THE INTERNATIONAL TELEGRAPH AND TELEPHONE CONSULTATIVE COMMITTEE

CCITT

SIXTH PLENARY ASSEMBLY

GENEVA, 27 SEPTEMBER - 8 OCTOBER 1976

ORANGE BOOK

VOLUME III-2

LINE TRANSMISSION

Published by the
INTERNATIONAL TELECOMMUNICATION UNION
GENEVA, 1977

THE INTERNATIONAL TELEGRAPH AND TELEPHONE CONSULTATIVE COMMITTEE

CCITT

SIXTH PLENARY ASSEMBLY

GENEVA, 27 SEPTEMBER - 8 OCTOBER 1976

ORANGE BOOK

VOLUME III-2

LINE TRANSMISSION

Published by the
INTERNATIONAL TELECOMMUNICATION UNION
GENEVA, 1977

ISBN 92-61-00351-6

CONTENTS OF THE CCITT BOOK
APPLICABLE AFTER THE SIXTH PLENARY ASSEMBLY (1976)

ORANGE BOOK

Volume I	— Minutes and reports of the VIth Plenary Assembly of the CCITT. — Resolutions and Opinions issued by the CCITT. — General table of Study Groups and Working Parties for the period 1977-1980. — Summary table of abridged titles of Questions under study in the period 1977-1980. — Recommendations (Series A) on the organization of the work of the CCITT. — Recommendations (Series B) relating to means of expression. — Recommendations (Series C) relating to general telecommunication statistics.
Volume II.1	— General tariff principles — Lease of circuits for private service: Series D Recommendations and Questions (Study Group III).
Volume II.2	— Telephone operation, quality of service and tariffs: Series E Recommendations and Questions (Study Group II).
Volume II.3	— Telegraph operations and tariffs: Series F Recommendations and Questions (Study Group I).
Volume III	— Line transmission: Series G, H and J Recommendations and Questions (Study Groups XV, XVI, XVIII, CMBD).
Volume IV.1	— Line maintenance and measurement: Series M and N Recommendations and Questions (Study Group IV).
Volume IV.2	— Specifications of measuring equipment: Series O Recommendations and Questions (Study Group IV).
Volume V	— Telephone transmission quality and telephone sets: Series P Recommendations and Questions (Study Group XII).
Volume VI.1	— General Recommendations relating to telephone switching and signalling: Series Q Recommendations and Questions (Study Group XI).
Volume VI.2	— Signalling System No. 6: Recommendations.
Volume VI.3	— Signalling Systems R1 and R2: Recommendations.
Volume VI.4	— Programming languages for stored-programme control exchanges: Series Z Recommendations.
Volume VII	— Telegraph technique: Series R, S, T and U Recommendations and Questions (Study Groups VIII, IX, X, XIV).
Volume VIII.1	— Data transmission over the telephone network: Series V Recommendations and Questions (Study Group XVII).
Volume VIII.2	— Public data networks: Series X Recommendations and Questions (Study Group VII).
Volume IX	— Protection: Series K and L Recommendations and Questions (Study Groups V, VI).

Each volume also contains, for its field and where appropriate:

— definitions of specific terms used;
— supplements for information and documentary purposes.

SECTION 7

GENERAL ASPECTS OF DIGITAL TRANSMISSION SYSTEMS; TERMINAL EQUIPMENTS

7.0 General

Recommendation G.701

FRAMEWORK OF THE SERIES G.700, G.800 AND G.900 RECOMMENDATIONS [1)]

(Geneva, 1972; amended at Geneva, 1976)

7.0 General

G.701 Framework of the Series G.700, G.800 and G.900 Recommendations.
G.702 Vocabularly of pulse code modulation (PCM) and digital transmission terms
G.703 General aspects of interfaces

7.1 Coding of analogue signals

G.711 Pulse code modulation (PCM) of voice frequencies
G.712 Performance characteristics of PCM channels at audio frequencies
G.71x PCM of other types of analogue signals (music, visual telephone, FDM blocks, television, etc.)
G.71y Other kinds of coding of analogue signals (e.g. Δ modulation)

7.2 General Recommendations on digital systems and paths

G.721 Hypothetical reference digital paths
G.72x Interconnection of digital paths using different techniques
G.72y Code conversion
G.72z Interference from the mains, etc.

7.3 Principal characteristics of primary multiplex equipment

G.731 Primary PCM multiplex equipment for voice frequencies
G.732 Characteristics of primary PCM multiplex equipment operating at 2048 kbit/s

---

[1)] Recommendations designated by a letter, followed by numbers (e.g. G.712) exist and are published. The titles of Recommendations designated by a letter, followed by numbers and another letter (e.g. G.71x) are intended to indicate the general plan for the drafting of Recommendations in the future.

G.733 Characteristics of primary PCM multiplex equipment operating at 1544 kbit/s

G.734 Characteristics of 2048 kbit/s frame structure for use with digital exchanges

G.73x Other primary multiplex, digital and PCM equipment.

7.4 Principal characteristics of second order multiplex equipment

G.741 General considerations on second-order multiplex equipment

G.742 Second-order digital multiplex equipment operating at 8448 kbit/s and using positive justification

G.743 Second-order digital multiplex equipment operating at 6312 kbit/s and using positive justification.

G.744 Second-order PCM multiplex equipment operating at 8448 kbit/s

G.745 Second-order digital multiplex equipment operating at 8448 kbit/s and using positive/zero/negative justification

G.746 Characteristics of 8448 kbit/s frame structure for use with digital exchanges

G.74x Other digital and PCM second order multiplex equipment

7.5 Principal characteristics of higher order multiplex equipment

G.751 Digital multiplex equipments operating at the third order bit rate of 34 368 kbit/s and the fourth order bit rate of 139 264 kbit/s and using positive justification

G.752 Characteristics of digital multiplex equipments based on a second order bit rate of 6312 kbit/s and using positive justification.

7.9 Other terminal equipments

SECTION 8

Digital networks

8.0 General

G.80x Overall aspects of the integrated services digital network (ISDN) and of dedicated networks

8.1 Organization of digital networks

G.811 Plesiochronous operation of international digital links

G.81x Other Recommendations on the synchronization of digital networks

G.81y Maintenance methods and alarms

8.2 Quality and availability targets

G.82x, G.82y Error rates, slip, jitter, distortions, etc.

G.82z Reliability and availability

8.x Signalling and switching in ISDNs

SECTION 9

Digital line sections

9.0 General

9.1 Transmission systems on cable

G.911 Cable systems operating at 8448 kbit/s

G.91x Digital transmission on FDM groups

9.5 Transmission systems on millimetric waveguides

9.6 Transmission systems on optical fibres

9.9 Digital transmission by radio

G.99x Digital transmission by radio relays

G.99y Digital transmission by satellite.

