_{\min} = \frac{m_{\min}}{n} = \frac{l}{n}\ 1\ d \sum_{i=0}^{e} \binom{n}{i} \tag{2}$$

$r_{\min}$ is to be increased to r, so that the number of test digits per block

$$m = r \cdot n \tag{3}$$

becomes integer, because a code word cannot contain fractions of a binary digit. As it is known, a code with the redundance r correcting e errors guarantees a probability for not recognizably erroneous code words of

$$P_{\text{crest}} = 1 - \sum_{i=0}^{e} \binom{n}{i} p^i (1-p)^{n-i} \tag{4}$$

under above presupposition, with an effective transmission rate

$$V_{\text{effc}} = 1 - r \tag{5}$$

With equations (2) to (5) one obtains therefore P_{crest} (n, p, V_{effc}) with the greatest possible transmission rate. This connection is presented in Figures 1 and 2.

By transmission over the telephone network in heavily disturbed channels a mean error rate of

$$P_s = 10^{-3} \tag{6}$$

is to be expected. However, errors appear here mainly in bursts. The frequency p_B of erroneous blocks therefore is smaller than for time independent error frequency; the mean error density in disturbed blocks, however, will become very high because of the error bundles and amounts from

$$d_m \approx 0.1 \text{ to } 0.01 \tag{7}$$

for the transmission duration for one block from

$$t_B = 50 \text{ ms to } 1000 \text{ ms} \tag{8}$$

Since the mean burst duration is about 500 ms, the employment of so-called burst error correcting codes is of no advantage. The attainable transmission rate by error correction through redundant codes can therefore approximately be read from Figures 1 and 2, when we set

$$p = d_m$$

and (9)

$$p_{\text{demanded}} = p_B \cdot p_{\text{crest}} = p_{B\text{ rest}}$$

In this connection p_{crest} is to be looked at as a conditioned probability, because a block contains a mean error density d_m with a probability P_B only, while for the other blocks $d_m = p \approx o$ is to be taken. Under the presupposition that approximately optimum codes can be constructed and be realized with acceptable effort, the following values are found for a transmission speed of

$$V_k = 1200 \text{ binary digits/s and } p_{\text{demanded}} = 10^{-8}: \tag{10}$$

$V_{\text{eff}} \approx 0.1$ at $t_B = 50$ ms $\rightarrow n = 60$, $d_m = 0.11$, $p_B \approx 10^{-2}$

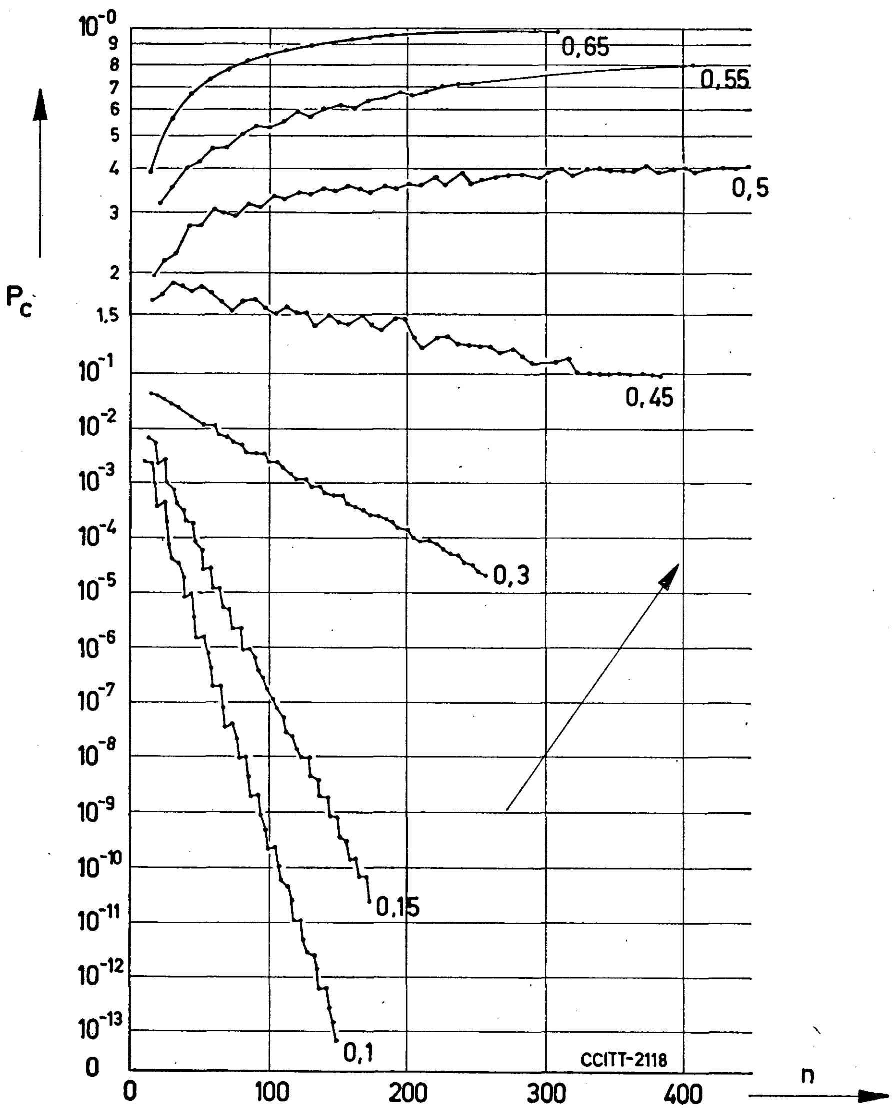
$V_{\text{eff}} \approx 0.8$ at $t_B = 1000$ ms $\rightarrow n = 1200$, $d_m = 0.013$, $p_B \approx 5 \cdot 10^{-2}$
(concerning p_B see reference [3]) (11)

By error correction through repetition of disturbed information one attains $V_{\text{eff}} \approx 0.8$ in telephone channels, so that the correction through redundant codes offers an advantage with respect to the transmission rate only with block lengths $>$ 1000 binary digits. Approximately optimum codes with $n > 1000$ and $r \approx 0.2$ cannot yet be realized today with economically acceptable expense, so that these procedures seem to make little sense.

2. *Means of error recognition*

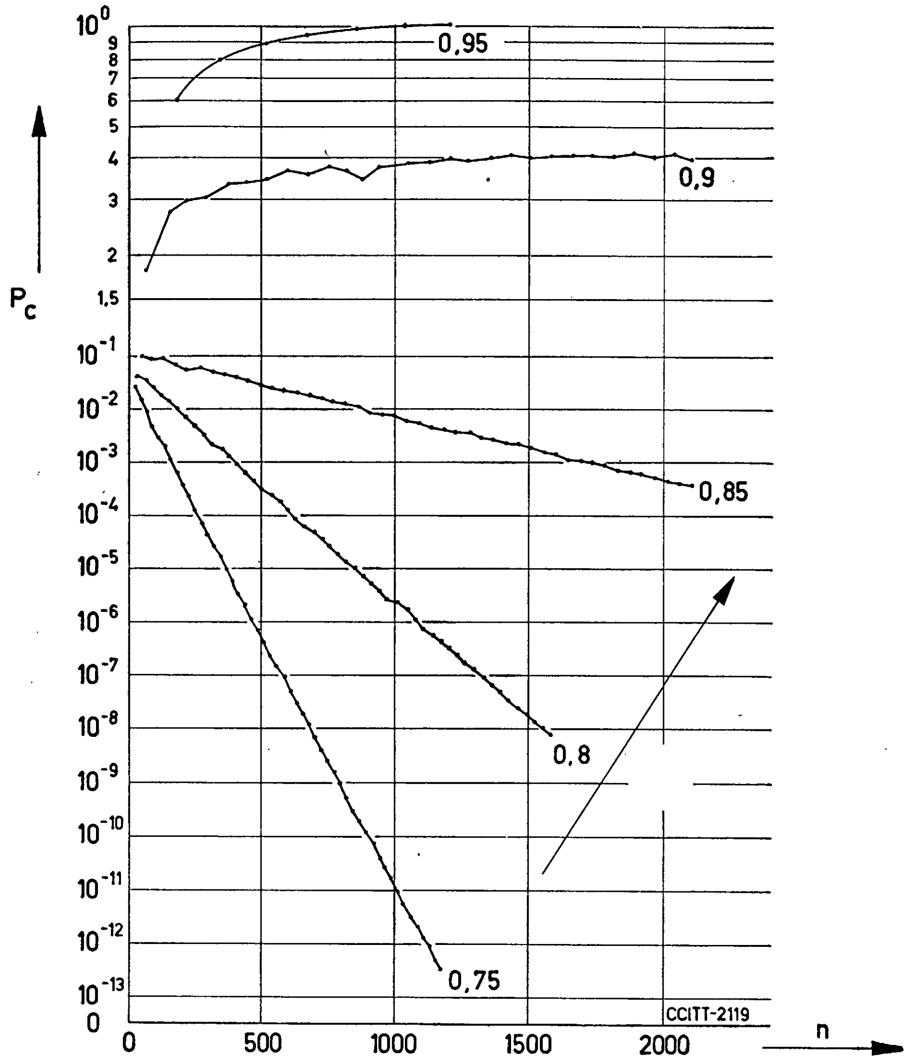
2.1 *Redundant codes*

In the next chapter error correction through codes shall be correlated to procedures with correction through repetition. Repetition systems require means for error recognition, therefore the effectiveness of redundant codes and of signal quality detectors shall be treated in short in the following, with respect to error recognition.



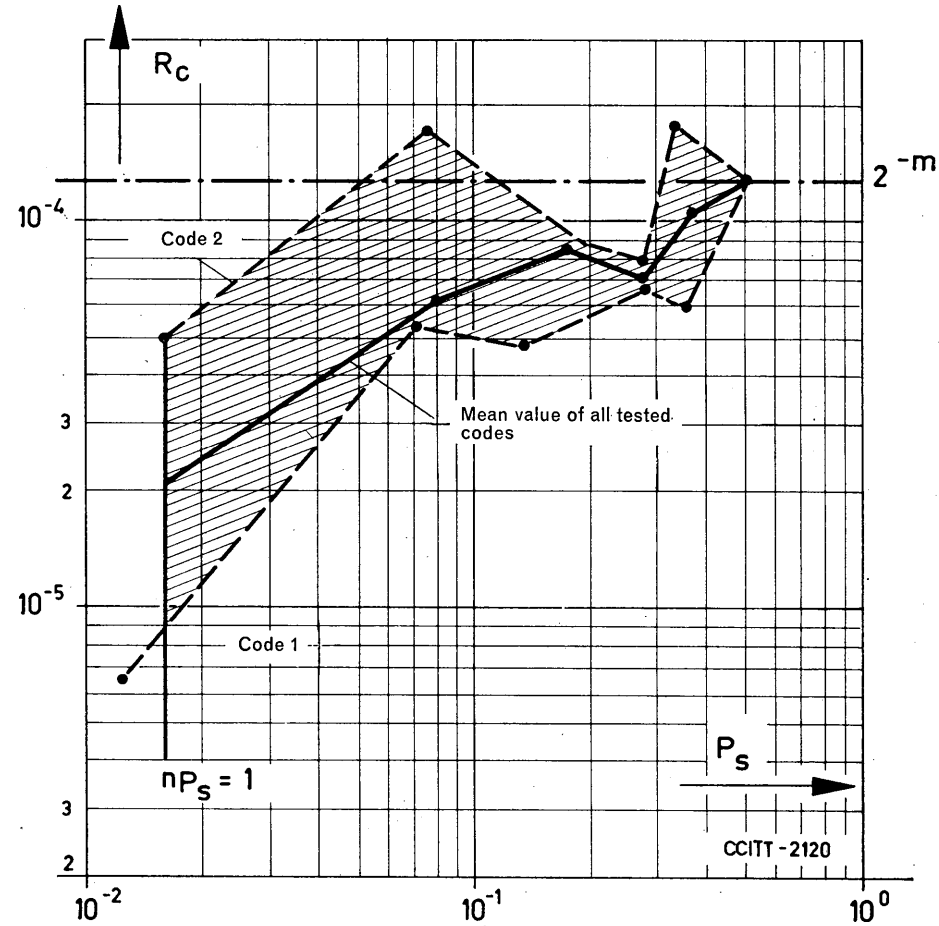
Probability of code words with errors after correction by optimal codes as a function of the blocklength by constant effective transmission rate V_{effc}. Statistically independent errors. Binary error probability $P = 0.11 = \text{constant}$.

FIGURE 1. — **Residual block error probability after error correction with optimal binary codes**



Probability of code words with errors after correction by optimal codes as a function of the blocklength by constant effective transmission rate V_{effc} Statistically independent errors; binary error probability $P = 0.013 =$ constant.

FIGURE 2. — **Residual block error probability after correction with optimal binary codes**



Reductions factor R_c of the block error probability as a function of the binary error probability P_s under conditions, as in telephone channels for codes with the following generator polynomials:

Block length $n = 63$ for all codes:

Code 1: $g(X) = 1+X^2+X^5+X^6+X^8+X^{13}$

Code 2: $g(X) = 1+X^2+X^5+X^7+X^8+X^{12}+X^{13}$

Code 3: $g(X) = 1+X^2+X^5+X^{10}+X^{13}$

Code 4: $g(X) = 1+X^2+X^3+X^5+X^6+X^7+X^9+X^{10}+X^{11}+X^{13}$

Code 5: $g(X) = 1+X^2+X^3+X^4+X^5+X^6+X^7+X^8+X^9+X^{11}+X^{12}+X^{13}$

FIGURE 3. — **Error reduction factor of different codes**

Significant for a redundant code is the reduction factor R_c [4] of the probability of non-recognizable errors (p_B) in the non-secured case, so that at error indication through the code, a remnant-error probability

$$p_{B\ \mathrm{rest}} = p_B \cdot R_c \tag{12}$$

remains. The reduction of non-recognizable errors has been treated theoretically in the literature at various places [5, 1, 6, 7].

We have investigated R_c by measurements for a series of binary codes, which are completely described by their block length n and the so-called generator polynomials $g(x)$ [1] (or [8]), and at data transmission under conditions as found in the telephone network, and we have found that for good codes the relation holds

$$R_c < 2^{-m} \quad \text{at} \quad n \leqslant 1000$$

The error reduction becomes smaller with increasing n at $m =$ const. and larger with falling p_s for $n \cdot p_s \leqslant 1$. The result of a measurement series with five different codes is shown in Figure 3. Displayed is the mean value

$$R_{c\ \mathrm{mean}}\,(p_s) = \frac{1}{5} \sum_{i=1}^{e} R_{ci}\,(p_s) \tag{13}$$

and R_c for the best and the poorest code. R_{ci} is the reduction factor of code i. It can be seen from Figure 3 for instance that

$$R_c \leqslant 10^{-1} \cdot 2^{-m} \text{ at } np_s \leqslant 1 \tag{14}$$

It shall be pointed out that bad codes exist also for which equation (14) does not hold.

2.2 *Signal quality detectors and combinations of signal quality detectors + code*

Signal quality detectors sound alarm when the distortions of a receiving signal exceed a permissible value. The reduction factor R_{st} of the probability of non-recognizable errors depends strongly on the block length n. We have measured $R_{st}\,(n)$ at transmissions over the telephone network for the amplitude-tolerance detector recommended by Marko [9] with different threshold distances S. The result is shown in Figure 4. Error indication results when the amplitude after the discriminator deviates more than $\pm\, S\%$ from the expected amplitude. Not each disturbance causes errors, but each disturbance indication leads to repetition. The frequency of error-free blocks indicated as disturbed corresponds therefore to the redundance r_{st} caused by a signal quality detector. The measured function $r_{st}\,(n, S)$ is indicated in Figure 5.

If a signal quality detector combined with code is used for error indication, the resulting reduction factor is

$$R_{st\ c} < R_{st} \cdot R_c \tag{15}$$

at conditions which can always be met in practice [10]. Figure 6 shows a measured curve for $R_{stc}\,(m)$, and for comparison $R_{st} \cdot R_c$ is drawn also. At $n = 35$ and $m = 8$ for instance one attains

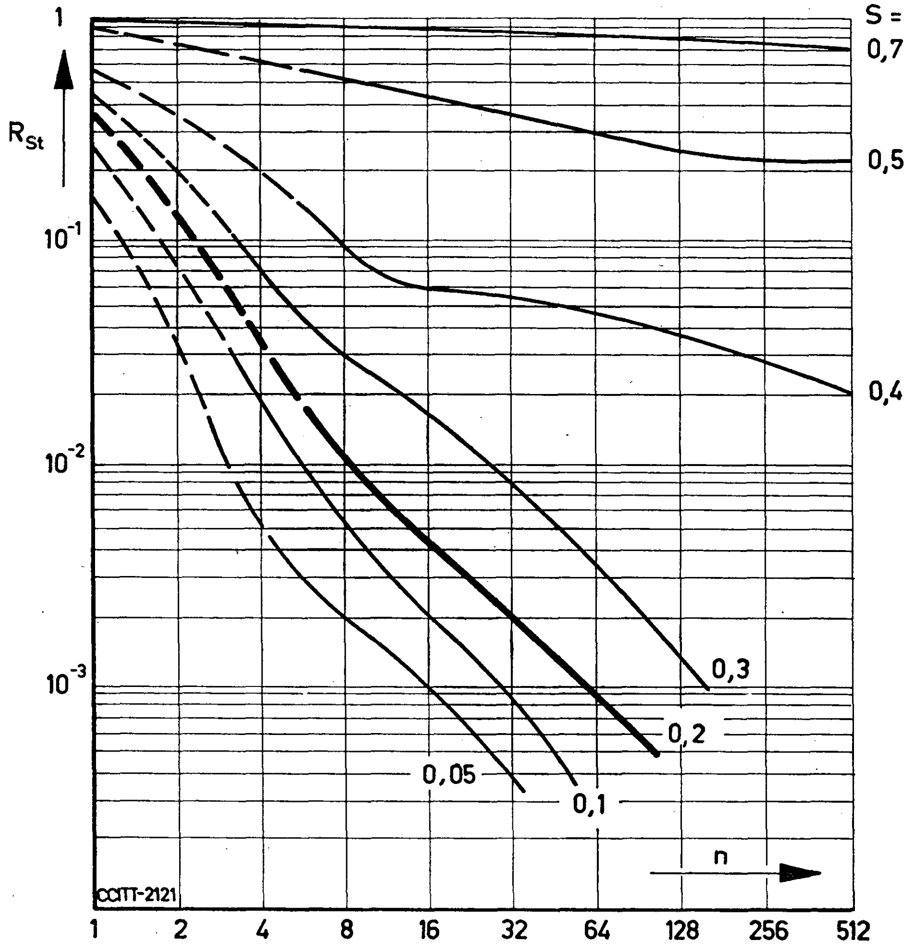
$$R_{st\ c} < 10^{-1} \cdot R_c \cdot R_{st} \tag{16}$$

If we substitute the numerical factors 10^{-1} by K_c and $K_{c\ st}$, in equations (14) and (16), it follows by combined employment of signal quality detector + code

$$p_{B\mathrm{rest}} = p_B \cdot K_c \cdot 2^{-m} \cdot K_{c\ st} \cdot R_{st} \tag{17}$$

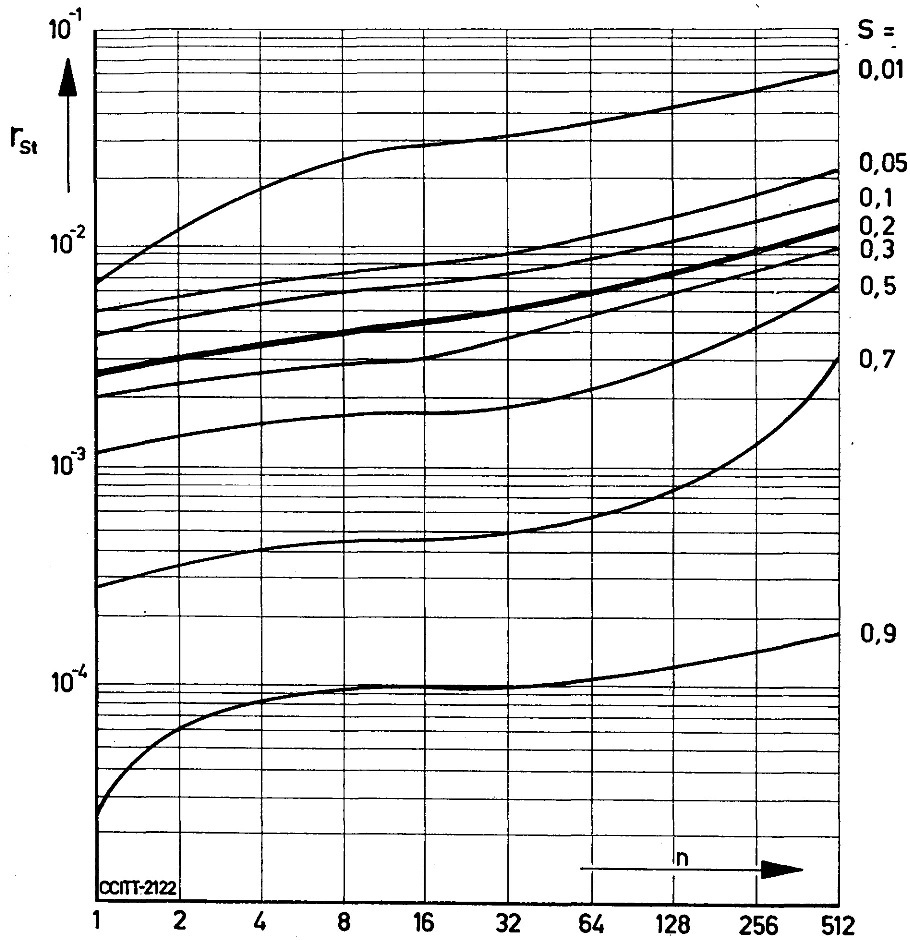
respectively the necessary number m of test digits per block

$$m = \log_2 \frac{p_B \cdot R_{st} \cdot K_c \cdot K_{c\ st}}{p_{B\mathrm{rest}}} \tag{18}$$



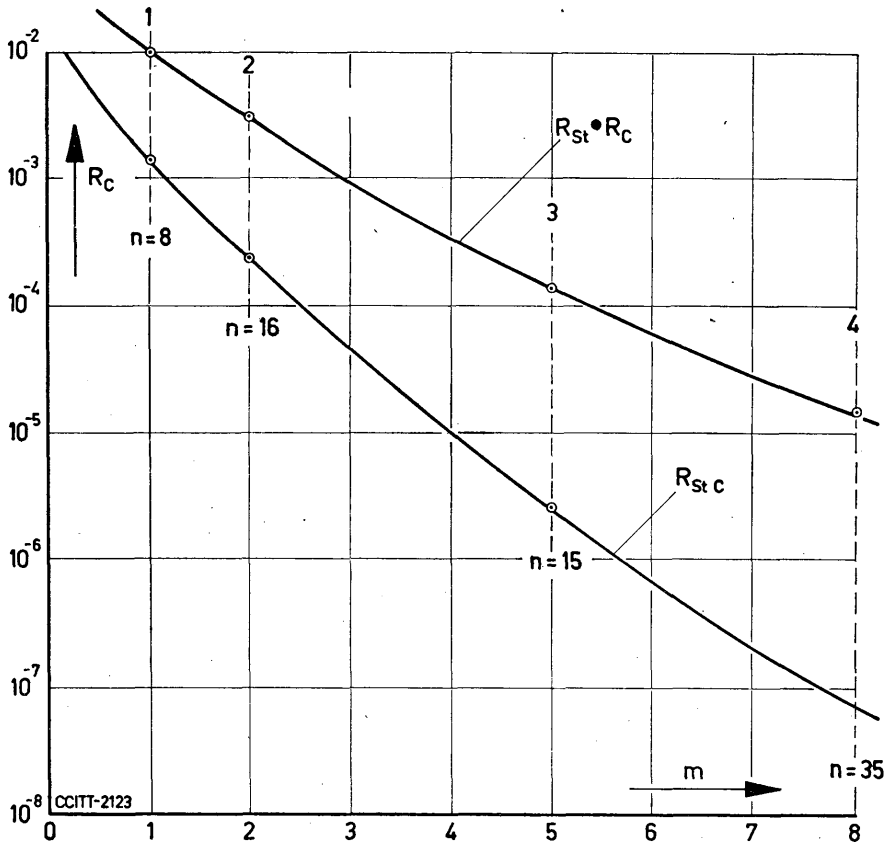
Reduction factor R_{st} of the block error probability as a function of the block length n by constant threshold S and disturbances of telephone channels for the amplitude-tolerance detector.
Binary f.m. with 1200 bit/s.
Receiving level $a_0 = - 30$ dBm0.

FIGURE 4. — **Block error reduction factor of a signal quality detector**



Redundancy R_{st} caused by the recognition of disturbances as a function of the block length n with constant threshold S and disturbances of telephone channels for the signal quality detector. Binary f.m. with 1200 bit/s. Receiving level $a_0 = -30$ dBm0.

FIGURE 5. — **Redundancy by blockwise disturbance indication**



Reduction factor R_{stc} of the block error probability of a combination of signal-quality-detector+code at disturbances of telephone channels as a function of the number m of parity-bits pro block

$R_{st} \cdot R_c$ = Product of the single-reduction-factors

Signal-quality-detector: amplitude-tolerance-detector binary error probability $P_s \approx 10^{-3}$

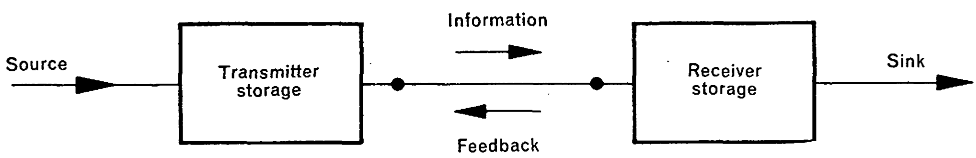
Generator polynomials: code 1: $g(X) = 1+X$

code 2: $g(X) = 1+X+X^2$

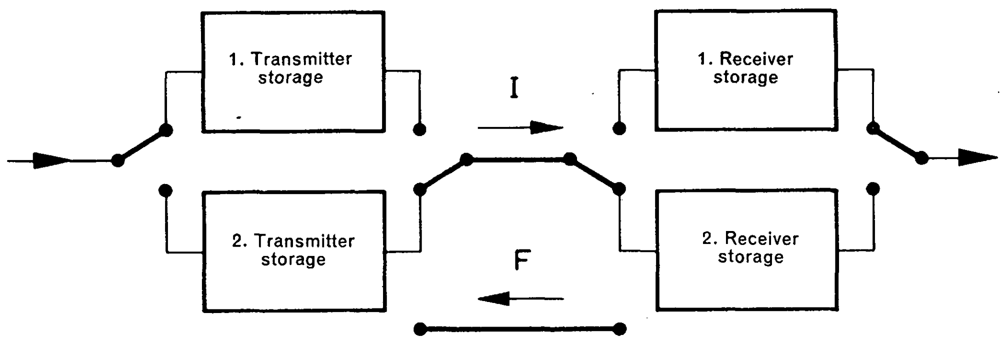
code 3: $g(X) = 1+X+X^4+X^5$

code 4: $g(X) = 1+X+X^3+X^4+X^6+X^8$

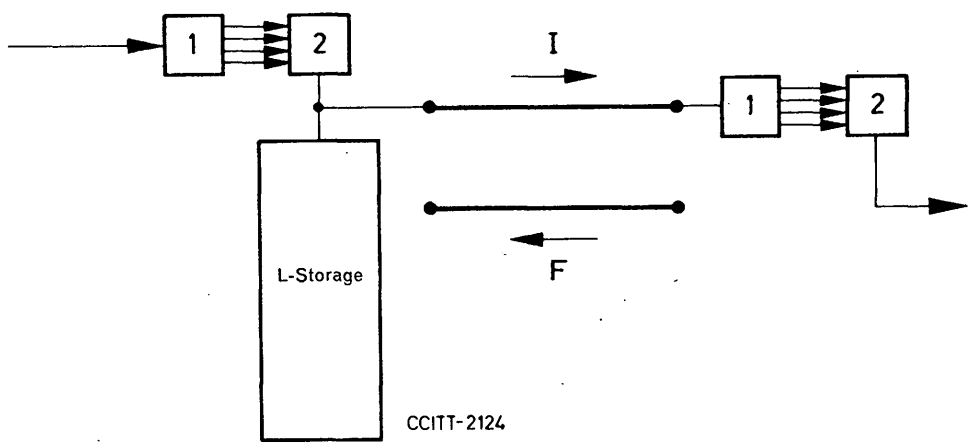
FIGURE 6. — **Error reduction factor of disturbance detector + code**



1) *Semi-duplex system*



2) *Storage change system*



3) *Loop delay time-controlled system*

FIGURE 7. — **Principles of systems organization**

For statistically independent errors the block error probability is

$$p_B = 1 - (1 - p)^n \tag{19.1}$$

$$p_B \approx p \cdot n \text{ at } np < 1 \tag{19.2}$$

Because of error bursts it is

$$p_B(n) \approx p_s \cdot n^{0.7} \tag{20}$$

at strongly disturbed telephone connections [3]. For the calculation of the effective transmission rate we use equation (6) as well as equations (18) to (20) with

$$K_c \cdot K_{st\ c} = 10^{-2} \tag{21}$$

R_{st} and r_{st} are taken from Figures 4 and 5 at a threshold distance $S = 0.2$. For $n \geq 512$ we set $R_{st} = 10^{-4}$ = const. and r_{st} will be extrapolated by means of the curves from Figure 5.

3. *Error correction by repetition of disturbed data blocks*

3.1 *System types*

In the following only systems with decision answer-back will be considered, since the German Federal Post Office offers modems with narrowband return-channel for transmission over the telephone network. For such systems three different principles exist, which are displayed symbolically in Figure 7 with respect of the storage organizations.

Semi-duplex system. — The oldest procedure works on the basis of the semi-duplex system: A block is transmitted and only after reception of decision answer-back (receipt) the next block resp. a repetition will be transmitted.

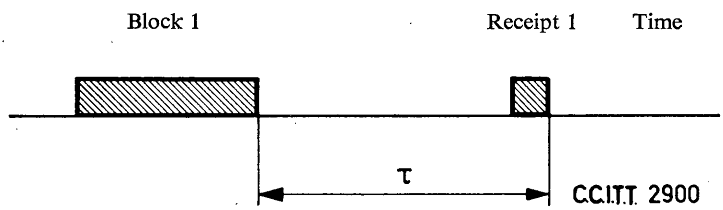


FIGURE 8. — **Definition of loop delay time** τ

The time between transmission end of one block and transmission begin of the next block shall be called loop delay time τ. In the semi-duplex system τ is effectively a time loss; it reduces the effective transmission rate. At least one storage device is necessary at each of the transmission and the reception sides for one data block.

Storage change system. — The loss time τ is avoided in systems based on the storage change principle, which was recommended by Marko [11] already in 1960: The blocks will be transferred to the channel without interruption alternately from the transmitter storage devices. During transmission of the i-th block the receipt comes in for the $(i - 1)$-th block. In case of "wrong"-receipt the disturbed and the next following block will be repeated. In order to recognize erroneous repetition, a sync block must be transmitted ahead of a repetition, such that the entire repetition act consists of three blocks. A variant by Girinsky and Roussel [12] works in this manner, while Marko makes use of three transmission storage devices and repeats five blocks in case of "wrong"-receipt. At the reception side one storage device picks up data from the channel, while the other transfers a block to the sink, respectively. Storage change systems require four storage devices with a total capacity of four data blocks. The transmission duration of one data block must at least be equal to the maximum possible loop delay time $\tau_{\max}$, therewith the smallest-possible block length is being determined.

Delay time controlled system. — The block length can be chosen in delay time controlled systems [13], [14] independently of the loop delay time. Those systems work with advantage with short blocks, and one attains therewith the smallest possible storage expenditure at uninterrupted transmission of data blocks: In storage 1 a data block is being prepared and taken up by storage 2 parallel between two transmission cycles. Storage 2 transfers the binary digits to the channel and at the same time to the L-storage, which can take up as many digits as can be transmitted during the maximum loop delay time. The transmitter measures the time from transmission end of the first block to recognition of the first receipt and knows then during the further transmission which of the incoming receipts is to be correlated to which block. At repetition first the sync-block, then the disturbed block, and finally the

$$L = \left(\frac{\tau}{t_B}\right) \text{ (approximated to integers)} \tag{22}$$

following blocks will be transmitted (t_B = transmission duration of one block). The repetition volume will therefore automatically be adjusted to the loop delay time of the respectively chosen connection. The storage devices 1 and 2 at the receiver side work like the corresponding storage devices of the transmitter. To avoid errors at the delay time measurements a series of provisions are possible [10]. Here only the system principle shall be explained in short.

3.2 *Effective transmission rates*

A repetition will be demanded by the receiver with the probability

$$p_f(n) = p_B(n) - p_{B\text{ rest}} + r_{st}(n) \approx p_B(n) + r_{ts}(n) \tag{23}$$

Most systems work in the manner that receiving a disturbed answer-back also introduces repetition. Therefore the probability $p_w(n)$ for the repetition of a block will be increased by $(1 - p_f(n)) \cdot p_{st} \approx p_{st}$, if p_{st} is the probability for a disturbed receipt. We obtain

$$p_w(n) \approx p_B(n) + r_{st}(n) + p_{st} \quad * \tag{24}$$

as a valid approximation as long as $p_B(n) + r_{st}(n) + p_{st} < 1$.

In the numerical calculations we set $p_{st} = 10^{-3}$. Since certainly $p_{st} < p_B(n) + r_{st}(n)$, deviations of this value will not substantially affect the calculations. It is known [15] that in semi-duplex operation each block of the source must be transferred to the channel

$$1 + \frac{p_w(n)}{1 - p_w(n)} \text{ times} \quad ** \tag{25}$$

The start of a block must be signed by n_{sync} binary digits in the semi-duplex procedure, so that, as a result, a redundancy of at least $m + n_{\text{sync}} + \tau \cdot v_k$ digits per block is created. With k information digits per block the fictive block length follows with

$$n^* = k + m + n_{\text{sync}} + \tau \cdot v_k \tag{26}$$

---

* It is presupposed that a non-recognizable falsification of a receipt occurs with negligible probability.

** Statistical independence of the events leading to repetitions is presupposed in the calculations. Because of the occurrence of error bursts in transmission over telephone channels this is true only in approximation.

As protection against erroneous repetition the blocks will be numbered by ε binary digits per number or by w binary digits in front of a repeated block. The first method causes a loss of

$$V_1 = \left(1 + \frac{p_w(n)}{1 - p_w(n)}\right) \cdot \varepsilon\,, \tag{27}$$

the second a loss of

$$V_2 = \frac{p_w(n)}{1 - p_w(n)} \cdot w \tag{28}$$

binary digits per block of the source. Therefore, as an average only

$$V_{\mathrm{eff\ S1}} = \frac{k}{\left(1 + \dfrac{p_w(n)}{1 - p_w(n)}\right) n^* + V_i} \tag{29}$$

$$i = 1 \text{ or } 2$$

binary digits of the source will be transmitted per binary digit of the transmission channel. For the numerical evaluation we set $n_{\mathrm{sync}} = w = 10$ binary digits. In [16] $n_{\mathrm{sync}} = 13$ is given. At $\varepsilon = 3$ (compare [16]) it is $V_{\mathrm{eff\ S1}} < V_{\mathrm{eff\ S2}}$ (for $n \leqslant 1000$) binary digits, so that we choose $V_{\mathrm{eff\ S2}}$ for comparisons.

At the storage change system the first repetition of a block will be initiated with the probability $p_w(n)$. At disturbance of the sync-block or the block originally to be repeated the whole repetition starts again. This happens with the probability $p_w(n) \cdot p_w(2n)$ with

$$p_w(2n) \qquad p_f(2n) + 2\,p_{st} \quad ^{**} \tag{30}$$

Each repetition effects 3 n binary digits, hence, after transmission of

$$\begin{aligned} & n + 3n(p_w(n) + p_w(n) \cdot p_w(2n) + p_w(n) \cdot p_w{}^2(2n) \ldots + p_w(n) \cdot p_w^{\infty}\,(2n)) \\ & = n\left(1 + 3\,\frac{p_w(n)}{1 - p_w(2n)}\right) \end{aligned} \tag{31}$$

binary digits, the sink contains in the average a block with k information digits.

According to equation (29) the effective transmission rate follows with

$$V_{\mathrm{eff}\ w} = \frac{k}{\left(1 + 3\,\dfrac{p_w(n)}{2 - p_w(2n)}\right) \cdot n} \tag{32}$$

If at the delay time controlled system the sync block or the originally disturbed block is disturbed, the repetition of the $L + 2$ blocks will proceed again ($L = \tau/t_B$ the 2 originates from the sync block and the disturbed block). For the delay time controlled system it follows, therefore, in the same manner as for the storage change system,

$$V_{\mathrm{eff}\,L} = \frac{k}{\left(1 + (L + 2) \cdot \dfrac{p_w(n)}{1 - p_w(2n)}\right) \cdot n} \tag{33}$$

---

* It is presupposed that a non-recognizable falsification of a receipt occurs with negligible probability.

** For p_{st} $(2 \cdot n)$ as an approximation $2 \cdot p_{st}\,(n_q) = 2 \cdot p_{st}$ is set, since the function $p_{st}\,(n_q)$ has not been ascertained yet. (n_q = number of binary digits per receipt).

For $\tau < t_B$ it is $L = 1$ and $V_{eff\ w} = V_{eff\ L}$. For distinction we call the block length of the storage change system n_w and the block length of the delay time controlled system n as hitherto. It follows then with equation (22)

$$n_w \geq \tau \cdot v_k = \frac{\tau}{t_B} \cdot v_k \cdot t_B = L \cdot n \tag{34}$$

For comparison, n in equation (32) must be substituted at least through $L \cdot n$. For the block length n_s of the semi-duplex system no relation exists to n, since n_s can be chosen freely. The expenses of a data-safeguarding system are determined essentially through the necessary storage expenditure. We choose therefore as basis for one of the comparisons equal storage expenditure for the semi-duplex and the delay time controlled system, i.e.

$$2\ n_s = 4\ n + (2 + L)\ n \tag{35}$$

The storage change system requires a storage capacity of 4 Ln, i.e.

$$\frac{4\ Ln}{4n + (2 + L)\ n} = \frac{4}{1 + \frac{6}{L}} \tag{36}$$

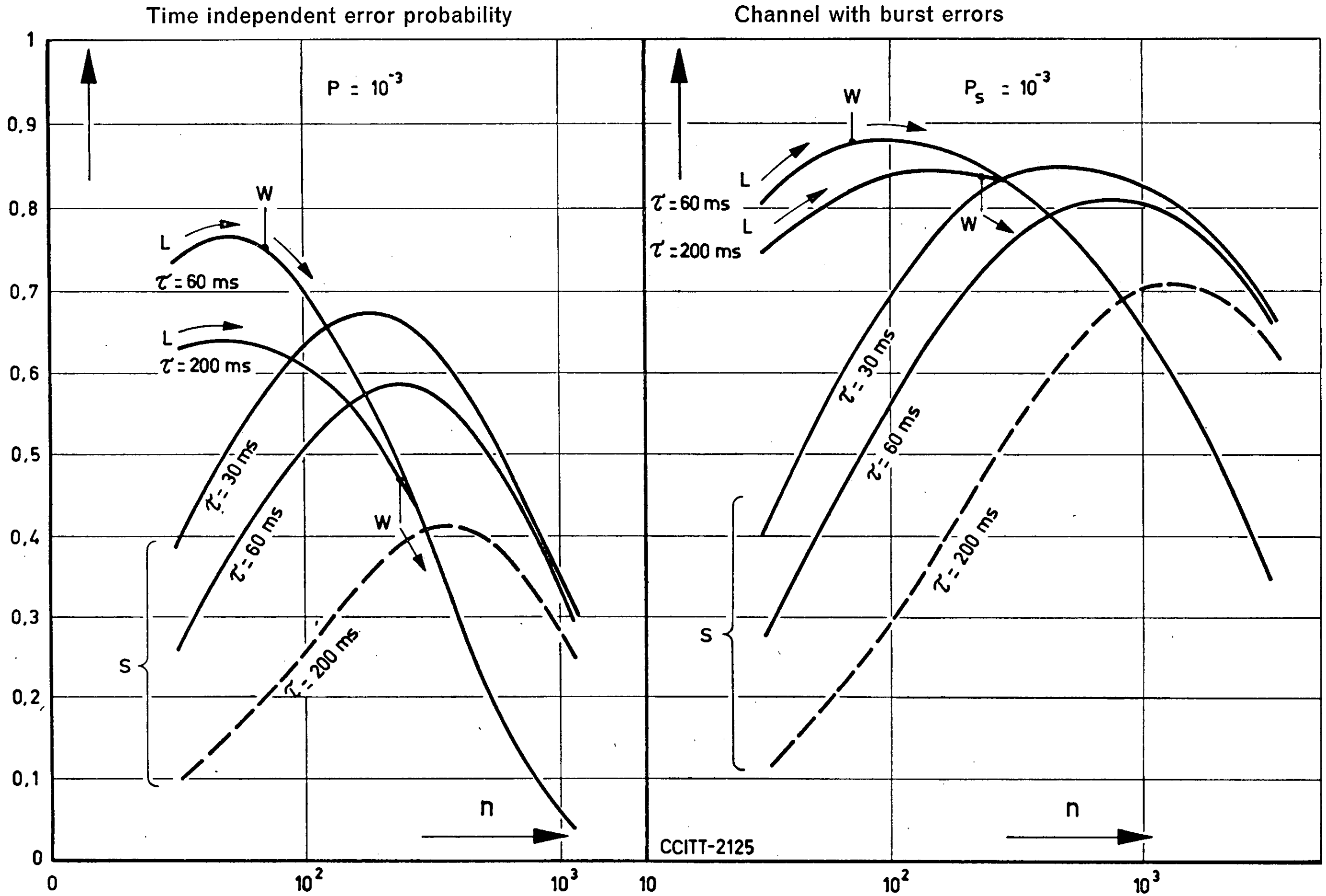
times as much as both the other systems. That means, at $L = 2$ the same storage expenditure is required for all three systems at this comparison.