Recommendation G.702

VOCABULARY OF PULSE CODE MODULATION (PCM) AND DIGITAL TRANSMISSION TERMS

(Geneva, 1972; amended at Geneva, 1976)

1. This Recommendation provides a vocabularly of terms and definitions which are appropriate to pulse code modulation and digital systems.

Some of the terms contained in the vocabulary already appear in the ITU *List of Definitions of Essential Telecommunication Terms* (2nd edition, 1961) and references to this List are given together with proposed new definitions where appropriate [1)].

In the interest of standardization in the drafting of documents the following abbreviations are recommended:

kbit/s,
Mbit/s,
Gbit/s.

To avoid misinterpretation of the use of the point (.) and the comma (,) in different languages to separate the whole and decimal parts, it is recommended that this should be avoided wherever possible. As an example, 2048 kbit/s is preferred to 2.048 (2,048) Mbit/s.

2. *Vocabulary of pulse code modulation (PCM) and digital transmission terms*

---

1) According to the conventions applied in this *List*, any term used, but not advised, is shown between square brackets, thus [ ].

Example: 3007 Parallel to serial converter [Dynamicizer]

Furthermore, any term which is in general use in addition to the principal term is shown between parentheses, thus ( ).

Example: 5010 Timing recovery (Timing extraction)

CONTENTS

2.1 General
2.2 Digital signals
2.3 Multiplexing in PCM
2.4 Frame alignment
2.5 Timing
2.6 Signalling in PCM
2.7 Audio performance
2.8 Codes
2.9 Digital networks
Alphabetical list of definitions contained in this Recommendation.

2.1 *General*

1001 **pulse code modulation (PCM)**

A process in which a signal is sampled, and the magnitude of each sample with respect to a fixed reference is quantized and converted by coding to a digital signal.

1002 **sample**

The value of a particular characteristic of a signal at a chosen instant.

1003 **sampling**

The process of taking samples, usually at equal time intervals.

1004 **sampling rate**

The number of samples per unit time.

1005 **working range**

The permitted range of values of an analogue signal over which a transmitting or other processing equipment can operate (see Figure 3/G.702).

1006 **quantizing**

A process in which samples are classified into a number of adjacent intervals each interval being represented by a single value called the quantized value (see Figure 3/G.702).

1007 **uniform quantizing**

Quantizing in which all the intervals are equal.

1008 **non-uniform quantizing**

Quantizing in which the intervals are not all equal.

1009 **reconstructed sample**

An analogue sample generated at the output of a decoder when a specified character signal is applied at its input. The amplitude of this sample is proportional to the quantized value of the corresponding encoded sample.

1010 **encoding**
coding

The generation of character signals to represent quantized samples.

1011 **encoder**
coder

A device for encoding signal samples.

1012 **uniform encoding**

The generation of character signals representing uniformly quantized samples.

1013 **non-uniform encoding**

The generation of character signals representing non-uniformly quantized samples (see Figure 2/G.702).

1014 **decoding**

A process in which one of a set of reconstructed samples is generated from the character signal representing a sample.

1015 **decoder**

A device for decoding character signals.

1016 **codec**

An assembly comprising an encoder and a decoder in the same equipment.

1017 **decision value**

A reference value defining the boundary between adjacent intervals in quantizing or encoding (see Figures 1/G.702 and 3/G.702).

1018 **virtual decision value**

Two hypothetical decision values, used in quantizing or encoding, located at the ends of the working range used, and obtained by extrapolation from the real decision values (see Figure 3/G.702).

1019 **encoding law**

The law defining the relative values of the quantum steps used in quantizing and encoding (see Figures 1/G.702 and 3/G.702).

1020 **segmented encoding law**

An encoding law in which an approximation to a smooth law (see Figure 2*a)*/G.702) is obtained by a number of linear segments (see Figure 2*b)*/G.702).

1021 **quantizing interval**

The interval between two adjacent decision values.

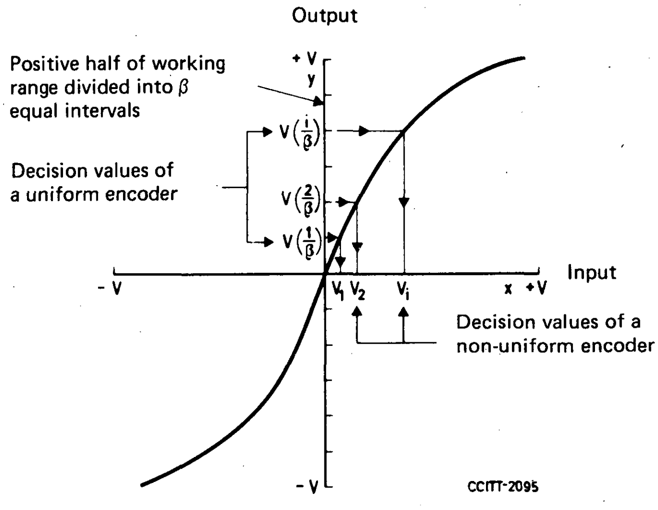
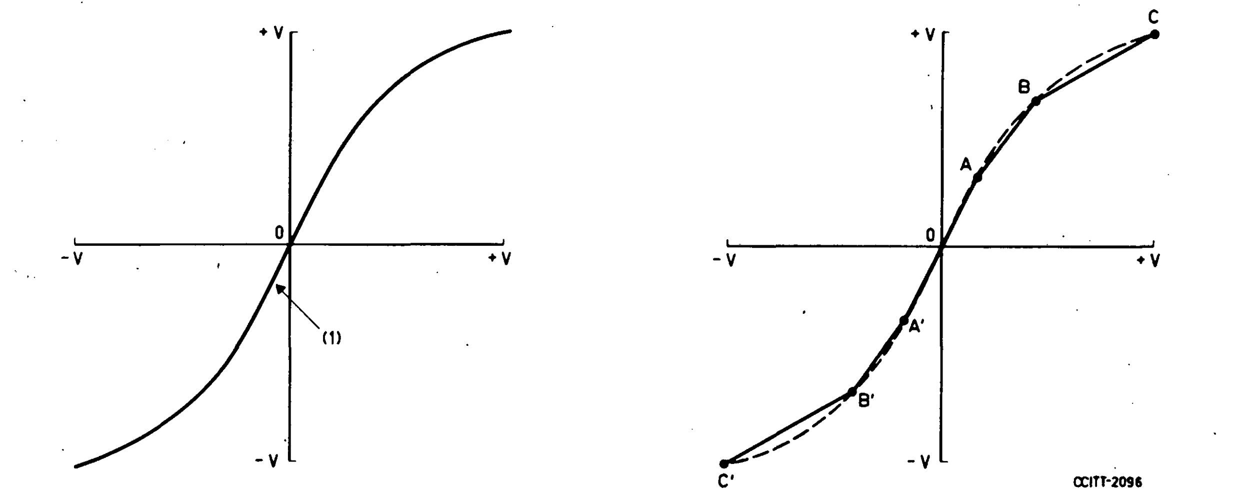


FIGURE 1/G.702 – **Relationship between the decision values of a uniform and a non-uniform encoding law**



a) Smooth characteristic

Note. – A central linear section (1), if present, must tangentially join on to the curved end-section.

b) Segmented characteristic

Note. – This particular characteristic has 5 linear segments: C′B′, B′A′, A′A, AB, BC.