First we calculate $V_{eff}\ (n)$ with equations (29), (32) and (33), hereby it is set $k = n - m$, and m is determined with equation (18). Difficulties appear in this comparison through the choice of τ. The modems with narrow-band return channel show a loop delay time of $\tau_M = 40$ ms at short-circuit operation. If we consider the feed-back duration, also, $\tau \geq 60$ ms should be an approximately correct limit. Semi-duplex systems work with wide-band return-channel, so that $\tau \geq 30$ ms appears correct here. The loop delay time τ will exceed 200 ms only seldom.

In Figures 9.1 and 9.2 $V_{eff}\ (n)$ is displayed therefore for $\tau = 30$, 60, and 200 ms. Since $\tau = 200$ ms is hardly to be expected with semi-duplex systems, the corresponding curves are drawn as dashed lines. $\tau = 30$ ms is inapplicable for systems with narrow-band return-channel. For Figure 9.1 a time-independent error frequency of $p = 10^{-3}$ is assumed and for Figure 9.2 a strong concentration of errors in bursts with $p_s = 10^{-3}$, so that $p_B(n) = 10^{-3} \cdot n^{0.7}$ is substituted according to equation (20). The optimum block lengths will be shifted to larger values with increasing tendency to bursts at the same time the effective transmission rates increase. The systems with narrow-band return-channel attain higher transmission rates at substantially shorter block lengths in comparison to the semi-duplex system. The storage change system will differ from the delay time controlled system in this comparison by the fact that it can be employed only by starting with a certain block length (signified by W), due to the loop delay time.

The effective transmission rate as a function of the delay time τ or $L = \frac{\tau}{t_B}$ respectively is shown in Figure 10. Thereby it is chosen $n = 64$, $n_w = L_n$ according to equation (34), and $n_s = n \left(3 + \frac{L}{2}\right)$ according to equation (35).

The curves for time-independent error frequency at $p = 10^{-3}$ are denoted by crosses. The other curves are valid for high burst concentration at $p_s = 10^{-3}$ and $p_B(n) = 10^{-3} \cdot n^{0.7}$. If—as done here—equal storage expense is considered for semi-duplex and delay time controlled systems, the semi-duplex system will fall off completely when compared to systems with narrow-band return channels. The semi-duplex system must work with large block lengths if it is to show good efficiency. At strong error burst concentration the two other systems are equivalent with respect to the effective transmission rate. The weaker the error burst concentration, the more favourable works the delay time controlled system, at constant mean error frequency. At a completely time-

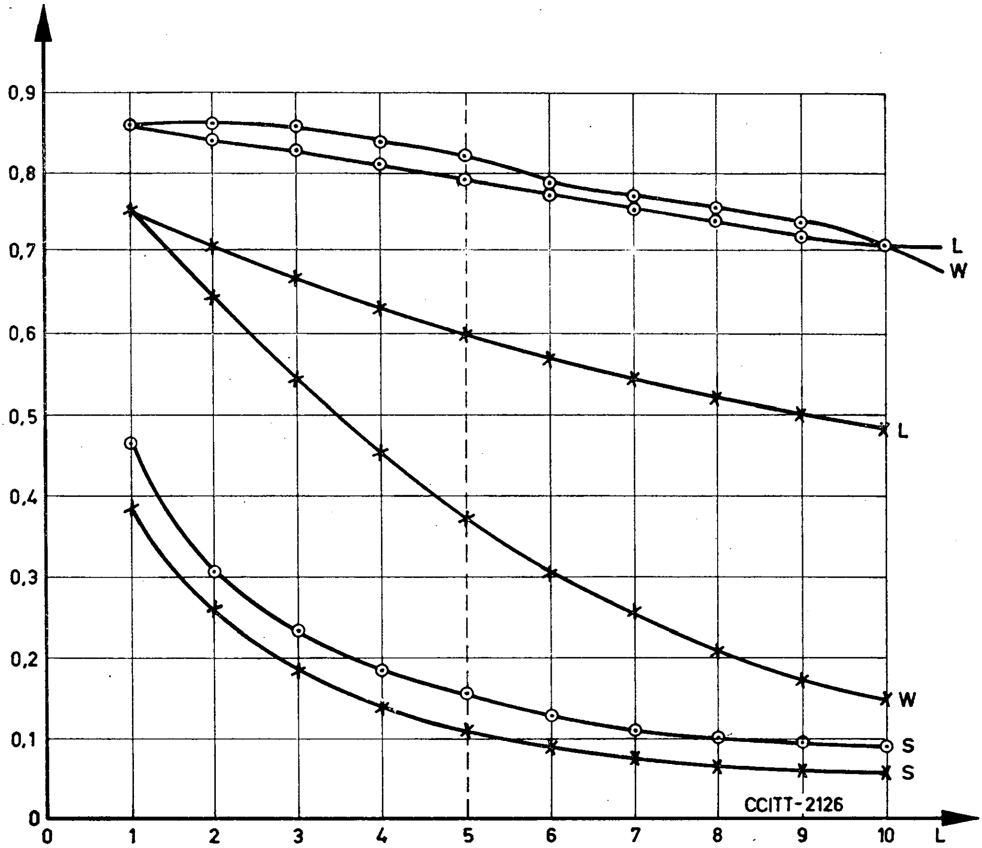


S = Semi-duplex system W = Storage changing system L = Loop delay time-controlled system

FIGURE 9.1 FIGURE 9.2

Effective transmission rate as a function of block length with *n* binary digits

independent error frequency the delay time controlled system is substantially superior to the other systems; for $L > 2$ it requires also less storage capacity than the storage change system. With increasing noise intensity the maximum of the effective transmission rate is shifted to smaller block lengths in repetition systems. Delay time controlled data transmission systems possess therefore a good efficiency also at high noise intensity. This is seen in Figure 11: $V_{eff}\ L,\ w \cdot 1^{(L)}$ are displayed with increasing noise intensity p as a parameter. At 1% error frequency and $L = 5$



Effective transmission rate V_{eff} as a function of the normalized loop-delay-time
$L = \tau / t_B$ ($t_B = 53$ ms $\quad n = 64$ with $V_k = 1200$ bits/s).

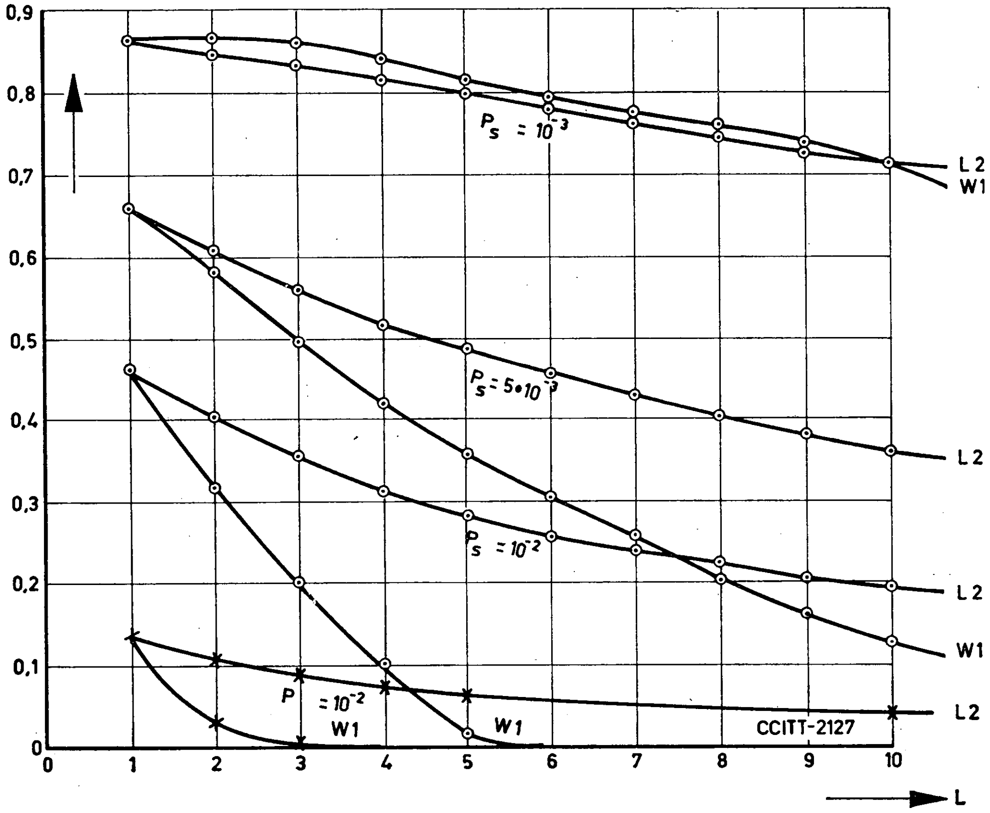
1. Time-independent error probability. P: 10^{-3} (—x—x—)

2. Channel with burst errors as in telephone channels. P: 10 (—o—o—)

Transmission rate: $V_k = 1200$ bits/s

Block length: $L \;\hat{=}\;$ Loop delay time-controlled system $n = 64$.
$W \;\hat{=}\;$ Storage changing system $n_w = n \cdot L$.
$S \;\hat{=}\;$ Semi-duplex system $n_s = n \cdot (3 + L/2)$.

FIGURE 10. — **Effective transmission rate as a function of the normalized loop-delay-time**



Effective transmission rate V_{eff} as a function of the normalized loop-delay-time $L = \tau/t_B$ (t_B = 53 ms) with:

1. Time-independent error probability. $P = 10^{-2}$ (—x—x—)
2. Channels with error bursts as in telephone channels
   P_s as parameter, transmission rate V_k = 1200 bits/s
   Block length n = 64 for the loop-delay-time-controlled system.
   $n_w = L \cdot n$ for the storage changing system.

FIGURE 11. — **Effective transmission rate as a function of the normalized loop-delay-time**

the effective transmission rate of the storage change system falls nearly to zero, while the delay time controlled system still works with an effective transmission rate of about 30%.

Finally, it can be stated that a very good channel economy is obtained at data transmission over the telephone network with repetition systems, when code plus signal quality detector for error recognition are employed. In all comparisons $P_{Brest} = 10^{-8}$ was presupposed, this gives a residual binary error probability of $P_{rest} \approx 10^{-9}$ for an error density of $d_m = 0.1$ in the disturbed block.

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Supplement No. 27

A.E.G. TELEFUNKEN. — *(Contribution COM Sp. A—No. 166—November 1967)*

SUGGESTIONS FOR TWO CODES WITH 63 BINARY DIGITS PER BLOCK

In Supplement No. 24 a system with error correction through repetition of disturbed blocks is suggested, which works with relatively short blocks. In Supplement No. 26 in the appendix it is shown that for transmission on telephone channels favourable block lengths are approximately 60 binary digits per block.

A number of codes with $n = 63$ binary digits per block have been tested, therefore, under conditions as exist at transmission over the telephone network. Tested were group codes which are characterized by generator polynomicals $g(X)$, i.e. which can be materialized by simple feedback shift registers. Thereby the codes:

Code 1): $g(X) = X^{12} + X^{10} + X^8 + X^5 + X^4 + X^3 + 1$
$n = 63$ binary digits per block
$m = 12$ test digits per block

Code 2): $g(X) = X^{10} + X^9 + X^7 + X^6 + X^5 + X^2 + X + 1$
n = 63 binary digits per block
m = 10 test digits per block

have shown to be particularly favourable. Code 1 is a Bose-Chaudhuri code. Measurements revealed that code 1 reduces the probability of non-recognizable block errors by a factor $Rc_1 \approx 1.5 \cdot 10^{-5}$, and code 2 by a factor $Rc_2 \approx 3.5 \cdot 10^{-5}$. It is recommended to employ one of those two codes for error recognition when the favourable block length of $n = 63$ binary digits per block is chosen for data transmission.

At the same time of the measurements of the error reduction factors Rc_1 and Rc_2, the signal quality detector has been tested which is described in Supplement No. 23. A reduction factor of the block error probability of $R_{st} \approx 10^{-3}$ resulted, at a redundance of $r_{st} \approx 2\%$.

Supplement No. 28

FEDERAL REPUBLIC OF GERMANY. — *(Contribution COM Sp. A—No. 17—August 1965)*

MEASUREMENT OF PHASE DISTORTION BETWEEN SUBSCRIBERS

It is common practice to indicate the group delay instead of the phase distortion (see [1]). This is to be attributed to the following reasons:

1. In the case of carrier telephone channels employing single sideband transmission with suppressed carrier, the phase between the input and the output of the channel gets lost. Direct assessment of phase distortion is therefore not possible.

2. When a signal, modulated on a carrier, is transmitted over a speech circuit, it is in practice sufficient to indicate attenuation distortion and group delay distortion for the assessment of the circuit quality with regard to its linear distortion [2]. If required, phase distortion may be derived by integration from this explanatory parameter, i.e. from the group delay/frequency response.

Two *explorative* series of tests were made to obtain a survey of the group delay distortion to be expected in the Deutsche Bundespost telephone network. The first series of tests was made on some twenty connections in the public telephone network, which were known "to be outside limits" [1] and the second series on some twenty connections (typical of the German telephone network) simulated in the laboratory. Both series of measurements were made with the help of a sweep frequency delay measuring set of Messrs. Siemens & Halske, which determines the group delay distortion from the phase shift of a sinusoidal oscillation (27.8 Hz) with which the frequency of the swept carrier is modulated [3].

[1] While carrying out measurements to determine the condition existing in the German telephone network with regard to distortion (see Supplement No. 29), these connections were found to be outside limits as far as the attenuation distortion is concerned. Owing to sections of heavily loaded circuits the bandwidth was comparatively narrow. At the time the group delay measurements were made—i.e. some 12 months later, some of these circuits had been replaced by carrier channels.

It became evident that group delay distortions are almost exclusively caused by carrier frequency channels and that the overall subscriber-to-subscriber group delay is, therefore, practically proportional to the number of carrier frequency channels connected in tandem in the connection.

Compared with that, the influence of loaded and unloaded circuits is but slight, particularly because in the German telephone network such type circuits are relatively short. In most cases the complete speech band (300-3400 Hz) is available even with loaded circuits. This is another reason why the increase of group delay distortion at the upper limit of the frequency band—which is typical of loaded circuits—is of little consequence to the German telephone network. As far as the German network is concerned, the number of connections comprising sections of circuits with a heavier load amounts to some 5-10% and is decreasing more and more.

In the following figures the group delay distortion as a function of frequency to be expected in the German switched telephone network is indicated by hatched areas.

The curves of Figure 1 were determined with a view to:

a) the results of group delay measurements carried out on some 20 real and 20 simulated connections; and

b) the fact that the majority of connections in the German telephone network does not comprise more than five (on the average about 2-3) carrier channels connected in tandem.

Connections with longer sections of heavily loaded circuits were not considered, since they constitute only a small percentage of the total number of possible connections.

The group delay distortion to be expected corresponds, on the average, to that of the dashed curve of Figure 1. Here it was assumed that the connection comprises three carrier telephone channels connected in tandem. This distortion is likely to be the maximum value to be expected on the national portion of an international connection.

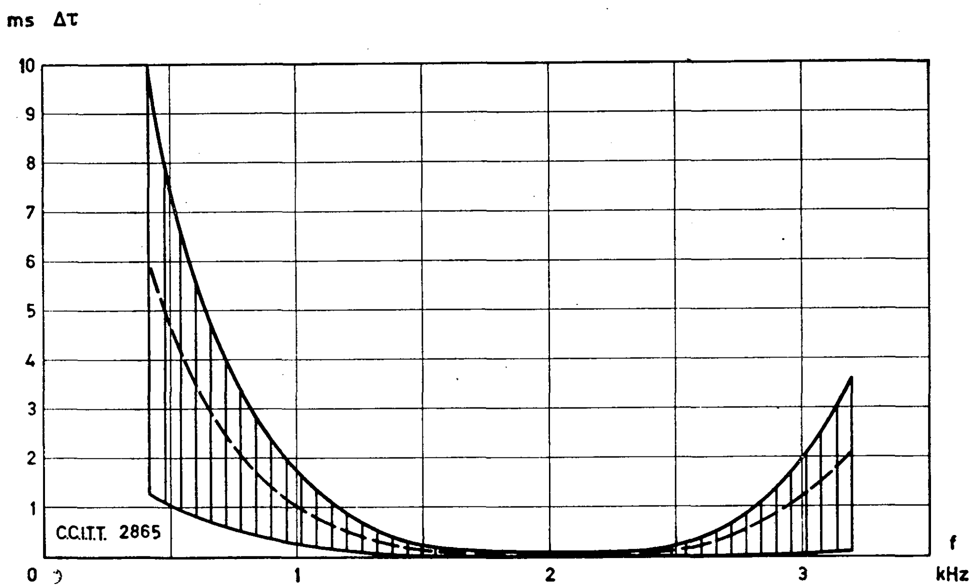

FIGURE 1. — **Deviation range of delay distortion $\Delta\tau$ in the switched telephone network**

(Connections comprising up to five carrier telephone channels in tandem, but no heavily loaded circuits)

Figure 2 a shows the deviation range of the group delay distortion of some connections, comprising no carrier channels but loaded circuits. In conclusion, Figure 2 b represents the deviation range of the group delay distortion of some connections only comprising unloaded circuits and exchange equipment.

It has to be pointed out once again that the group delay distortion plotted in the figures can only give an indication for an assessment of the actual conditions, since the number of test circuits does not allow more accurate statements to be made.

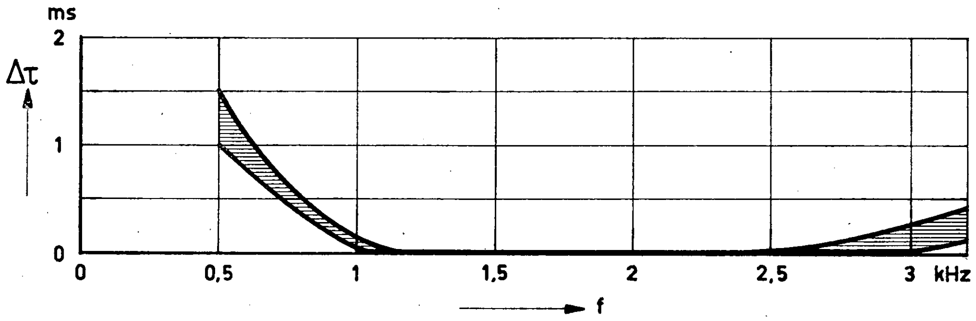

FIGURE 2 a. — **Deviation range of delay distortion $\Delta\tau$ of several connections simulating typical connections comprising unloaded local circuits and loaded regional circuits, but no carrier channels**

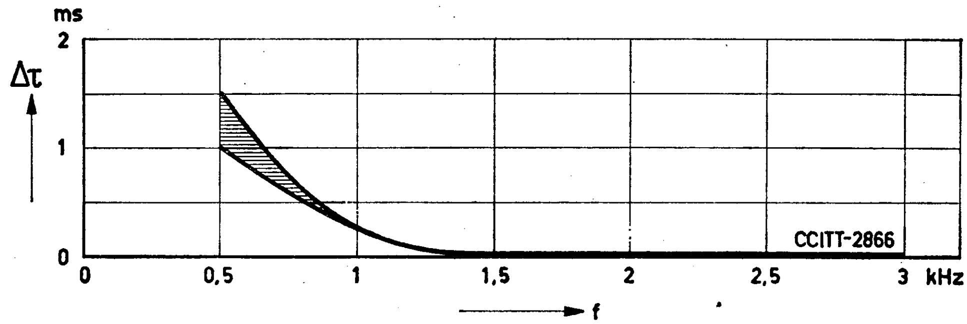

FIGURE 2 b. — **Deviation range of delay distortion $\Delta\tau$ of several connections simulating typical connections in the local network comprising unloaded circuits only**

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FEDERAL REPUBLIC OF GERMANY. — *(Contribution COM Sp. A—No. 18—August 1965)*

SUBSCRIBER-TO-SUBSCRIBER LOSS

1. *General*

The quality of data transmission in the public telephone network depends, amongst others, on the overall loss and the attenuation distortion occurring between subscribers' telephone stations. While the overall loss must be considered when laying down the transmitting level and the sensitivity of the receiver incorporated in the data transmission equipment, the rating of a compromise equalizer requires the attenuation distortion to be known.

When the distribution of the attenuation distortion (i.e. the bandwidth) of the connections in the public telephone network is known, it is possible to estimate the probability with which a connection lends itself to a data transmission rate of 1200 bits/s.

That is why in 1963 the Deutsche Bundespost carried through an extensive test programme with a view to determining the statistical distribution of the overall loss and the attenuation distortion between telephone stations of the public network. Although the assessment of the network quality requires the knowledge of the delay distortion and the noise voltages (particularly, pulse type ones) these parameters could not be measured because there were not yet suitable measuring sets available at that time.

2. *Test programme*

2.1 *Selection of measuring points*

Some 200 telephone stations distributed all over the area of the Federal Republic of Germany were selected as measuring points. The selection was made with a view to the volume of trunk traffic in the different trunk exchange areas. Hence, for instance, 15 % of the 200 measuring points were in a tertiary (regional switching centre) area, whose trunk traffic amounts to 15 %. Within the tertiary areas distribution to terminal exchanges of different type was effected according to the same principle. In the selection of subscribers' stations, no importance was attached to the fact whether their connections to the long-distance network were particularly favourable or unfavourable. This was deliberately left to chance.

2.2 *Measuring and evaluating methods*

From each of the 200 telephone stations a connection was set up to a telephone station at Darmstadt (extension in the laboratory of the Fernmeldetechnisches Zentralamt). Then, the overall loss as a function of frequency was measured on this connection in both directions of transmission. Thereafter, the connection was released and established again by the station at the Fernmeldetechnisches Zentralamt. Thus, the connection was frequently set up on a different route. The overall loss as a function of frequency was measured also on this new connection in both directions of transmission. In this way it was possible to measure overall loss and attenuation distortion in both directions on a total of 400 connections. The results are plotted in Figures 1 to 5, curve *a*.

A disadvantage of this test programme was that at one end all connections terminated on the same telephone station. In order to obtain a true representative cross-section of the telephone network, it would have been necessary to measure the connections between the different telephone stations, but this would have required a large number of staff and a lot of time.

But, in order to get as close to the actual conditions as possible, the test results were additionally evaluated according to the following method:

All calls incoming to Darmstadt and not originated in the close vicinity of Darmstadt were routed via Frankfurt. Loss and attenuation distortion between the four-wire through-connection point at Frankfurt and the Darmstadt telephone station were the same for all incoming calls. Now, the 200 telephone stations were grouped to form random pairs. Only those pairs were evaluated, further, whose geographical position was such that a call between the two partners had to be established via Frankfurt. For each of these pairs the overall loss of the calls to Darmstadt was added. Double the amount of the overall loss between Frankfurt and Darmstadt was then subtracted from this sum. Moreover, the attenuation distortion of a carrier channel was additionally subtracted, because normally these calls would have been through-connected in the basic group at Frankfurt.

Some 800 pairs were formed and evaluated in this way.

The results of these investigations are plotted in Figures 1 to 5, curve *b*.

3. *Results*

3.1 *General*

As stated above, the loss as a function of frequency was measured for all connections (terminated with 600 ohms). These attenuation curves were analysed from angles being particularly interesting to data transmission in the public telephone network. The results of this analysis were plotted in a probability network, so that the measured values on a straight line have a Gaussian distribution.

The solid curve (curve *a*) applies always to the calls tested between the Fernmeldetechnisches Zentralamt Darmstadt and the 200 different subscribers' stations. The dashed curve (curve *b*) applies always to some 800 calls set up between about 200 different subscribers' stations. These values were determined according to the method outlined under 2.2 above. While, as far as the deviation of the loss values is concerned, the solid curves apply to calls set up between one specific subscriber and a great number of other subscribers, the dashed curves indicate the deviation of the loss values to be expected between any two stations of the network.

Since the curves plotted in Figures 1, 2 and 3 are in rough approximation based on a Gaussian distribution, the arithmetic mean and the standard deviation were determined graphically. These values give a good survey and can be used when making a comparison with the results obtained by other administrations. It should be noted that these values include the loss and attenuation distortion of the subscribers' lines, part of which are rather long.

3.2 *Overall loss at 800 Hz*

Figure 1 shows the integral distribution curves of the overall loss at 800 Hz.

	Curve *a*	Curve *b*
Arithmetic mean $\bar{a}$	2.08 Np	2.33 Np
Standard deviation S	0.4 Np	0.6 Np
Loss, not exceeded on 95% of the connections $a_{(95\%)}$	2.8 Np	3.34 Np

3.3 *Overall loss at 1700 Hz*

Figure 2 shows the integral distribution curves of the overall loss at 1700 Hz.

	Curve *a*	Curve *b*
Arithmetic mean $\bar{a}$	2.24 Np	2.64 Np
Standard deviation S	0.5 Np	0.7 Np
Loss, not excceded on 95% of the connections $a_{(95\%)}$	3.22 Np	3.93 Np

3.4 *Attenuation distortion between 800 Hz and 2500 Hz*

Figure 3 shows the integral distribution curves of the attenuation distortion Δa between 800 Hz and 2500 Hz.

	Curve *a*	Curve *b*
Arithmetic mean Δa	0.45 Np	0.7 Np
Standard deviation S	0.34 Np	0.52 Np
Attenuation distortion, not exceeded on 95% of the connections $a_{(95\%)}$	1.19 Np	1.72 Np

3.5 *Upper frequency limit for an attenuation distortion $\Delta a = 6$ dB*

Figure 4 shows the integral distribution curves of the upper frequency limit, having an attenuation distortion of 6 dB (= 0.69 Np) at 800 Hz. The value plotted on the ordinate indicates the percentage of connections, where the upper (6 dB) frequency limit was below the value shown on the abscissa. Hence, for 50% of all connections the upper frequency limit was $\leqslant$ 3.0 kHz (curve *a*) and $\leqslant$ 2.6 kHz (curve *b*) respectively.

For 10% of all connections it was $\leqslant$ 2.1 kHz (curve *a*) and $\leqslant$ 1.6 kHz (curve *b*) respectively.

3.6 *Upper frequency limit for an attenuation distortion $\Delta a = 10$ dB*

Figure 5 shows the integral distribution curves of the upper frequency limit, having an attenuation distortion of 10 dB (= 1.15 Np) at 800 Hz.

For 50% of all connections it was $\leqslant$ 3.3 kHz (curve *a*) and $\leqslant$ 3.25 kHz (curve *b*).

For 10% of all connections it was $\leqslant$ 2.65 kHz (curve *a*) and $\leqslant$ 2.2 kHz (curve *b*).

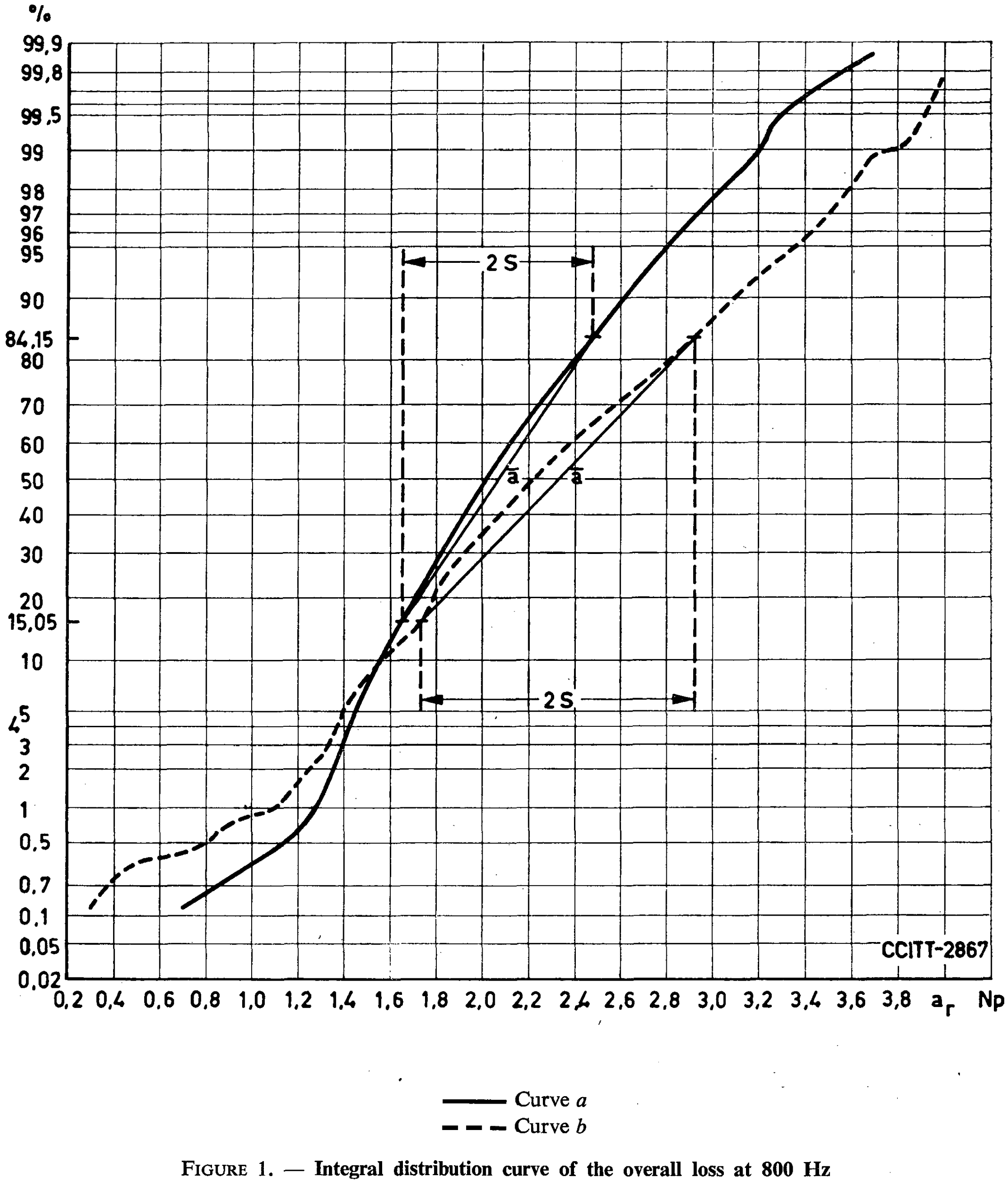
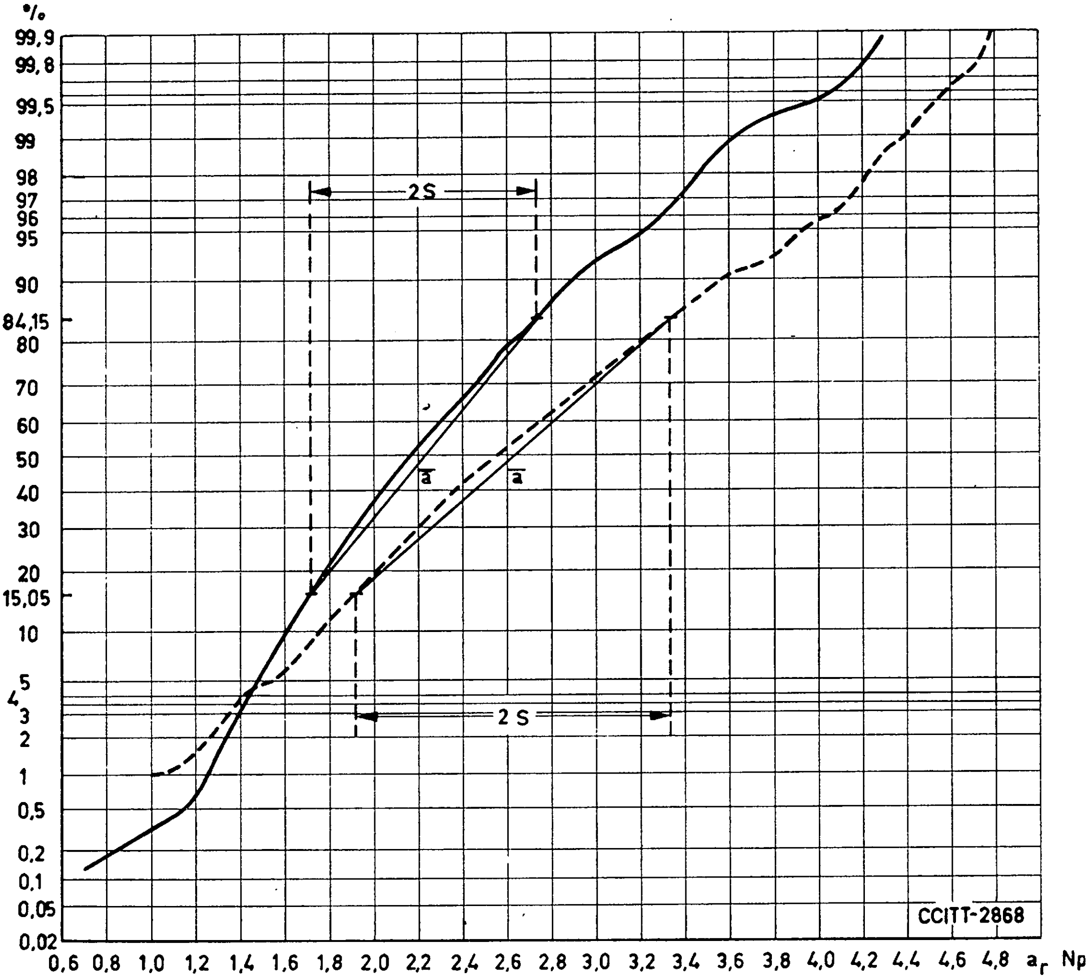
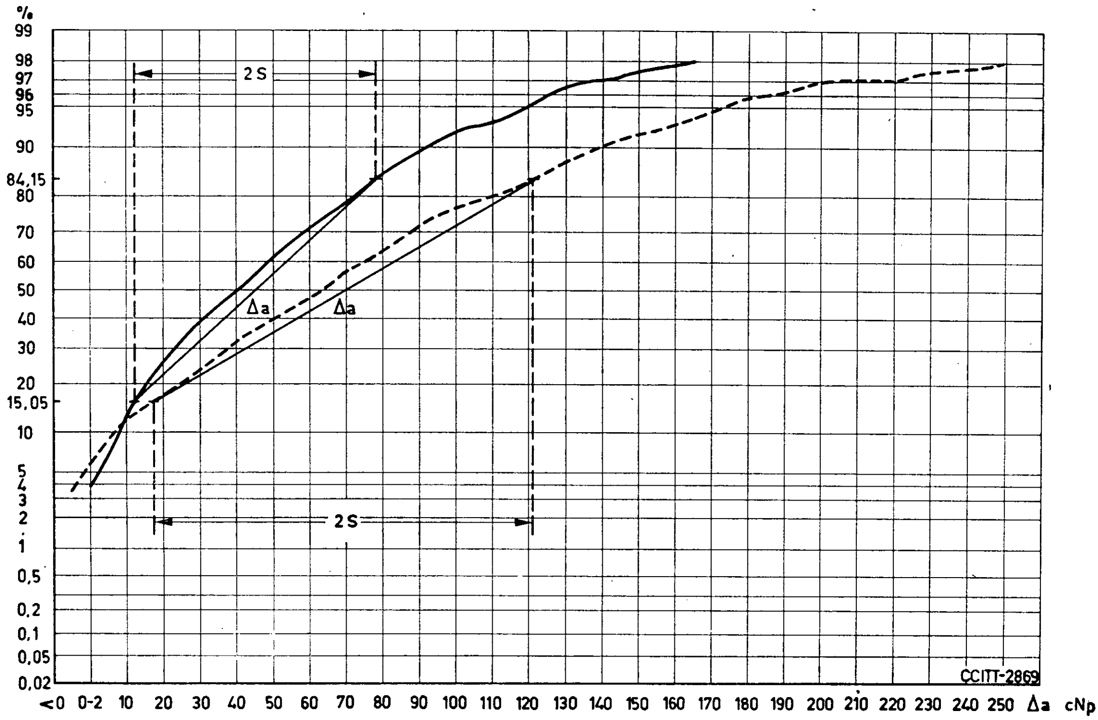


FIGURE 1. — **Integral distribution curve of the overall loss at 800 Hz**



—— Curve *a*
– – – Curve *b*

FIGURE 2. — **Integral distribution curve of the overall loss at 1700 Hz**



—— Curve *a*
– – – Curve *b*

FIGURE 3. — **Integral distribution curve of the attenuation distortion Δa between 800 Hz and 2500 Hz**

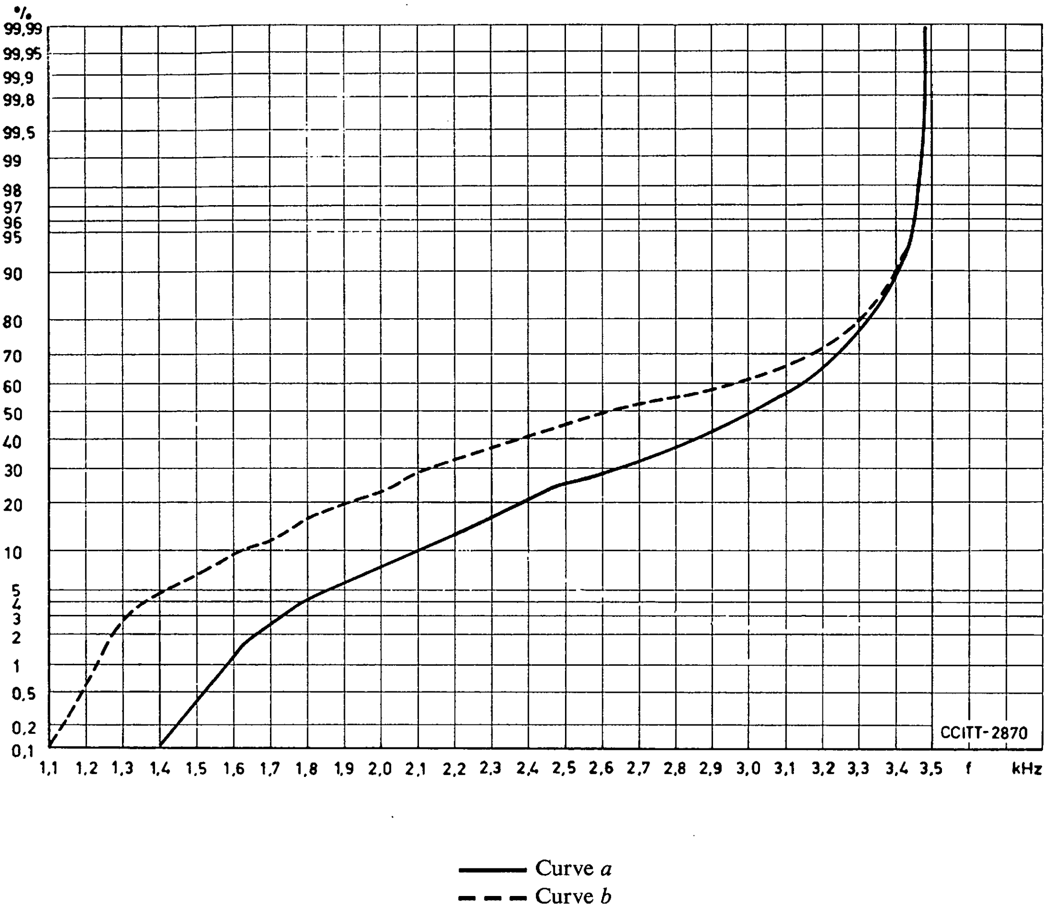


FIGURE 4. — Integral distribution curve of the frequency f, at which the attenuation distortion $\Delta a = 6$ dB

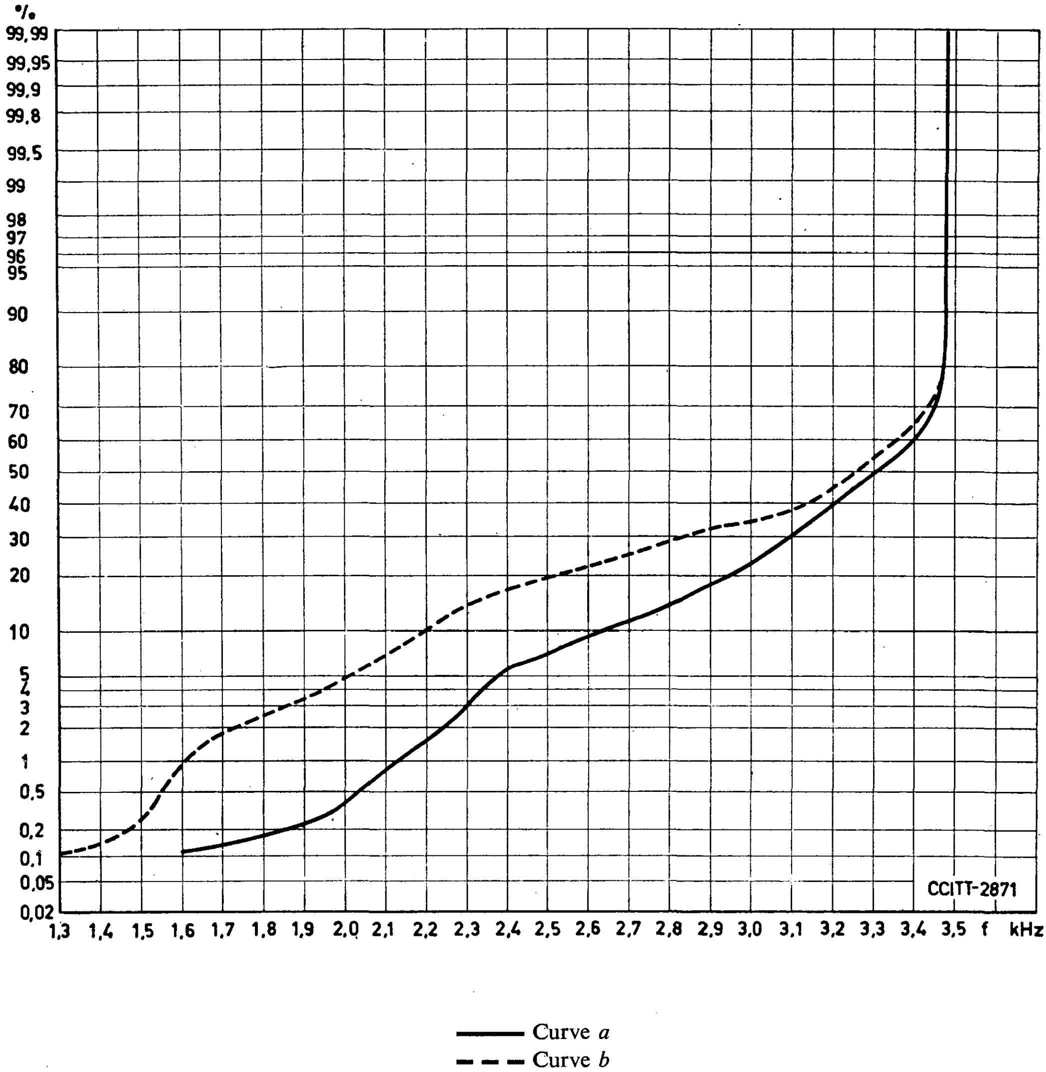


FIGURE 5. — **Integral distribution curve of the frequency *f*, at which the attenuation distortion $\Delta a = 10$ dB**

NETHERLANDS. — *(Contribution COM Sp. A—No. 49—November 1965)*

SUBSCRIBER-TO-SUBSCRIBER LOSS IN INTERNATIONAL TELEPHONE CONNECTIONS

A complete international telephone connection consists of three parts: an international chain and two national systems. In part A of this contribution the statistical distribution is given of the Netherlands national system of an international connection.