FIGURE 2/G.702 – **Non-uniform encoding laws**

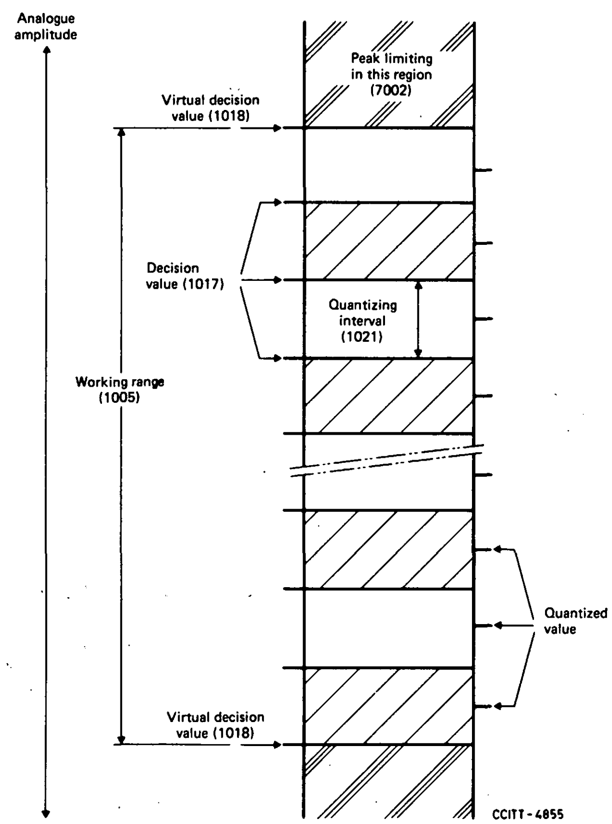


FIGURE 3/G.702 – **Illustration of terms associated with quantizing (1006)**

2.2 *Digital Signals*

2001 **digit** [replaces 53.02 [2)]]

A member selected from a finite set.

Note 1. – In digital transmission, a digit may be represented by a signal element, being characterized by the dynamic nature, discrete condition and discrete timing of the element, e.g. it may be represented as a pulse of specified amplitude and duration.

Note 2. – In equipment used in digital transmission, a digit may be represented by a stored condition being characterized by a specified physical condition, e.g. it may be represented as a binary magnetic condition of a ferrite core.

Note 3. – The context of the use of the term should be such as to indicate the radix of notation. (The meaning of "digit" in Notes 1, 2, and 3 translates into French as "élément numérique".)

Note 4. – In telephone subscriber numbering, a digit is any of the numbers 1, 2, 3 ... 9 or 0 forming the elements of a telephone number (Recommendation Q.10). (This meaning of "digit" translates into French as "chiffre".)

2002 **digital signal**

A signal constrained to have a discontinuous characteristic in time and a set of permitted discrete values.

2003 **digit position**

The position in time or space into which a representation of a digit may be placed.

2004 ***n*-ary digital signals**

Digital signals in which a signal element may assume *n* discrete states.

2005 **binary figure**

One of the two figures (i.e. 0 or 1) used in the representation of numbers in binary notation.

2006 **binary digit**[replaces 53.01 [2)]]

A member selected from a binary set.

Note 1. – Bit is an abbreviation for binary digit.

Note 2. – In the interest of clarity, it is recommended that the term "bit" should not be used in two-condition start-stop modulation instead of "unit element".

2007 **equivalent bit rate**

In a line coded signal, the number of binary digits that can be transmitted in a unit of time.

Note. – The point to which the equivalent bit rate is referred may be either real or hypothetical.

2008 **character signal**

A set of signal elements representing a character, or in PCM representing the quantized value of a sample.

Note. – In PCM, the term "PCM word" may be used in this sense.

2009 **significant instants of a digital signal**

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

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

[2)] Such numbers refer to the *List of Definitions of Essential Telecommunication Terms*, Part 1 (2nd edition, 1961). Numbers 51.01 et seq. are to be found in the 2nd Supplement to the *List*, entitled *Data Transmission*.

2010 **decision instant of a digital signal**

The instant at which a decision is taken by a receiving device as to the probable value of a signal element.

2011 **digit rate**

The number of digits per unit time.

Note 1. – An appropriate adjective, as indicated in definition 2004, should precede the word "digit", for example, binary digit rate. (This may be abbreviated to "bit rate" in accordance with 2006.)

Note 2. – In the interests of clarity it is recommended that this term should not be used to express the symbol rate on the line.

2012 **digital error**

A single digit inconsistency between the transmitted and received signals.

2013 **jitter**

Short-term variations of the significant instants of a digital signal from their ideal positions in time.

2014 **regeneration**

The process of recognizing and reconstructing a digital signal so that the amplitude, waveform and timing are constrained within stated limits.

2015 **regenerator**

A device which performs signal regeneration.

2016 **regenerative repeater**

A device which performs signal regeneration together with ancillary functions.

2017 **decision circuit**

A circuit which decides the probable value of a signal element.

2018 **equivalent binary content**

The content, expressed in binary terms, of a signal generated by a digital source.

Note. – The point to which the equivalent binary content is referred may be either real or hypothetical.

2019 **redundant *n*-ary signal**

A digital signal whose elements can assume n discrete states and where the average information transmission capacity is less than $\log_2 n$.

Note. – The percent redundancy R, of an n-ary digital signal, is given by:

$$[1 - r_e/(r_d \cdot \log_2 n)] \cdot 100$$

where r_d is the symbol rate of the n-ary signal and r_e is the equivalent bit rate.

This may also be expressed in terms of the number of binary digits which can be transmitted by an element of a particular line code. Examples are:

AMI (37% redundant), 1 binary digit per element;
4B3T (16%redundant), 1.33 binary digit per element.

2020 **symbol rate**

The reciprocal of the unit interval in seconds. (This rate is expressed in bauds.)

Note. – This definition is the same as 31.27, "modulation rate". The term "symbol rate" is preferred in the case of line transmission of digital signals.

2.3 *Multiplexing in PCM*

3001 **highway**

A common path or a set of parallel paths over which signals from a plurality of channels pass with separation achieved by time division.

3002 **channel gate**

A device for connecting a channel to a highway, or a highway to a channel, at specified times.

3003 **primary block**
(American: **digroup**)

A basic group of PCM channels assembled by time division multiplexing.

Note. – The following conventions could be useful:

Primary block μ – a basic group of PCM channels derived from 1544 kbit/s PCM multiplex equipment.

Primary block A – a basic group of PCM channels derived from 2048 kbit/s PCM multiplex equipment.

3004 **frame**

A set of consecutive digit time slots in which the position of each digit time slot can be identified by reference to a frame alignment signal.

The frame alignment signal does not necessarily occur, in whole or in part, in each frame.

3005 **multiframe**

A set of consecutive frames in which the position of each frame can be identified by reference to a multiframe alignment signal.

The multiframe alignment signal does not necessarily occur, in whole or in part, in each multiframe.

3006 **subframe**

A sequence of non-contiguous sets of digits assembled within a frame, each set being repeated at n times the frame repetition rate where n is an integer > 1.

3007 **parallel to serial converter**
(American: **serializer**)
[dynamicizer]

A device which converts a group of digits, all of which are presented simultaneously, into a corresponding sequence of signal elements.

3008 **serial to parallel converter**
(American: **deserializer**)
[staticizer]

A device which converts a sequence of signal elements into a corresponding group of digits, all of which are presented simultaneously.

3009 **PCM multiplex equipment**

Equipment for deriving a single digital signal at a defined digit rate from two or more analogue channels by a combination of pulse code modulation and time division multiplexing (multiplexer) and also for carrying out the inverse function (demultiplexer).

The description should be preceded by the relevant equivalent binary digit rate, e.g. 2048 kbit/s PCM multiplex equipment.

3010 **digital multiplex equipment**

Equipment for combining, by time division multiplexing (multiplexer) a defined integral number of digital input signals into a single digital signal at a defined digit rate and also for carrying out the inverse function (demultiplexer).

Note. – When both functions are combined in one equipment at the same location the abbreviation "MULDEX" may be used to describe this equipment.

3011 **digital multiplex hierarchy**

A series of digital multiplexes graded according to capability so that multiplexing at one level combines a defined number of digital signals, each having the digit rate prescribed for a lower order, into a digital signal having a prescribed digit rate which is then available for further combination with other digital signals of the same rate in a digital multiplex of the next higher order.

3012 **service digits** [replaces 53.23 [2)]]
(housekeeping digits)

Digits which are added, normally at regular time intervals to a digital signal to enable the equipment associated with that digital signal to function correctly, and possibly to provide ancillary facilities.

3013 **digital filling**

The addition of a fixed number of digits to a digital signal to change the digit rate from its existing nominal value to a higher predetermined nominal value.

Note. – The added digits will not be used to trasmit information.

3014 **justification**
(pulse stuffing)

A process of changing the rate of a digital signal in a controlled manner so that it can accord with a rate different from its own inherent rate, ususally without loss of information.

3015 **positive justification**
(positive pulse stuffing)

In digital multiplexing, the provision of a fixed number of dedicated time slots (normally at regular intervals) in the output digital signal, these time slots being used to transmit either information from the tributaries, or no information, according to the relative digit rates of the individual tributaries and the output digital signal.

3016 **negative justification**
(negative pulse stuffing)

In digital multiplexing, the controlled deletion of digits from the tributary digital signal so that the digit rates of the individual tributaries correspond to a rate determined by the multiplex equipment. The deleted information is transmitted via a separate low-capacity time slot.

3017 **positive/negative justification**
(positive/negative pulse stuffing)

A combination of positive and negative justification in which justifying digits are transmitted (positive) or information bits are deleted (negative) at each justification opportunity.

3018 **positive/zero/negative justification**

A combination of positive and negative justification in which non-information bits are transmitted (positive) or information bits are deleted (negative) only when it is essential to do so to avoid loss or mutilation of information.

3019 **justifying digit**
(stuffing digit)

A digit inserted in a justifiable digit time slot when that time slot does not contain an information digit.

3020 **justifiable digit time slot**
(stuffable digit time slot)

A digit time slot which may contain either an information digit or a justifying digit.

3021 **justification service digits**
(stuffing service digits)

Digits which transmit information concerning the status of the justifiable digit time slots.

3022 **nominal justification rate**
(nominal stuffing rate)

The rate at which justifying digits are inserted (or deleted) when both the tributary and the multiplex digit rates are at their nominal values.

3023 **maximum justification rate**
(maximum stuffing rate)

The maximum rate at which justifying digits can be inserted (or deleted).

3024 **justification ratio**
(stuffing ratio)

The ratio of the actual justification rate to the maximum justification rate.

3025 **transmultiplexer**

An equipement which transforms signals derived from frequency-division-multiplex equipment (such as group or supergroup) to time-division-multiplexed signals having the same structure as those derived from PCM multiplex equipment (such as primary or secondary PCM multiplex signals) and vice versa.

2.4 *Frame Alignment* [3)]

4001 **frame alignment**

The state in which the frame of the receiving equipment is correctly phased with respect to that of the received signal.

4002 **frame alignment signal**

The distinctive signal used to enable frame alignment to be secured.

4003 **bunched frame alignment signal**

A frame alignment signal in which the signal elements occupy consecutive digit time slots.

4004 **distributed frame alignment signal**

A frame alignment signal in which the signal elements occupy non-consecutive digit time slots.

4005 **frame alignment recovery time**

The time that elapses between a valid frame alignment signal being available at the receive terminal equipment and frame alignment being established.

Note. – The frame alignment recovery time includes the time required for replicated verification of the validity of the frame alignment signal.

4006 **out-of-frame alignment time**

The time during which frame alignment is effectively lost. That time will include the time to detect loss of frame alignment and the alignment recovery time.

---

3) Similar definitions are applicable to multiframe alignment.

2.5 *Timing*

5001 **timing signal**

A cyclic signal used to control the timing of operations.

5002 **reference clock**

A clock [4] of high stability and accuracy which is used to govern the frequency of a network of mutually synchronizing clocks of lower stability. The failure of such a clock does not cause loss of synchronism.

5003 **master clock**

A clock [4] which generates accurate timing signals for the control of other clocks and possibly other equipments.

5004 **time slot**

Any cyclic time interval which can be recognized and defined uniquely.

5005 **channel time slot**

A time slot starting at a particular phase in a frame and allocated to a channel for transmitting a character signal and possibly in-slot signalling or other information.