To know the extent of the loss in the complete subscriber-to-subscriber connection the data concerning the other two component parts should also be known.

As long as these data are unknown, the determination in part B of the subscriber-to-subscriber loss had to be based on some suppositions. As soon as these data will be known it is possible to provide an exact distribution by means of the results obtained from part A.

A. *Statistical distribution of the loss in the Netherlands national system of an international connection*

To avoid that measurements had to be taken at subscribers' premises the following procedure was followed. A great many measurements (about 1000 in total) were made between various exchanges in the Netherlands to which subscribers are connected on the one hand, and the virtual switching point [1] of the international circuit in the international centre of either Amsterdam or Rotterdam on the other hand.

The loss distribution of the local network was next calculated from the data that were known about the length and diameter of the cables connecting the subscriber to the exchanges. Combination of these two provided the distribution of the loss from subscriber to international centre. Figure 1 shows by way of example this distribution at a frequency of 800 Hz. The average m and the standard deviation s of this distribution were determined. As this was also done at a number of other frequencies, m and s could be drawn as functions of the frequency f (Figure 2).

In these Figures 1 and 2 it was assumed that all the Netherlands subscribers contributed to the same extent to the international traffic.

This is certainly not the case for the *telephone* traffic: by far the greater part of the international telephone traffic originates from subscribers connected to district exchanges. If this is taken into account, the loss statistics will be different; the values thus found are given in Figures 3 and 4.

The above examples have been given, because it is not yet clear how the traffic distribution will be for phototelegraphy and data transmission.

[1] According to Recommendation G.141, A: the point of the international circuit where the nominal relative levels at the reference frequency are: at the emission — 3.5 dB; at the reception — 4.0 dB.

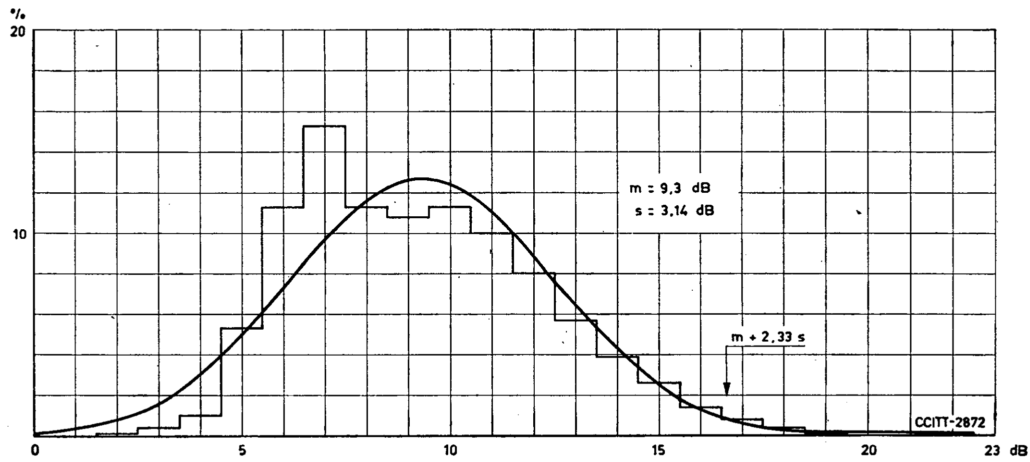


FIGURE 1. — Distribution of the loss in the Netherlands national system of an international connection in case each subscriber contributes to the same extent to the traffic f = 800 Hz

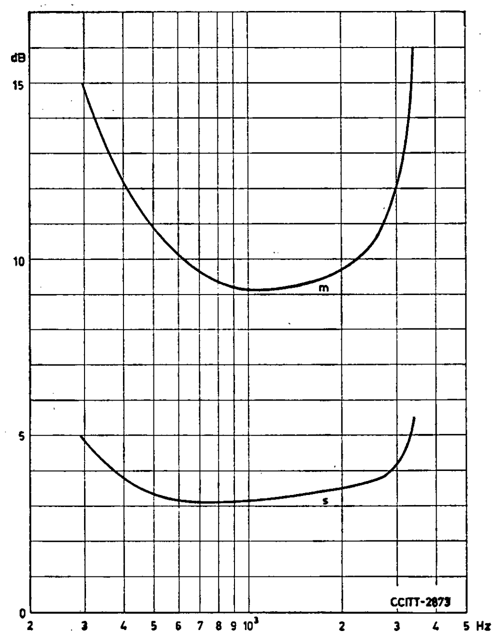


FIGURE 2. — Average value m and standard deviation s of the loss in the Netherlands national system of an international connection in case each subscriber contributes to the same extent to the traffic

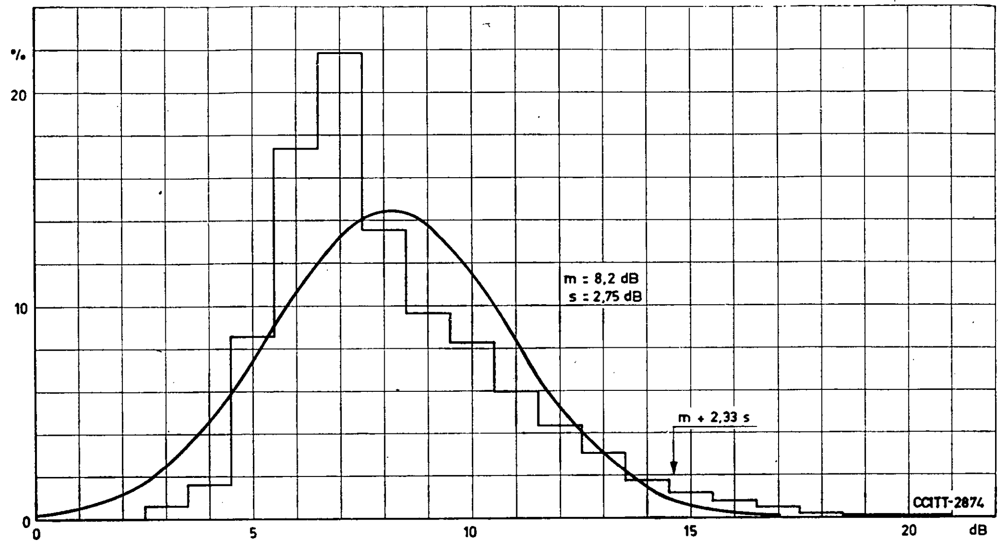


FIGURE 3. — Distribution of the loss in the Netherlands national system of an international connection, taking into account the real distribution of the international telephone traffic over the subscribers $f = 800$ Hz

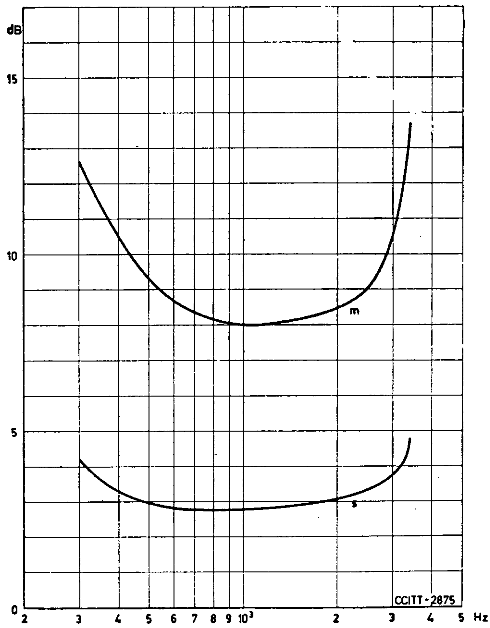


FIGURE 4. — Average value *m* and standard deviation *s* of the loss in the Netherlands national system of an international connection, taking into account the real distribution of the international telephone traffic over the subscribers

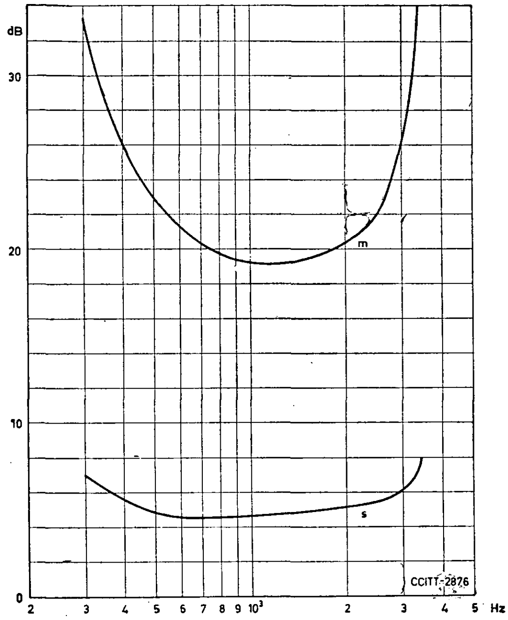


FIGURE 5. — **Average value *m* and standard deviation *s* of the subscriber-to-subscriber loss in an international connection in case each subscriber contributes to the same extent to the traffic**

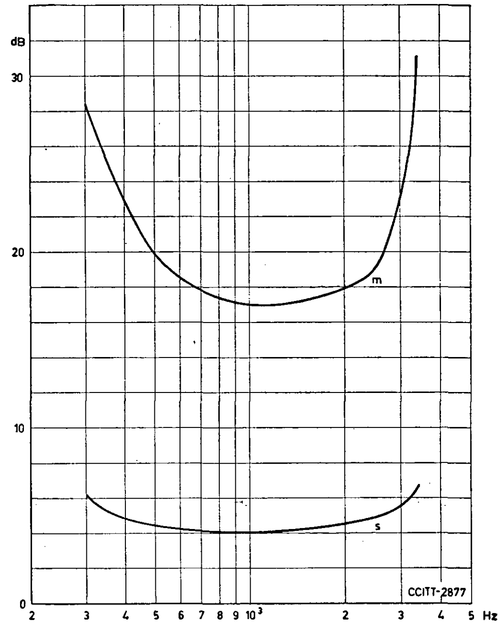


FIGURE 6. — **Average value *m* and standard deviation *s* of the subscriber-to-subscriber loss in an international connection, taking into account the real distribution of the international telephone traffic over the subscribers**

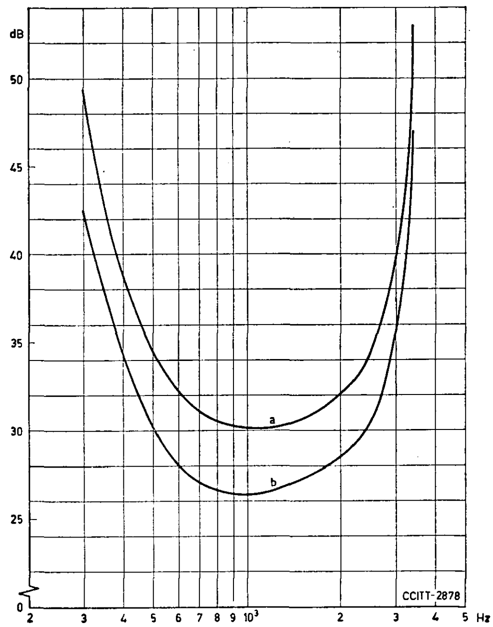


FIGURE 7. — **Upper limit of the subscriber-to-subscriber loss in an international connection which is exceeded in 1% of the cases**

a) If each subscriber contributes to the same extent to the traffic

b) Taking into account the real distribution of the international telephone traffic over the subscribers.

B. *Subscriber to subscriber loss*

The following calculations are based on the suppositions:

a) that the loss distribution in the other national system is equal to that in the Netherlands system;

b) that the international chain consists of two circuits. Each circuit has an average loss of 0.5 dB at 800 Hz, increasing to 1 or 2 dB at the band limits, and a standard deviation of 1 dB.

Figures 5 and 6 indicate the average value and standard deviation of the loss obtained for the complete circuit by means of adding up the averages and variances [1] of the component parts.
In the case of a normal distribution the upper limit of the loss which is exceeded in 1% of the cases lies at $m + 2.33 \cdot s$.
Although it appears from Figures 1 and 3 that the distributions are not normal, this will nevertheless be assumed, so as to simplify matters.
The upper limit thus obtained is shown in Figure 7.

Supplement No. 31

STUDY GROUP XVI. — *(Geneva Meeting, 6-10 June 1966)*

CHARACTERISTICS OF WORLD-WIDE CONNECTIONS WITH REGARD TO DATA TRANSMISSION

Introduction

The following text is forwarded to Special Study Group A. Study Group XVI will continue to collect information to amplify it.

1. *Attenuation distortion*

In order to make a reliable estimate of the attenuation distortion likely to be encountered in practice on a maximum world-wide connection from subscriber to subscriber it is necessary to know, or estimate, the distortion contributed by the principal components of such a connection. These components are as follows:

1.1 *Two subscribers' lines*

The average length of a subscriber's line is probably between ½ to 1½ miles (0.8 to 2.4 km) routed on unloaded cable pairs, often of mixed gauge and introducing an attenuation of say 4 to 5 dB (4.6 to 5.8 dNp) at 1600 Hz. The maximum amount of unloaded cable pair is typically such as will introduce an attenuation of 10 dB (12 dNp) at, say, 1600 Hz. Longer lengths will probably be loaded.

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[1] Square value of the standard deviation.

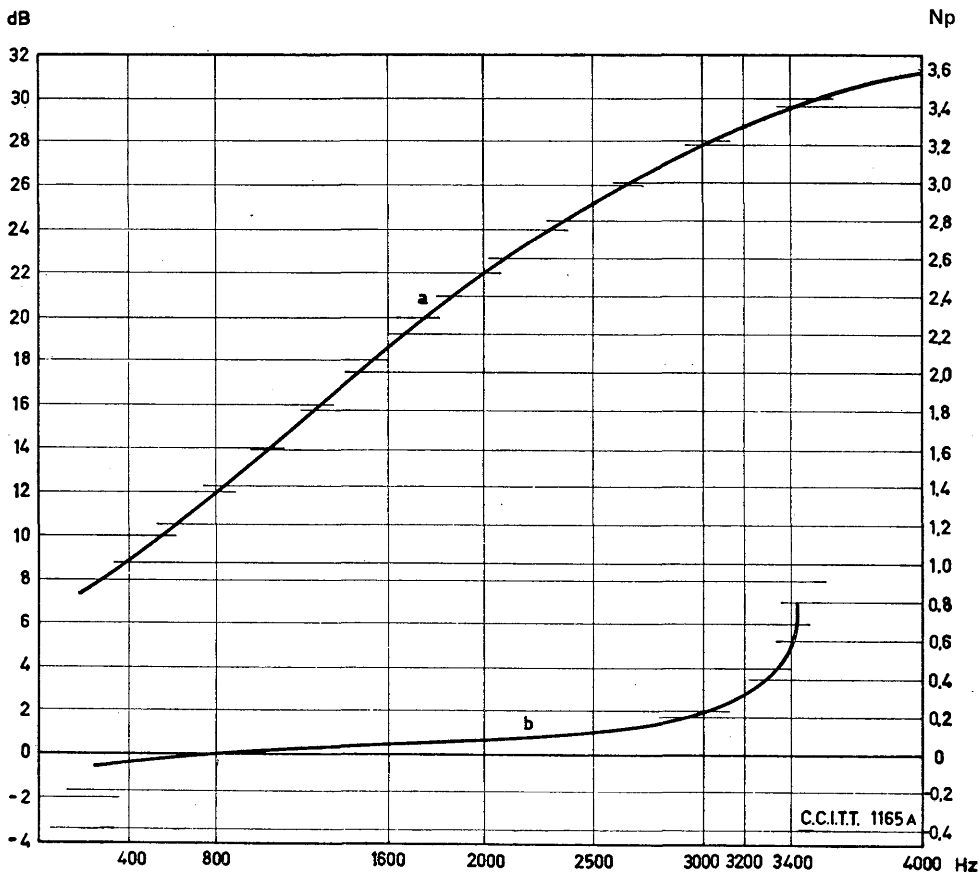
1.2 *Two junction circuits*

These circuits connect the local exchange to the trunk centre and typically are routed on loaded or unloaded cable pairs, each introducing attenuation in the range 3 to 6 dB (3.5 to 7 dNp) at 1600 Hz. In some connections these junctions will be provided on short-haul FDM or PCM line transmission systems.

Figure 1 shows the insertion loss-frequency characteristic between 600-ohm resistances of an unloaded cable pair dimensioned to introduce 19 dB (or 2.2 Np) at 1600 Hz. The 19 dB (2.2 Np) is composed as follows:

— two average subscribers' lines, 4.5 dB each = 9 dB

— two average junction circuits, 5 dB each = 10 dB



a = Twice the insertion loss between 600-ohm resistances of 14 000 yards (12.8 km) of unloaded cable (20 lb; 19 gauge; 0.9 mm) the attenuation at 1600 Hz being 19 dB (2.19 Np).

b = Unequalized attenuation of 50 miles (80 km) of loaded cable relative to the attenuation at 800 Hz (88 mH, 2000 yards; 1.818 km, 20 lb; 19 gauge, 0.9 mm; 0.066 nF/mile, 41 nF/km).

FIGURE 1. — **Characteristics of loaded and unloaded cable circuits — No allowance for exchange equipment**

Figure 1 also shows the unequalized attenuation distortion of 50 miles (80 km) of loaded cable (20 lb, 19 gauge, 0.9 mm; 88 mH; 2000 yards, 1.818 km; 0.066 μF/mile, 41 nF/km).

The recommendations of Section 1 of the *Blue Book*, Volume III, recognize the presence of up to 100 miles (160 km) of loaded cable in national networks. The ordinates of the curve are proportional to the assumed length.

1.3 *A four-wire chain of 12 carrier circuits*

Study Group XV has expressed the opinion that the attenuation-frequency characteristic of a chain of twelve carrier circuits will probably be within the limits given in the table below, provided that the channel-translating equipments comply with the requirements of Recommendation G.232.

TABLE

Likely limits of the attenuation-frequency characteristic of 12 carrier circuits each with one pair of 4-kHz channel equipements conforming to Recommendation G.232. (No allowance is made for exchange apparatus.)

Frequency range (Hz)	Range of loss relative to that at 800 Hz
Less than 400	Not less than − 2.2 dB (− 2.5 dNp) otherwise unspecified
400- 600	+ 4.3 to − 2.2 dB (+ 5 to − 2.5 dNp)
600-2400	+ 2.2 to − 2.2 dB (+ 2.5 to − 2.5 dNp)
2400-3000	+ 4.3 to − 2.2 dB (+ 5 to − 2.5 dNp)
3000-3200	+ 8.7 to − 2.2 dB (+ 10 to − 2.5 dNp)
Greater than 3200	Not less than − 2.2 dB (− 2.5 dNp) otherwise unspecified

Some long international circuits use 3-kHz-spaced channel equipment, so there will be no signal transmitted above about 3100 Hz on connections incorporating such a circuit. Furthermore, such circuits sometimes have 3 pairs of channel-translating equipments.

The nominal loss at 800 Hz of the chain of 12 circuits between the two-wire points of the terminating sets at the terminating trunk exchanges will be of the order of 10 dB (or 12 dNp); the standard deviation will be of the order of 3 to 4 dB (3.5 to 4.6 dNp).

1.4 *Fifteen telephone exchanges*

The distortion introduced by the fifteen telephone exchanges (two local exchanges, two terminal trunk centres and eleven intermediate trunk exchanges, national and international) is difficult to assess, there being very little information available [1]. The combined distortions of the subscribers' feeding bridges, the terminating sets at the terminal trunk centres, the switching and signalling equipments, and any tie-cables between repeater stations and telephone exchanges, on such a maximum connection will be considerable.

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[1] See Annex.

2. *Group delay and group delay distortion*

Study Group XV has estimated the combined group delay characteristic of the channel-translating equipments in a chain composed of 12 carrier circuits using typical figures supplied by administrations.

This characteristic together with the corresponding characteristic for 50 miles (80 km) of loaded cable is given in the table below.

Group delay (ms)

Frequency (Hz)	12 pairs of channel-translating equipments	50 miles (80 km) of loaded cable [1]
300	50	4.5
400	35	4
2000	14	4
3000	22	5.5
3400	41	6

The group delay distortion introduced by unloaded cable pairs and the usual type of exchange apparatus is generally negligible compared with group delays of the magnitude shown in the table.

Hence the group delay distortion of a world-wide connection is likely to be of the order of 36-38 ms at 300 Hz and between 27-35 ms at 3400 Hz, the higher figures being taken if there is 200 miles (320 km) or so of loaded cable in the connection.

The absolute group delay at a frequency of 800 Hz or so can be estimated for a particular connection from the following:

National extensions

The group delay to the most distant subscriber is unlikely to exceed:

12 + (0.0064 × distance in statute miles) ms,

12 + (0.004 × distance in kilometres) ms.

International circuits

1) Terrestrial, including submarine, cables: assume a velocity of propagation of 100 statute miles/ms (160 km/ms).

2) *Satellites* (single hop)

Medium-altitude satellite—mean altitude 8750 miles (14 000 km): 110 ms
High-altitude satellite—mean altitude 22 500 miles (36 000 km): 260 ms.

[1] 20 lb, 19 gauge, 0.9 mm; 88 mH/2000 yards; 1.818 km; 0.066 μF/mile, 41 nF/km.

3. *Magnitude of echoes accompanying the data signal*

In Annex 1 of the *Blue Book*, Volume III, an estimate is made of the stability of the four-wire chain of circuits of a world-wide connection. The method of calculation described in that annex can be used to estimate the magnitude of the first listener echo accompanying the signal at the data receiver. As a first approximation the ratio of the signal to first listener echo at the terminal trunk exchange serving the data receiver is twice the stability[1] of the four-wire chain assuming substantially equal balance return losses at the two terminating sets and also assuming that they are the principal echo sources.

The attenuation distortion suffered by the echo signal is not now a function solely of the transmission losses of the circuits and exchanges but also of the balance return loss characteristics presented at the terminating sets. In this regard it is important to note that the worst values of balance return loss usually occur at the edges of the telephone band, outside the likely band of interest for data transmission. Hence, the value of balance return loss may be assumed to be greater than the stability balance return loss, perhaps approaching the value of echo balance return loss, although this latter is more strictly related to the subjective effect of echo on telephone users.

4. *Thermal and intermodulation noise* (hissing)

The objectives for the noise power at a zero relative level point on international hypothetical reference carrier circuit 2500 km long on a cable or radio-relay systems are given in Recommendation G.222. Other noise objectives are given for circuits on open-wire lines, tropospheric forward-scatter systems and communication satellite systems in Recommendations G.311, G.444 and G.445 respectively.

Objectives are given for hourly-mean, the one-minute-mean, and the 5 ms-integrated noise powers. The one-minute-mean and the 5 ms-integrated noise objectives apply particularly to radio-relay systems whereas cable f.d.m. systems are adequately described by the hourly-mean alone.

It must be noted that the durations mentioned (hour, minute, 5 ms) refer to the method of measurement or calculation. They do not in any way purport to describe the characteristics of the noise being measured. In the particular case of radio-relay systems the C.C.I.R. is of the opinion that high-level noise attributable to propagation phenomena (e.g. fading) is characterized by duration of the order of seconds or tens of seconds and that high-level noise from other sources (e.g. clicks from power supplies and switching apparatus) is reduced to negligible proportions by the design of the radio-relay system.

For circuits less than 2500 km, one may assume without serious error that the magnitudes of the hourly-mean and the one-minute-mean noise powers are directly proportional to length. In the case of the 5 ms-integrated noise power it is the incidence which may be considered to be proportional to length (i.e. the percentage of the month or hour).

Provisional noise objectives for very short circuits on carrier systems are given in Recommendation G.125, and those for circuits much longer than 2500 km are given in Recommendation G.153. These objectives are being studied by Special Study Group C.

It may be of interest to note that the objectives given in Recommendation G.222 are suitable for existing systems of telephone signalling and also for frequency-modulated voice frequency telegraphy at 50 bauds. Amplitude-modulated voice-frequency telegraphy at 50 bauds is more susceptible to noise and Recommendation G.442 gives a more stringent objective for the 5 ms-integrated noise power.

[1] The quantity "twice the stability" is numerically equal to the quantity referred to as "loop loss" symbol *M*) in Annex 1 of the *Blue Book*, Volume III.

5. *Impulsive noise*

Little is known quantitatively about impulsive noise on international circuits. Special Study Group C has recently specified a suitable instrument to measure impulsive noise [1]. For information, impulses equalling the peak value of a sinusoid with a power level of — 30 dBm0 or — 3.5 Npm0 are not uncommon on channels of modern carrier systems. Impulses at lower levels are correspondingly more numerous.

6. *Spurious signals at discrete frequencies*

When 16-channel groups are routed on group links provided by f.d.m. systems based on 4-kHz-spaced virtual carrier frequencies it sometimes requires special effort to reduce the 1-kHz and 2-kHz "whistles" to an acceptable level.

7. *Short interruptions*

Supplement No. 10 to Volume IV of the *Blue Book* gives the results of the recent series of measurements of interruptions on international circuits. As a first approximation the incidence of the interruptions can be assumed to be proportional to circuit length. However, there cannot reasonably be interruptions of this type on that part of a circuit or connection provided on undersea plant or satellite paths.

8. *Sudden phase changes*

In f.d.m. systems these often occur when carrier supplies are changed over. Any phase-change from 0° to 180° can be expected. It is doubtful if such phase changes ever occur unaccompanied by a short break or indeed if they are even distinguishable from them.

9. *Unwanted side-frequencies derived from power supplies* (hum-modulation)

One cause of this is the presence of spurious signals on the carrier supply at the power supply frequency (and its harmonics and sub-harmonics). Values of signal-to-side frequency ratios lie typically in the range 45 to 55 dB (or 5.2 to 6.3 Np) per circuit, although much lower ratios have been encountered. This topic is to be investigated in the current study period.

10. *Frequency error*

This is due to carrier asynchronism and it is unlikely that an error greater than 2 Hz per circuit will be introduced under normal conditions. Study Group XV is considering (under Question 12/XV) the error to be expected on a chain of 12 circuits.

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[1] See Recommendation V.55.

11. *Non-linear distortion*

11.1 *F.d.m. systems*

Cross modulation products generated within the transmitted band of f.d.m. systems are usually at an insignificantly low level, provided the total power level of the compound signal is sufficiently below the overload point of the channels.

11.2 *P.c.m. systems*

It is inherent in p.c.m. line transmission that sinusoidal signals will be accompanied by intermodulation products which are somewhat higher in level than those generated in a conventional f.d.m. carrier circuit and of course considerably higher in level than those in a physical audio pair circuit. The actual signal/distortion ratio for a particular p.c.m. system is a function of the number of coding digits, the companding law and the signal level. For systems currently in operation, or being studied, a maximum signal/distortion ratio of about 30 dB (35 dNp) could be attained over a range of signal levels from 0 to 30 dBm0 (0 to 35 dNpm0). At levels above or below this range, the signal/distortion ratio falls fairly rapidly to, typically, 10 dB (11.5 dNp). The distortion products are of the type 2A, A + B and 2A − B. On international connections there could be two such circuits and, more exceptionally, four.

ANNEX

Transmission characteristics for an international automatic exchange

Extract from the Reply by Study Group XI
(New York Meeting, 14-22 April, 1966) to Question 4/XI

Reply

Study Group XI proposes the following recommendations:

General

The proposed values for the parameters listed in the various points of the question are intended as design objectives for new exchanges. It would be desirable to apply similar design ob .ctives to new national four-wire transit exchanges.

It is proposed that in the present context international automatic exchange should be defined in principle as all four-wire electromechanical exchanges for service at CTI, CT2 and CT3 centres.

The recommendations may also be applicable to electronic exchanges having metallic contact crosspoints.

The measuring points are A2 and D2 as shown on Figure 11 of Recommendation Q.45 of the *Blue Book* (Volume VI) (Figure 13 of Recommendation G.142, *Blue Book*, Volume III).

. . . .

Loss-frequency distortion referred to 800 Hz, A to D

It is considered that wetting must be retained as a possibility.

The transmission loss measured over the frequency bands indicated shall not differ from that measured at 800 Hz[1] by more than the values stated.

300- 400 Hz	– 0.2 dB to + 0.5 dB
400-3400 Hz	– 0.2 dB to + 0.3 dB

Delay distortion

The delay distortion measured over the band 500-3000 Hz shall not exceed 100 microseconds.

Supplement No. 32

UNITED KINGDOM. — *(Extract from Contribution COM Sp. A—No. 136—October 1967)*

MEASUREMENT OF PHASE DISTORTION AND TRANSMISSION LOSS BETWEEN SUBSCRIBERS

1. *Introduction*

The United Kingdom Administration is currently conducting a test programme to investigate characteristics of the public switched telephone network of special interest to data transmission. Information relevant to Question 1/A, point T, and to a lesser extent point S, is given in this contribution.

2. *Test procedure and equipment*

The present tests include, among other things, collection of information on the following parameters:

a) Attenuation/frequency characteristics;

b) Echo delay time and echo relative level;

c) Group delay/frequency characteristics.

No results on point c) are, as yet, available.

In order to obviate the necessity for transporting large quantities of measuring equipment, magnetic tape recorders are being used since this permits measurement on any particular circuit without duplicate measuring equipment. The test signals are reproduced from standard test tapes at the transmitting end of the circuit and the received signals are recorded at the other end, the records being subsequently analysed at a central point. This standardized measurement procedure ensures that a maximum number of tests can be made in a given time and also provides a permanent record of results. Tests are conducted between 600-ohm terminations.

Attenuation/frequency and group delay/frequency characteristics. — The test signal consists of selected discrete tones from 200 Hz to 3500 Hz, amplitude-modulated at 25 Hz, and transmitted sequentially at a fixed level. At the receiver, the recorder is calibrated with respect to level by application of a local reference tone prior to each test. Transmission delay is determined by comparing the phase of the received envelope with that of a similar locally generated reference signal.

[1] 1000 Hz is an acceptable alternative reference frequency.

At both ends of the circuit the 25-Hz modulating frequency is derived from a high stability 200-kHz radio transmission. Since it is required only to determine the variation with frequency of the group delay, the reference signal does not have to be absolute.

Echo delay time and echo relative level. — The test signal consists of an unmodulated tone gliding from 200 Hz to 3500 Hz in 20 seconds. The echo delay time and relative level are derived from the cyclic variations in the attenuation/frequency curves.

Selection of routes. — Tests have so far been conducted within and between three of the ten telecommunication Regions into which the United Kingdom is divided. These three are i) The London Telecommunications Region, ii) The South Western Telecommunications Region (centred on Bristol) and iii) The North Western Telecommunications Region (centred on Manchester).

To ensure a typical selection of routes in a Region, a computer programme was used which analysed details of possible interconnections in that Region, and selected the routes by a partially random process. This ensures that the majority of possible types of routing within the network structures are included in the tests. Two tests per connection (one in each direction) and four connections per route (two set up from each end) are generally made. The number of routes and the number of tests so far conducted in each Region are given below. Typical route distances within Regions would be from 5 to 80 kilometres. The inter-Regional routes were within the range 240 to 320 kilometres.

	Number of routes	Number of tests
London	60	480
South Western	28	224
North Western	28	224
South Western-North Western (inter-region)	18	144
Totals . . .	134	1072

Tests are conducted with the test equipment located in the exchanges in question using standard subscribers' exchange equipment. Local ends of various typical lengths, to represent the known distribution of line lengths, are simulated by the insertion of networks between the test and exchange equipments.

3. *Results*

The full analysis of the measurements taken in this series of tests is a task not yet completed but preliminary results are shown in Figures 1 to 4 and provide some useful information on the characteristics relevant to data transmission of the public switched telephone network. Time has not yet been found to apply weighting factors to account for the distribution of line plant etc. The results cannot be regarded as complete until this weighting process has been carried out.

3.1 *Subscriber to subscriber loss*

Figure 1 shows the cumulative distribution of insertion loss at 1700 Hz for the South Western, North Western and London Telecommunications Regions together with results of tests between the South Western and North Western Regions. The median insertion loss at 1700 Hz is approximately 16 dB and the standard deviation is approximately 4 dB. These figures may be compared with those obtained by the Federal German Administration and described in Supplement No. 14.

Figure 2 shows the envelope formed by sixty of the attenuation/frequency characteristic for London test series. The graph also shows some of the characteristics which define the envelope. The mean attenuation at 1700 Hz has been normalized for this figure. To complete the graphs

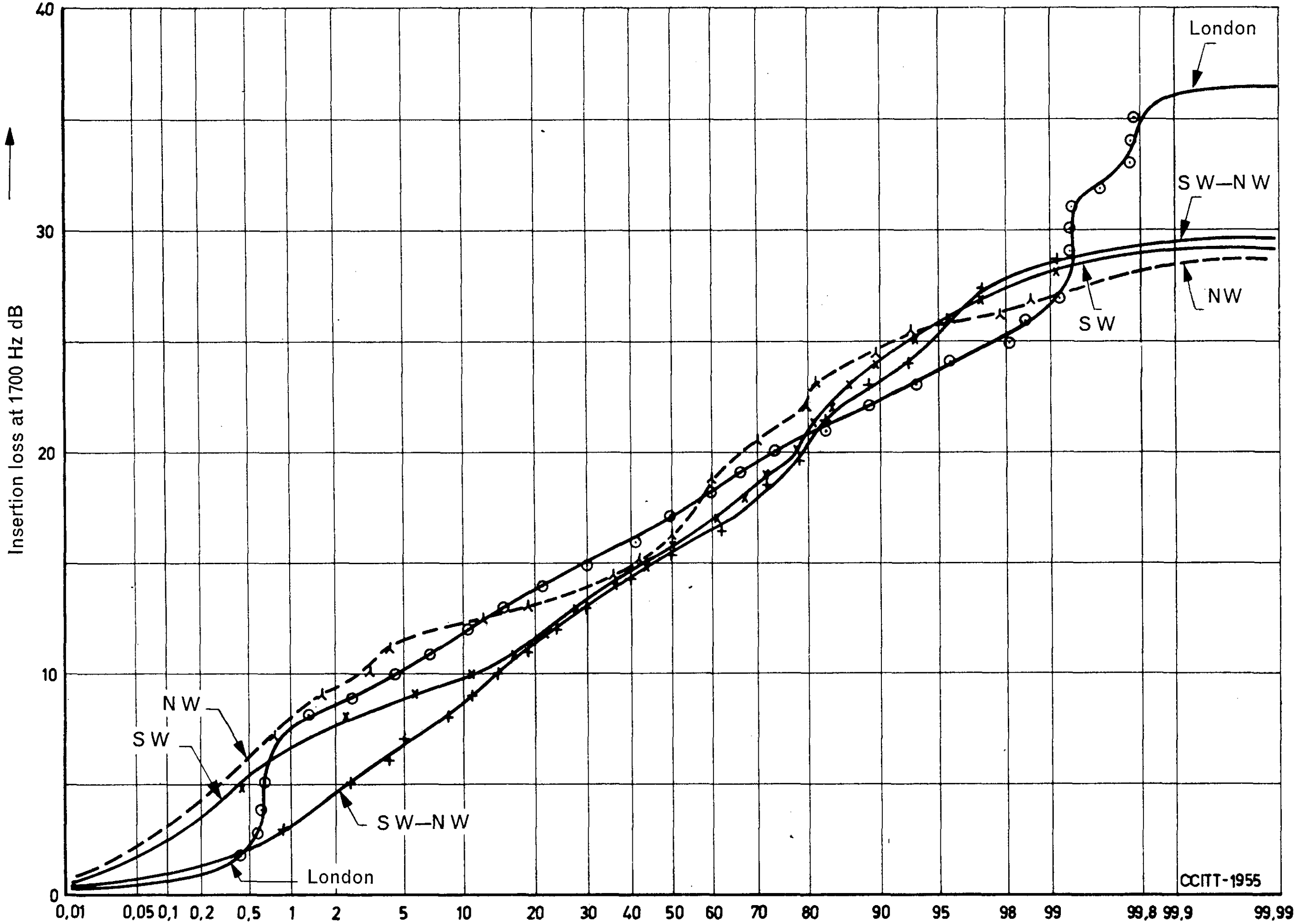


FIGURE 1. — **Percentage of connections having insertion loss at 1700 Hz ≤ value shown**

Cumulative distribution of insertion loss at 1700 Hz for: a) London Region; b) South Western Region; c) North Western Region; d) South Western Region—North Western Region tests

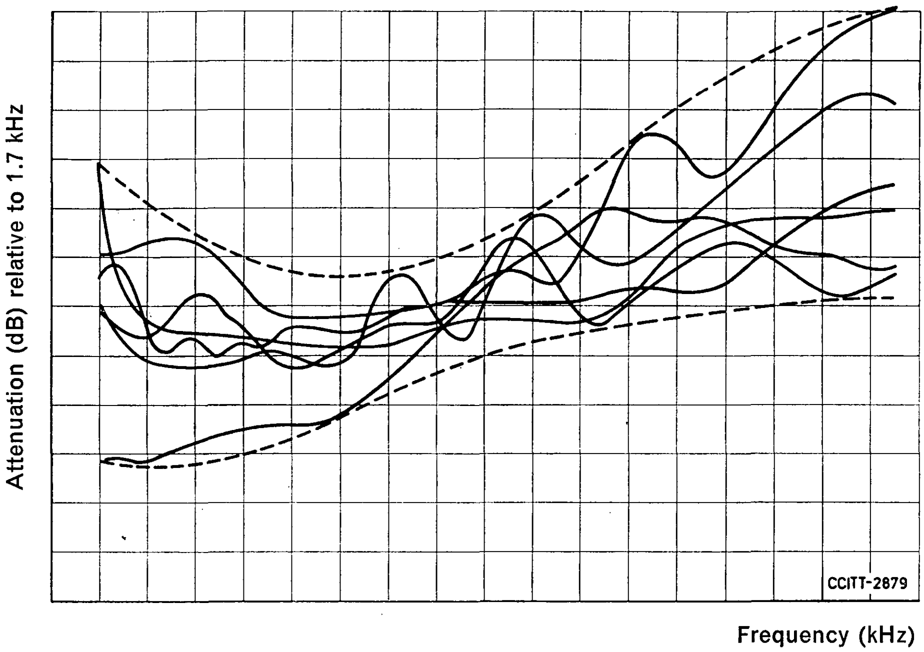


FIGURE 2. — **Typical attenuation characteristic forming envelope (London test series)**

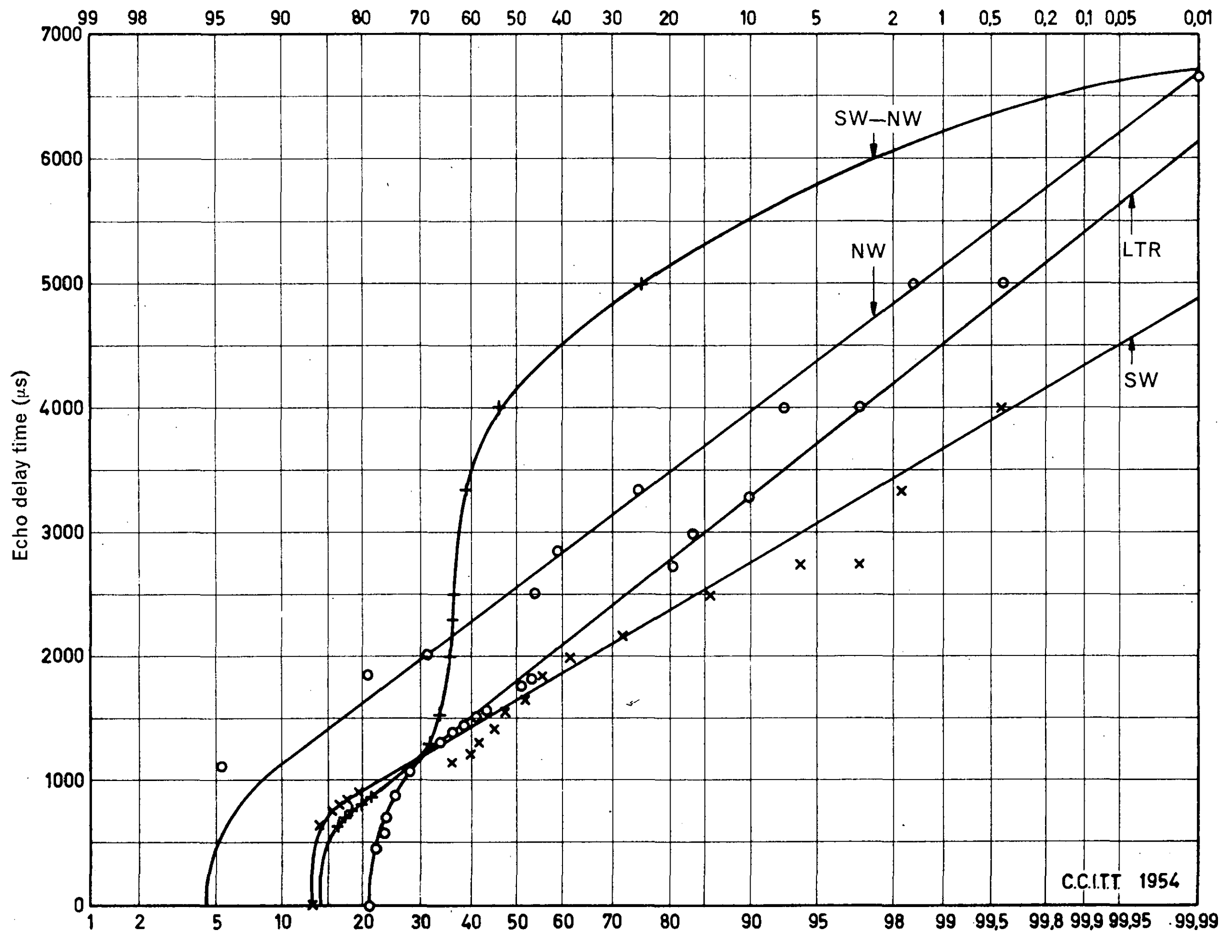
between the discrete tone frequencies, reference is made to the gliding tone measurements used for the echo tests.