Note. – Where appropriate a description may be added, for example "telephone channel time slot".

5006 **signalling time slot**

A time slot starting at a particular phase in each frame and allocated to the transmission of signalling.

5007 **frame alignment time slot**

A time slot starting at a particular phase in each frame and allocated to the transmission of a frame alignment signal.

5008 **digit time slot**

A time slot allocated to a single digit.

5009 **retiming**

Adjustment of the intervals between corresponding significant instants of a digital signal, by reference to a timing signal.

5010 **timing recovery**
(timing extraction)

The derivation of a timing signal from a received signal.

---

[4] In these definitions "clock" is taken with the general meaning of Definition 51.10 and it is assumed that where replicated sources are used for security reasons, the assembly of these is regarded as being a single clock.

For information, Definition 51.10 is reproduced below:

51.10 Clock

Equipment providing a time base used in a transmission system to control the timing of certain functions such as the control of the duration of signal elements, the sampling, etc.

5011 **isochronous**

A signal [5)] is isochronous if the time interval separating any two significant instants is theoretically equal to the unit interval or to a multiple of the unit interval.

Note. – In practice, variations in the time intervals are constrained within specified limits.

5012 **anisochronous**

A signal [5)] is anisochronous if the time interval separating any two significant instants is not necessarily related to the time interval separating any other two significant instants.

5013 **synchronous**

Two signals [5)] are synchronous if their corresponding significant instants have a desired phase relationship.

5014 **synchronization**

The process of adjusting the corresponding significant instants of two signals [5)] to obtain the desired phase relationship between these instants.

5015 **homochronous**

Two signals [5)] are homochronous if their corresponding significant instants have a constant, but uncontrolled, phase relationship.

5016 **mesochronous**

Two signals [5)] are mesochronous if their corresponding significant instants occur at the same average rate.

Note. – The phase relationship between corresponding significant instants usually varies between specified limits.

5017 **plesiochronous**

Two signals [5)] are plesiochronous if their corresponding significant instants occur at nominally the same rate, any variation in rate being constrained within specified limits.

Note 1. – Two signals having the same nominal digit rate, but not stemming from the same clock [4)] or homochronous clocks, are usually plesiochronous.

Note 2. – There is no limit to the phase relationship between corresponding significant instants.

5018 **heterochronous**

Two signals [5)] are heterochronous if their corresponding significant instants do not necessarily occur at the same rate.

Note 1. – Two signals having different nominal digit rates, and not stemming from the same clock or from homochronous clocks [4)] are usually heterochronous.

Note 2. – Terms 5011 to 5018 are based on the following Greek roots:

iso = equal
syn = together
homo = same
meso = middle
plesio = near
hetero = different

[5)] In these definitions "signal" is taken with the general meaning of Definition 02.27.

5019 **synchronous network**

A network in which the clocks [4] are controlled so as to run, ideally, at identical rates, or at the same mean rate with limited relative phase displacement.

Note. – Ideally the clocks [4] are synchronous, but they may be mesochronous in practice. By common usage such mesochronous networks are frequently described as synchronous.

5020 **non-synchronous network**
(asynchronous network)

A network in which the clocks [4] need not be synchronous or mesochronous.

2.6 *Signalling in PCM*

6001 **signalling**

The exchange of electrical information (other than by speech) specifically concerned with the establishment and control of connections, and management, in a communication network.

6002 **speech digit signalling**

Signalling in which digit time slots primarily used for the transmission of encoded speech are periodically used for signalling.

6003 **in-slot signalling**

Signalling associated with a channel and transmitted in a digit time slot permanently (or periodically) allocated in the channel time slot.

6004 **out-slot signalling**

Signalling associated with a channel but transmitted in one or more separate digit time slots not within the channel time slot.

6005 **common channel signalling**

A signalling method using a link common to a number of channels for the transmission of signals necessary for the traffic via these channels.

6006 **channel associated signalling**

A signalling method in which the signals necessary for the traffic carried by a single channel are transmitted in the channel itself or in a signalling channel permanently associated with it.

2.7 *Audio Performance*

7001 **load capacity**
(overload point)

In PCM, the level expressed in dBm0, of a sinusoidal signal the positive and negative peaks of which coincide with the positive and negative virtual decision values of the encoder.

7002 **peak limiting**

In PCM, the effect caused by the application to an encoder of an input signal whose value exceeds the virtual decision values of the encoder (see Figure 3/G.702).

7003 **quantizing distortion**

The distortion resulting from the process of quantizing.

7004 **quantizing distortion power**

The power of the distortion component of the output signal resulting from the process of quantizing.

2.8 *Codes*

8001 **pulse code**

A code giving the equivalence between the quantized value of a sample and the corresponding character signal.

8002 **line code**

A code chosen to suit the transmission medium and giving the equivalence between a set of digits generated in a terminal or other processing equipment and the pulses chosen to represent that set of digits for line transmission.

8003 **alternate mark inversion signal (AMI)**
(bipolar signal)

A pseudo-ternary signal, conveying binary digits, in which successive "marks" are normally of alternative, positive and negative, polarity but equal in amplitude and in which "space" is of zero amplitude.

8004 **alternate mark inversion violation**
(bipolar violation)

A "mark" which has the same polarity as the previous "mark" in the transmission of AMI signals.

8005 **modified alternate mark inversion**

An AMI signal which does not strictly conform with alternate mark inversion but includes violations in accordance with a defined set of rules.

Examples of such signals are HDB, B6ZS.

8006 **disparity**

The digital sum of a set of *n* signal elements.

8007 **digital sum**

In a multilevel pulse code, the sum of pulse amplitudes from some arbitrary time-origin to the last transmitted pulse at the time considered, the amplitude unit being chosen in such a way that adjacent levels differ by one unit.

8008 **digital sum variation**

The difference between the maximum and the minimum possible digital sum in any coded sequence of a given code.

8009 **balanced code**

A code whose digital sum variation is finite (balanced codes have no d.c. component in their frequency spectrum).

8010 **paired-disparity code**
(alternative code)
(alternating code)

A code in which some or all of the digits or characters are represented by two assemblies of digits, of opposite disparity, which are used in a sequence so as to minimize the total disparity of a longer sequence of digits.

Note. – An alternate mark inversion signal is an example of a paired-disparity code.

8011 **PCM binary code**

A pulse code in which the quantized values are identified by binary numbers taken in order.

Note. – This term should not be used for line transmission.

8012 **symmetrical binary code**

A pulse code derived from a binary code in which the sign of the quantized value positive or negative, is represented by one digit and in which the remaining digits constitute a binary number representing the magnitude.