3.2 *Echo delay time and echo relative level*

Figure 3 shows the distribution of echo delay time and Figure 4 the echo relative level for the 3 Regions previously mentioned and also for tests between the South Western and North Western Regions. The figures refer only to centre of the frequency band in question, approximately 1200-2300 Hz.

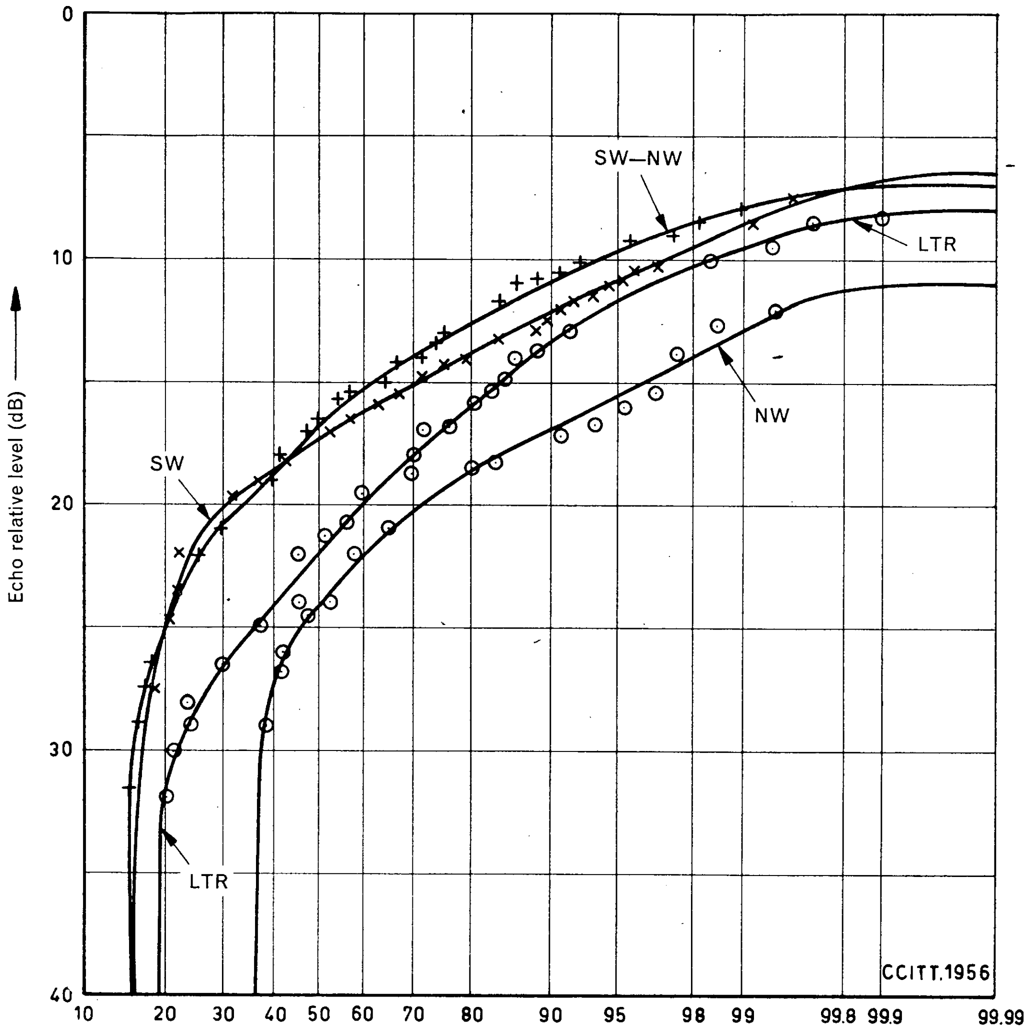
Although no results for phase distortion are yet available these figures are of interest since echo affects both attenuation and phase characteristics. In particular it causes ripples which can become the dominant factor if the echo level is high. System tolerance to echo level will depend on the modulation technique employed. Supplement No. 36 discusses the effect of echo levels on peak individual distortion for a modem conforming to C.C.I.T.T. Recommendation V.23 and it is shown that any signal/echo ratio less than 20 dB will adversely effect performance of this modem in this respect.

It is worthy of note that only 20% of test connections were virtually echo-free, the median value of echo relative level being 20 dB with 10% of tests worse than 12 dB. It would be interesting to observe results obtained from the networks of other administrations.



Cumulative distribution of echo delay time for: a) London Region; b) South Western Region; c) North Western Region; d) South Western Region—North Western Region tests

FIGURE 3. — **Percentage of connections having echo delay time ≤ value shown**



Cumulative distribution of echo relative level for a) London Region
b) South Western Region
c) North Western Region
d) South Western Region—North Western Region tests

FIGURE 4. — **Percentage of connections having echo relative level ≤ value shown**

Supplement No. 33

U.S.S.R. — *(Contribution COM Sp. A—No. 145—October 1967)*

METHODS AND RESULTS OF MEASUREMENTS OF ENVELOPE DELAY/FREQUENCY CHARACTERISTICS OF TELEPHONE CHANNELS

1. At present it is common practice to evaluate the harmful effect of the phase distortion by the irregularity of delay/frequency characteristics. However, it should be borne in mind that this value does not adequately define the harmful effect on the pulse transmission.

It becomes evident from the theory of "paired echoes" that echo-pulse amplitude is dependent on the irregularity of phase characteristic [1] and hence on the value of the irregularity of envelope delay characteristic divided by the number of fluctuations of this characteristic in the passband. Thus, the echo-pulse amplitude which determines the adverse effect of phase distortion is also dependent *on the form* of envelope delay characteristic.

The question of a search for a more accurate parameter should be left under study. The convenience of its use in operation is to be taken into account. For the present in the Soviet Union the irregularity of the envelope delay/frequency characteristic is measured and standardized as recommended in the *Blue Book*, Volume IV.

2. In the U.S.S.R. statistical measurements of envelope delay/frequency characteristics of non-switched carrier frequency telephone channels with different numbers of links were carried out.

The measurements were made with the help of the measuring set employing the Nyquist method.

For a considerable group of channels measurements were carried out to determine:

— average characteristics of a group of channels and their most probable deviation range as well as the characteristic summation law for channels connected in tandem;

— stability of characteristics in time;

— dependence of the influence of envelope delay characteristic on the quality of service.

3. *The averaged characteristic, its deviation range and summation law*

One of the main problems in data transmission system design is the determination of the probable deviation range of envelope delay/frequency characteristics for a group of channels in operation. This problem is particularly urgent in a switched network where a temporary established connection may comprise any one of the channels.

To determine the deviation range a series of tests was made in a considerable group of non-switched telephone channels of 24-channel carrier systems.

Out of 174 channels tested 70 channels had a single link, 32—two, 22—four, 25—six and 25—eight links.

Furthermore, the characteristics of 300 telephone channels set up by means of a loop connection on the channel-translating bay (without using the transmission path) were analysed.

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[1] H.A. Wheeler: The interpretation of amplitude and phase distortion in terms of paired echoes; *Proc. of I.R.E.*, June 1939, Volume 27, No. 6, p. 359-385.

The tests carried out showed that channel filters were the principal elements to determine the envelope delay. Some additional distortion is introduced by group and line filters. This is particularly noticeable in the edge channels of 12-channel groups. Characteristics of these channels strongly differ from those of the rest.

The deviation range of both the filter characteristics and envelope delay characteristics is determined by the production tolerances of filter elements which are of random character. Therefore, it would be natural to assume that the envelope delay at every single frequency is a random value having the Gaussian law of distribution.

Figure 1 represents histograms plotted by using the results of measurements of 300 channels looped at the exchange (shown by dotted lines) and 70 channels having a single link (shown by solid lines). Bearing in mind the insufficiency of the statistical data available the conformity to the Gaussian law of distribution may be admitted to be good.

Choice of the probable deviation range was based on the requirement that 95 per cent of the channels be included within the prescribed limits. Then

$$\Delta\tau_f = \tau_m \pm K\sigma,$$

where $\Delta\tau_f$ = deviation range of envelope delay at a given frequency,

τ_m = mean time value (normal process mathematical expectation),

σ = normal process r.m.s. deviation,

K = factor corresponding to 95% ($k = 2.04$).

Figure 2 shows the averaged envelope delay of single-link channels and permissible deviation range of the envelope delay for various channels.

For n channels (links) connected in tandem, by using the Gaussian law of distribution, the average value of envelope delay may be found

$$\tau_{m \cdot n} = n\tau_m,$$

as well as r.m.s. deviation

$$\sigma_n = \sqrt{n}\sigma$$

for any single frequency.

Figures 3 and 4 indicate the averaged curves and permissible deviation ranges of the envelope delay for four and six links connected in tandem.

The examination showed that 92% of the channels tested having 2, 4, 6 and 8 links were within the deviation ranges determined by the specified methods.

4. *Stability of characteristics*

To determine the stability of envelope delay characteristics within a long period of time the same channels having 1, 2, 4, 6 and 8 links were tested. Measurements were carried out in the frequency range between 0.6 and 3.2 kHz with the measuring set having the accuracy in the order of 20 µs.

The analysis of test results reveals certain instability of characteristics in time. The absolute instability:

$$\Delta t = t_{\max} - t_{\min}$$

during the whole measurement period in the frequency range of 1200-2400 Hz (the operating range at 1200 bauds) does not exceed 80 µs, with the maximum irregularity in this range for six links being equal to:

$$t_{\max} = 800 \ \mu s.$$

Thus, the relative instability $\Delta t/t_{\max}$ is about 0.1. The relative instability in the frequency range of 600-3200 Hz is also approximately equal to 0.1 ($\Delta t < 0.5$ ms; $t_{\max} < 5.5$ ms).

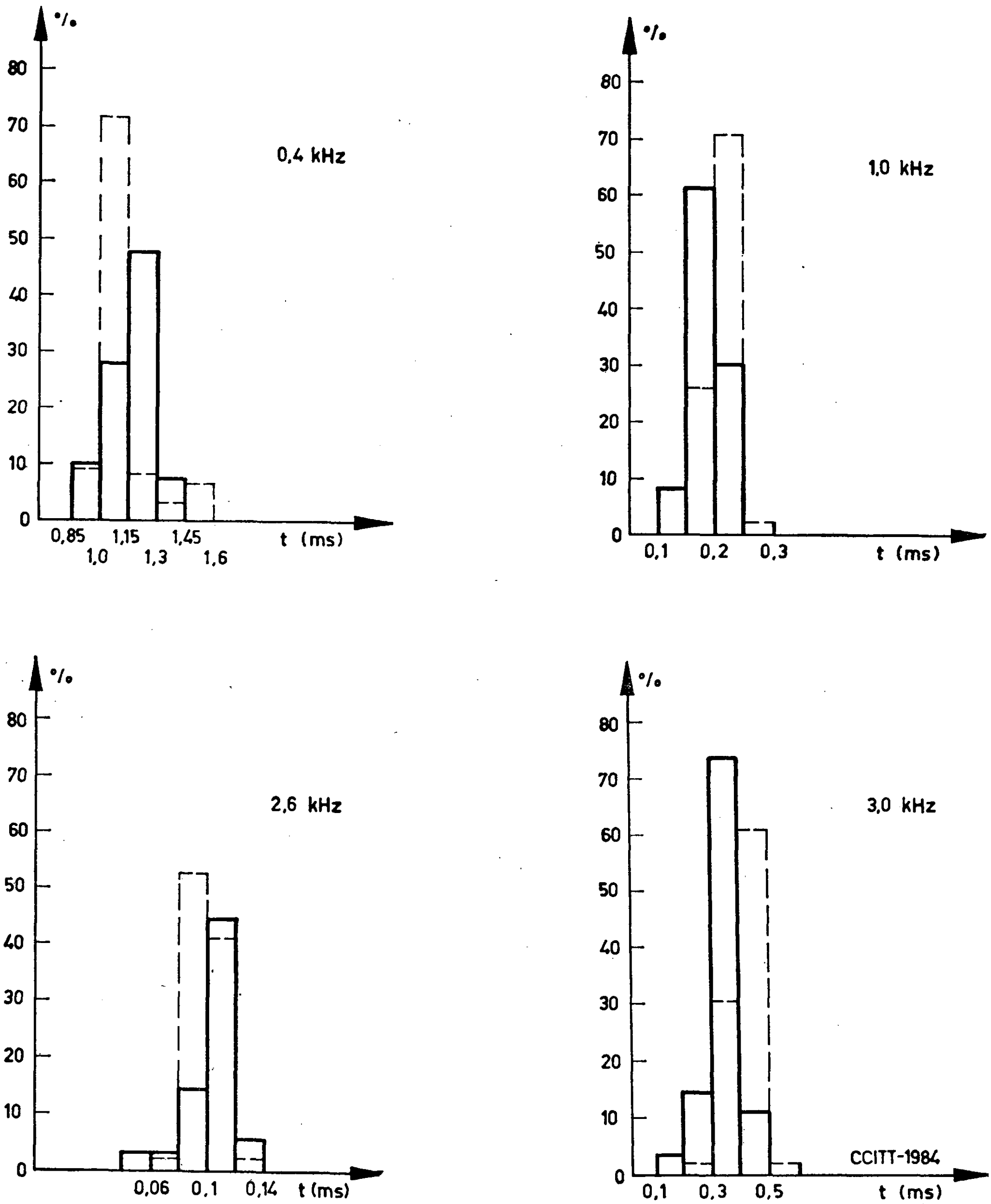
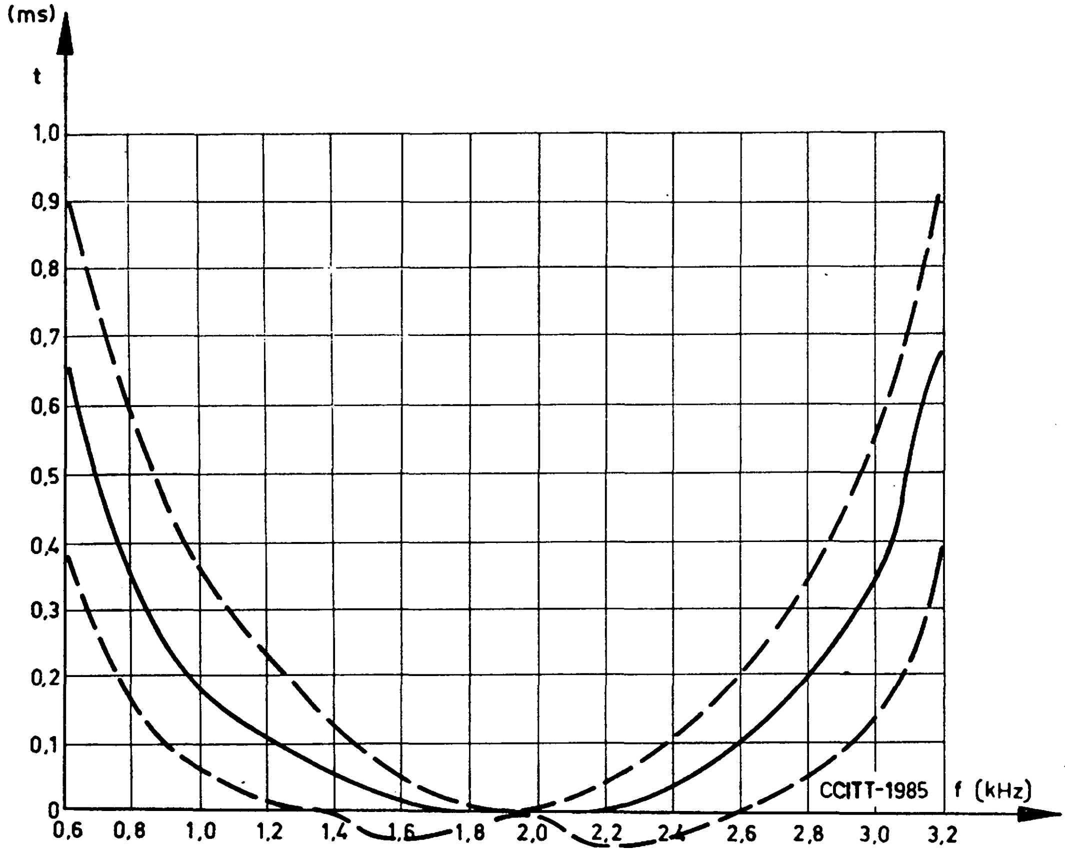
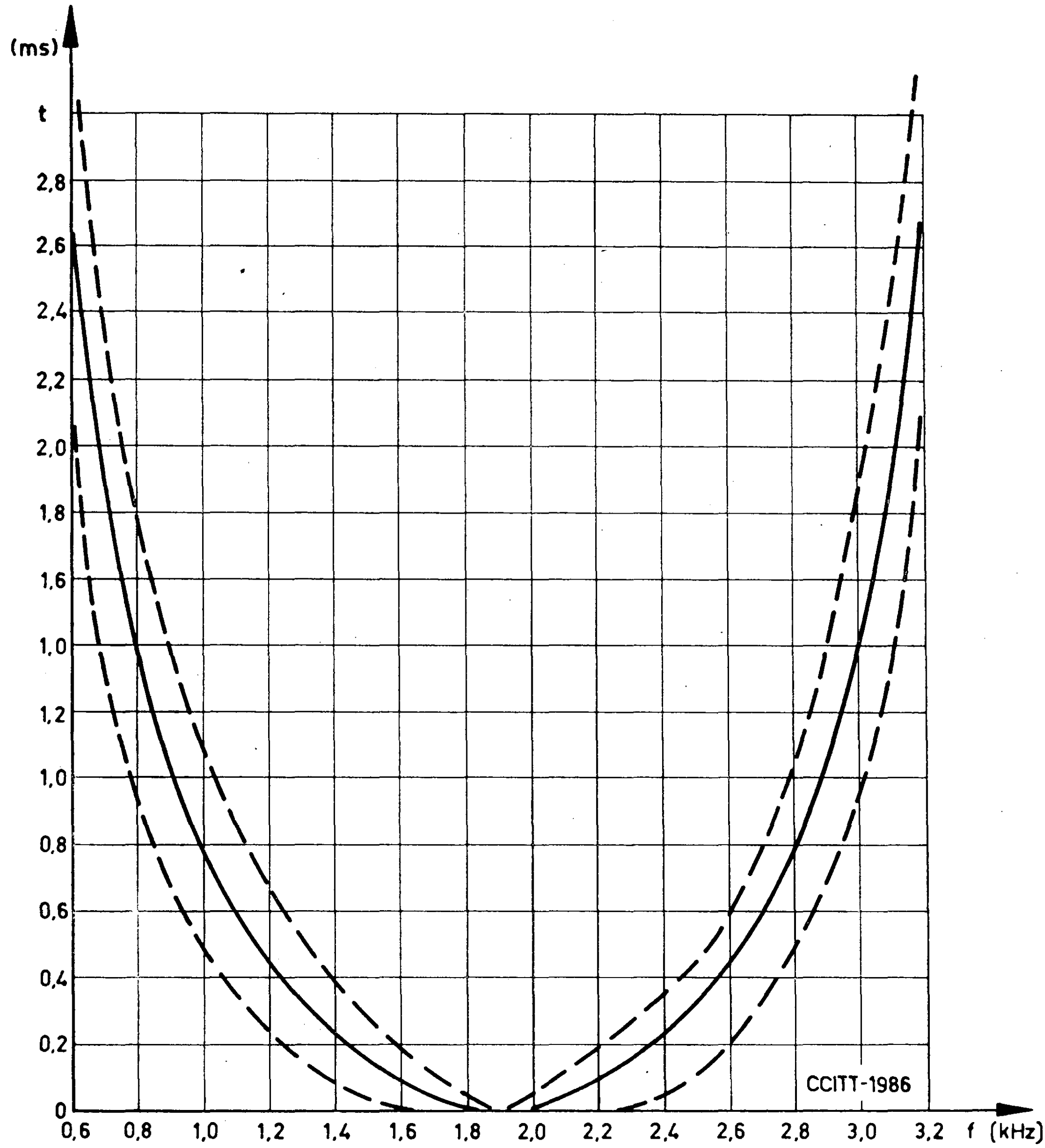


FIGURE 1. — Deviation range of envelope delay at four frequencies



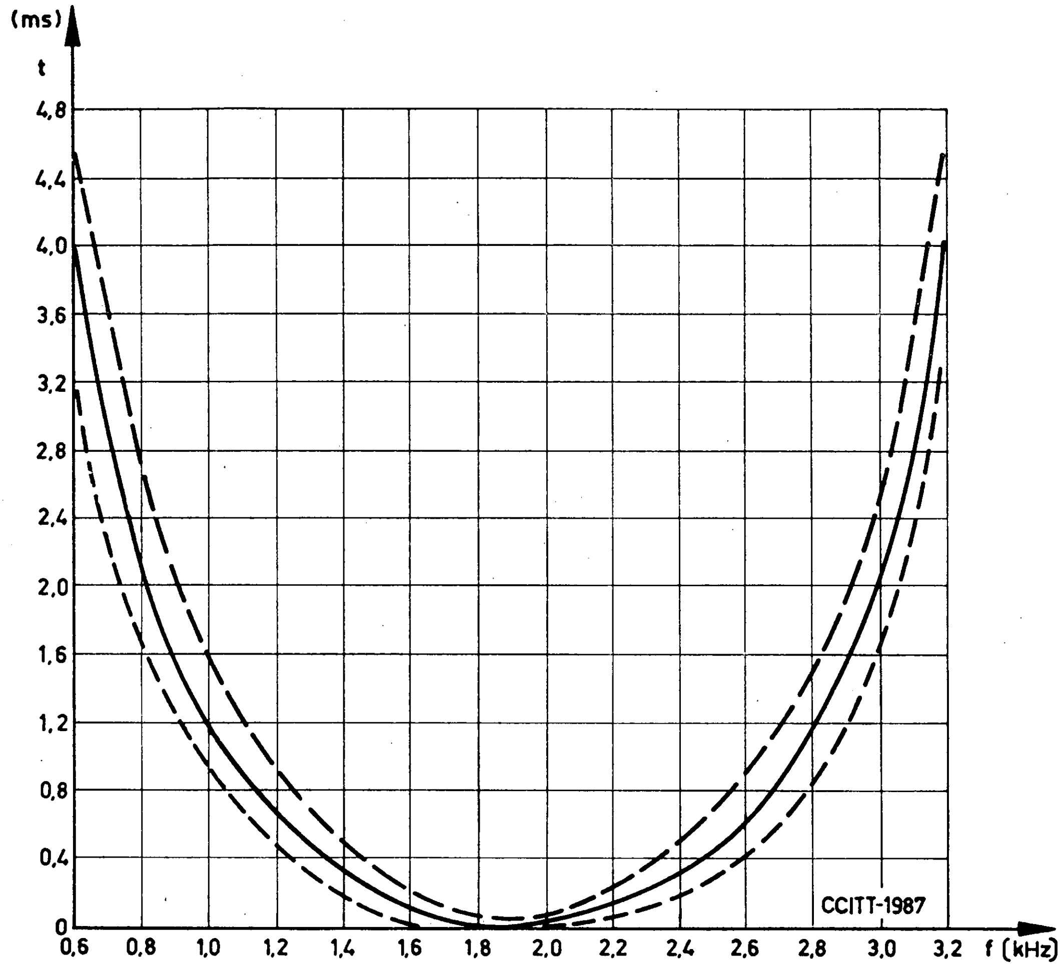
f (kHz)	1,3	1,4	1,6	1,9	2,2	2,4	2,5
t (ms)	0,09	0,06	0	0	0	0,05	0,07
Δt (ms)	0,08	0,06	0,05	0	0,05	0,07	0,08

FIGURE 2. — Deviation range on envelope delay of one link



f (kHz)	1,3	1,4	1,6	1,9	2,2	2,4	2,5
t (ms)	0,35	0,24	0,1	0	0,1	0,2	0,3
Δt (ms)	0,17	0,15	0,1	0	0,1	0,15	0,15

FIGURE 3. — **Deviation range of envelope delay of four links**



f (kHz)	1,3	1,4	1,6	1,9	2,2	2,4	2,5
t (ms)	0,5	0,35	0,15	0	0,15	0,3	0,45
Δt (ms)	0,2	0,15	0,1	0	0,1	0,15	0,2

FIGURE 4. — **Deviation range of envelope delay of six links**

Studies showed that the instability of every channel in time was noticeably less than probable deviation range of envelope delay at the same frequency in a group of channels. Relation between these values does not exceed the amount:

$$\frac{\Delta t}{2\sigma} = 0.25$$

in the frequency range of 1200-2400 Hz and

$$\frac{\Delta t}{2\sigma} = 0.5$$

in the frequency range of 600-3200 Hz.

This is not the case with the equalized channel. Here the absolute instability is approximately the same (since the instability of equalizing devices is approximately zero), whereas envelope delay distortion is greatly reduced. From the above one may conclude that there is no danger of failure of an unequalized channel due to the instability in time and vice versa that it is necessary to design the exact (fine) equalization system so as to ensure sufficiently frequent fine adjustment in the operation to eliminate the effects of instability of channel envelope delay.

5. *Influence of envelope delay on data transmission quality*

Measurements of the influence of envelope delay on data transmission were carried out with the help of data transmission equipment employing the two-phase differential modulation with transmission rate of 1200 bauds.

"Loop around" tests were conducted on the cable routes used for 24-channel carrier telephone system with different numbers of links.

Testing procedure was to measure the value of the flat noise that was to be injected into the channel to obtain bit error rate 10^{-5}. At the same time telegraph distortion of pulses was measured.

Figure 5 shows the measured dependence of difference between signal and noise levels (with bit error rate 10^{-5}) on the number of links. Figure 6 represents the measured dependence of telegraph distortions on the number of links.

As seen from Figure 5, the value of level difference increases from 0.7 Np to 1.5 Np when changing from 1 to 10 links. The particularly rapid impairment takes place when the number of links exceeds 6. When the number of links exceeds 10 the operation becomes impracticable.

One may conclude that the undistorted transmission of digital information at 1200 bauds may be carried out with no additional equalization of phase and frequency distortions in a channel when the number of links does not exceed 6.

Conclusions

1. The statistical measurements of the envelope delay have proved the hypothesis of its distribution at a given frequency according to the Gaussian law which makes it possible to calculate the expected mean value $t_{m \cdot n}$ and the probable deviation range σ_n for n channels connected in tandem.

2. The time instability of the characteristics of a single channel is considerably lower than the probable deviation range in a group of channels. This permits, as far as the non-equalized channel is concerned, to take no account of the characteristic instability in time. The equalization of a channel may be brought to such values that the instability in time be within the permissible tolerances determined by the given signalling rate.

3. The relation between the S/N ratio and the number of links is practically linear only up to a certain limit (at signalling rate of 1200 bauds and with phase modulation this corresponds to approximately six links). Further increase of the number of links leads rapidly to the impairment of noise immunity. Therefore, phase equalization of the channel becomes essential.

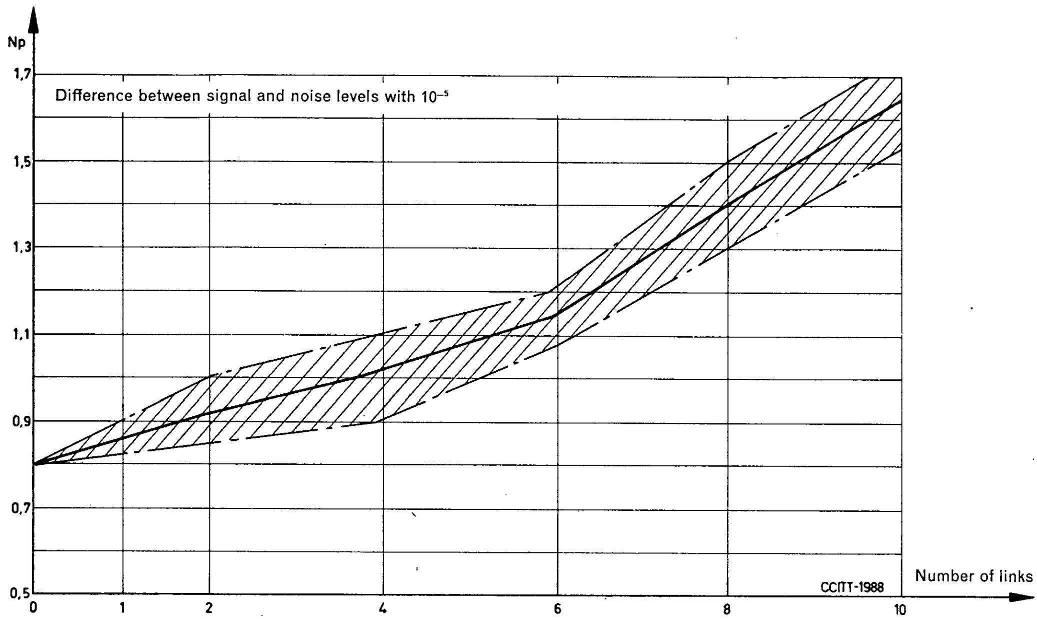


FIGURE 5. — **Effect of number of links on S/N ratio**

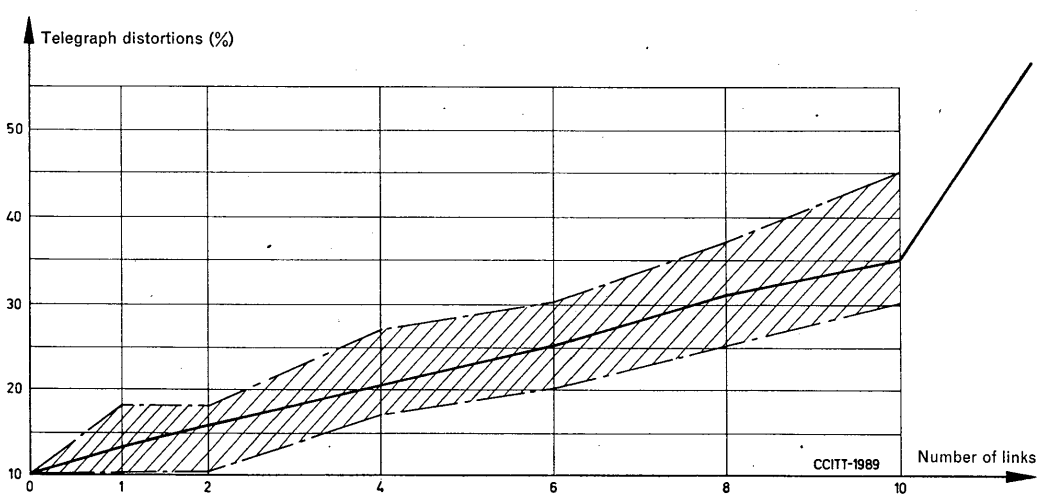


FIGURE 6. — **Effect of number of links on telegraph distortions**

AMERICAN TELEPHONE AND TELEGRAPH Co.—*(Contribution COM Sp. A—No. 18 b—December 1967)*

FUTURE STUDY OF CHARACTERISTICS OF SUBSCRIBER LINES USED FOR DATA TRANSMISSION OVER THE TELEPHONE NETWORK

The subscriber-to-subscriber data transmission performance over the telephone network is affected by the quality of all the lines and trunks which are switched together to form the overall connection. The number of possible alternative routings and combinations of circuits is very large for most calls. However, the line which connects the subscriber to the first switching centre is utilized in every call. Therefore special attention should be given to the data transmission characteristics of this line. Special routing or selection of lines may be needed to provide satisfactory transmission on a sufficiently high percentage of the subscriber's calls.

As a possible guide for the study of the characteristics of subscriber lines, the following has been extracted from the practices used by the Telephone Companies in the U.S.A.

The specific design recommendations are given in Appendix A. Installation measurements are shown in Appendix B. The following paragraphs discuss the requirements in detail. Two different designs are given, one for low-speed (up to 300 b.p.s.) and one for high-speed services.

Insertion loss

The purposes of insertion loss objectives are to fix the data level relative to voice levels and ensure that extremely high loss loops (i.e. subscriber lines) do not cause degradations that would place undue transmission requirements on the network itself. The insertion loss recommendation is 10 dB, maximum. Pads and sensitivity adjustment in the modems permit adjustment for smaller values.

Insertion loss should be calculated. For mixed gauge loaded facilities, an approximation should be made by summing the individual insertion losses or by the exact calculations for insertion loss (manually or computer).

Slope

The upper test frequency is 2750 Hz. The 2750-Hz tone will serve as both an upper test frequency and a compandor holding tone for noise measurements in connection with data services. Test lines and generators are under development. The 2750 Hz was selected primarily to avoid interference with standard SF signalling circuits, and to provide a test tone whose frequency was high enough to permit a good gain slope evaluation.

This upper test frequency applies only to high speed. If 1000 Hz actual measured loss (AML) agrees with the calculated expected measured loss (EML), it is reasonably safe to assume that the f_2 losses (2200 Hz) will be acceptable. The few trouble cases where this is not true do not justify the expense of making the measurement on each installation. The 2750-Hz EML should be recorded on all circuit layout cards for plant use in trouble analysis, but note, *no* slope requirement applies for low speed.

Slope measurements are required for high-speed operation. These are particularly important for the 202-type data sets (comparable to V.23 modems), but not as important to the 201 (4-phase) data sets which operate over wide variations in other transmission parameters, almost independently of slope.

Impulse noise

Impulse noise objectives have been relaxed. Although it is possible to relax low-speed objectives 1-3 dB more than high speed, it was felt that more experience should be gained with the new objectives and measurement procedures before any distinction is made. Additional studies by Bell Laboratories would also be required to determine the exact amount of relaxation possible.

The operation with the new objectives will be as follows. If 15 or less counts are observed in 15 minutes at 60 dBrn0VB, the requirements are met. If over 15 counts are observed, make another 15-minute measurement of the loop only. If the objectives are not met, correct or reassign the facilities or equipment. If the loop measurements are good, the service through the office is still questionable and end-to-end error tests using the 900-series data test set are mandatory. If the error tests are good, the service can be given to the customer. If not, verify that the trouble is caused by the office and provide RX lines as required.

All measurements over 15 counts to the quiet termination should be sent to the appropriate engineering office to aid in determining where mitigation or survey work should begin. If over 100 counts are recorded, the office is almost certainly in trouble and would be unsuitable for general Data-phone service. This does not mean that particular installations which are successful should be removed or that new installations cannot be made, but only that an investigation of the causes should be made at an early date.

All of the measurements are to be made with the voiceband (*vb*) or C-message weighting filter [1]. The narrowband filters are no longer required, as studies show they provided little additional information about the condition of the circuit under test. Measurements should be made during normal day or early evening working hours; it is no longer necessary to restrict the measurements to the busy hour.

Measurements are to be made to the quiet termination in the local central office. Within the central office, each path will exhibit a different number of counts at the counting levels used in the objectives. In other words, the office will have a distribution with some paths having higher numbers of impulses than others. If the contribution by the facilities is assumed to be constant (and it is over long periods of time—15 minutes and longer), then the variation in counts registered

[1] The C-message weighted filter is used in the 6F and 6H impulse counters and attenuates 1 dB more than the VB filter so the objectives must be reduced by 1 dB.

during any one measurement interval on an individual loop will depend upon the intra-office path of the connection to the quiet termination.

Since some doubt will still exist if the measurements fall in the grey area (15-100 counts in 5 minutes), the logical course of action is to determine the effect on service (which naturally is the purpose of the objective in the first place). This is done by making the end-to-end error tests.

Return loss

In recent weeks, the Bell Laboratories have studied return loss requirements. The tests indicate that the 12-dB first listener echo requirement for data sets is still valid. However, end-to-end simulation combining loops and trunks confirms that the 12-dB first listener echo requirement can be met without additional loop treatment. Therefore, there will no longer be a specific return loss requirement for Data-phone loops. Return loss is still an important parameter, especially to high-speed data transmission, but the troubles are usually isolated to trunks, improperly installed repeaters or poorly balanced hybrids and the like, rather than to the loop facilities.

Message circuit noise

No special Data-phone loop requirements are needed for message circuit noise.

Envelope delay distortion

Only the high-speed (above 300 bps) loop design has an envelope delay requirement. Computer model simulation has shown this to be an important and sensitive parameter. Companies should carefully check that computations are made on each loop and corrections made where required.

Transmitting levels

The transmitting levels at the serving office should be met as closely as possible. In cases of doubt, use the next lower outputs, wherever possible. These levels are required to prevent serious overloading of radio and other systems as data services grow. Basic design calls for a − 16 dBm0, e.g., − 32 dBm at the MOD-IN of standard − 16, + 7 carrier systems, loading in a voice channel. Half-duplex signals could statistically operate at − 13 dBm0 and full duplex at − 16 dBm0. The use of reverse channels and frequency diversity such as is found in the 100 Series of data sets (comparable to V.21 modems) has increased the carrier loading problem. As noise levels and system improvements permit, the data level will be lowered toward the long-range objectives.

Remote exchange (RX) design

Whenever the local office or serving trunks are unsuitable to provide data service as indicated by actual end-to-end measurements with the 900 Series data test sets, and by transmission measurements, a remote exchange line or special exchange code is required. This bypasses the normal routing or normal office. (Note to Sp. S.G.A.: This situation has very rarely occurred.)

In the past, the remote exchange line was designed as if it were a local loop. Now, the assumption will be made that the remote exchange line will include the distortion of a toll connecting trunk. The RX routing should take the customer to an office that will provide better network access from a transmission point of view. If the toll connecting trunk distortion is not bypassed by the RX line, then local loop design rules prevail.

Foreign exchange (FX)

Foreign exchange (FX) poses a very difficult problem. It is difficult to design an FX that may be hundreds of miles long to Data-phone loop design criteria. However, if the local design criteria are not met, Data-phone service may not be successful to all stations in the network.

Installation measurement

Appendix B lists the installation measurements required for Data-phone. For low speed, only 1000 Hz loss and message circuit noise are required. For high speed, impulse noise and 2750-Hz loss measurements are specified in addition. This should greatly simplify installation procedures. In case of trouble, the more complete objectives found in Appendix A should be met.

Miscellaneous considerations

A few miscellaneous items should also be mentioned.

— Touch-tone installations require no special tests or handling. They are to be installed as any ordinary business or residence telephones. In SxS office areas, polarity guards will be required to overcome battery reversals that prevent end-to-end signalling after connection is established.

— Behind-the-PBX installations should still be discouraged, especially where high-speed operation is involved. Error performance will generally not be as good if the network is accessed through a PBX, due to impulse noise in the PBX.

— Where one location is receiving from many locations throughout the country, make certain the loop design to this station is as good as economically possible.

— Each Telephone Company should maintain proper records of loop design and the measurements made. Circuit layout cards on all data installations should be issued and adequate files maintained in both Engineering and Plant offices. Failure to keep proper records is one of the most serious data and special services problems.

APPENDIX A

Station loop objectives

	Type data set (see below)	
	Low-speed	High-speed
Max. insertion loss in dB		
1000 Hz	10.0	10.0
2750 Hz	not critical	13.0
Max. slope	not critical	3.0 dB
Impulse noise—no more than 15 counts in 15 minutes at:		
— Non-compandored or compandored facilities with – 10 dBm0 holding tone	60 dBrn0VB [1]	60 dBrn0VB
— Compandored facilities N1 or ON with expandor disabled	40 dBrn0VB	40 dBrn0VB
— N2 and N3 at the DODG jacks (–7.5 TLP)	60 dBrnVB	60 dBrnVB
Message circuit noise: Meet voice objectives		
Envelope delay distortion 1000-2400 Hz	not critical 1	100 microseconds 2
Transmitting level at serving central office in dBm	100 series F1 –15 ± 1 F2 –10 ± 1 Other than 100 series – 10	– 10
FX, RX	*RX* (served from other than the local office)	*FX*
Attenuation distortion 1000-2750 Hz	5.0 dB	6.0 dB
Envelope delay distortion 1000-2400 Hz	300 microseconds	600 microseconds
Impulse noise: 15 counts in 15 minutes at 69 dBrn0VB use a – 10 dBm0 holding tone if compandored carrier is in the layout.		

[1] If C-message weighting is used, reduce all objectives by 1 dB.

APPENDIX B

Data-phone station loop installation measurements

The Data-phone and TWX loop requirements were included in the previous table. On installation only the following measurements need be made. In cases of trouble, Appendix A lists the additional parameters which should be measured as indicated by the type of trouble.

	Type data set [1]	
	Low-speed	High-speed
Insertion loss:		
1000 Hz	YES	YES
2750 Hz	NO	YES
Impulse noise [2]	NO	YES
Return loss	NO	NO
Message circuit noise	YES	YES
Envelope delay distortion	NO	NO

Supplement No. 35

SIEMENS & HALSKE. — *(Contribution COM Sp. A—No. 59—November 1965)*

TELEGRAPH DISTORTION CAUSED BY ATTENUATION DISTORTION AND ENVELOPE DELAY DISTORTION

1. *Basic considerations*

With noise voltages disregarded, the distortion of the shape of a data signal is caused chiefly by attenuation and delay distortions.

Signal shape distortion mostly involves telegraph distortion, which must not exceed a certain limit if transmission errors are to be avoided. Attenuation and delay distortions thus set a limit to the maximum possible modulation rate.

Siemens & Halske's laboratories have conducted measurements in order to study the effects of attenuation and delay distortion on telegraph distortion as a function of modulation rate.

2. *Test conditions*

2.1 *Transmission path*

Several typical telephone-type circuits commonly used in the German postal telephone network (such as non-loaded local circuits, circuits comprising carrier sections, and combinations of local

[1] Type 2 measurements should be made for FX or RX.

[2] TLP corrections are to be made using the average of the 1000 Hz and 2750 Hz AML readings.

circuits and carrier sections) were simulated (cf. Supplement No. 28). The attenuation and delay distortions, being the characteristic quantities of these circuits, were measured.

2.2 *Measurement of telegraph distortion*

Isochronous telegraph distortion [1] of test signals transmitted in the frequency-modulation mode in accordance with Recommendation V.23 (1300-2100 Hz) was measured. In order to eliminate the bias distortion, the receiver was adjusted for minimum telegraph distortion. The period of observation covered about 10 seconds.

2.3 *Test signal*

The test signals used include both the text as recommended in R.51 and also a pseudo-random text with a block length of 1024 bits. Both yielded the same test results.

3. *Test results*

Isochronous telegraph distortion δ as a function of the modulation rate v is shown in Figures 1, 2 and 3. In the lower part of these graphs the characteristics of the circuits are shown. Figure 1 indicates the telegraph distortion with increasing attenuation distortion, the latter having the characteristics of non-loaded cables used in local telephone circuits. Its influence on the telegraph distortion is very small.

Figure 2 shows the telegraph distortion resulting primarily from the delay distortion, the latter having the characteristics of 0, 3 or 5 carrier sections with short lengths of non-loaded local circuits in tandem. As expected, the telegraph distortion increases with the increase of delay distortions. But there is a maximum of telegraph distortion in the range from about 600 to 1200 bauds, which also increases considerably with the increase of delay distortions. This maximum is caused by the overshoot oscillations resulting from the delay distortions of the circuit.

Figure 3 represents the telegraph distortions caused by circuits comprising considerable lengths of non-loaded local cable and 3 carrier sections. It shows that attenuation distortion gives rise to additional telegraph distortion, provided that there are also delay distortions.

These test results may differ to some extent depending on the modem involved.

4. *Conclusions*

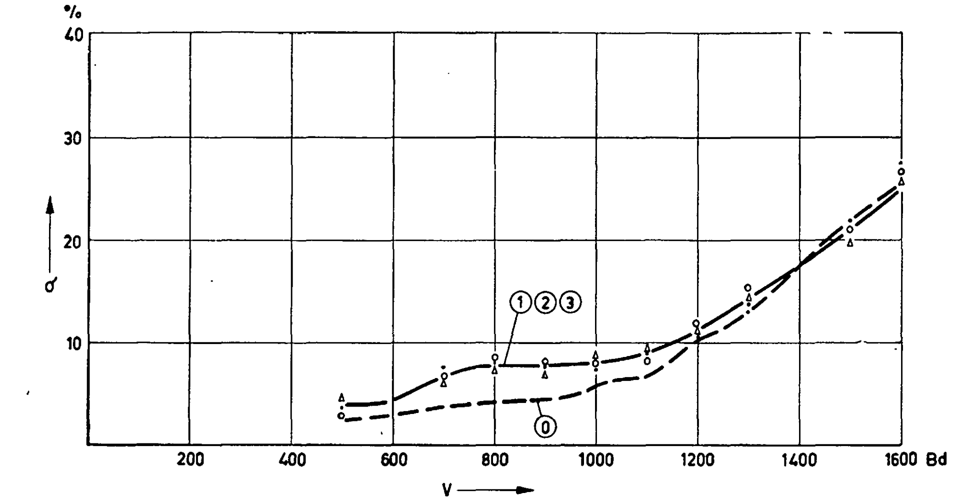
Re: Question U

In studying Question U, it would be appropriate to consider the effects of the attenuation and delay distortions as described above.

Re: Question W, Points 1 (Isochronous or start-stop transmission) and 2 (Modulation rates)

A modem which corresponds to the C.C.I.T.T. Recommendation V.23 should work with modulation rates of *up to* 600-1200 bauds. Besides it should permit both synchronous and asyn-

[1] See *List of definitions of essential telecommunications terms*, Part I, 1961, definition 33.07.

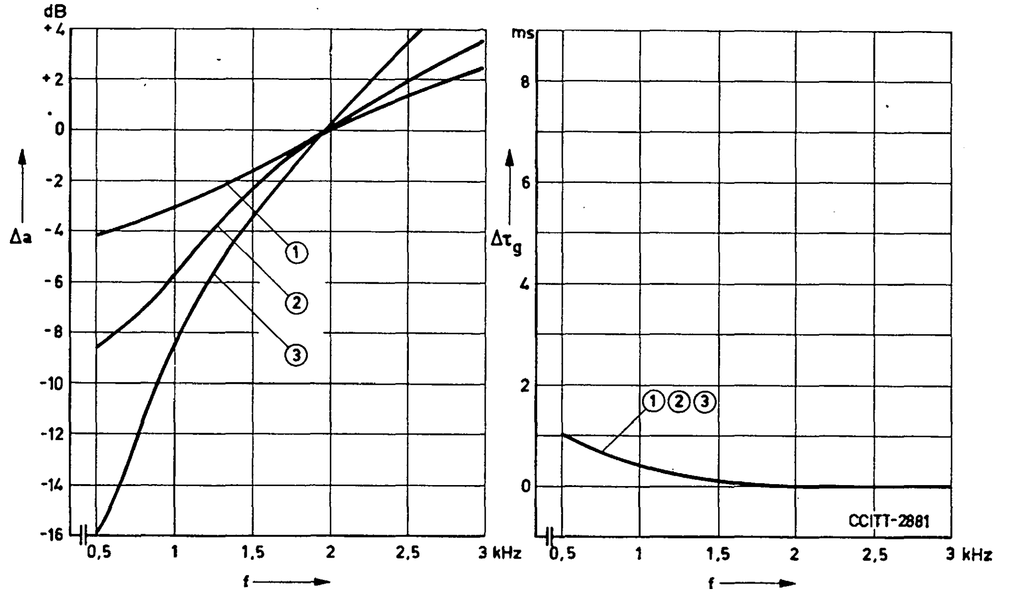


Isochronous telegraph distortion (σ) as a function of modulation rate (v)

Modem: FM (1700±400) Hz

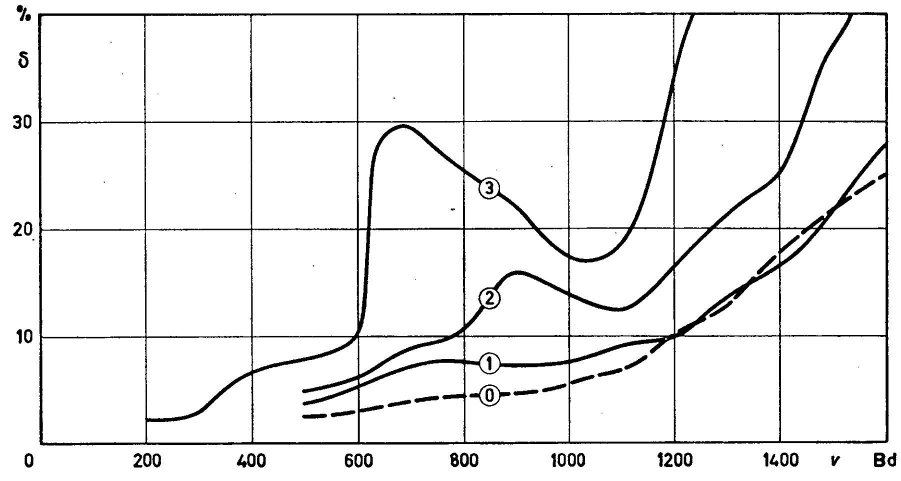
⓪ without line

①②③ non-loaded local lines of different length



Attenuation distortion Δa *(f)* and envelope delay distortion $\Delta \tau g$ *(f)* of the lines measured

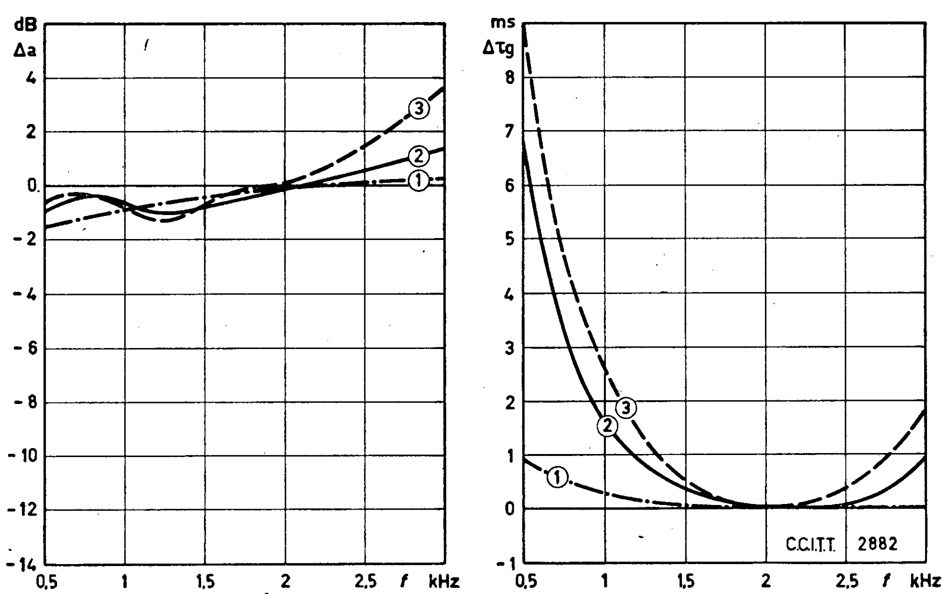
FIGURE 1. — **Effect of attenuation distortion**



Isochronous telegraph distortion (σ) as a function of modulation rate (v)

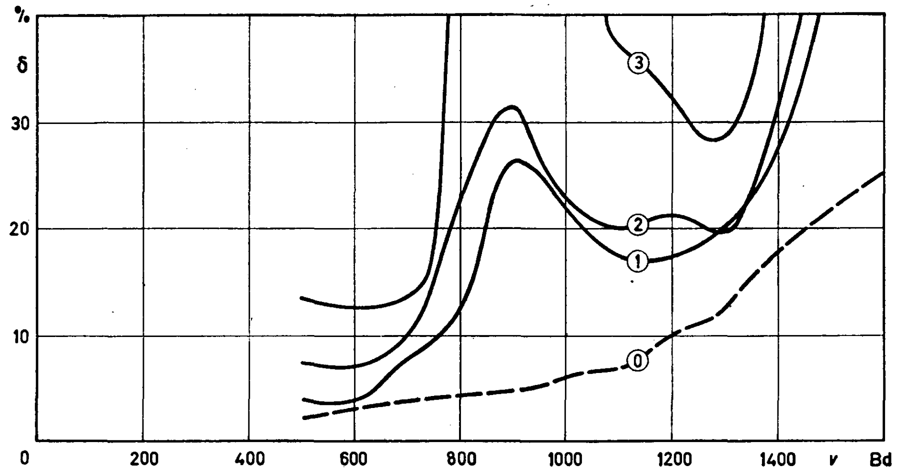
Modem: FM (1700±400) Hz

⓪ Without line
① Short non-loaded local lines without carrier section
② Short non-loaded local lines with three carrier sections
③ Short non-loaded local lines with five carrier sections



Attenuation distortion $\Delta\, a\, (f)$ and envelope delay distortion $\Delta\, \tau g\, (f)$ of the lines measured

FIGURE 2. — **Effect of envelope delay distortion**

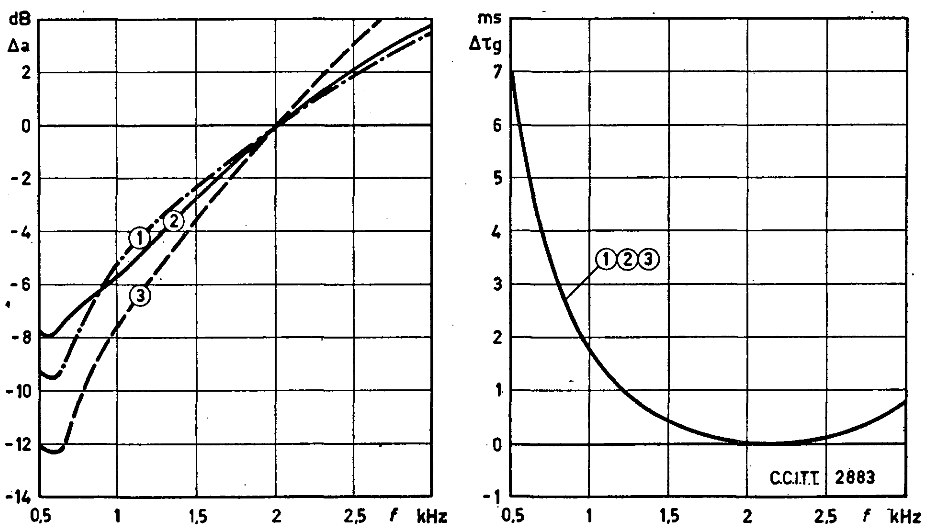


Isochronous telegraph distortion (σ) as a function of modulation rate (ν)

Modem: FM (1700±400 Hz)

⓪ without line

①
② longer non-loaded local lines with three carrier sections
③



Attenuation distortion *Δ a (f)* and envelope delay distortion *Δ* τg *(f)* of the lines measured

FIGURE 3. — **Combined effects of attenuation and envelope delay distortion**

chronous transmission. If this latitude in the choice of transmission mode and modulation rate is to be retained, it would be appropriate not to confine the tests to the maximum modulation rates of 600 and 1200 bauds.

Re: Question W, point 6 (Telegraph distortion measurement)

The error rate is indeed the essential criterion for the quality of a transmission system (modem + line) as far as the user of data transmission equipment is concerned. It should be borne in mind, however, that bit errors are exclusively the result of noise impairment, as long as the attenuation and delay distortion of the circuit at all permit data transmission at a given modulation rate. It would certainly be possible to arrive at a generally applicable assessment of the quality of a transmission system if the measurement of error rate were carried out at various noise levels. This method is, however, very time-consuming. In contrast to this—considering that the telegraph distortion is dependent on attenuation and delay distortion—the measurement of telegraph distortion will very quickly yield a qualitative picture of the essential characteristics of the transmission system (modem + line). The degree of telegraph distortion thus obtained is, amongst other factors, a measure of the permissible signal-to-noise ratio, as the noise voltages give rise to additional telegraph distortion which is superimposed on the basic distortion produced by the attenuation and delay distortion of the circuit itself.

The relationship between the telegraph distortion and the occurrence of bit errors can only be analysed more accurately if further tests are made under actual service conditions.

If a satisfactory relationship can be found out, the administrations could derive benefits for their maintenance services from the measurement of unit element distortion because—and this is only one of the reasons—the relevant measuring equipment has already been adopted by other service branches.

Supplement No. 36

UNITED KINGDOM. — *(Extract from Contribution COM Sp. A—No. 93—January 1967)*

MAINTENANCE METHODS

PART I

. . . .

4.2 *Modem design*

Concerning modem design, the United Kingdom standardized modems have specified maximum "back-to-back" isochronous distortions, but for the purpose of easy comparison in Table I it is assumed that these specified figures can be divided by two and represented by peak individual distortion. These modems have also been tested on artificial networks representing a) three telephone carrier channels connected in tandem and b) 100 miles of standard loaded cable (20/88/1.136). These distortion figures are all shown in Table I.

TABLE I

Peak individual distortion performance of the Datel modems Nos. 1 and 2

Modem	Transmission rate (bauds)	Peak individual distortion (%)		
		Back-to-back (maximum)	Three tandem carrier channels	100 miles loaded cable
No. 1 (V.23)	75	3	5	4
	600	4	5	5
	1200	8	11	17
No. 2 (V.21)	200 (1080 Hz)	4	3	3
	200 (1750 Hz)	4	5	4

Regarding the distortion which is met in practice the Datel modem No. 1 was tested over 14 long-distance and five short (within London) routes on 50 connections via the switched network. At 600 bits/second the maximum individual distortion reading was 25% and the average for all connections was 10%, and at 75 bits/second the corresponding readings were 10% and 5% respectively. The 1200 bits/second transmission rate is not guaranteed on this network, but only one of these connections was unworkable at this rate and the others gave distortion values of 35% maximum and 18% average. Experience with the Datel modem No. 2 is more limited, but a series of 15 connections gave 6% maximum and 3½% average and 13% maximum and 6% average for channels 1 and 2 respectively.

When international data links are set up on the general switched telephone network it may be assumed that the international line characteristics will fall within the limits given by Recommendation M.61. However at the two-wire to four-wire conversion points it is not possible to completely dation M.61. However at the 2-wire to 4-wire conversion points it is not possible to completely match the line impedances and this produces echoes. The effect of echoes is to produce ripples in both the attenuation and group delay characteristics of the line and these ripples can have a marked influence on the peak individual distortion. In a series of measurements on a single telephone carrier circuit the following results were obtained, using the Datel modem No. 1 at 600 and 1200 bits/second.

TABLE II

Individual distortion produced by various signal/echo ratios

Signal to first listener echo/signal ratio (dB)	Peak individual distortion (%)	
	600 bits/s	1200 bits/s
25	6	9
20	6	10
15	7½	14
10	13	25
7½	20	34
6	50	50

Thus it can be seen that in these tests any signal/echo ratio less than 20 dB affected the individual distortion.

In further tests on unidirectional circuits in which the circuit characteristics may not have remained within the M.61 limits for the severe cases, the degree of individual distortion, using Datel modem No. 1, was found to be as shown in Tables III and IV. No noticeable echo was present in either case.

TABLE III

Peak individual distortion produced by transmission over standard loaded cable (20/88/1.136)

Cable length (miles)	Peak individual distortion (%)	
	600 bits/s	1200 bits/s
0	4	7
50	4	9
100	5	17
150	7	50
300	13	—
400	11	—

TABLE IV

Peak individual distortion produced by tandem telephone carrier channels

Number of tandem carrier channels	Peak individual distortion (%)	
	600 bits/s	1200 bits/s
1	4	7
2	5	8
4	6	15
6	5	39
10	7	—
12	15	—
14	20	—

4.3 *Leased lines*

Referring to the case of leased lines, and bearing in mind that on ordinary quality lines a transmission capability of 1200 bits/second is desirable, Tables III and IV suggest that the distortion will rise undesirably at this rate if more than four tandem carrier channels or 100 miles of standard loaded cable are included in the circuit.

A further case of leased lines which requires study is multipoint circuits. The United Kingdom Administration intends to conduct tests on this type of circuit, but no results are yet available.

In consideration of the foregoing, the United Kingdom Administration suggests that no distortion limits shall be made for the present, but that target figures of 30% at 600 bits/s on the general switched telephone network and at 1200 bits/s on point-to-point leased lines be provisionally adopted. The United Kingdom proposes that the study of distortion limits should continue.

PART II

Error performance of V.23 modems compared to impulse noise measurements

1. *Introduction*

In consideration of this question, the United Kingdom Administration has instituted a series of tests to find any correlation that may exist between the error performance of modems to Recommendations V.21 and V.23 and impulse noise as measured by the impulse noise measuring instrument proposed by Special Study Group C (cf. Rec. V.55). This contribution gives preliminary results obtained with the Datel modem No. 1 (to V.23), using line simulating networks and a recorded sample of impulse noise. The tests will continue with field measurements which will necessarily take considerable time to complete. The full results will therefore be published at a later date.

It is hoped that a further contribution will be made reporting the results obtained in a similar series of tests with the Datel modem No. 2 (to V.21).

2. *The sample of noise*

The noise sample used for these tests was a 10-minute tape recording consisting of two periods of 4 minutes each and a 2-minute period, each being a selected noise except from a different long-distant automatic connection of one hour duration on the general switched telephone network. A reference tone was recorded before the noise samples were taken, and using this reference the noise was held throughout all these tests to the levels which occurred when the recordings were made.

This noise sample is conveniently described by impulse counts obtained for various threshold settings as shown in Figure 1. The C.C.I.T.T. specified filter (see Recommendation V.55) included in the impulse counter and another included filter corresponding to the backward data channel were used for these measurements, and also the receive filter of the Datel modem No. 1A forward or backward channel together with no filter (i.e. the flat bandwidth condition) in the impulse counter. The bandwidths of the impulse counter filters to the — 3 dB points were 600-3000 Hz and 300-500 Hz respectively and those of the Datel modem No. 1 filters were 950-2850 Hz and 350-490 Hz respectively. The modem filters are thus somewhat narrower than those in the impulse counter and Figure 1 shows that, as expected, the narrower bandwidths gave the lower impulse counts.

The forward channel filters have approximately ten times the bandwidth of their respective backward channel filters, and if uniform spectrum noise had been present, the curves obtained on the two channels would be about 10 dB apart. This is the case for counts in excess of 100, but for counts below this value, the curves for the forward and backward channels approach one another. This may indicate that in this noise sample the low level impulse noise has an approximately uniform spectrum, but that the high level impulses have most of their energy concentrated at the lower end of the audio band.

3. *Simulating networks and test procedures*

The signals transmitted from a modem were passed through a frequency-flat attenuator and a line simulating network and the impulse noise was added to the signals at a point between the simulating network and the receiving modem. All level measurements and impulse counts were conducted at the line terminals of the receiving modem. The noise level was kept constant and

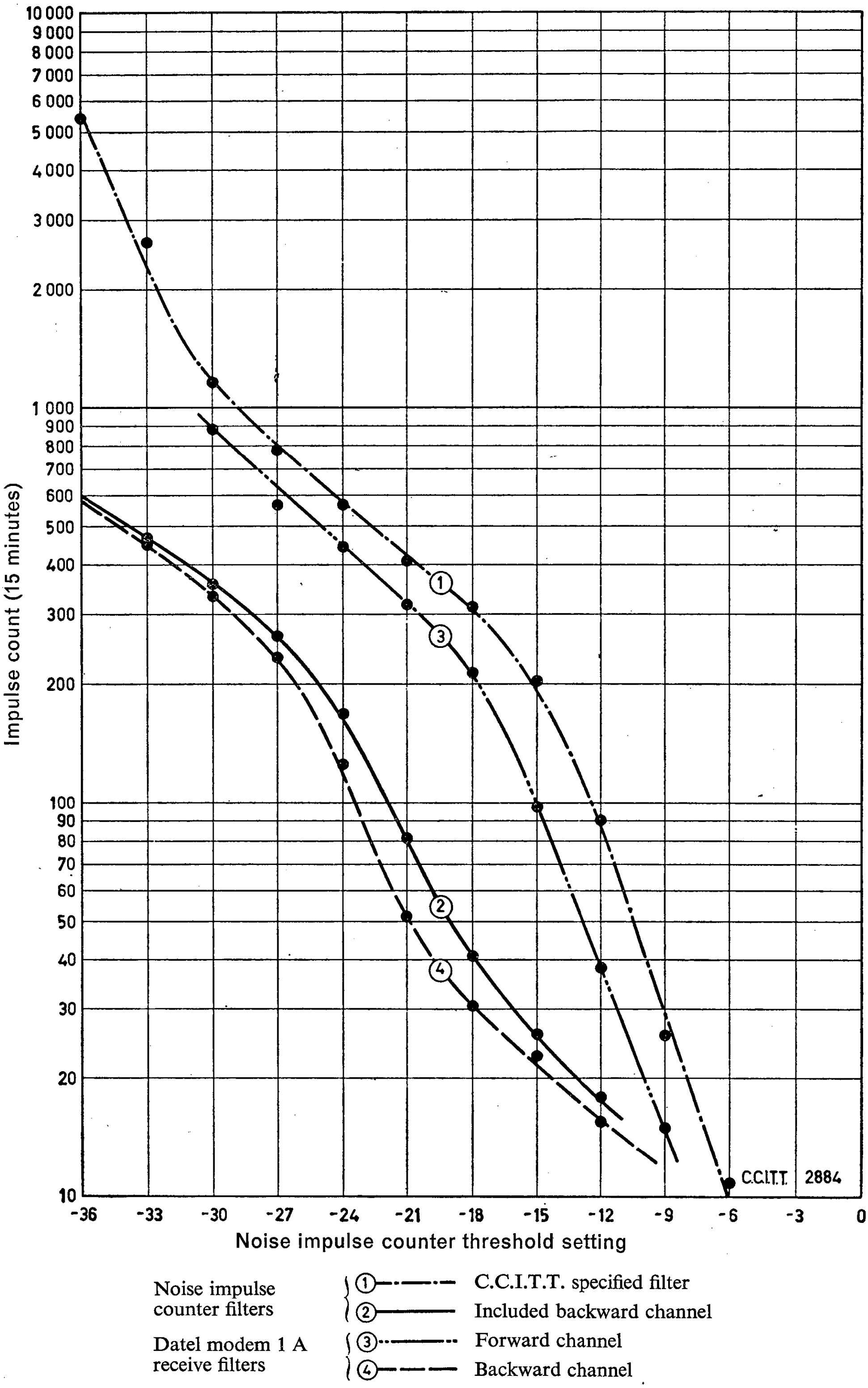


FIGURE 1. — **The sample of noise—impulse noise count versus impulse counter threshold**

only the signal level was varied. The errors were counted using the tester and 511-bit pattern (cf. Rec. V.52). The line simulating networks represented:

a) three telephone carrier channels connected in tandem, and

b) 100 miles of standard loaded cable (20/88/1.136). In addition, tests were also conducted using only the frequency-flat attenuator. On Figures 2-7, the results obtained using these have been marked "Na", "Nb" and "Flat" respectively.

All tests were carried out over 10-minute periods owing to the duration of the noise sample, but the error and impulse counts have been multiplied by 1.5 as if they had been obtained over 15-minute periods, this being justified owing to the good repeatability of the results.

4. *Review of results*

Figures 2, 3 and 4 show the relationships which were obtained between the error counts using the 511-bit pattern at data signalling rates of 600, 1200 and 75 bits/s respectively and the impulse count. To obtain this relationship the appropriate impulse counter filter was used and the counter threshold was set to the same value (in dBm) as the level of the received data signal as measured with a 1:1 data pattern.

Figures 5, 6 and 7 show the same results but plotted versus the threshold setting or received data signal level. The dotted lines show the effect of varying the signal level with respect to the fixed noise level and the solid lines are taken from Figure 1 and superimposed for the sake of completeness. In studying these results it is stressed that they are only for a particular sample of impulse counter, a particular sample of noise and a single modem, and it may not be permissible to draw general conclusions from them. However, it is noticeable that for the region between 10 and 100 counts per 15 minutes, the error count and the impulse count do not differ from each other by more than a factor of about 2. For counts greater than 100, the error count increases more rapidly than impulse count. It is suggested that this is partly due to an increasing proportion of impulses causing error bursts rather than single errors and partly due to an increasing number of instances of saturation of the electromechanical impulse counter.

5. *Field tests at fixed thresholds*

In addition to the tests detailed above the United Kingdom Administration is conducting field tests on leased lines using fixed threshold levels. The investigation has not yet proceeded far enough to establish a criterion for maintenance purposes.

6. *Comments*

Impulse counts could be used in at least two ways, e.g. a) as an acceptance test for the lines or b) as a substitute for an error rate measurement. In case b), measurement of the received signal power level would be necessary in order to set the impulse counter threshold prior to commencing the count. The United Kingdom Administration recommends that attention be concentrated on case a) rather than on case b) since case b) is best dealt with by a direct error count as given by testers conforming to Recommendation V.52, and these testers are now available. Furthermore, use of these testers involves no received signal level measurement.

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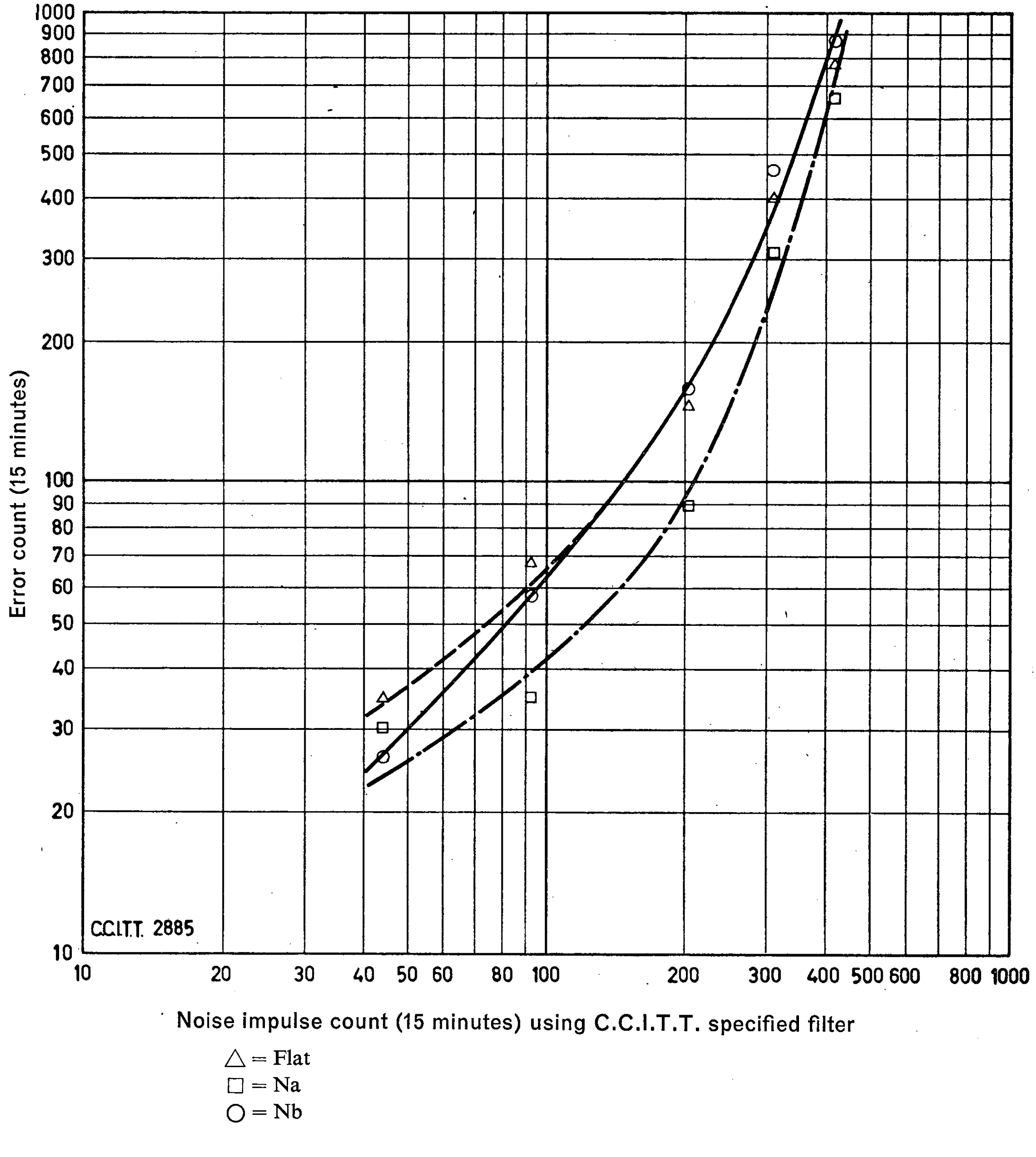


FIGURE 2. — **Error count versus noise impulse count using Datel modem No. 1A and a recorded sample of impulse noise**

(The threshold level of the noise impulse counter was set to the value of the received data signal level. The data signalling rate was 600 bits/s using the specified 511-bit pattern.)

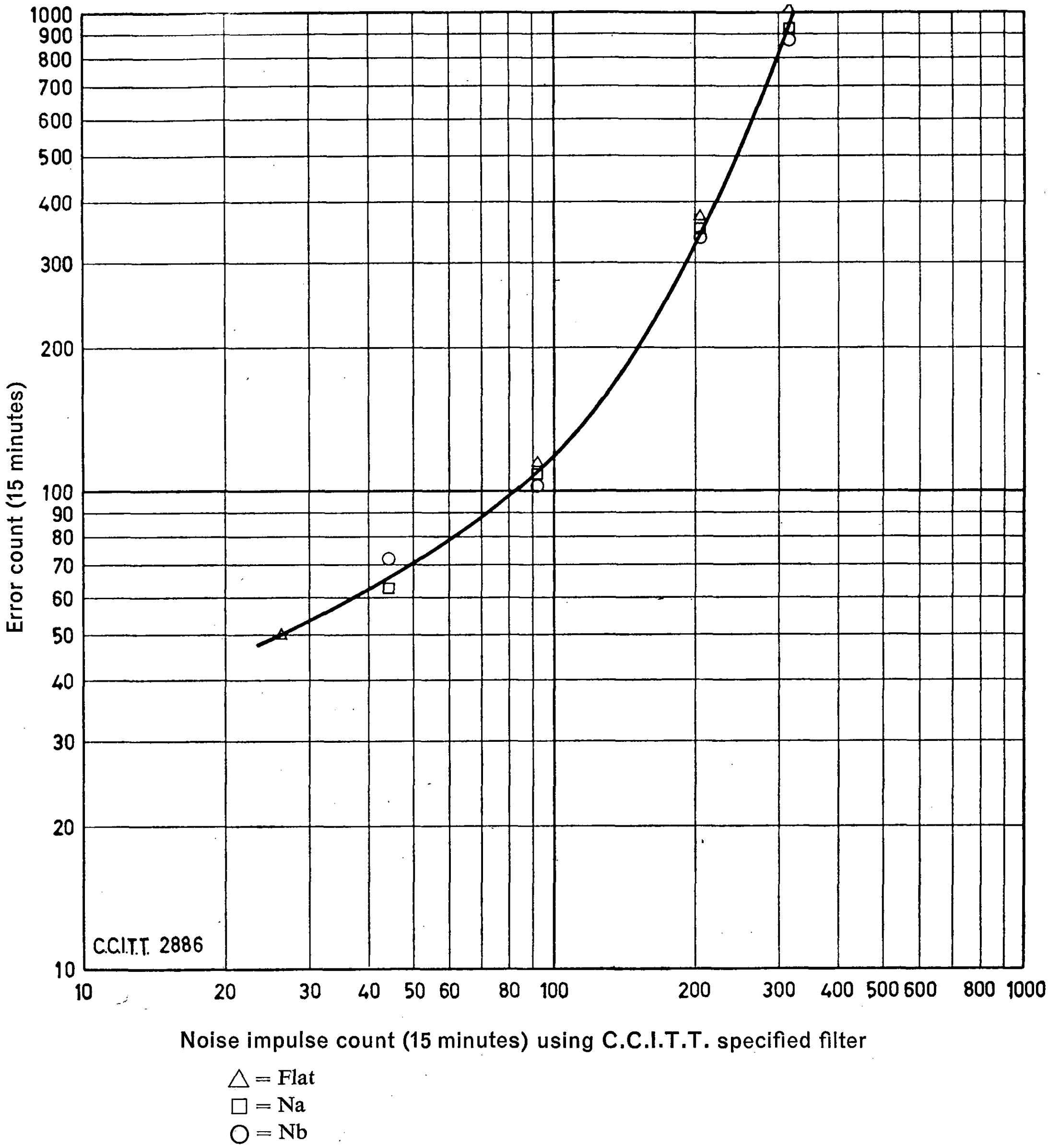
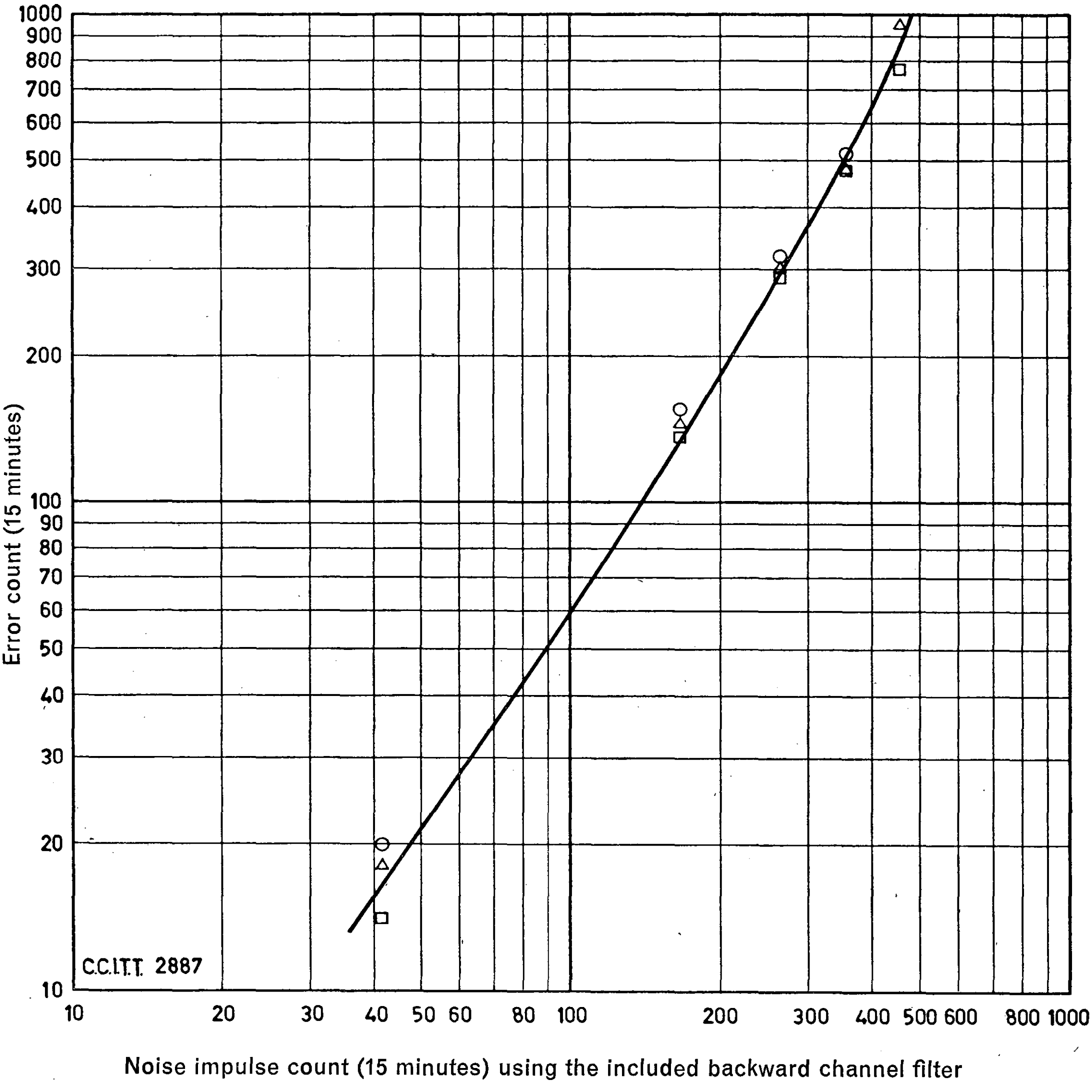


FIGURE 3. — **Error count versus noise impulse using Datel modem No. 1A and a recorded sample of impulse noise**

(The threshold level of the noise impulse counter was set to the value of the received data signal level. The data signalling rate was 1200 bits/s using the specified 511-bit pattern.)



△ = Flat
□ = Na
○ = Nb

FIGURE 4. — **Error count versus noise impulse count using Datel modem No. 1A and a recorded sample of impulse noise**

(The threshold level of the noise impulse counter was set to the value of the received data signal level. The data signalling rate was 75 bits/s using the specified 511-bit pattern.)