Note 1. – In a particular symmetrical binary code, the order of the digits and the use made of the symbols 0 and 1 in the various digit positions must be specified.

Note 2. – This term should not be used for line transmission.

8013 **code conversion**

Conversion of character signals or groups of character signals in one code into corresponding signals or groups of signals in another code.

2.9 *Digital Networks*

9001 **regenerator section**

A regenerator and its preceding transmission path.

9002 **digital section** [6)]

The whole of the means of transmitting and receiving between two consecutive digital distribution frames (or equivalent) a digital signal of specified rate.

Note 1. – A digital section forms either a part or the whole of a digital path.

Note 2. – Where appropriate, the bit rate should qualify the title.

Note 3. – The description always applies to the combination of "go" and "return" directions of transmission, unless stated otherwise.

9003 **digital path**

The whole of the means of transmitting and receiving a digital signal of specified rate between those two digital distribution frames (or equivalent) at which terminal equipments or switches will be connected. Terminal equipments are those at which signals at the specified bit rate originate or terminate.

Note 1. – A digital path comprises one or more digital sections.

Note 2. – Where appropriate, the bit rate should qualify the title.

Note 3. – The description always applies to the combination of "go" and "return" directions of transmission, unless stated otherwise.

Note 4. – Digital paths interconnected by digital switches form a digital connection.

9004 **bit sequence independence**

A digital path or digital section is bit sequence independent at its specified bit rate when its design objectives permit any sequence of bits at that rate, or either equivalent, to be transmitted.

Note. – Practical transmission systems which are not completely bit sequence independent may be described as quasi bit sequence independent. In such cases the limitations should be clearly stated.

9005 **digital switching**

A process in which connections are established by operations on digital signals without converting them to analogue signals.

[6)] Figure 4/G.702 gives examples of digital sections, digital paths, digital line sections, etc.

9006 **integrated digital network**

A network in which connections established by digital switching are used for the transmission of digital signals.

9007 **integrated services digital network**

An integrated digital network in which the same digital switches and digital paths are used to establish connections for different services, for example, telephony, data, etc.

9008 **unilateral control**

A synchronization control system between exchanges A and B is unilateral if the clock [4)] at exchange A controls that at exchange B but B does not control A.

9009 **bilateral control**

A synchronization control system between exchanges A and B is bilateral if the clock [4)] at exchange A controls that at exchange B and the clock at exchange B controls that at exchange A.

9010 **single-ended control**

A synchronization control system between exchanges A and B is single-ended if phase error signals used to control the clock [4)] at a particular exchange are derived only from a comparison of the phase of the incoming digital signals and the phase of the internal clock of the exchange.

9011 **double-ended control**

A synchronization control system between exchanges A and B is double-ended if phase error signals used to control the clock [4)] at a particular exchange are derived from comparison of the phase of the incoming digital signal and the phase of the internal clock at both exchanges.

9012 **analogue control**

A synchronization control system is characterized as having analogue control if the relationship between the actual phase error between clocks [4)] and the error signal device is a continuous function (at least over a limited range).

9013 **linear analogue control**

An analogue system in which the functional relationships are of simple proportionality.

9014 **amplitude quantized control**

A synchronization control system is characterized as having amplitude control if the functional relationship between actual phase error and derived error signal includes discontinuities.

Note. – In practice this implies that the working range of phase errors is divided into a finite number of subranges and that a unique signal is derived for each subrange whenever the error falls within a subrange.

9015 **time quantized control**

A synchronization control system is characterized as having time quantized control if the error signal is derived or utilized only at a number of discrete instants which may or may not be equally spaced in time.

9016 **mutually synchronized network**

A network synchronizing arrangement is mutually synchronized when each clock [4)] in the network exerts a degree of control on all others.

9017 **democratic (mutually synchronized) network**

A mutually synchronized system is democratic when all clocks [4)] in the network are of equal status and exert equal amounts of control on the others, the network operating frequency (digit rate) being the mean of the natural (uncontrolled) frequencies of the population of clocks.

9018 **hierarchic (mutually synchronized) network**

A mutually synchronized system is hierarchic when some clocks [4)] exert more control than others, the network operating frequency being a weighted mean of the natural frequencies of the population of clocks.

9019 **despotic (synchronized) network**

A network synchronizing arrangement is despotic when a unique master clock [4)] exists with full power of control of all other clocks.

9020 **oligarchic (synchronized) network**

A network synchronizing arrangement is oligarchic when control is in the bands of a selected few clocks [4)], the remainder being controlled by these.

9021 **digital line section**

Two consecutive line terminal equipments, their interconnecting transmission medium and the in-station cabling between them and their adjacent digital distribution frames (or equivalents) which together provide the whole of the means of transmitting and receiving between two consecutive digital distribution frames (or equivalents) a digital signal of specified rate.

Note 1. – Where appropriate, the bit rate should qualify the title.

Note 2. – Line terminal equipments may include the following:

- regenerators
- code converters
- scramblers
- remote power feeding
- fault location
- supervision

Note 3. – The description always applies to the combination of "go" and "return" directions of transmission unless stated otherwise.

Note 4. – A digital line section is a particular case of a digital section.

9022 Digital block

The combination of a digital path and associated digital multiplex equipments.

Note. – The bit rate of the digital path should form part of the title.

9023 **Digital line path**

Two or more digital line sections interconnected in tandem in such a way that the specified rate of the digital signal transmitted and received is the same over the whole length of the line path between the two terminal digital distribution frames (or equivalents).

Note. – Where appropriate, the bit rate should qualify the title.

9024 **Digital radio section**

Two consecutive radio terminal equipments and their interconnecting transmission medium which together provide the whole of the means of transmitting and receiving, between two consecutive digital distribution frames (or equivalents), a digital signal of specified rate.