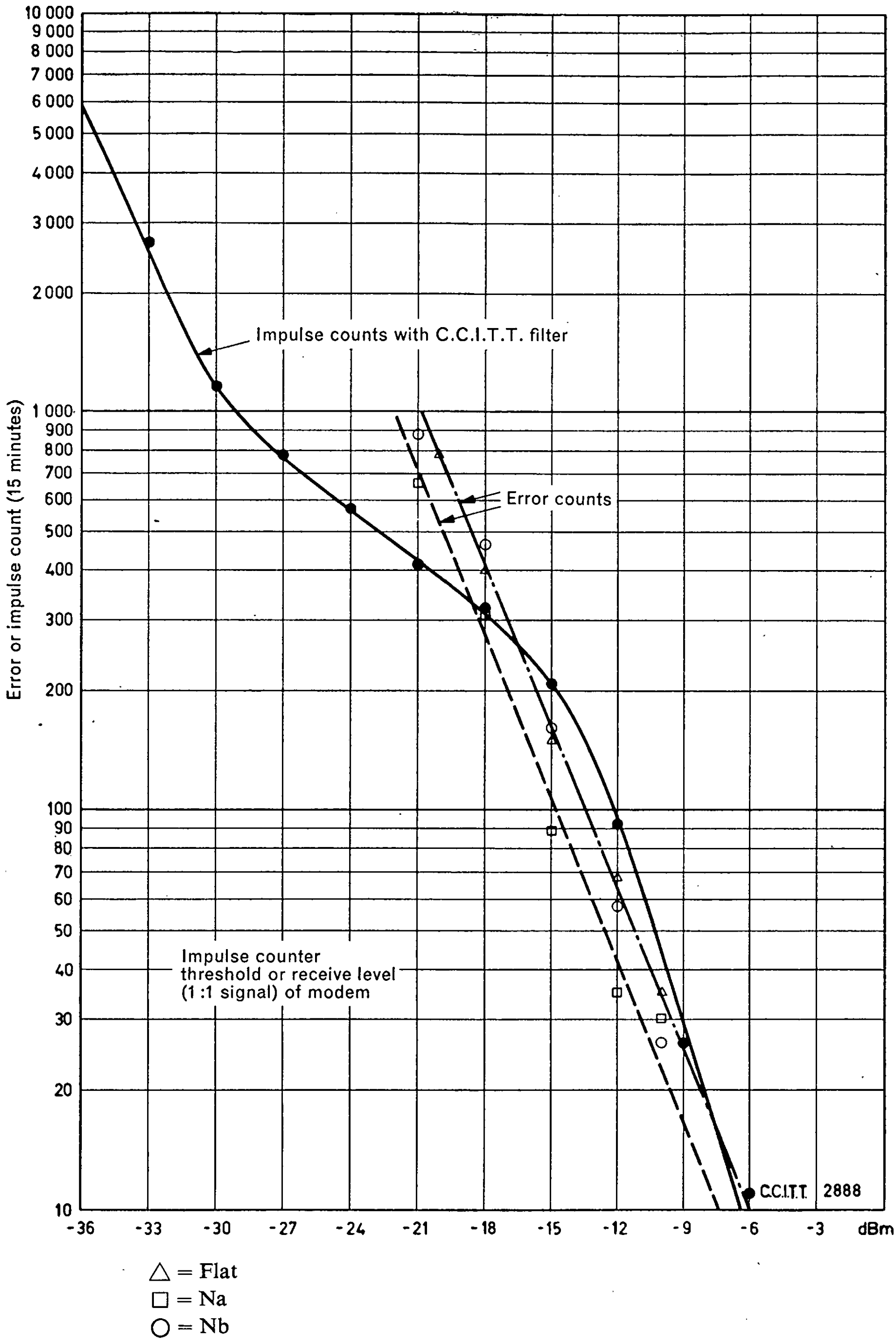
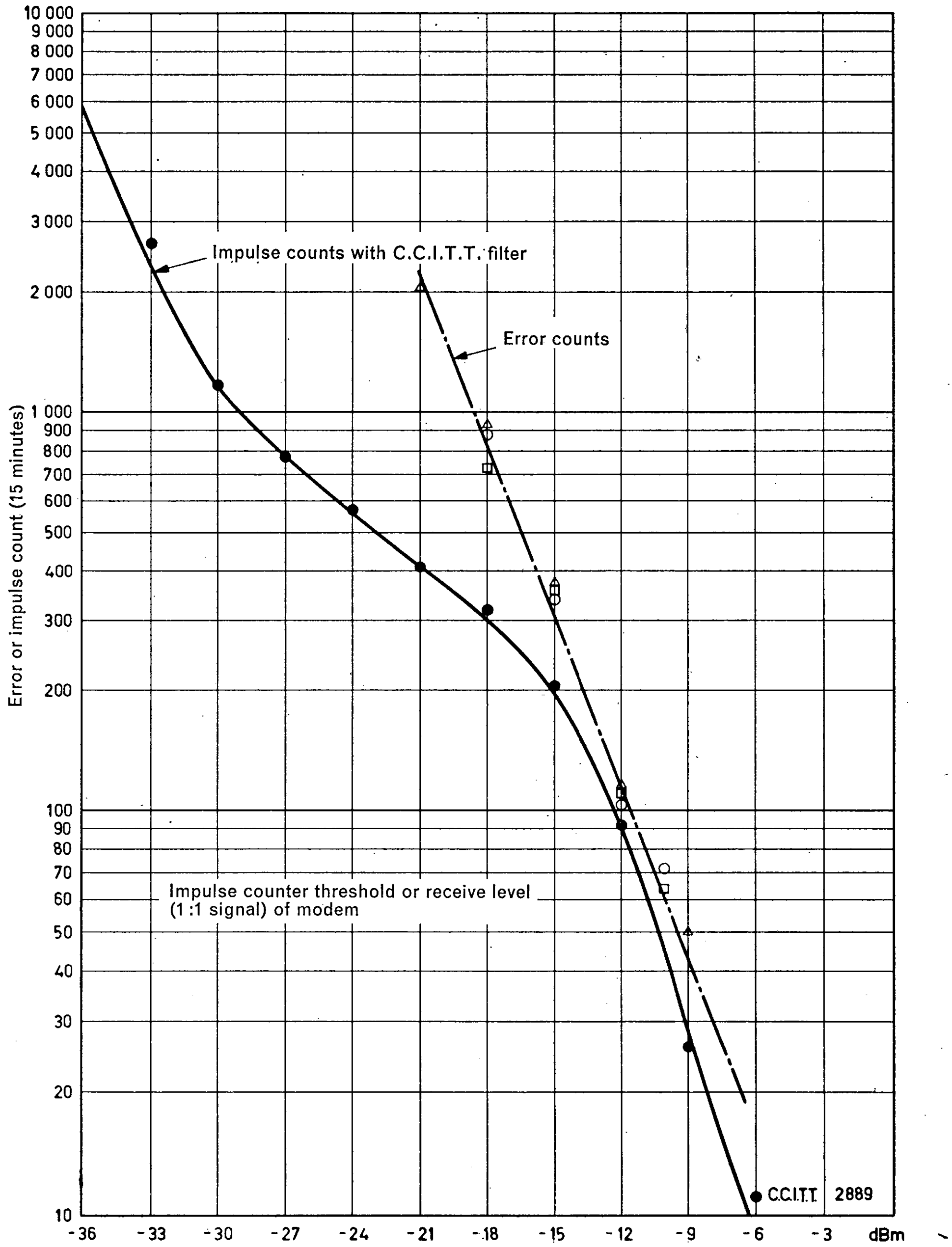


FIGURE 5. — **Error count of impulse noise count versus received data signal level using Datel modem No. 1A and a recorded sample of impulse noise at a fixed level**

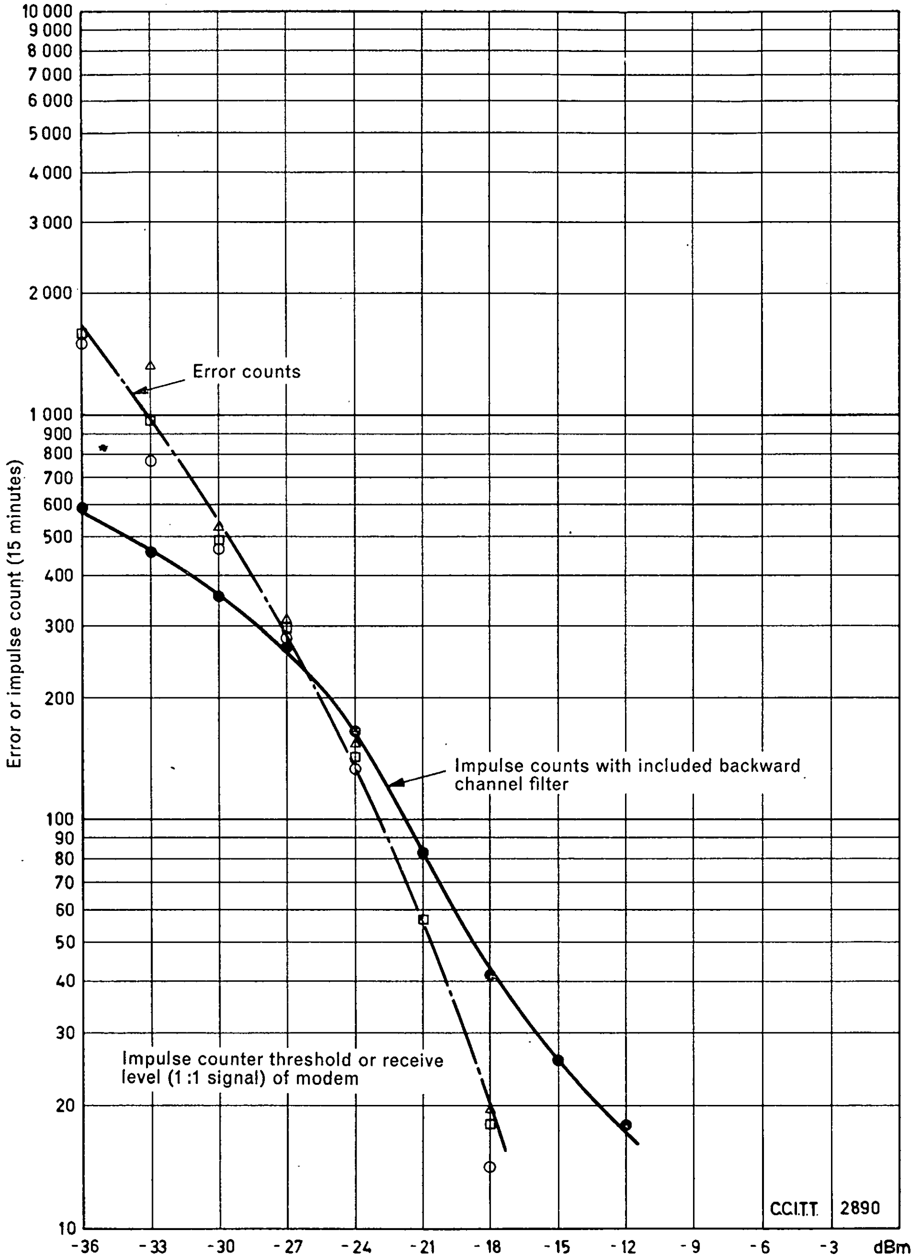
(The data signalling rate was 600 bits/s using the 511-bit pattern.)



△ = Flat
□ = Na
○ = Nb

FIGURE 6. — **Error count or impulse noise count versus received data signal level using Datel modem No. 1A and a recorded sample of impulse noise at a fixed level**

(The data signalling rate was 1200 bits/s using the 511-bit pattern.)



△ = Flat
□ = Na
○ = Nb

FIGURE 7. — **Error count or impulse noise count versus received data signal level using data modem No. 1A and a recorded sample of impulse noise at a fixed level**

(The data signalling rate was 75 bits/s using the 511-bit pattern.)

AMERICAN TELEPHONE AND TELEGRAPH COMPANY. — *(Contribution COM Sp. A— No 113—July 1967)*

USE OF IMPULSE NOISE COUNTERS

1. *Introduction*

For several years impulse noise counters with characteristics similar to those presented in COM Sp. A—No. 66 (see Recommendation V.55) have been used by A.T. and T. Co. in the maintenance of the telephone network. This contribution summarizes recent experience in the interpretation of impulse noise measurements and outlines general measurement procedures and techniques for trunks and offices.

II. *General*

For the many different modulation techniques employed in data transmission, the peak value of a noise pulse has been found to be the most useful single criterion for estimating its error-producing potential.

The relationship between the peak amplitude of a noise pulse and its error-producing potential is easily displayed by means of empirically derived impulse noise performance curves. These are plots of the expected number of bit errors per impulse versus the r.m.s. data signal to impulse noise peak ratio expressed in dB as

$20 \log_{10}$ (r.m.s. signal/peak noise amplitude).

The shape of a particular curve depends upon the modulation scheme, the type of operation (synchronous or asynchronous), and the amount of other transmission impairments in the system. Several of these curves are shown in Figures 1 and 2. They show that the expected number of errors per noise pulse varies from less than 10^{-3} to 10 depending upon the signal-to-noise ratio. The value of signal-to-noise ratio at the point where one of the curves crosses the ordinate value of 10^{-3}, the abscissa in Figures 1 and 2, is sometimes defined as the threshold of susceptibility to impulse noise.

Since the expected number of errors is determined by the peak amplitude of the pulse, the distributions of the peaks of noise pulses encountered on connections are important. These have been found to be adequately described by an exponential, which when used to relate the expected number (during some specified time) of impulses at one level to the expected number at another level, may be written as

$$n_2 = n_1 \, e - \frac{(l_2 - l_1)}{Mm} \tag{1}$$

where:
- n_1 = expected impulses exceeding level l_1 (in dBrn)
- n_2 = expected impulses exceeding level l_2
- m = the inverse "slope" of the line expressed in dB per decade
- $1/M = \log_e (10)$

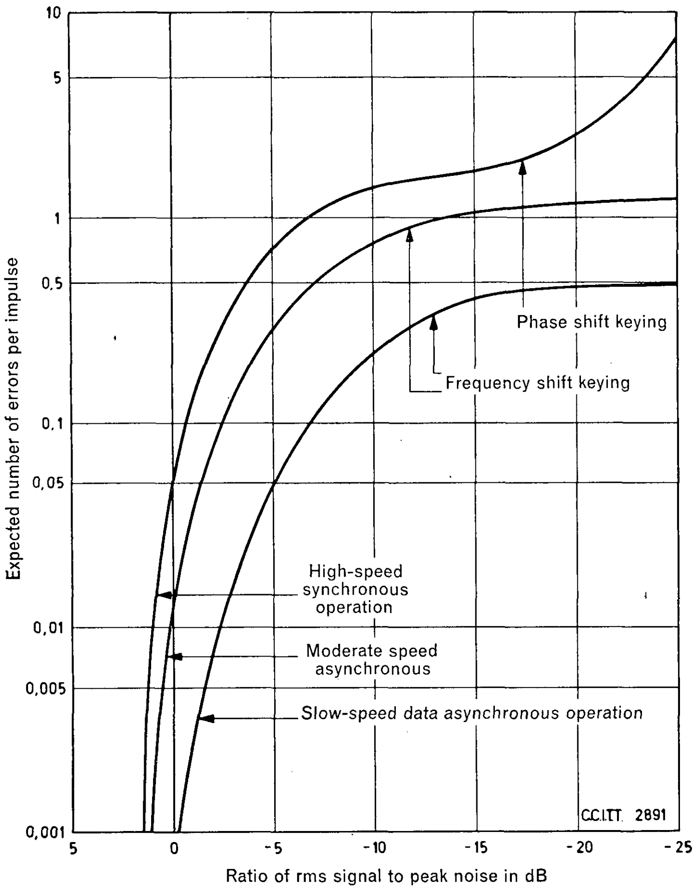


FIGURE 1. — **Variations in impulse noise performance curves with bit rate**

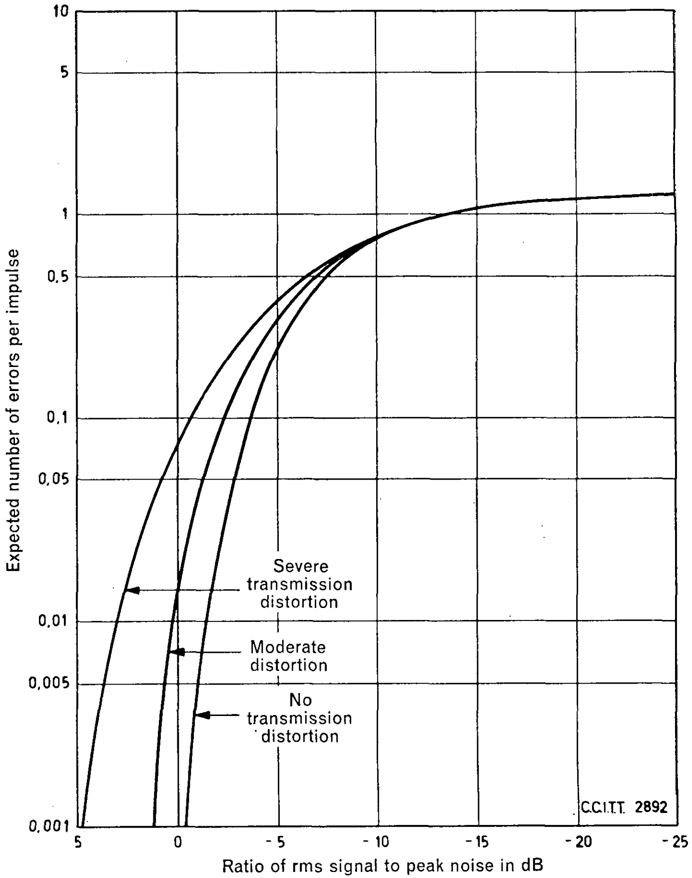


FIGURE 2. — **Variations in impulse noise performance curves with added transmission distortions**

If we let $n_1 = 10n_2$ (one decade in expected number of impulses), which implies $l_2 > l_1$, and take the logarithm of both sides of the expression, which is equivalent to plotting on semi-log paper, as is commonly done, and solve for m, we have $m = l_1 - l_2$ or, $1 = -1/m\,(l_2 - l_1)$. The first form shows that m is a negative number, and the second form shows m explicitly as the inverse slope of a linear relation. Common usage drops the minus sign and the "inverse" and we speak simply of the "slope" of the distribution. Values of slope observed on transmission facilities range from 1 to 33 with about 95% of them falling within the range 2 to 10 and with an expected value of about 7.

In switching equipment, slope shows a more restricted range but an expected value of about 15. If an impulse noise distribution is measured above a level corresponding to the threshold of susceptibility for a given data set and specified transmission impairments, the measured distribution and the impulse noise performance curve can be used to predict the expected number of errors that the noise will cause. To do this, the impulse noise performance curves shown in Figures 1 and 2 are represented mathematically by a double exponential,

$$E = a \exp (b \exp (-cl))$$

where: E = expected number of errors per impulse

a, b, c = constants determined by the data set and other impairment conditions

l = level in dB above the critical S/N ratio.

The impulse noise peak amplitude distribution is measured (using multiple impulse counters) with the most sensitive counter set at a level corresponding to the critical S/N ratio and other levels are referred to that one. l_1 in equation (1) becomes zero, l_2 becomes a variable as in equation (2). The derivative of equation (1) (normalized) then expresses the density function of impulse noise peaks above the critical S/N level. The expected number of errors in a given time period is found by weighting the performance curve with the density function of noise peaks and integrating over the total range of levels, above the critical one, in which pulse peaks occur. If the derivative of equation (1) is called $f(p, l)$ and equation (2) called $f(E, l)$ then functionally:

$$\text{Expected errors} = T_N \int_0^\infty f(p, l)\, f(E, l)\, \mathrm{d}\, l,$$

where N is the quantity used to normalize equation (1) and T is the total number of noise pulses during the same observation interval.

Controlled tests using recorded noise, provide a check on the accuracy of this technique. Figure 3 summarizes the results of such tests. Predicted number of errors are plotted against actual number of errors. The straight line on the figure represents perfect prediction.

To predict error rates by the method just described, one needs to know two parameters related to the noise: the expected number of impulses at some level and the slope of the distribution. By making some approximations and with some loss in accuracy, the slope of the distribution can be ignored. Figure 4 shows noise distributions, taken over 30 minutes, with slopes from 2 to 20 dB/decade, adjusted along the ordinate in such a way that each of them would give rise to exactly the same expected error rate, 10^{-5} in this case. The threshold of susceptibility of the corresponding impulse noise performance curve was + 4 dB. Note that the number of impulses which exceed that s/n ratio varies from 44 to 7000 per half-hour, depending upon the slope. Note further, however, that the range of impulses for these equally interfering distributions is much smaller at S/N of -1 dB, where it is only 24 to 78. By choosing a point near the middle of this

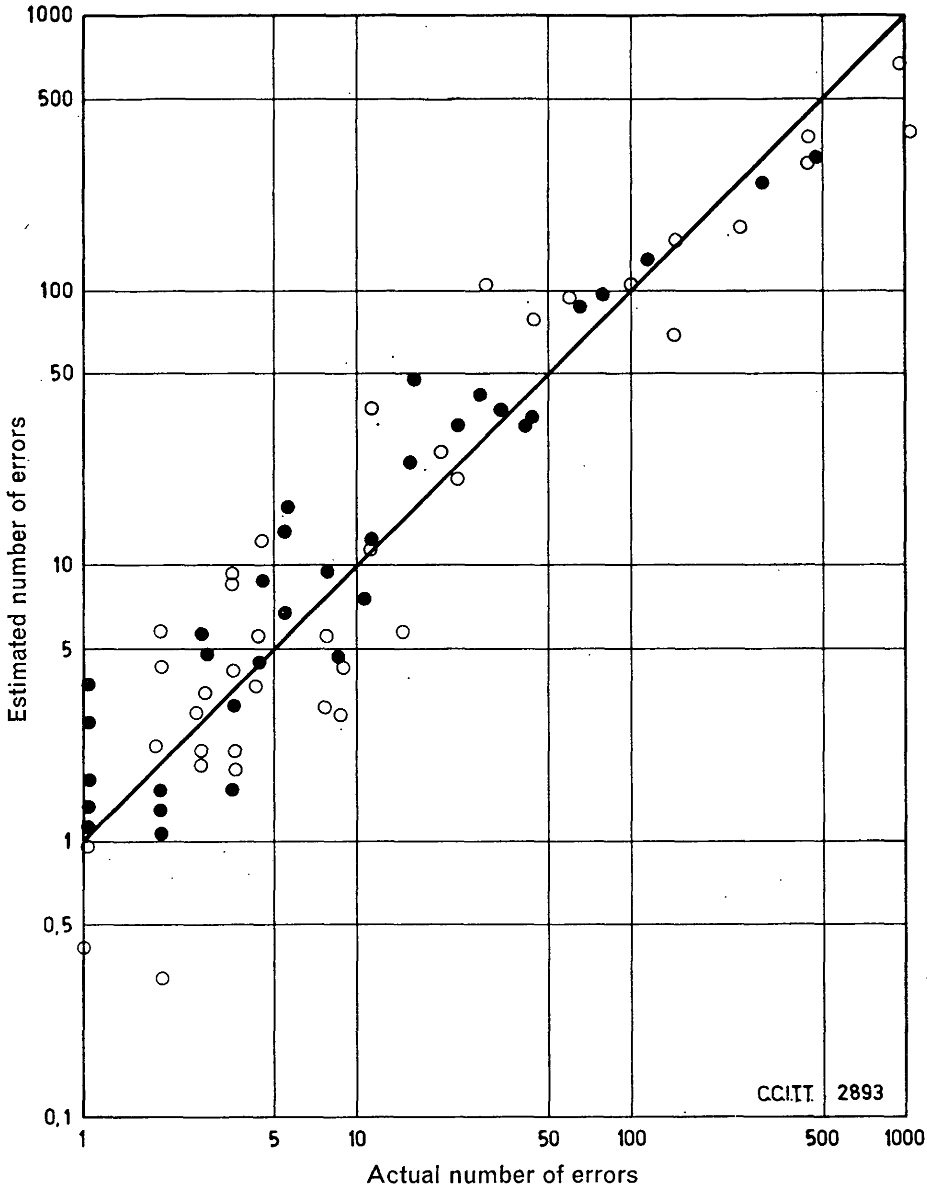


FIGURE 3. — Plot of estimated vs. actual number of errors caused by impulse noise during 30-minute period; bit rates of 2000 and 2400 per second

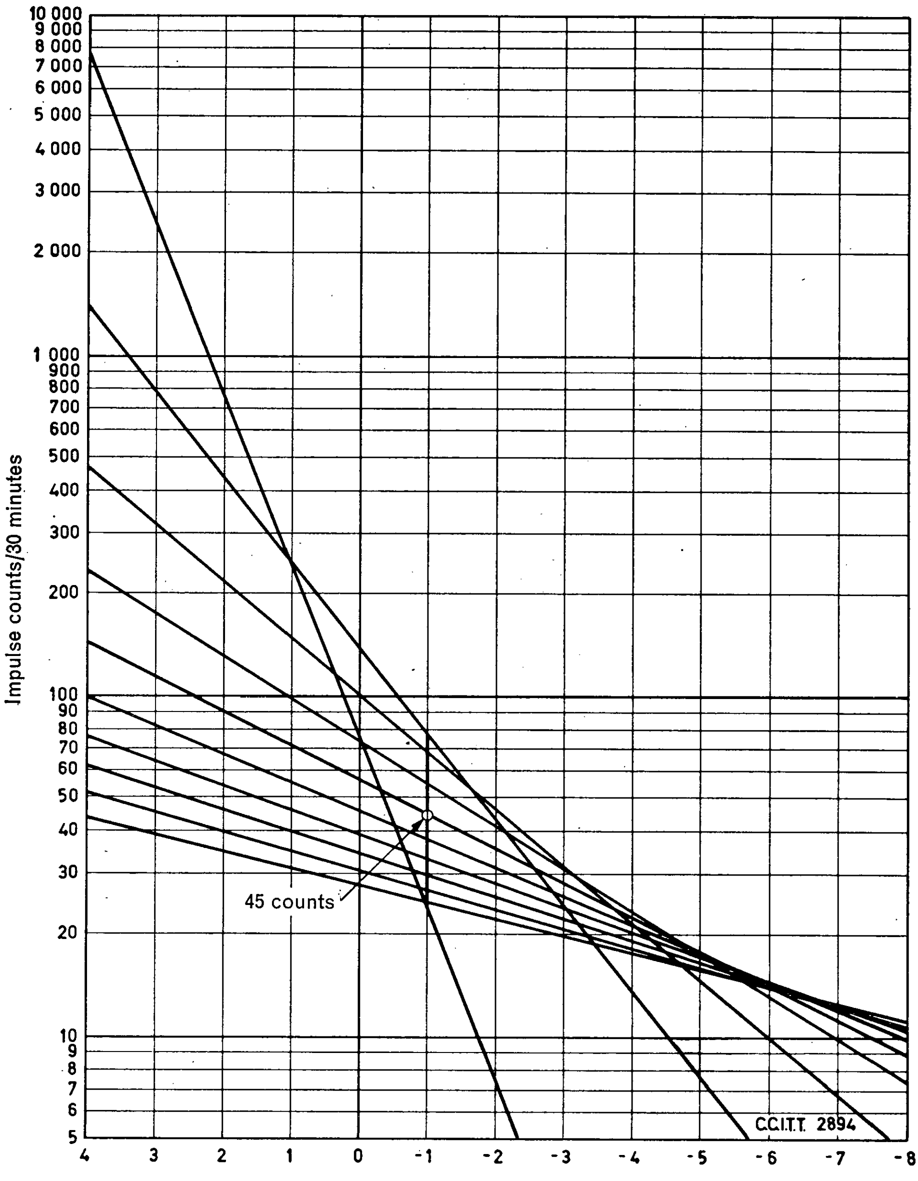


FIGURE 4. — R.m.s. signal to impulse noise peak in dB

range, 45, as indicated on the drawing, it is possible to be within a factor of 2 of the actual number of impulses, regardless of the slope of the distribution. As other transmission parameters change and the impulse noise performance curve changes, other families of equally interfering noise distributions arise. The number 45 is a good compromise for all situations encountered with the higher speed (1200-2400 bits/s) voice-band data sets. The impulse noise objectives currently in use by the A.T. & T. Co. are based on this number.

In practice, there is still another uncertainty which must be considered. This is due to the use of electromechanical counters in impulse noise measuring apparatus. Due to the finite counting rate of the counters, some noise pulses are not recorded. The percentage of pulses missed ranges from zero to about 85, with an average of 34%. The large percentages arise when the noise being measured is due primarily to cross modulation products or high background noise. Correcting the 45 count for the average number of pulses missed leads to 30 counts per half-hour, or 5 in 5 minutes, as read on impulse noise measuring sets. In transmission work relating to voice-bandwidth data, the impulse noise level is defined as that level at which the expected rate of occurrence of noise pulses is equal to one per minute. For satisfactory data transmission the impulse noise level should be at least two or more dB below the r.m.s. data signal level at the receiving station set. The actual required signal to impulse noise level varies over a range of about 6 dB (+ 3 to −3.5) depending upon other transmission impairments, but if 2 dB S/N is met, then impulse noise may be ruled out as a significant contributor to an excessive error rate (say poorer than 10^{-4}) in almost all cases. A few exceptions exist. If nearly all of the measured noise counts are greater than 5 dB above the signal level (see Figure 1), then each noise pulse will cause at least one error and the error rate may be excessive. This situation is occasionally found in step-by-step switching machines.

In tests made on customer-dialled connections where the data signal level was adjusted after measurements of impulse noise and other impairments, to give a predicted error rate of 10^{-5}, the average observed error rate was 1.1×10^{-5} and the standard deviation of the ratio of actual error rate to 10^{-5} was 0.67. The approximations made in deriving the single criterion of impulse noise evaluation were thereby shown to yield usable engineering results.

III. *Measurement techniques and objectives*

1. *Trunks*

For purposes of general trunk routine measurements, trunks are considered to exist in distinct populations. Each population is considered to consist of all those trunks which have common end points (office or switching machine), regardless of the routing or make-up of individual members. Objectives are established on the mean noise activity within a trunk group. Measurements are made by terminating the far end of a trunk in its characteristic impedance, connecting the impulse noise counter to the near end, and counting the impulses that occur in five minutes at a specified level. The trunk group is acceptable if at least 50% of the measurements register 5 counts or less. If any one measurement exceeds 100 counts, further detailed measurements are made on that trunk to determine whether the impulse noise on the trunk is actually excessive or whether that one measurement suffered from the occasional rare burst of crosstalk or deep fade of a radio system and should therefore be ignored.

The levels specified for the measurements are dependent upon the trunk length and its facility make-up as shown in Table 1. It is emphasized that these levels apply to trunks which are components of connections with both end points within the continental United States only and are not necessarily satisfactory for intercontinental service.

TABLE I

Recommended measurement levels for impulse noise objectives

Length of trunk in miles	VF repeatered and non-repeatered	Compandored or mixed compandored non-compandored carrier	Non-compandored carrier
Measurement levels in dBrn0VB			
0- 60	55	69	59
61- 125	55	69	59
126- 250	55	69	60
251- 500		69	60
501-1000		69	60
1001-2000		69	62
> 2000		69	65

These recommended levels were determined on the basis of noise performance of trunks according to length and facility type. They have been tested, via simulation, and have been found to be adequate by current standards for acceptable data transmission.

When compandored carrier systems are included in the trunk make-up a compandor holding tone is transmitted from the far end at a level of −10 dBm0. This tone, at a frequency of 2750 Hz is filtered out ahead of the impulse counter by a very narrow slot filter.

It is expected that sampling techniques will be incorporated in the future to reduce the total time required for trunk group evaluation.

2. *Switching offices*

The measurement of impulse noise in switching machines is accomplished by establishing connections between pairs of spare terminals on the distributing frame served by the switching machine. Since the possible number of pairs of terminals is extremely large, two stage sampling techniques are employed in the selection of spare pairs from which estimates of the noise in the entire machine can be made. The elements within each primary unit are the terminal pairs within the primary units of the first stage. A sample size of four primary units with 10 elements in each unit has been found sufficient to characterize the noise in an entire switching machine, i.e., a complete step-by-step office, or one marker group in a crossbar machine. Once a sample has been selected, one additional spare terminal is selected in each primary unit and assigned a directory number. It is designated as the called station. All other elements selected in each primary unit are designated calling stations. Connections are established one at a time from each calling station to the called station within the primary unit. When the connection is established, the called station terminals are terminated in the nominal characteristic impedance and the impulse noise counter connected through a suitable holding coil to the calling station terminals. A five-minute measurement is made and the total count recorded. As long as an equal number of measurements are made in each primary unit, the mean of the office can be estimated from the mean of the total sample, so, as with trunks, if at least 50 % of the measurements have 5 counts or fewer, the switching machine impulse noise is acceptable.

The measurement level for switching office impulse noise must be low compared with the level required for trunks which terminate at the office. It has been found that a level of 50 dBrnBV can be attained in crossbar type switches. This low a level is difficult to achieve in step-by-step switching equipment. Since most step-by-step switches in the Bell System are in exchange rather than inter-trunk use, the inherently higher noise of these switches is recognized and the exchange office measurement level is 60 dBrnVB. The exchange office thus is allocated a major share of the total impulse noise on complete connections.

Supplement No. 38

UNITED KINGDOM. — *(Extract from Contribution COM Sp. A—No. 136— October 1967)*

SPECIFICATION OF FREQUENCY/LOSS CHARACTERISTICS, PHASE DISTORTION AND IMPULSE NOISE LIMITS FOR CIRCUITS LEASED FOR DATA TRANSMISSION; MEASURING TECHNIQUES TO CHECK THESE SPECIFICATIONS

The characteristics of international leased circuits for data transmission are given in Recommendation M.89 *(Blue Book,* Volume IV). In this Recommendation two classes of leased circuit are detailed as follows:

a) Ordinary telephone circuits having the characteristics given in Recommendation M.61, and

b) Special quality circuits

In order to determine the maximum data transmission rate possible on any given circuit, the United Kingdom considers that it would be useful for Special Study Group A to specify limiting circuit characteristics for use with standardized modems, particularly modems complying with Recommendation V.23 for 1200 bits/second transmission rate. The special quality circuit detailed in Recommendation M.89, paragraph 3, is unnecessary for transmission at 1200 bits/second. On the other hand Recommendation M.61 details only circuit characteristics between international test centres.

The United Kingdom Administration therefore proposes that Special Study Group A should request Study Group IV to consider additions to Recommendation M.61 or M.89, as is convenient to Study Group IV which would detail the limiting characteristics, measured between customers' premises, for leased circuits to be used for data transmission at signalling rates up to 1200 bits/second using modems complying with Recommendation V.23. The additions which the United Kingdom proposes should be made to the leased circuit. Recommendations are set out in paragraphs 1 to 5 below and were determined by the United Kingdom during tests carried out at 1200 bits/second with modems complying with Recommendation V.23.

1. *Overall loss at 800 Hz*

The overall loss for circuits for data transmission at rates up to 1200 bits/second when using modems complying with Recommendation V.23 not to exceed 13 dB between two-wire terminals at customers' premises.

2. *Attenuation/frequency distortion*

The variation with frequency of the overall loss of a circuit, used for data transmission at rates up to 1200 bits/second when using modems complying with Recommendation V.23 not to exceed the limits given in Figure 1 over the frequency range 300 and 3000 Hz between two-wire terminals at customers' premises.

3. *Group delay/frequency distortion*

The group delay/frequency distortion for circuits to be used for data transmission at rates up to 1200 bits/second when using modems complying with Recommendation V.23 to be within the hatched lines on Figure 2 when measured between customers' premises.

4. *Signal/echo ratio*

The ratio of signal to echo received at customers' premises, arising from signal leakage across the four-wire of the two-wire/four-wire terminating sets, to be greater than 18 dB for circuits to be used for data transmission at rates up to 1200 bits/second using modems complying with Recommendation V.23.

(The signal-to-echo ratio is the sum of the overall losses in each direction and the balance return loss of the two-wire line connected at each four-wire terminating set, assuming that the GO to RETURN crosstalk ratio of the four-wire section is not less than 43 dB).

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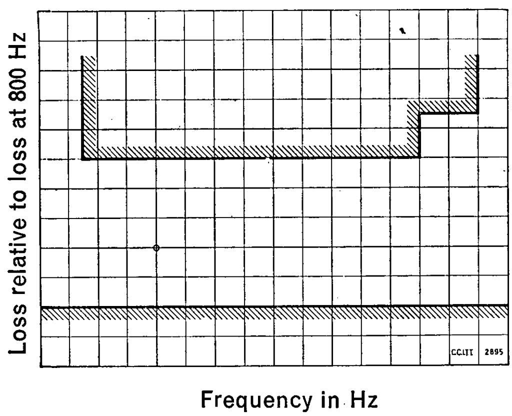


FIGURE 1. — **Proposed objective for attenuation/frequency characteristic of two-wire or four-wire leased circuits for data transmission at 1200 bit/s using modem complying with Recommendation V.23**

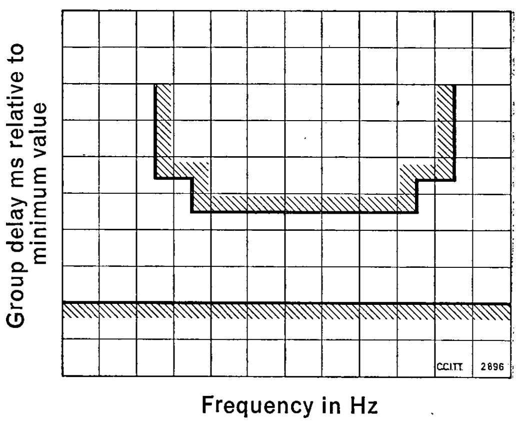


FIGURE 2. — **Proposed objective for group delay frequency characteristic of two-wire or four-wire leased circuits for data transmission at 1200 bits/s using modem complying with Recommendation V.23**

Supplement No. 39

FRANCE. — *(Contribution COM Sp. A—No. 183—December 1967)*

SPECIFICATION OF AN ORDINARY TELEPHONE CIRCUIT FOR DATA TRANSMISSION

A specialized ordinary telephone link, which can be set up either with two-wire or with four-wire terminals (in the latter case, the two directions of transmission are separate from end to end) shall have the following characteristics:

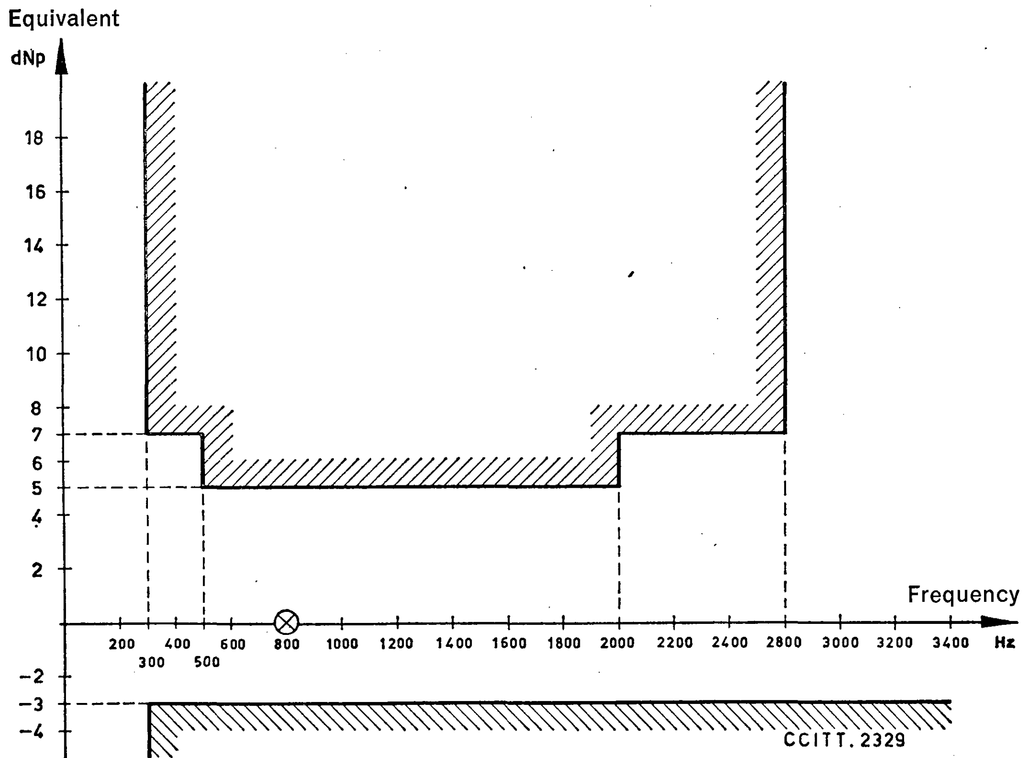
1. The equivalent at 800 Hz, measured at the user's installation between the two ends of the link, shall not exceed 7 dNp in links with four-wire terminals and 15 dNp in links with two-wire terminals.

2. The variation of the equivalent at 800 Hz shall not exceed ± 5 dNp.

3. The variation with the frequency of the circuit equivalent in relation to the equivalent at 800 Hz shall not exceed the limits shown in the following figure.

4. The delay/frequency distortion shall not exceed 1 millisecond in the 1000-2400-Hz band and 1.5 millisecond in the 800-2600-Hz band.

5. The mean psophometric voltage measured across the terminals of a non-reactive 600-ohm resistor and referred to a point of zero relative level shall not exceed 12 millivolts.



Limits for the variation with frequency of the equivalent of the ordinary telephone circuit

Supplement No. 40

AUSTRALIA. — *(Contribution COM Sp. A—No. 115—August 1967)*

MEASUREMENTS OF IMPULSIVE NOISE AT SYDNEY INTERNATIONAL SWITCHING CENTRE

1. *Introduction*

This document outlines the results of impulsive noise measurements made at the Sydney International Switching Centre.

The objectives of the measurements were:

i) To measure the number of impulsive noise counts at different threshold levels on a number of connections through the exchange during the busy hour and at other times of lesser traffic density.

ii) To attempt to determine a relationship between exchange activity and the number of noise impulses counted.

These objectives are best achieved by a large number of measurements on a large number of connections. The number of measurements was restricted to several hundred by the limited time available.

2. *The Sydney International Switching Centre*

The Sydney International Switching Centre is a crossbar exchange to which are currently connected 63 international circuits (both-way), 56 one-way junction circuits from the Australian network, and 53 one-way junction circuits to the Australian network.

C.C.I.T.T. No. 5 signalling is used on all international circuits and loop-disconnect decadic signalling is used on the junction circuits.

3. *Measuring instrument used*

The instrument used was an STC 74258-A Impulse Noise Counter which conformed to the Special Study Group C specification (Recommendation V.55).

4. *Method of measurement*

The measurements were made at the points labelled A5 and D3 on C.C.I.T.T. Figure 11 of Recommendation Q.45 in Volume VI of the *Blue Book*. There are no compandors in the exchange and echo suppressors were excluded.

a) *Measuring impulsive noise*

The impulse noise counter was connected to the exchange as shown in Figure 1. The counter was set to measure unweighted impulsive noise and to present a 600-ohm termination to the exchange.

Three different types of connection can be established through the exchange: a transit international call, a direct international call outgoing from Australia, a direct international call incoming to Australia. Impulsive noise counts were made on the GO and RETURN paths for one connection of each type at thresholds between −60 dBm0 and −20 dBm0 (signal level of −2 dBm is 0 dBm0); the counts were made during the busy period when the traffic was about 31 E and were repeated later when traffic dropped to 21 E and again at about 15 E.

Counts were also made on two other connections of each type.

b) *Measuring exchange activity*

The number of seizures of common control equipment was taken to be a measure of exchange activity. Accordingly, a counter was connected to exchange common control equipment and the number of common control equipment seizures counted during the impulsive noise measurement periods.

5. *Results*

a) *Impulsive noise counts per hour at different threshold levels*

The measurement periods were:

Threshold dBm0	– 60	– 55	– 50	– 45	– 40	– 35	– 30	– 25	– 20
Measurement period minutes .	2	2	5	5	5	10	10	10	10
Typical number of noise counts	250	80	50	10	4	4	1	0	0

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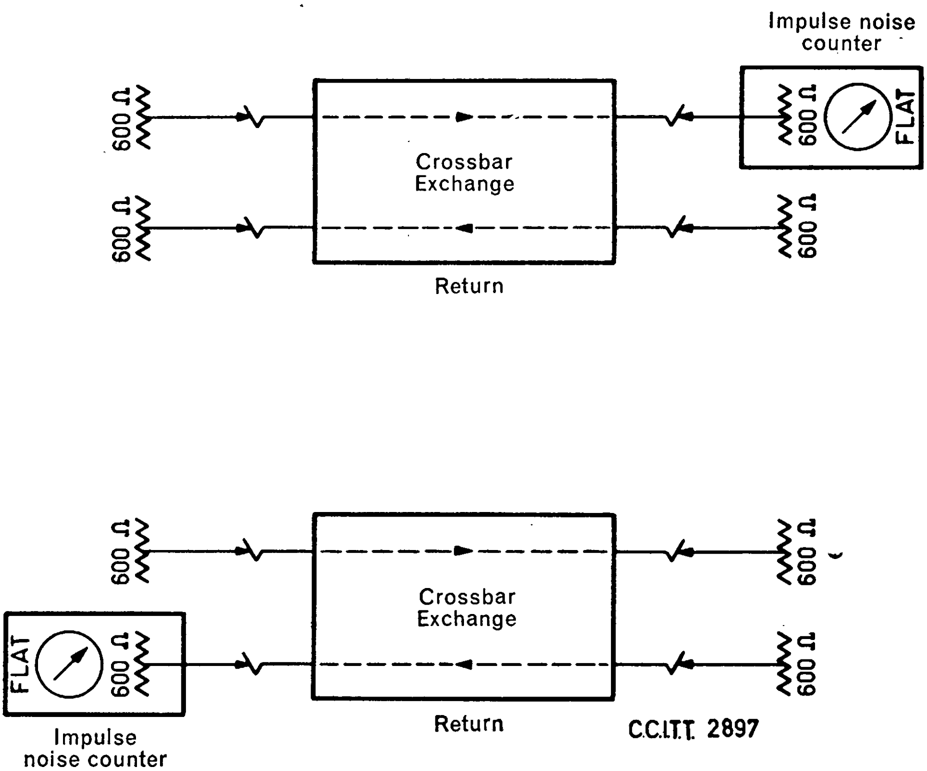

FIGURE 1. — **Measurement of impulsive noise on the go and return speech paths for a connection through the exchange**

b) *Impulsive noise counts per seizure of common control equipment at different threshold levels*

The impulsive noise counts and the common control seizure counts were made at each of the three traffic loadings (about 31 E, 21 E, and 15 E) for the GO and RETURN paths of three connections. The number of impulsive noise counts per common control seizure was not the same at each traffic loading for a given threshold. In other words, a proportionality between impulsive noise counts and exchange activity was neither proved nor disproved. It is apparent that the determination of such a relationship requires a larger number of measurements and a more precise measure of exchange activity than the seizure of common control equipment.

However, the ratio of the number of noise counts to the number of common control seizures (the ratio was computed by dividing the sum of the three noise counts by the sum of the three seizure counts) is plotted as a function of threshold. This graph shows that impulses above about −40 dBm0 are due to causes other than the progress of the average call through the exchange and may be due to coincidences of conditions and isolated occurrences of excessive mutual coupling. The threshold at which there is one impulsive noise count per common control seizure (or some other unit of activity) is perhaps useful as a figure of merit for the exchange.

Additional measurements were made on three other connections (one transit, one direct outgoing, and one direct incoming) and these measurements also followed the same general pattern.

6. *Conclusion*

a) *Number of impulsive counts per hour at different threshold levels:*

i) At −20 dBm0, there were always less than 30 counts/hour;

ii) At −30 dBm0, there were always less than 150 counts/hour;

iii) At −40 dBm0, there were always less than 1000 counts/hour.

b) *Relationship between impulsive noise counts (at a given threshold) and exchange activity:*

i) Well-defined proportionality between impulsive noise counts and common control equipment seizures was not apparent: this led to the belief that the seizure of common control equipment is not a sufficiently accurate measure of exchange activity to produce such a relation.

ii) Noise impulses exceeding −40 dBm0 occurred less frequently than common control equipment seizures.

Supplement No. 41

U.S.S.R. — *(Contribution COM Sp. A—No. 144—October 1967)*

STUDY OF SHORT INTERRUPTIONS IN TELEPHONE CHANNELS DESIGNED FOR DATA TRANSMISSION

1. Variation of overall telephone circuit loss on cable routes is the consequence of a great number of various causes and, as a rule, is in conformity to the normal (Gaussian) law of distribution. However, recent studies have shown that there are short and long level variations (mostly

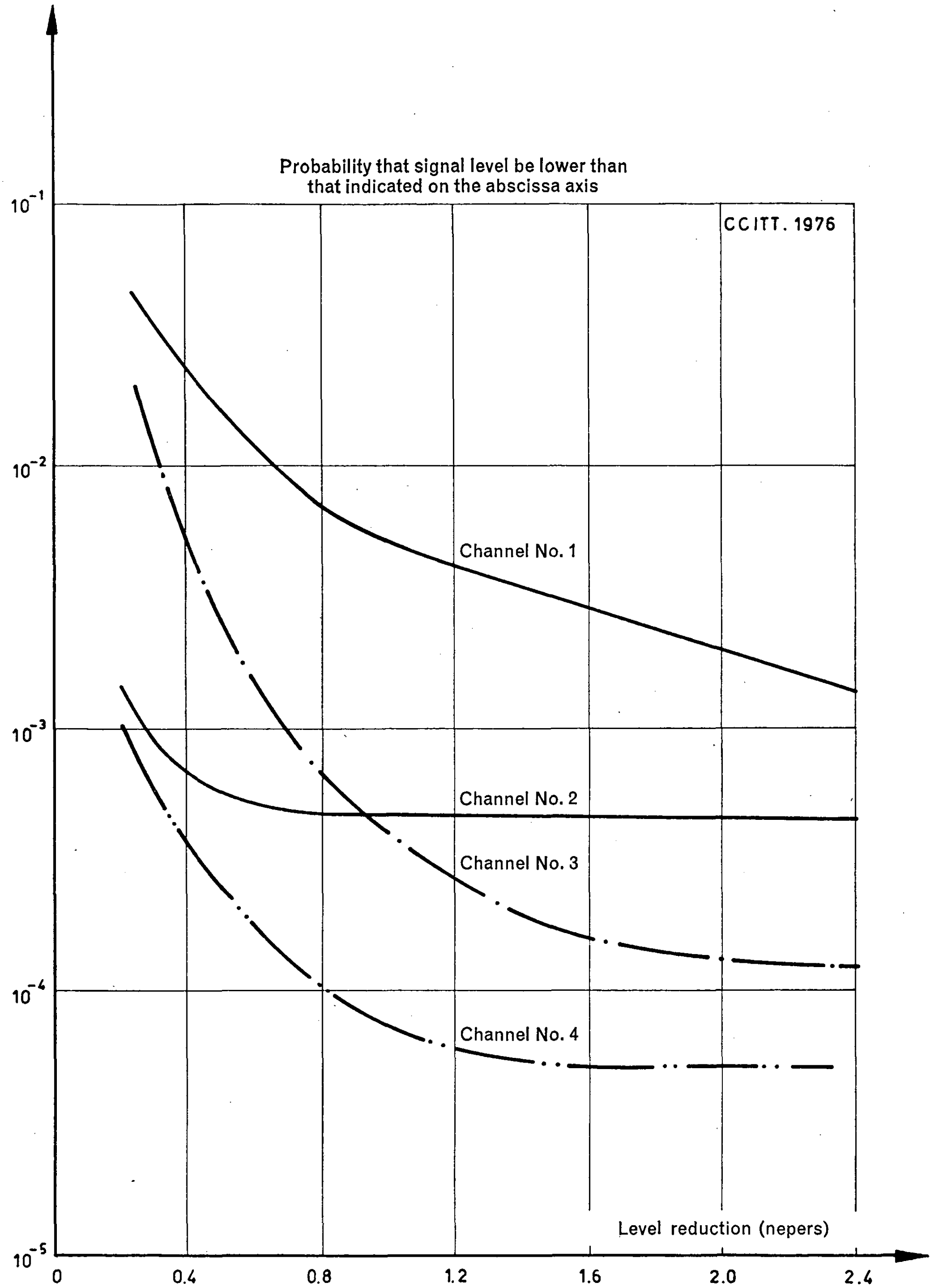

FIGURE 1. — **Examples of distribution of level reductions**

reductions) which do not obey this law. These level variations are one of the main factors to which errors in data transmission channels and failures of channels are attributable.

Therefore, the indication by COM Sp. A concerning the necessity to study short interruptions endangering data transmission is opportune.

2. In the U.S.S.R. measurements of the distribution of level reductions according to value and duration were carried out on telephone channels of the trunk network with the help of an impulsive noise and interruption analyser. The block diagram and brief description of the device are given in Annex I.

Analysis of the results of measurements lasting several hundred hours revealed that the probability of pilot level being lower than the given threshold varied with time very considerably and was substantially different on various routes.

Figure 1 shows several examples of distribution of total level reduction duration as a function of level reduction value on various long-distance non-switched channels (3000-4000 km).

All reductions including those of more than 300 ms were taken into account. Most of the curves show a sloping equiprobable section in the area where the level reduction is more than 1.5 Np, which is explained by a great share of complete level drops (more than 2.0 to 2.5 Np).

The operation feasibility of data transmission equipment is limited. Level reductions of 2.0 Np and more usually give rise to an increase in error rate. At the same time there is a very low probability that smaller values of level reductions (less than 1 to 2 Np) would produce errors in transmitted information.

Bearing in mind these considerations, it would be advisable to take interest only in considerable level reductions (more than 2.0 Np) which may be provisionally called "*interruptions*".

Thus, while measuring the possibility to use a telephone channel for data transmission at an average rate (600 to 2400 bauds) it is possible to measure solely the probability of total interruption duration P_b instead of plotting the whole distribution curve of level reduction values.

Using the measured value of interruption probability, one may approximately determine the usefulness of the channel for data transmission and the error rate in it. During an interruption, i.e. when there is no information, the provisional error probability is equal to ½; hence one may calculate the error rate due to interruptions according to formula:

$$P_e = \frac{1}{2} P_b.$$

The degree of inaccuracy of this assessment may be reduced by excluding the short interruptions of less than a single unit interval which produce errors with low probability. Furthermore, interruptions of more than 300 ms may be neglected considering that they do not influence the error rate but give rise to a reduction of system reliability (occurrence of failures).

3. Better accuracy, as indicated above, could be obtained by using the interruption distribution vs. duration curve. Furthermore, knowing the duration of interruptions it is possible to locate them and eliminate their causes.

Interruption duration measurements carried out on various non-switched channels of the U.S.S.R. trunk network showed that the distribution of interruption durations is very nearly the same for channels of various lengths and for channels on various routes. An example of interruption distribution vs. duration on one of the channels is shown in Figure 2 (curve a).

The curve illustrates that 75% of interruptions lasted less than 150 ms. Therefore, it is to be understood that channels intended for telephone conversations during which interruptions of less than 0.15 s are hardly perceptible cannot always be used for data transmission. This circumstance urgently demands the individual testing of each channel intended for data transmission at intermediate rate even despite the fact that this channel may be successfully used for telephone calls.

If only the interruptions influencing the error rate had to be taken into account, the studies may be confined to interruptions lasting more than 300 ms (curve b in Figure 2).

4. The interruption characteristics of a telephone channel considered above give an average assessment within the long periods of time. However, this is insufficient to determine the possibility of channel use for data transmission within sufficiently short periods. To obtain such an assessment it is necessary to describe the distribution of interruption occurrence in time. The assumption which was in force before, concerning the independence of interruption flow, had to be revised (together with the assumption of independence of errors). Measurements carried out in the U.S.S.R. on a large number of point-to-point and switched channels in trunk and local networks showed that there was a strong tendency to interruption grouping.

This may be illustrated by an example of distribution of intervals between successive interruptions measured on several channels of one of the main routes (see Figure 3). It is known that the distribution of intervals between independent events of the Poisson flow should obey the exponential law of the form:

$$F(\chi) = 1 - e^{-\lambda\chi},$$

where λ is a parameter of the Poisson law.

Figure 3 reveals a considerable difference between the experimental and exponential curves which rejects the hypothesis of the independence of interruptions.

It was assumed while plotting the experimental curve in Figure 3 that the intervals between interruptions were much longer than the interruptions themselves; thus it was possible to reduce an interruption to a point and to consider it as "an event".

Measurement of the distribution of duration of intervals between interruptions was carried out with the help of the device described in Annex II.

5. The analysis of Figure 3 makes it possible to state the hypothesis that interruptions are grouped in "bursts" representing the independent events which obey the Poisson law.

By a burst of interruptions we mean several interruptions (in the special case a single interruption) the intervals between which do not exceed the burst-build-up criterion "τ". Burst-build-up criterion has to be chosen so as to ensure a maximum degree of independence for the bursts and at the same time a minimum number of independent interruptions combined in bursts. It may be assumed that the burst-build-up criterion τ lies in Figure 3. On the abscissa axis at least behind the last point of curve inflection ($\tau = 10$ seconds).

Experimental studies of the flow of interruption bursts with the burst-build-up criterion $\tau = 10$ seconds on various channels of the U.S.S.R. trunk network showed that the bursts were independent and obeyed the Poisson law.

Several sections with various parameters λ are shown in Figure 4. Measurements were conducted with the help of the interruption burst counter described in Annex III.

During long test periods (of several hundred hours) the Poisson parameter is a variable quantity which is an indication of an unstationary Poisson flow. However, it proved to be possible to divide the burst flow which was on the whole unstationary into a number of stationary sections, several hours long, within which the parameter remained a constant quantity:

$$\lambda(t) = \lambda.$$

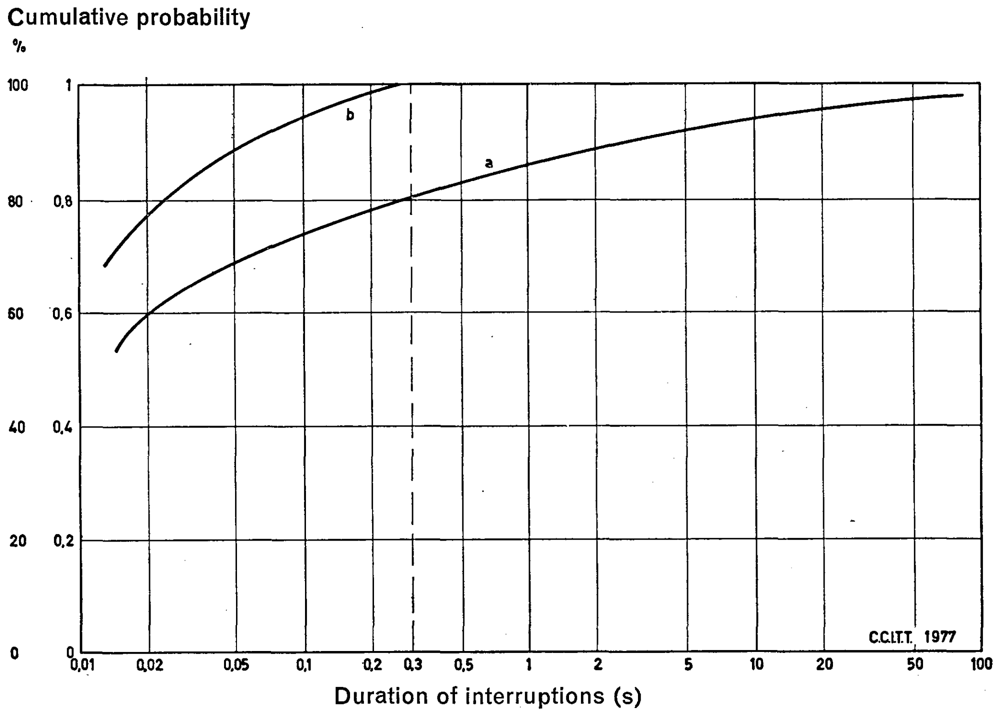

FIGURE 2. — **Example of cumulative distribution of interruption durations**

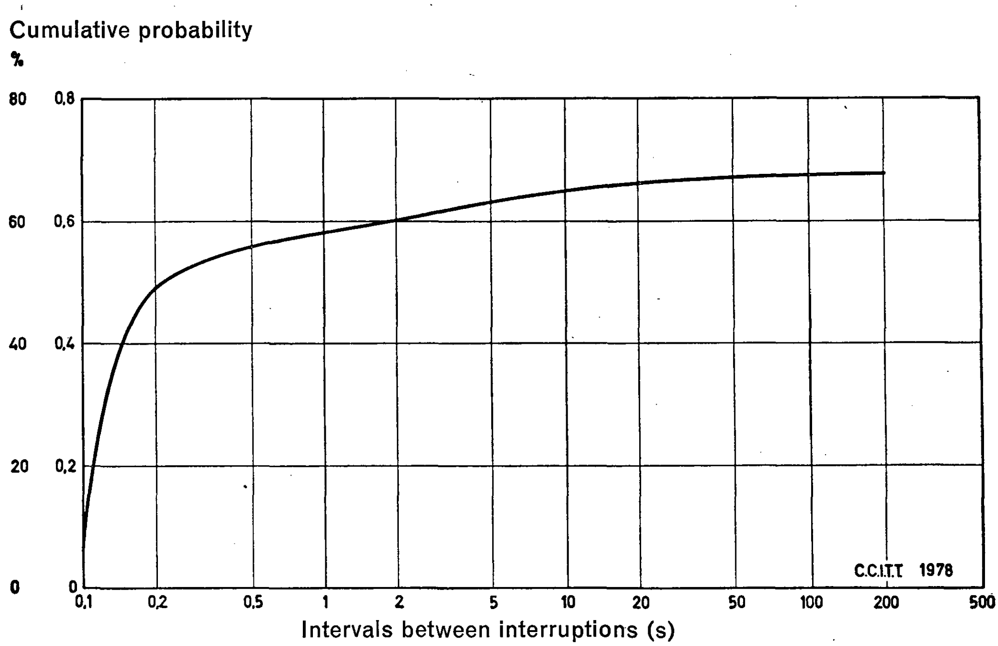

FIGURE 3. — **Example of a function of distribution of intervals between interruptions**

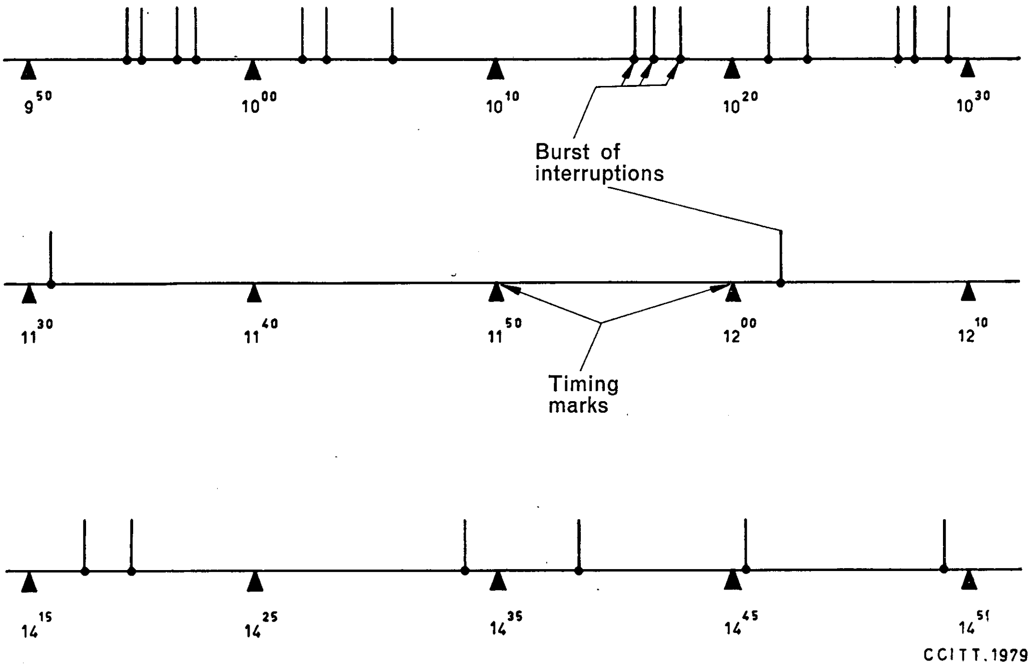

FIGURE 4. — **Examples of distribution of interruption bursts in time**

This circumstance may be very useful since the stationary Poisson flow can easily be assessed by its sole parameter λ. Even the short measurement of this parameter enables to assess with a certain degree of probability the state of a channel within a stationary period lasting many hours.

The use of similar procedure to measure the flow of interruptions that have not formed bursts does not enable to predict the flow intensity in the same easy way because the flow is unstationary.

6. The determination of interruption burst distribution presents certain interest. It may be obtained by the measurement of two characteristics:

— interruption number distribution in a burst;

— burst duration distribution.

These two distribution functions plotted from the results of measurements of long-distance cable channels on trunk lines in the U.S.S.R. are shown in Figures 5 and 6.

Number of interruptions in bursts was determined with the help of the device described in Annex II whereas the duration of bursts was measured with the help of the device described in Annex III. In all cases the burst-build-up criterion adopted was equal to 10 seconds.

As seen from Figure 5, the number of interruptions in bursts is sufficiently large (approximately 30% of bursts have more than 5 interruptions). Figure 6 indicates that the duration of bursts is considerable (approximately 30% of bursts are longer than 10 s).

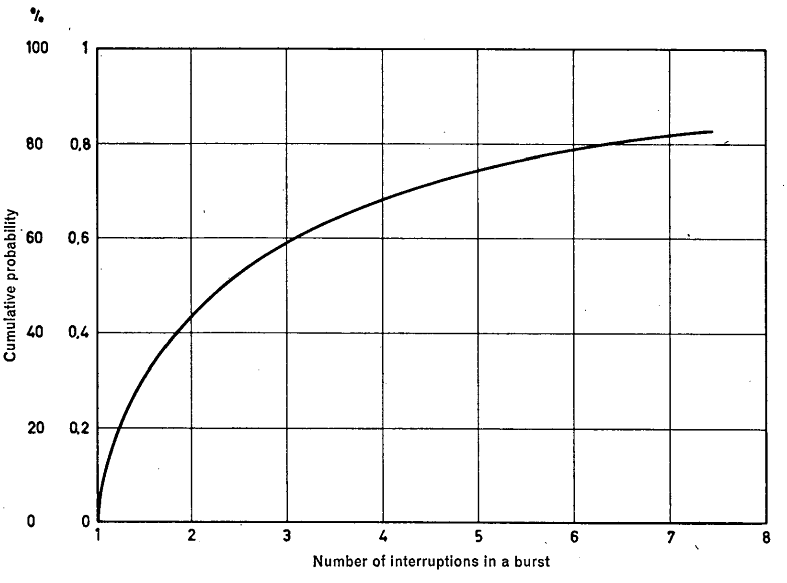

FIGURE 5. — Example of a function of distribution of interruption number in a burst

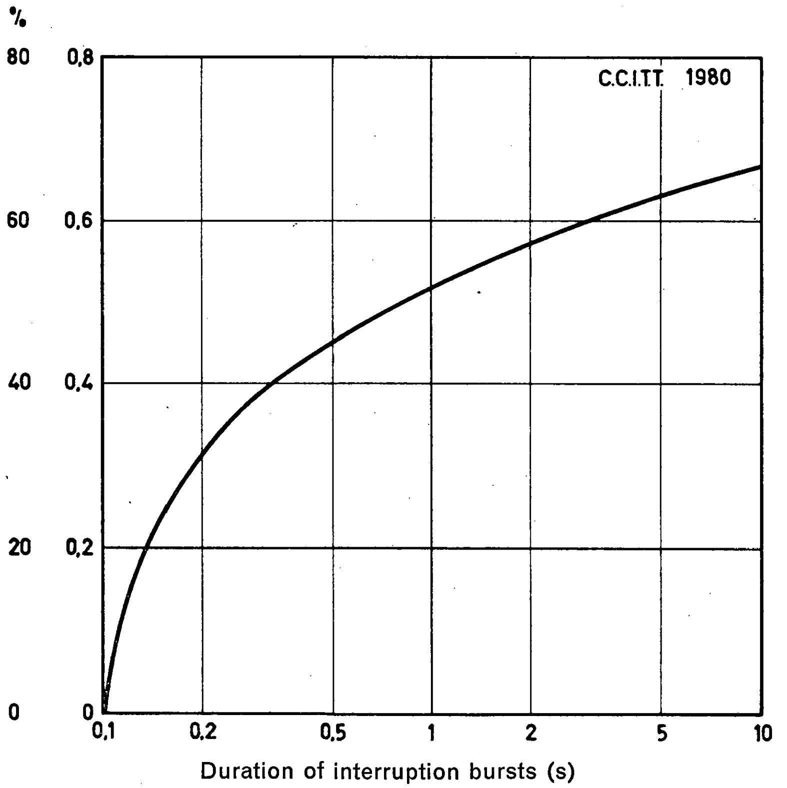

FIGURE 6. — Example of a function of distribution of duration of interruption bursts

ANNEX I

Interruption analyser

The analyser is intended for the study of the distribution of interruption in telephone channels. The analyser makes it possible to obtain the distribution of values of signal level variations and the distribution of level reduction duration at one of the thresholds.

The device provides for multichannel application. The measurement results appear in binary notation on an electronic display which contains 20 electronic counters. Every counter can register up to 2^{20} pulses. Measurement results are photographed at the operator's will or automatically with an interval of 30 minutes or under the action of an overflow signal from one of the counters.

The analyser (Figure 7) comprises three principal units: level reduction analysing unit, duration analysing unit and counter unit.

The principles of analyser operation are as follows. Pilot signal passes from the channel to the input of the amplifier A_1. The amplified signal is applied to the modulator M where the signal frequency band is translated in the frequency band of 16.6 to 19.2 kHz. After detection by detector D and filtration by filter F_2 the signal reaches the threshold circuits TC_0—TC_7.

Signal band translation makes it possible to detect easily even the shortest interruptions (0.3 ms and less).

Triggers Tr_1—Tr_7 together with threshold circuits and NOT circuits give shape to the pulses being equal to the period of pilot level reduction. These pulses are applied to the first inputs of AND_1—AND_7 circuits, while the pulses from oscillator O_2 are applied to their second inputs. From AND circuit inputs the pulses pass on to the counters C_1—C_7 which register the total time of pilot level reductions below each of the thresholds.

The duration of level reduction is analysed at the output of one of the AND circuits. Switch S_1 selects the threshold analysed. Pulses from the AND circuit output are applied through gate G_1 to the register R and simultaneously switch off the reset pulses from register R by means of gate G_2.

Under the action of pulses the register prepares in succession AND_{12}—AND_{24} circuits. The number of pulses determines the duration of a period when the threshold is exceeded and defines the index of the AND circuit prepared as a result of the arrival of these pulses. At the end of the period when the threshold is exceeded, the pulses at the AND circuit output disappear. As a result the gate G_2 is on and the first pulse from the oscillator O_2 resets the register R to zero. Reset pulses, after their passage through the AND circuit, are registered by counters C_8 to C_{20}.

If the number of pulses applied to the register exceeds its capacity the register is stopped by AND_{25} circuit and G_1 until the arrival of a reset pulse.

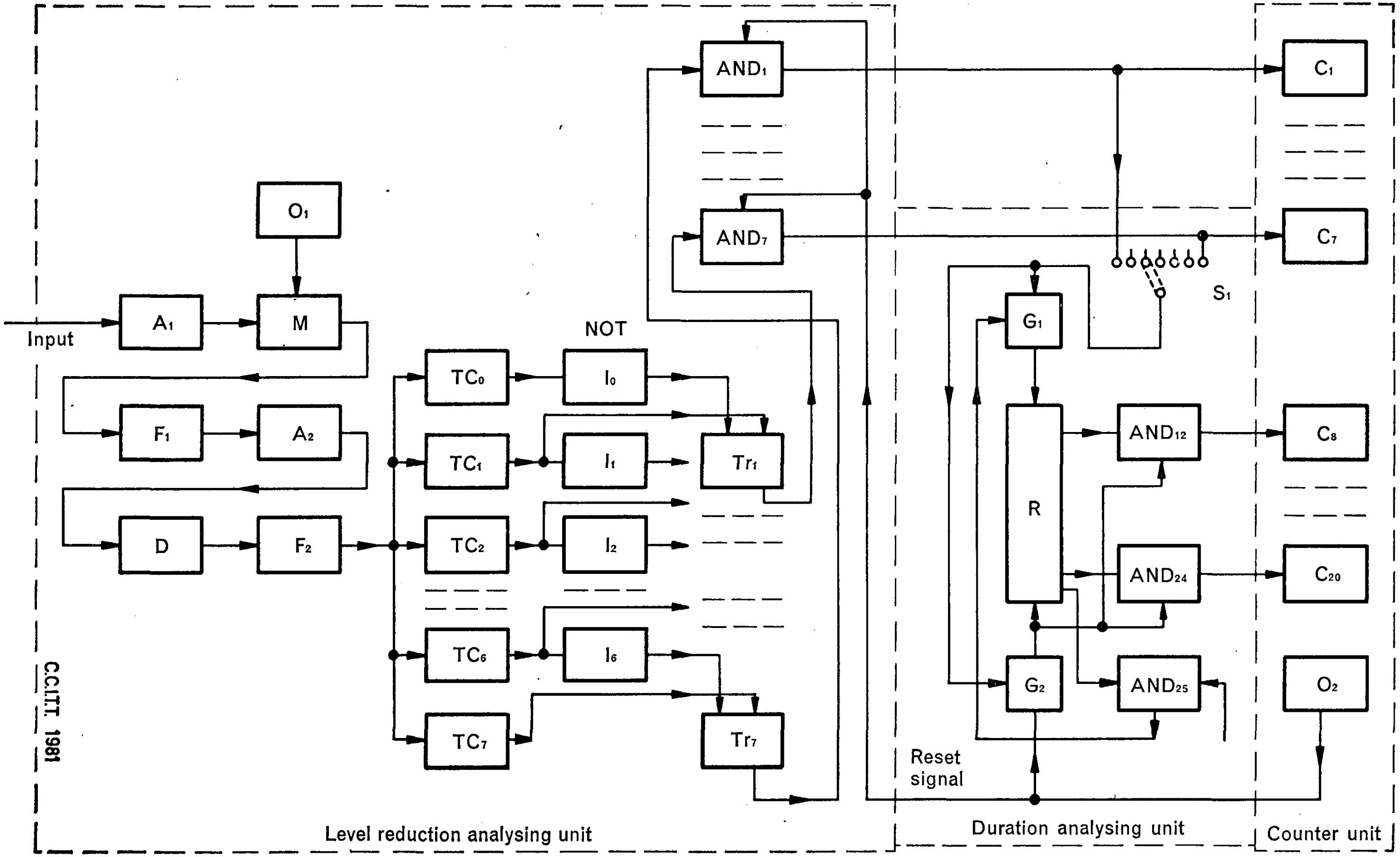

FIGURE 7. — Interruption analyser block-diagram

ANNEX II

Device for measuring the duration of interruptions and intervals between them

To obtain the distribution of duration of interruptions and intervals between them, a device was used in conjunction with a 20-channel electronic display described in Annex I.

Figure 8 shows the block diagram of the device.

When there is a pilot level reduction, the amplitude selector shapes a pulse which is applied through the differentiating circuit DC to the squaring circuit SC. The squaring circuit makes it possible for the first trigger of the commutator to be driven by both the leading and trailing edges of the pulse generated by an amplitude selector. The commutator comprises binary counters (triggers Tr_1—Tr_5) and a decoder.

The pulse corresponding to the moment of decrease in signal amplitude below the selected threshold passes to the input of the first trigger Tr_1. The trigger turns over. The states of triggers Tr_1—Tr_5 indicate the arrival of one pulse.

The states of commutator triggers are determined by a decoder. From the decoder output pulses are applied to the AND circuit inputs. Timing frequency pulses are permanently applied to the second inputs of AND_1—AND_{20} circuits (with pulse repetition rate equal to 300 ms); after passing the corresponding AND_1—AND_{20} coincidence circuits these pulses are applied to the pulse counters. At the end of interruption the pulse corresponding to the moment of pilot level recovery in the channel is applied to the input of trigger Tr_1 from the amplitude selector output. At this moment the potential is applied from the decoder circuit to the AND_2 circuit. Counter 2 begins the counting of timing pulses. Counter 1 stops the counting.

Thus, using the first counter reading and knowing the timing pulse repetition rate one can determine the interruption duration; using the readings of the second counter it is possible to determine the intervals between interruptions. The counter capacity is 2^{20}. Total number of counter is 20.

The circuit operates similarly when the next interruption occurs. Counter 2 stops its work whereas counter 3 begins counting.

At the moment of arrival of the 21st pulse the display is photographed, triggers and counters are reset. The display comprises a clock the readings of which are photographed.

When one of the counters is overflowed in the case of a long interruption or long interval a photograph is taken. Under these circumstances triggers and counters are reset by a counter overflow signal. The period of time from the moment a photograph is taken up to the moment the device is ready for work is approximately 10 s.

The camera used has an automatically operated shutter for 10-15 pictures.

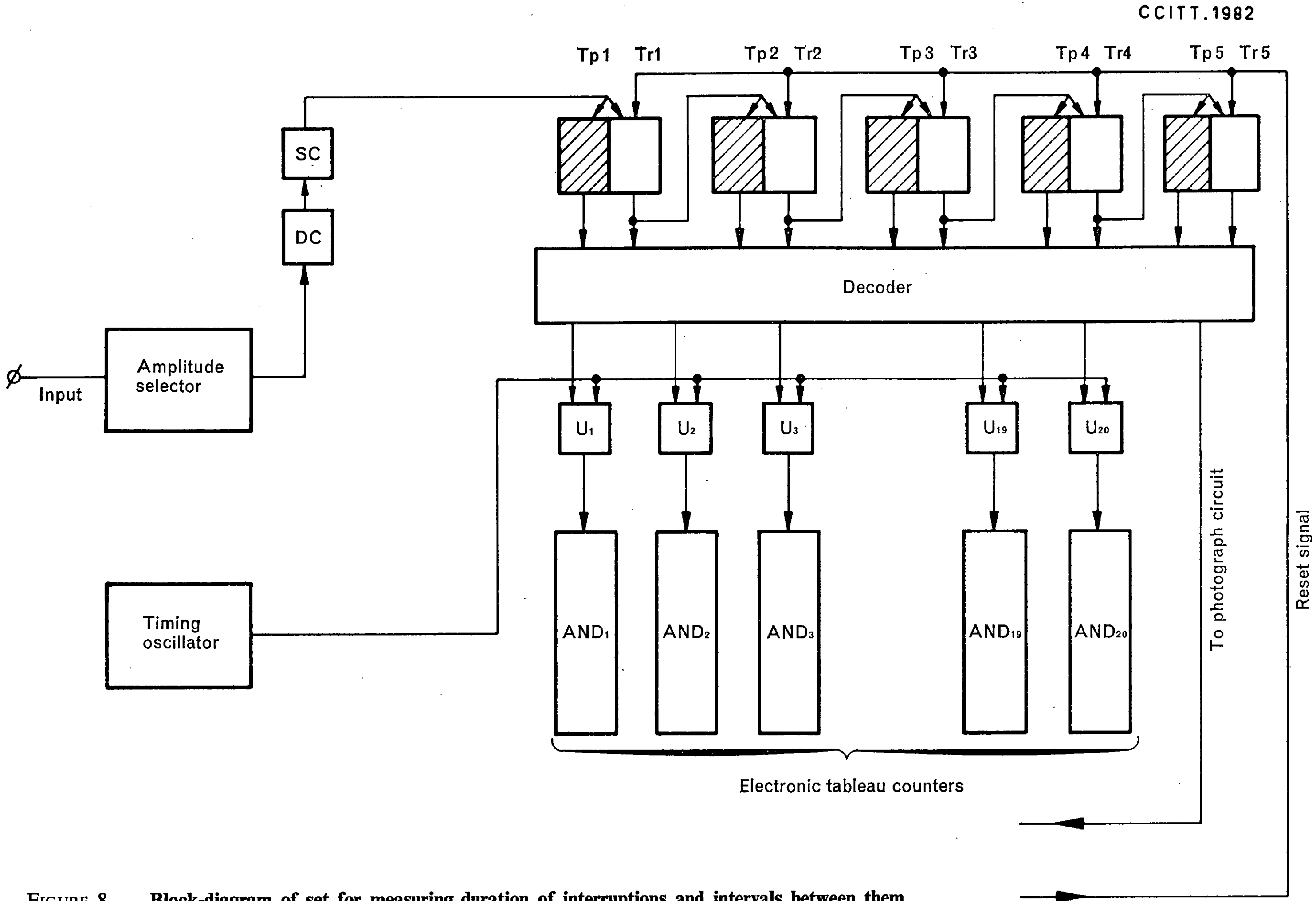

FIGURE 8. — **Block-diagram of set for measuring duration of interruptions and intervals between them**

ANNEX III

Interruption burst recorder

To enregister bursts of interruptions a self-recording unit with a sufficiently high tape speed and with low inertia of recording instrument is employed. The function of shaping a signal which is applied to the self-recording unit is performed by an interruption burst shaping circuit.

By an interruption burst we mean a sequence of interruptions in which the interval between two successive interruptions does not exceed the prescribed value (the so called burst-build-up criterion).

Block diagram of the device is shown in Figure 9.

The input signal is applied to an amplifier and then to a threshold circuit. When the reduction of the signal level is more than 2 Np during the time period covering more than 0.5 ms the threshold circuit is operated. A Schmidt trigger shapes the pulse the duration of which is equal to the level reduction duration in the channel.

From the output of the Schmidt trigger the signal is applied to the delay network which gives shape to a signal having the duration $t+\tau$, where:

t = interruption duration,

τ = burst-build-up criterion.

As a result of this, separate interruptions are grouped into interruption bursts. From the output of the Schmidt trigger the signal leading edge is applied to the second delay network which introduces a delay $\tau_1 = 0.3$ s. From the output of the Schmidt trigger and delay network τ_1 the signals pass to the input of time comparator. At the comparator output a signal appears having a delay equal to 0.3 s. The two signals received through resistors R_1 and R_2 are applied to the amplifier input and then to a self-recording unit.

As a result of circuit operation, a sequence of interruptions every one of which does not exceed 0.3 s. is converted in a signal with amplitude A. If there are interruptions of more than 0.3 s., the signal amplitude at the amplifier output reaches the value 2A in 0.3 s.

Thus, with interruptions of less than 300 ms, a step of unit amplitude is registered on the tape, which is an indication of a burst of short interruptions, whereas with interruptions of more than 300 ms the amplitude of a step is doubled.

Duration of bursts of interruptions and the moments of their occurrence may be determined with the help of the tape with an accuracy which is determined by the tape speed plus certain irregularity of its feed.

If necessary, time marks may be recorded on the additional track of the self-recording unit.