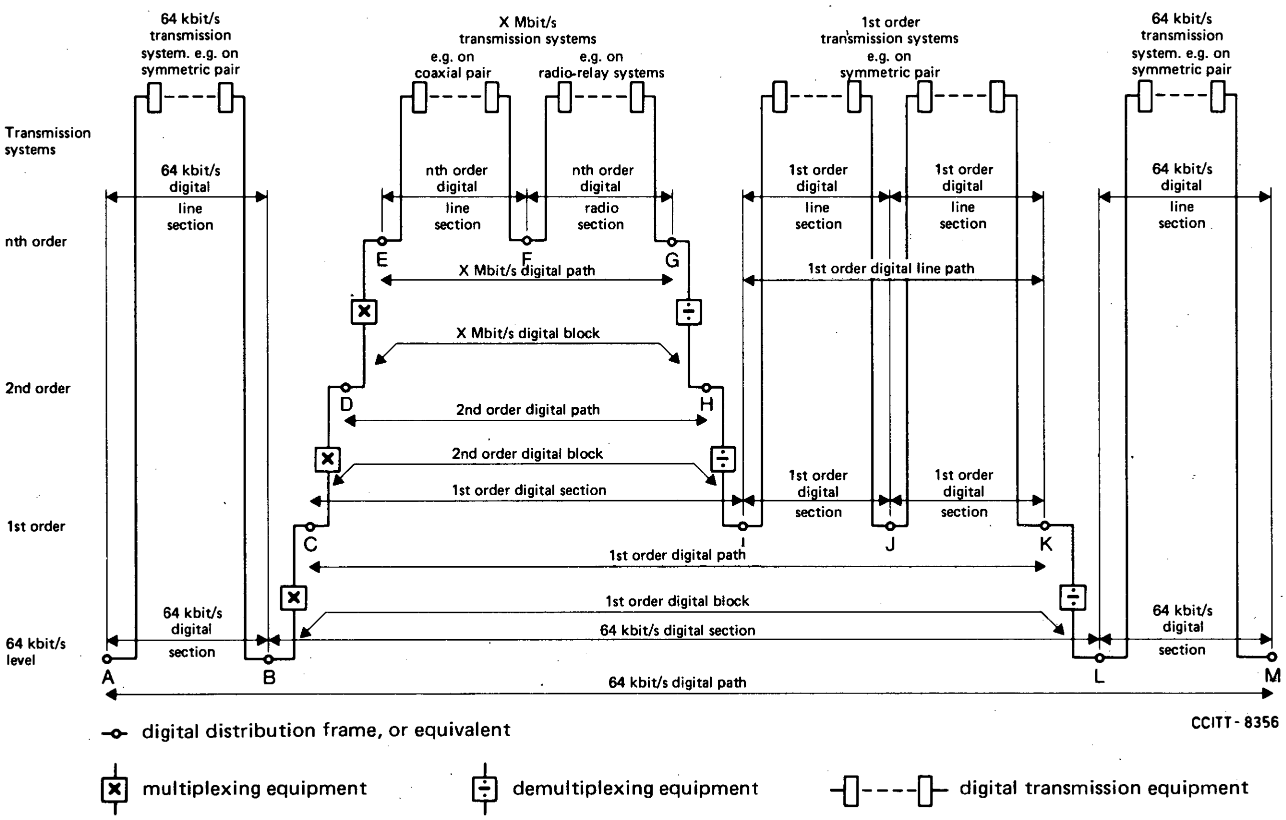
Note 1. – Where appropriate, the bit rate should qualify the title.

Note 2. – The description always applies to the combination of "go" and "return" directions of transmission, unless stated otherwise.

Note 3. – A digital radio section is a particular case of a digital section.

9025 **Digital radio path**

Two or more digital radio sections interconnected in tandem in such a way that the specified rate of the digital signal transmitted and received is the same over the whole length of the radio path between the two terminal digital distribution frames (or equivalents).



Note 1. – Digital line and radio sections may be at digit rates which are either hierarchical or non-hierarchical.
Note 2. – A-B is a 64 kbit/s digital line section, which is a particular case of a 64 kbit/s digital section.
Note 3. – A-M is a 64 kbit/s digital path which comprises three 64 kbit/s digital sections, A-B, B-L and L-M.
Note 4. – F-G is an X Mbit/s digital radio section which forms part of an X Mbit/s digital path E-G.
Note 5. – C-I is a 1st order digital section which contains a 2nd order digital path D-H.
Note 6. – I-K is an example of a digital line path.

FIGURE 4/G.702 – **Examples of digital path, digital section, digital line section, etc.**

Alphabetical list of definitions contained in this Recommendation

8005 Alternate Mark Inversion (modified)
8003 Alternate Mark Inversion Signal
8004 Alternate Mark Inversion Violation
8013 Code Conversion
1016 Codec
1011 Coder
1010 Coding
6005 Common Channel Signalling
1017 Decision Value
2017 Decision Circuit
2010 Decision Instant of a Digital Signal
1015 Decoder
1014 Decoding
9017 Democratic (Mutually Synchronized) Network
3008 (Deserializer: American)
9019 Despotic (Synchronized) Network
2001 Digit
2011 Digit Rate
5008 Digit Time Slot
9022 Digital Block
2012 Digital Error
3013 Digital Filling
9023 Digital Line Path
9021 Digital Line Section
9025 Digital Radio Path
3010 Digital Multiplex Equipment
3011 Digital Multiplex Hierarchy
9003 Digital Path
9025 Digital Radio Path
9024 Digital Radio Section
9002 Digital Section
2002 Digital Signal
8007 Digital Sum
8008 Digital Sum Variation
9005 Digital Switching
3003 (Disgroup: American)
8006 Disparity
4004 Distributed Frame Alignment Signal
9011 Double-ended Control
3007 [Dynamicizer]
1011 Encoder
1010 Encoding
1019 Encoding Law
2018 Equivalent Binary Content
2007 Equivalent Bit Rate
3004 Frame
4001 Frame Alignment
4002 Frame Alignment Signal
4005 Frame Alignment Recovery Time
5007 Frame Alignment Time Slot
5018 Heterochronous
9018 Hierarchic (Mutually Synchronized) Network
3001 Highway
5015 Homochronous
3012 (Housekeeping Digits)
6003 In-slot Signalling
9006 Integrated Digital Network
9007 Integrated Services Digital Network
5011 Isochronous
2013 Jitter
3020 Justifiable Digit Time Slot
3014 Justification
3024 Justification Ratio
3021 Justification Service Digits
3019 Justifying Digit
8002 Line Code
9021 Line Section Digital
9013 Linear Analogue Control
7001 Load Capacity
5003 Master Clock
3023 Maximum Justification Rate
3023 (Maximum Stuffing Rate)
5016 Mesochronous
8005 Modified Alternate Mark Inversion
3005 Multiframe
9016 Mutually Synchronized Network
2004 *n*-ary Digital Signals
3016 Negative Justification
3016 (Negative Pulse Stuffing)
3022 Nominal Justification Rate
3022 (Nominal Pulse Stuffing)
5020 Non-synchronous Network
1013 Non-uniform Encoding
1008 Non-uniform Quantizing
9020 Oligarchic (Synchronized) Network
4006 Out-of-Frame Alignment Time
6004 Out-slot Signalling
7001 (Overload Point)
8010 Paired-Disparity Code
3007 Parallel to Serial Converter
8011 PCM Binary Code
3009 PCM Multiplex Equipment
7002 Peak Limiting
5017 Plesiochronous
3015 Positive Justification
3017 Positive/Negative Justification
3017 (Positive/Negative Pulse Stuffing)
3015 (Positive Pulse Stuffing)
3018 Positive/Zero/Negative Justification
3003 Primary Block
8001 Pulse Code
1001 Pulse Code Modulation (PCM)
3014 (Pulse Stuffing)
1006 Quantizing
7003 Quantizing Distortion
7004 Quantizing Distortion Power