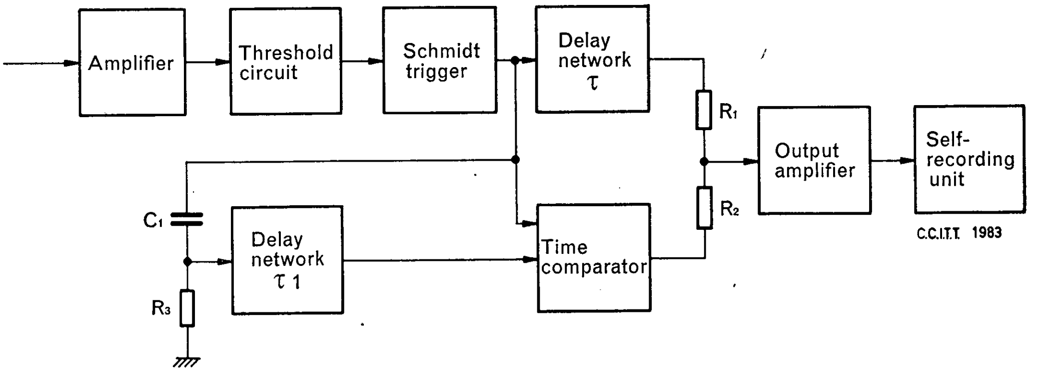

FIGURE 9. — **Block-diagram of the device for registration of interruption bursts**

SIEMENS A.G. — *(Contribution COM Sp. A—No. 98—February 1967)*

AUTOMATIC ORIGINATING AND ANSWERING OF CALLS IN THE TELEPHONE NETWORK

Summary

In view of the widespread use of dial switch selection, a simplified automatic calling equipment is put forward for discussion for this type of application. In this case the time sequence of the dial pulses is directly controlled by the data terminal equipment and, in contrast to the any-number equipment, this simplified version results in a considerable reduction of the equipment complication. Furthermore, less interface circuits are required. The functions attending the dialling procedure as well as the functions of the interface circuits will not be subject to major alterations.

In addition, it is pointed out that our experience also has shown that in certain cases the equipment requirements can be reduced by the use of one-number automatic calling equipment.

A. *Any-number A.C.E. with serial presentation of the digit signals*

1. In the Report on the Brussels meeting of Special Study Group A the operation and the interface of an automatic calling equipment (A.C.E.) are described. This A.C.E. permits selection of any desired number, the signals interchanged via the interface being independent of the selection method used on the telephone line.

2. Because of the widespread use of dial switch selection, it seems justified to consider a simplified version of the A.C.E. for this special application: the time sequence of the dial pulses is directly determined by the data terminal equipment and passed on to the A.C.E. via one circuit. For this purpose, circuit NB 1 can be used; circuits NB 2, NB 4 and NB 8 are not employed.

This results in a reduction of the A.C.E. equipment complication which considerably exceeds the reduction that could, for example, be achieved by the use of a one-number A.C.E. (see section B).

Such an automatic selection method has already been approved for national telegraph networks and is presently being discussed in C.C.I.T.T. Study Group Sp. A in connection with Question 1/A, Point F.

3. Even in the case of this simplified A.C.E., the functions attending the selection of a call largely correspond to those described in the Brussels report, Appendix I (pp. 115 to 117).

3.1 The data terminal equipment is responsible for making the decision to place a data call.

3.2 Upon receipt of a call request, the A.C.E. requests a telephone line from the telephone office.

3.3 The telephone office provides the A.C.E. with an indication that a line has been provided (for instance, dial tone).

3.4 The A.C.E. indicates to the data terminal equipment that a line has been provided for the data call by requesting that the dial pulses are presented on the interface signalling lead.

3.5 The data terminal must know the telephone number of the called station and be able to present the dialling pulses in proper time sequence to the A.C.E. The data terminal must also know when the number contains subsets between which are separations (for instance, second dial tone) and insert appropriate time delays.

3.6 The A.C.E. accepts the dialling pulses from the data terminal equipment and converts them to a form electrically suitable for signalling on the telephone line.

3.7 The data terminal equipment must know when all dialling pulses in the telephone number have been presented to the A.C.E.

3.8 The following functions are the same as those described in the Brussels report, page 116, points 10 to 16.

4. The interface used for the simplified A.C.E. (Figure 1) corresponds to that shown in the Brussels report, Appendix II (pp. 119 to 133), with the exception of the following modifications:

- The possibility of terminating the selection by transmitting the EON code combination, as mentioned under paragraph 1.5.2, is modified (see circuit DPR);
- the four digit signal circuits are reduced to one circuit for dial pulse control;
- the definitions of circuits PND and DPR are extended.

In the case of operation with the simplified A.C.E., the following definitions apply for these circuits:

Circuit NB 1—Digit signal

Direction: To the automatic calling equipment.

The signals transmitted over circuit NB 1 are directly used for keying the loop current in the telephone line.

In the ON condition, the loop is interrupted and in the OFF condition, the loop is closed. The signals offered over circuits NB 1 are interpreted by the A.C.E. only if DPR is simultaneously in the ON condition.

Circuit PND—Present next digit

Direction: From the automatic calling equipment

The ON condition on circuit PND indicates that the telephone line requested by the ON condition on circuit CRQ has been through-connected to the exchange and that the desired subscriber number can be dialled. The ON condition is maintained until circuit CRQ has been restored to the OFF condition.

Circuit DPR—Digit present

Direction: To the automatic calling equipment

The ON condition on circuit DPR indicates that dial pulses are transmitted over circuit NB 1. The ON condition must be applied before the first dial pulse (OFF condition) is transmitted over circuit NB 1 and must be maintained until the last dial pulse has been transmitted. During the rest of the time, circuit DPR is held in the OFF condition.

While the ON condition prevails on circuit DPR, the v.f. signal detection feature in the A.C.E. is short-circuited so that the input resistance on the circuit side of the A.C.E. is suitable for dialling purposes.

When circuit CRQ is in the ON condition while the OFF condition prevails on circuit DPR, the d.c. input resistance on the circuit side of the A.C.E. is 130 to 200 ohms. The impedance measured as balance return loss across 600 ohms is $\geqslant$ 13 dB within a frequency range of 300 to 3400 Hz.

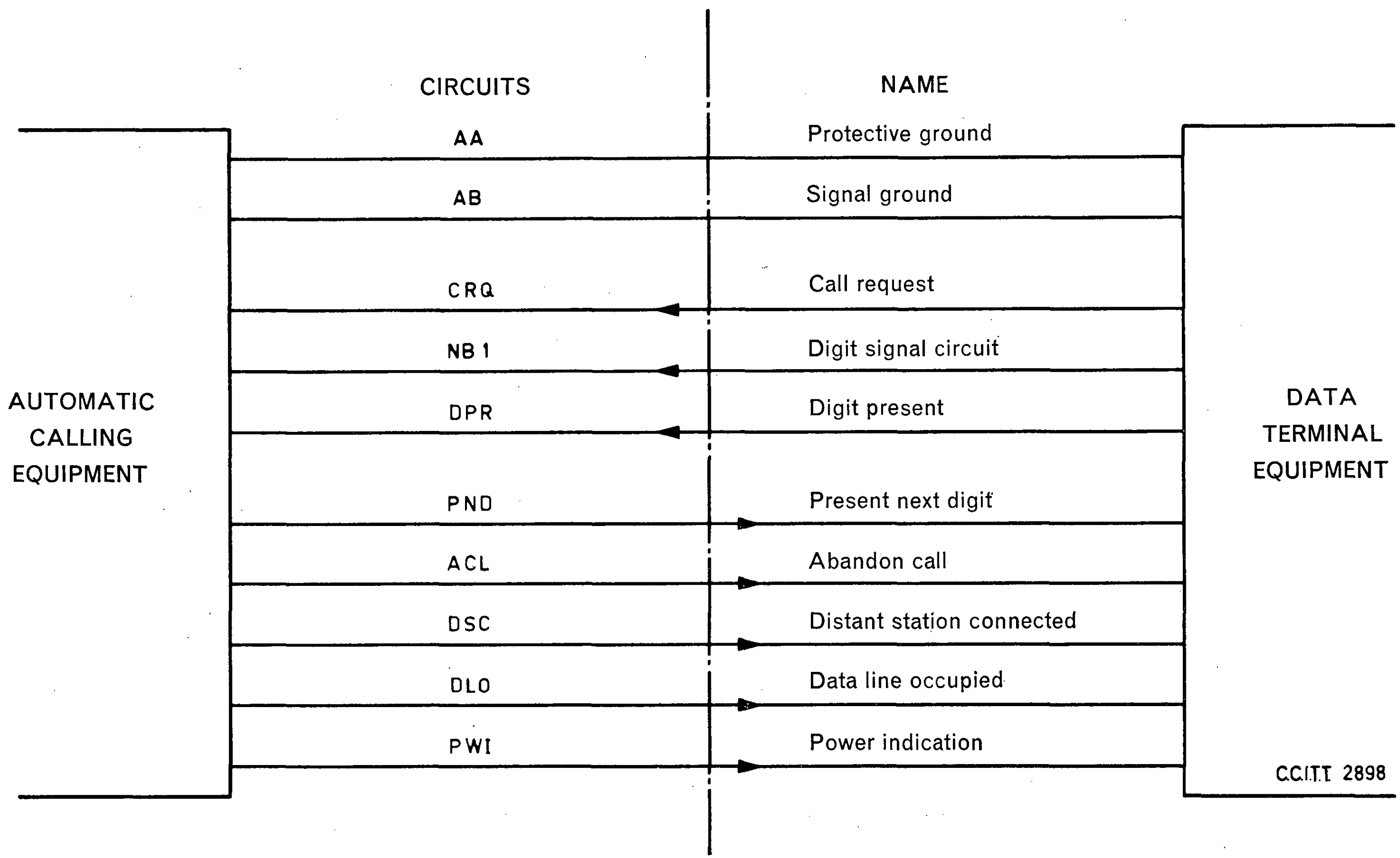

FIGURE 1. — **Signal interchange point**

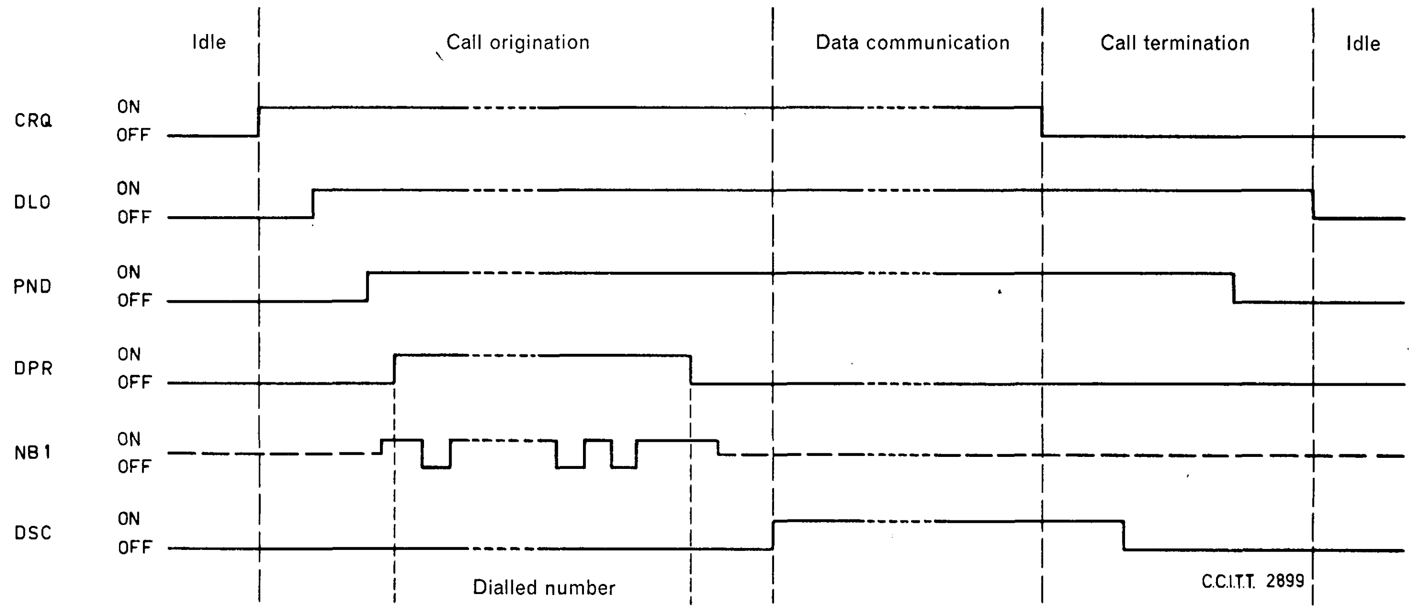

FIGURE 2. — **Conditions of signal interchange circuits on an automatically originated call with answer detection and where circuit CRQ is used to terminate the call**

The transition from the ON to the OFF condition on circuit DPR can be interpreted by the A.C.E. as an EON criterion.

The signalling conditions occurring on the interface circuits are shown in Figure 2.

B. *One-number A.C.E.*

In the C.C.I.T.T. *Blue Book*, Volume VIII, page 707 et seq., the repertory-number A.C.E. and the one-number A.C.E. are described in addition to the universal any-number A.C.E. Both types use the same calling procedure as the any-number A.C.E. but do not call for a digit-by-digit pick-up and have a simpler interface.

Our experience has shown that there are simple stations at the periphery of data systems where the use of a one-number A.C.E. reduces the equipment complication of these stations as compared with the use of an any-number A.C.E.

Supplement No. 43

ISO/TC 97/SC 6. — *(Contribution COM Sp. A—No. 203—June 1968)*

DRAFT I.S.O. RECOMMENDATION FOR: ASSIGNMENT OF CONNECTOR PIN NUMBERS FOR INTERCHANGE CIRCUITS BETWEEN DATA TERMINAL EQUIPMENT AND DATA COMMUNICATION EQUIPMENT WHERE C.C.I.T.T. RECOMMENDATION V.24 APPLIES

Scope

This recommendation applies to the interface between data terminal equipment and data communication equipment either modems or automatic calling equipment.

Although the connector still remains to be the subject of a recommendation by IEC/TC48 it is generally accepted to be a 25-pin connector with separate connectors provided for the modem (or telex service) and for the automatic calling equipment interfaces. The male connector (plug) is associated with the data terminal equipment and the female connector (socket) is associated with the data communication equipment.

Pin assignments are recommended for the following equipments:

A. Modem to Recommendation V.21

B. Modem to Recommendation V.23

C. Modem for 2400 bits per second

D. Future modems

E. Telex

F. Other telegraph services

L. Automatic calling over telephone networks

M. Automatic calling over telex networks

Pin number	Circuit numbers in C.C.I.T.T. Recommendation V.24								
	Modem interfaces				Telegraph interfaces			Automatic calling interfaces	
	V.21 A	V.23 B	2400 bps C	Future D	Telex E	Other F		Telephone L	Telex M
1	101	101	101	101	101	101		212	212
2	103	103	103	103	103	103		211	211
3	104	104	104	104	104	104		205	205
4	105	105	105	105				202	202
5	106	106	106	106	106	106		210	210
6	107	107	107	107	107	107		213	213
7	102	102	102	102	102	102		201	201
8	109	109	109	109	109	109		F	F
9	N	N	N	N	N	N		N	N
10	N	N	N	N	N	N		N	N
11 Note 2	126	120	120	120				F	F
12		122	122	122				F	F
13		121	121	121				204	204
14		118	118	118				206	206
15		Note 1	113 or 114	113 or 114				207	207
16		119	119	119				208	208
17		Note 1	115	115				209	209
18	F	F	F	F	F	F		F	F
19	F	F	F	F	F	F		F	F
20	108/1 108/2	108/1 108/2	108/1 108/2	108/1 108/2	108/2	108/2		F	F
21								F	F
22	125	125		125	125	125		203	203
23		111		111				F	F
24 Note 2								F	F
25	F	F	F	F	F	F		F	F

N = Circuit reserved for national use

F = Circuit reserved for future international recommendation

Note 1: Where signal timing is provided in the data communication equipment to C.C.I.T.T. Recommendation V.23, pin No. 15 will be used for circuit 114 and pin No. 17 will be used for circuit 115.

Note 2: See Appendix A for proposed alternate use of pin Nos. 11 and 24.

The descriptions of the interchange circuits are given below for reference.

Circuit No.	Designation
101	Protective ground or earth
102	Signal ground or common return
103	Transmitted data
104	Received data
105	Request to send
106	Ready for sending
107	Data set ready
108/1	Connect data set to line
108/2	Data terminal ready
109	Data channel received line signal detector
111	Data signalling rate selector
113	Transmitter signal element timing (data terminal equipment source)
114	Transmitter signal element timing (data communication equipment source)
115	Receiver signal element timing (data communication equipment source)
118	Transmitted backward channel data
119	Received backward channel data
120	Transmit backward channel line signal
121	Backward channel ready
122	Backward channel received line signal detector
125	Calling indicator
126	Select transmit frequency
201	Signal ground or common return
202	Call request
203	Data line occupied
204	Data station connected
205	Abandon call
206	Digit signal (2^0)
207	Digit signal (2^1)
208	Digit signal (2^2)
209	Digit signal (2^3)
210	Present next digit
211	Digit present
212	Protective ground or earth
213	Power indication

APPENDIX A

The following change to the use of pins Nos. 11 and 24 is proposed:

— Modems to Recommendation V.23

— Modem for 2400 bits per second

— Future modems

a) Pin No. 11 shall be spare

b) Circuit 120 shall be placed on pin No. 24.

If this change is approved the draft recommendation will be modified accordingly.

Explanatory notes

Pin No. 11, which is allocated to circuit 120, is currently being used in some equipments for circuit 111. This dual use of pin No. 11 will be a cause of difficulty in implementing the proposed standard. It is therefore suggested that, to facilitate the use of the draft recommendation, pin No. 11 be made spare and circuits 111 and 120 be allocated to other pins.

In the draft recommendation circuit 111 is shown connected to pin No. 23.

Pin No. 24, which at present is spare, could be used for circuit 120. This arrangement would enable the draft recommendation to be incorporated in current equipment by strapping as follows:

circuit 120	strap pins Nos. 11 and 24
circuit 111	strap pins Nos. 11 and 23

This would be an interim arrangement until all equipment has been changed and pin No. 11 would become spare.

Supplement No. 44

CHILE TELEPHONE COMPANY. — *(Contribution COM Sp. A—No. 173—December 1967)*

USE OF DIGITAL TRANSMISSION

1. *Introduction*

The increasing scale of introduction of p.c.m. for voice transmission justifies urgent examination of the relationship of a digital network based on this mode to future handling of data, text and other non-voice information. It may be some years before comprehensive switched networks based on integrated digital transmission and switching are widely available, but incorporation on a considerable scale seems likely in the relatively near future.

2. *Economic impact of digital transmission*

The main motivation for use of this new mode for data is the potentiality for *very great reductions in the cost* of transmission and switching and the ready availability of the capacity to handle greatly increased bit rates.

The p.c.m. structure will be based on a voice channel with a digital transmission rate of 56 kb/s or thereabouts (ignoring any capacity added for signalling). To a first approximation this capacity is available for the cost of a speech connection.

For the lower-speed applications a derived low-speed network based on time division of the speech channel or even on by-products, such as spare signalling capacity, could readily be made available. This could provide, for example, a 2000-b/s capacity for the same order of cost as, or even significantly less than, that of the present 50-baud connection.

These gains in cost and some associated gains in performance and reliability are real but imply the acceptance of some changes in attitude to such issues as transparency. There is need to anticipate these new restraints and new relationships so that as the use of digital transmission increases the potential cost and performance gains may be realized with minimal disturbance to established practices. So far as possible the digital network should be so organized as to place the fewest impediments in the way of its use for non-voice traffic.

3. *Transmission aspects*

3.1 *Completely digital network*

With the p.c.m. class of transmission available data communication will be possible and economical at all sorts of speeds up to 1.5 Mb/s or more. The two most important categories are:

a) Data on a 56-kb/s switched voice network, i.e. voice channel slot assumed to have 8 bits/slot and 8-kHz repetition rate.

b) Data on a 2000-b/s switched network for telegraphy and low-speed data.

Figure 1 shows the essential form of either of these networks as they would appear in a network employing integrated p.c.m. throughout. Between terminal exchanges A and B there is a fully synchronous, full duplex transmission capability operating at 56 kb/s or 2000 bits/s. Distortion as such is not a factor in this network since the t.d.m. processes involved in both transmission and switching are inherently regenerative. There will be some finite error rate. More experience is needed to establish this but there is reason to hope for something of the 1 in 10^6 order. However, owing to imperfect clock correlation, there will be a risk of a new type of error (1 in 10^7, or better) in the form of the total omission or repetition of a bit. This will be referred to as slip. This main network also, by virtue of its fully synchronous mode, carries the effective equivalent of transmitted clock, i.e. no matter what the nature of the user intelligence the receiving terminal is constantly aware of the exact bit rate.

Finally, whatever the user information rate, transmission over the main network will be at 56 kb/s or 2000 kb/s precisely.

3.2 *Hybrid network*

Figure 2 shows a modification of this network in which between p.c.m. areas a link is included based on the traditional f.d.m. transmission. This could be literally f.d.m. (e.g. v.f. telegraph) or it could be t.d.m. derived from a wide-band capability such as 48 kb/s on the 60-108 kHz group. This network, which we shall call the hybrid network, presents additional interface problems at the p.c.m./-f.d.m. and f.d.m./p.c.m. boundaries. This type of case is inevitable during a long transition phase and implies certain added restraints which are considered in section 5 following.

3.3 *Subscriber line transmission*

The tails or subscriber lines are individual channels, not multiplexed. Transmission can be arranged in a variety of modes ranging from relatively conventional telegraph modes to such methods as dipulse or phase-modulated base-band. The latter guarantees a high clock content and would be particularly useful for higher-speed data.

Consideration is also required of such questions as the choice between full duplex and switched simplex working.

The main problem, however, is that of the speed restraints involved in the interfacing with the main t.d.m. synchronous network.

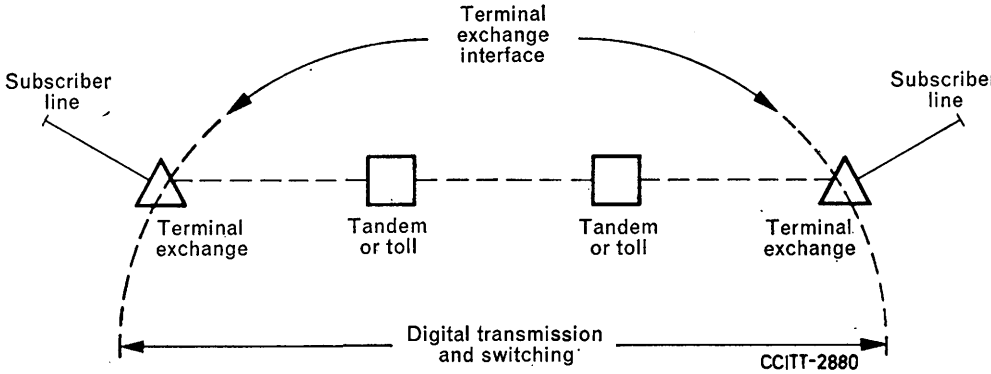

FIGURE 1

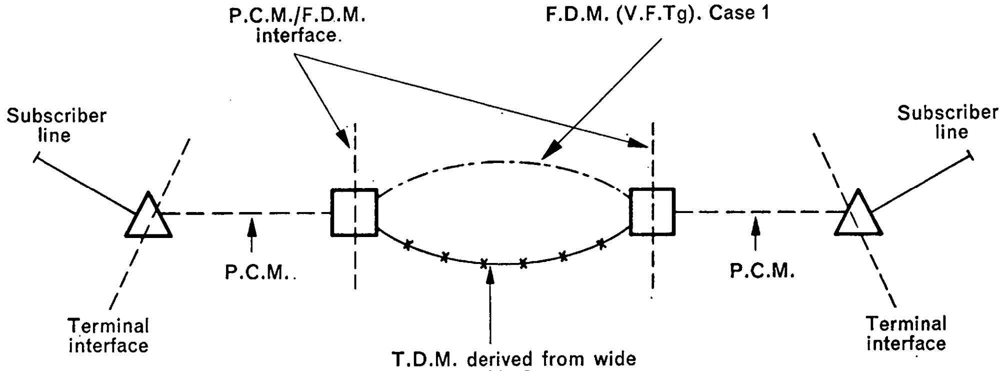

FIGURE 2

4. *Organization of the subscriber line/local exchange interface*

4.1 *Switched voice network*

In this case the network transmission is not continuous but consists of transmitting 7-bit characters at the sampling rate of the network; 8 kHz has been assumed. A frame, inherent in the system, is therefore imposed on the data for their transmission over the network. This framing facilitates the handling of certain problems related to the control of redundancy.

Data speeds may be handled which have no integral relationship with 56 kb/s. However, there are two reservations.

Firstly, occasional slip occurring in the network can cause loss or duplication of data unless the data speed is sufficiently low that there is sufficient redundancy to control errors due to this cause.

Secondly, the final delivery to the receiving party is dependent on a speed-controlled oscillator, or equivalent device, settling down to a rate determined by the mean rate of arrival of true information bits. If delivery at a steady rate is to be ensured this requires fairly long time constants and the initial settling down period could be disadvantageously long for switched connections.

The speed stabilization difficulty indicates the desirability of not allowing, for general use, an indefinite range of speeds but confining general use to a few selected speeds. Interface units would thus have to cover slight clock discrepancies but not wide ranges. In this way a wide range of users may be satisfied by relatively simple cheap interfaces. Special users could still be accommodated but they would bear the cost of the special interface units needed for non-standard speeds.

Three very suitable modes of data transmission over this network can be identified:

1. Synchronous transmission at 48 kb/s, data terminal stations receiving timing signals from the data network. Data are transmitted over the network in the form of 6-bit "characters", a seventh bit in each time slot being used in a fixed pattern, e.g. 00110011..., to indicate the occurrence of slip. Errors due to slip would still occur but phase could be held.

2. Isochronous transmission at a nominal rate of about 48 kb/s, data terminal stations using independent timing. The number of data bits inserted in each character is variable, the seventh bit being used as a base for a redundancy marking pattern which indicates the actual number of data bits.

3. Isochronous transmission at a nominal speed of about 16 kb/s. Six data bits are sent in each character, this being repeated three or more times. The seventh bit is used (alternately 0 and 1) to indicate the character repetitions. This mode is immune to errors caused by slip. The triple redundancy also provides a means of control of ordinary transmission errors by majority vote correction.

4.2 *Telegraph and low-speed data switched network*

The 2000-bits/s network is isochronous and has no natural frame which is in any way determined by the system. This fact influences the particular way in which the interface between the subscriber's line and the data network is realized. However, the same problems of slip and of accommodating data speeds not integrally related with the natural speed of the network exist as in the case of the 56-kb/s network.

Data speeds up to 200 b/s. — It is to be expected that initially, at any rate, a high proportion of users would fall in this category, using telegraph type machinery at 50, 100 or 200 bauds. At these speeds a completely asynchronous interface is feasible. At 200 bauds, for example, each unit of data corresponds to 10 system bits. If there is less than 10% overall distortion on the sending subscriber line each data bit is represented on the system by either 9, 10 or 11 bits in patterns which will mean some transitions involving no distortion and some 10%. With 5% distortion on the receiving line the resultant 15% overall would be satisfactory. It would still be satisfactory if on rare occasions a further 10% is added by slip.

Lower speeds still, e.g. 100 or 50 bauds, would be even more secure.

Data speeds in the range 200-500 b/s approx. — In this range the distortion build-up due to slip is more damaging. However, the redundancy is still sufficient for the effects of slip to be controlled so long as there is an integral speed relationship between the nominal speed of data transmission and 2000 b/s. Thus at 400 b/s the interface unit would count 5 bits between each possible transition. To allow for minor speed discrepancies it could quite cheaply be arranged to use a modified count of 4 or 6 at infrequent intervals when it is established that a sufficiently marked discrepancy has arisen between mean transitions incoming from the subscriber's line and regenerated. This could produce an assurance of mainly zero distortion with occasional shifts of 20%. If the subscriber accepts the system clock there will be no distortion at all. It should be noted that this method implies isochronous transmission. This is not consistent with present-day start-stop based on mechanical machines but it could be an aim even for start-stop on a next generation of machines.

Data speeds in the range 500 b/s to 2000 b/s — Users who can tolerate the effects of slip use the system at 2000 kb/s provided they are willing to accept the system clock.

If they cannot accept the system clock then interface units which can create and recognize a frame of variable length or variable length of information content would be needed. The speed would have to be dropped to allow room for the redundancy needed to define this frame. The net result of this is likely to be an interface cost disproportionate to the very cheap transmission and switching which is a primary aim.

There are considerable arguments, therefore, in the case of data transmission above 200 b/s, for data transmission to be limited to those rates which have an integral relationship with 2000 b/s and for data terminal equipments to accept timing from the network.

5. *Transitional phase of a hybrid network*

Despite the present interest in digital communication techniques a homogeneous network based on integrated digital transmission and switching such as was discussed in the foregoing will not materialize for some considerable time.

A transitional phase is envisaged where certain links of the 56-kb/s and 2000-b/s network are provided over conventional facilities. A single 48-kb/s link would, according to present concepts, correspond to a comfortable fill of the 60-108-kHz group link. For the 2000-b/s network 200-b/s links could be provided either by conventional v.f. telegraph means or by time division of relatively wide-band units, e.g. 240 channels from a 48-kb/s modem link. Alternatively, a 48-kb/s link could provide 24-2000-b/s channels.

Such a hybrid network is shown in principle in Figure 2. Two modes are illustrated one in which the analogue network links use f.d.m. (e.g. v.f. telegraph) and the other in which t.d.m. is used to derive an appropriate speed of bit stream from a higher capacity, e.g. telegraph channels from a 48-kb/s stream on a 60-108-kHz group link. The general problems are as for the homogeneous digital network discussed in section 4 above but are accentuated by the additional interfaces which are required.

48-kb/s network. — For this traffic the proposed mode of usage of the p.c.m. network admits of small speed adjustments by manipulation of redundancy. If f.d.m. derived links have the same capability either by pure f.d.m. methods or by a "stuffing" process in t.d.m. derivation there are no major new problems created by the hybrid situation.

2000-b/s network. — Presumably the dominant users of this network would be operating at 200 b/s. It might be reasonable, in that case, because of economic considerations that the links derived from conventional f.d.m. facilities be limited to 200 bauds. There would thus be geographical limitations, during the transitional stage, on the use of the network at higher speeds.

If the links provided were 200 bauds v.f.t.-type circuits there would be no interface problems but the cumulative distortion encountered would be greater than for the completely digital network. However, regeneration at the interconnection points could be effected on the basis of ten system bits to one user bit, slight speed discrepancies being adjusted by an occasional count of 9 or 11 instead of 10. To achieve this on a code insensitive basis would demand an isochronous stream at a rate of 200 bauds or one integrally related to it. This is inconsistent with conventional start-stop but could be organized at source on future machines and be arranged at the initial subscriber line interface unit for existing machines.

The use of 200-b/s t.d.m. links would reintroduce problems of synchronization requiring more complex interface units. In addition, there would be risk of additional errors due to slip. These could be alleviated, however, if the t.d.m. links were somewhat faster than 200 b/s so as to permit the accommodation of small speed variations by means of a "stuffing" technique. The cost of this could be held to quite a low figure provided the operating speeds were limited to a nominal 200 b/s and simple fractions thereof. An alternative, more extravagant in bandwidth, would be to continue, for 200 bauds operation, to employ "bulk redundancy", i.e. to make the channel speed, say, 600 b/s. Here again, if code sensitivity is to be avoided, close speed regulation would be needed to avoid ambiguous "slip" when long successions of marks or spaces are transmitted.

The use of this t.d.m.-based network by present-day alphabet No. 2 50-baud terminal machines would demand code sensitive regeneration at the subscriber line/network interface to produce a bit stream isochronous at 200 b/s.

Where in order to provide the possibility of services faster than 200 b/s, links of 2000 b/s are provided, either by means of f.d.m. or t.d.m. techniques, the interface for 200-b/s connections could be asynchronous. For the 2000-b/s service the interface problems will be similar to those discussed above for 200 bauds operation over 200-baud intermediate links.

6. *Summary of conclusions*

The main conclusion is that, by various special processing techniques, a wide range of data speeds could be accommodated over the digital network which is in course of provision for voice purposes. However, in the interests of economy, simplicity and high performance, it is most desirable to confine general usage to certain selected speeds and to hold these speeds in close relationship to certain subdivisions of the basic system speed. In the ultimate, users should be encouraged to achieve a perfect match by using the system clock.

The modes in which data transmission might most naturally be effected over this network are:

Voice band network

a) 48 kb/s nominal (say 47.99 $\pm$ 0.01 %)
The 7th bit of each character is used for occasional indication of a reduction from 6 to 5 user bits.

b) 48 kb/s synchronous using system clock.
Using 7th bit with 00110011 pattern to indicate slip. Information would occasionally be lost but phase could be held. (Possibly most appropriate for facsimile.)

c) 16 kb/s nominal.
Triple redundancy is used for majority vote correction of random errors and for guard against slip. (7th bit would alternate between 1's and 0's on successive *user* characters.)

2000-b/s network

a) Use synchronously at 2000 b/s for users who can tolerate occasional slip.

b) Use at 200 bauds with quasi regeneration. Start-stop to be modified if need be to an isochronous mode. User should preferably use system clock or the initial user interface unit should execute isochronous regeneration.

c) 50 bauds. Use the 200-b/s network with interface regenerators where necessary.

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SOME PRINCIPAL CONSIDERATIONS CONCERNING THE FUTURE DEVELOPMENT OF A SWITCHED NETWORK FOR DATA TRANSMISSION

1. *Objections against existing possibilities*

An objection which can in general be raised against the use of the public telephone network for switched data transmission connections is that use is made of equipment which, both from the transmission technique and the switching technique point of view, was typically designed for telephony. The designers of the equipment aimed only at analogous transmission of the power spectrum of speech signals. Form fidelity in signal transmission was hardly taken into consideration. With regard to transmission the following points are of special importance:

a) The slight, but for pulse transmission very inconvenient, frequency drift occurring when transmission takes place via carrier circuits.

b) Differences in attenuation, group delay distortion, and loss/frequency distortion in different circuits.

c) The restriction with regard to the use of the frequency band imposed by signalling tones and pilots.

As concerns exchange techniques, apart from the fact that the interference levels are often rather high for data transmission, the exchanges have been typically designed for setting-up telephone connections between people, audible signals being used in various stages of the setting-up of the connection. It is difficult, if not impossible, to adapt the properties of the equipment to data transmission requirements.

As a result of the earlier-mentioned transmission properties of the telephone network, data modems are rather complex and therefore expensive apparatus, which in the present system of data transmission via the public telephone network are required per data-subscriber.

Some of the disadvantages mentioned above for the telephone network also exist in the telex network, be it only as far as transmission is concerned. Up to rather large distances the price of a telex line is therefore almost entirely determined by the multiplex telegraph terminal equipment.

2. *Some demands and desires with regard to a future switched data network*

In view of the disadvantages of the use of the telephone transmission path, it is desirable to build up a special network, the word network meaning the total of transmission and switching equipment. In doing so the properties required for digital signals can be fully taken into account. A point of particular importance is the demand to keep the transmission from subscriber to first concentration point as simple as possible (direct current transmission). Further, for multiplex techniques time division seems the cheapest solution, which also opens up good possibilities for the switching technique.

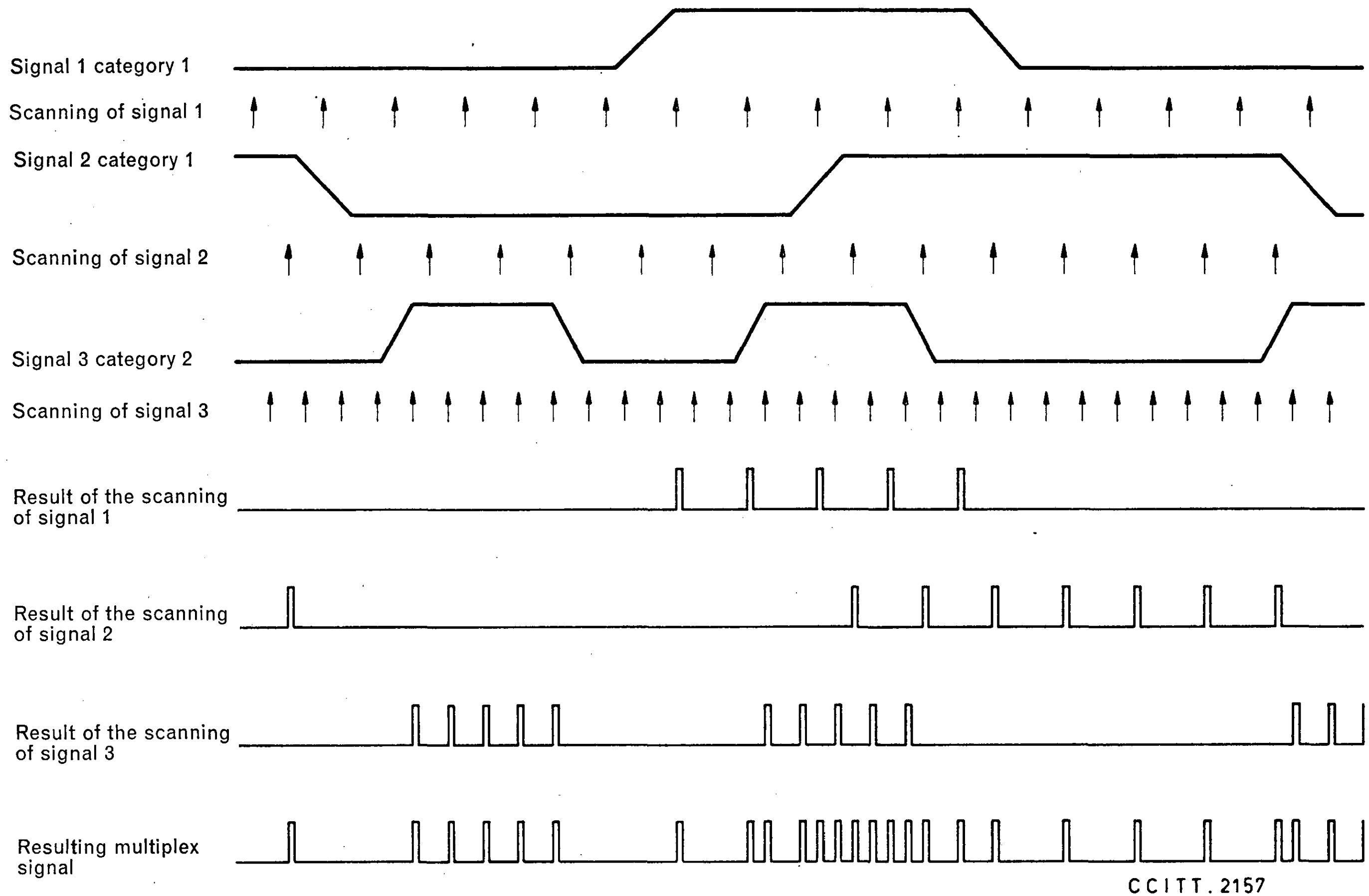
As the data network will probably be used by a great variety of customers, various categories of speed must be enabled to gain access to the transmission system. This may or may not be effected by means of the same switching equipment.

The subscribers must be enabled to use any speed below the maximum speed fixed for the category to which they belong.

3. *A possibility for meeting the demands and desires mentioned in section 2*

A possibility for a separate digital network may be found in special digital lines, the telephone carrier terminal equipment being bypassed. If coupling through modulation with the telephone carrier network should be inevitable, it should at any rate only occur on such a network level that a proper amount of multiplexing and possibly reduction has been achieved. In that case the complexity of the modulation needed does not weigh very heavily, calculated per subscriber.

Avoiding the telephone carrier terminal equipment does not necessarily mean that the same line repeaters as for telephony cannot be used. It might then be desirable that the C.C.I.T.T. should



Principle of multiple scanning

reserve in the line-transmission systems a separate (modest) place for digital transmission beside telephone transmission.

If the introduction of p.c.m. in telephony, which is actually a digital transmission technique, is going to be carried through, a certain integration of digital transmission techniques might be possible. It should be observed here that application of p.c.m. in telephony necessitates a greater bandwidth with respect to f.d.m. systems due to the coding of the telephone signal amplitude. This disadvantage does not exist, however, where the use of digital lines for data transmission is concerned, as it is in general characteristic of data signals to be offered in a bivalent form. As for the switching means needed, if a time-division technique is chosen, use could be made of the results of the developments taking place in the field of time-division switching technique for telephony.

For multiplexing information flows of various speeds on a time-division basis, the principle of multiple scanning may be applied. In this technique a digital signal to be multiplexed is scanned several times per digit at a fixed scanning rate. The number of scannings depends on the digit length and the rate category to which the signal is considered to belong. (See figure.) The result of the scannings is transmitted over the digital line at a fixed rate and either in coded form or not [1]. As after the scanning all signals to be multiplexed are of the same rate, multiplexing on a time-division basis can then be easily realized.

Time-division switching is also made possible in this way.

If the connection between transmitter and receiver consists of a number of circuits built up in the manner described above and connected to each other on a time-division basis, special attention should be given to the limitation of the distortion value. One of the causes of this distortion may be that the rates on successive transmission paths differ slightly, due to inevitable tolerances. In the data transmission network discussed here, the various transmitting and receiving installations will tolerate widely different distortions. This makes the distortion problem a very complex one, requiring further studies. However, various solutions leading to the desired result are possible. Different categories of rate can be realized by making the switching and multiplex equipment function in such a manner that several time slots can be assigned to a connection, so that lines are produced the maximum rate of which is a multiple of the maximum rate of a basic line (e.g. 200 bauds).

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[1] Literature: TRAVIS and YAEGER: Wide- band data on T1 carrier; *B.S.T.J.*, Vol. XLIV, October 1965, pages 1567 ff.