1021 Quantizing Interval
1009 Reconstructed Sample
5002 Reference Clock
2014 Regeneration
2016 Regenerative Repeater
2015 Regenerator
9001 Regenerator Section
5009 Retiming
1002 Sample
1003 Sampling
1004 Sampling Rate
1020 Segmented Encoding Law
3008 Serial to Parallel Converter
3007 (Serializer: American)
3012 Service Digits
6001 Signalling
5006 Signalling Time Slot
2009 Significant Instants of a Digital Signal
9010 Single-ended Control
6002 Speech Digit Signalling
3008 [Staticizer]
3020 (Stuffable Digit Time Slot)
3015 (Stuffing)
3019 (Stuffing Digit)
3024 (Stuffing Ratio)
3021 (Stuffing Service Digits)
3006 Subframe
2020 Symbol Rate
8012 Symmetrical Binary Code
5014 Synchronization
5013 Synchronous
5019 Synchronous Network
2004 Signals (*n*-ary Digital)
9015 Time Quantized Control
5004 Time Slot
5010 (Timing Extraction)
5001 Timing Signal
5010 Timing Recovery
3025 Transmultiplexer
1012 Uniform Encoding
1007 Uniform Quantizing
9008 Unilateral Control
1018 Virtual Decision Value
1005 Working Range

Recommendation G.703

GENERAL ASPECTS OF INTERFACES

(Geneva, 1972; amended at Geneva, 1976)

1. *Interface at 1544 kbit/s*

1.1 Interconnection of 1544 kbit/s signals for transmission purposes is accomplished at a digital distribution frame.

1.2 The signal shall have a bit rate of 1544 kbit/s ± 50 parts per million (ppm).

1.3 One symmetrical pair shall be used for each direction of transmission. The distribution frame jack connected to a pair bringing signals to the distribution frame is termed the in-jack.

The distribution frame jack connected to a pair carrying signals away from the distribution frame is termed the out-jack.

1.4 Test load impedance shall be 100 ohms, resistive.

1.5 An AMI (bipolar) code shall be used. Connecting line systems require suitable signal content to guarantee adequate timing information. This can be accomplished either by scrambling or by permitting not more than 15 spaces between successive marks and having an average mark density of at least 1 in 8.

1.6 The shape for an isolated pulse measured at either the out- or in-jack shall fall within the mask in Figure 1/G.703 and meet the other requirements of Table 1/G.703. For pulse shapes within the mask, the peak undershoot should not exceed 40% of the peak pulse (mark).

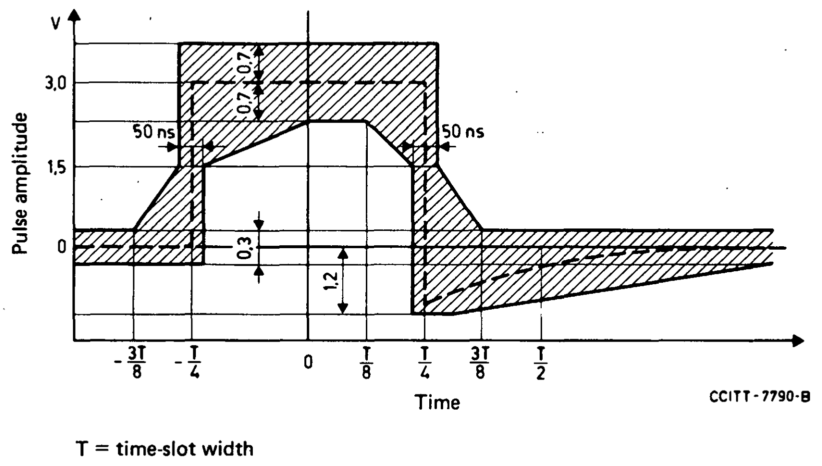


FIGURE 1/G.703 – **Pulse mask for interface at 1544 kbit/s**

1.7 The voltage within a time slot containing a zero (space) shall be no greater than either the value produced in that time slot by other pulses (marks) within the mask of Figure 1/G.703, or ± 0.1 of the peak pulse (mark) amplitude, whichever is greater in magnitude.

TABLE 1/G.703 – **Digital interface at 1544 kbit/s**[a]

Location		Digital distribution frame
Bit rate		1544 kbit/s
Pair(s) in each direction of transmission		One symmetric pair
Code		AMI[b]
Test load impedance		100 ohms resistive
Nominal pulse shape		Rectangular
Signal level[c]	Power at 772 kHz	+12 dBm to +19 dBm
	Power at 1544 kHz	At least 25 dB below the power at 772 kHz

[a] The pulse mask for 1st order digital interface is shown in Figure 1/G.703.
[b] See 1.5 in the text.
[c] The signal level is the power level measured in a 3-kHz bandwidth at the in-jack for an "all ones" pattern transmitted.

2. *Interface at 6312 kbit/s*

2.1 Interconnection of 6312 kbit/s signals for transmission purposes is accomplished at a digital distribution frame.

2.2 The signal shall have a bit rate of 6312 kbit/s ± 30 ppm.

2.3 One symmetrical pair of characteristic impedance 110 ohms, or one coaxial pair of characteristic impedance of 75 ohms shall be used for each direction of transmission. The distribution frame jack connected to a pair bringing signals to the distribution frame is termed the in-jack. The distribution frame jack connected to a pair carrying signals away from the distribution frame is termed the out-jack.

2.4 Test load impedance shall be 110 ohms resistive or 75 ohms resistive as appropriate.

2.5 A pseudo-ternary code shall be used as indicated in Table 2/G.703.

TABLE 2/G.703 – **Digital interface at 6312 kbit/s**[a]

Location	Digital distribution frame	
Bit rate	6312 kbit/s	
Pair(s) in each direction of transmission	One symmetric pair	One coaxial pair
Code	B6ZS[b]	Scrambled AMI[c]
Test load impedance	110 ohms resistive	75 ohms resistive
Nominal pulse shape	Rectangular, shaped by cable loss (see Figure 2/G.703)	Rectangular (see Figure 3/G.703)
Signal level	For an "all ones" pattern transmitted, the power measured in a 3-kHz bandwidth should be as follows: 3156 kHz : 0.2 to 7.3 dBm 6312 kHz : –20 dBm or less	3156 kHz : 6.2 to 13.3 dBm 6312 kHz : –14 dBm or less

[a] The pulse mask for 2nd order digital interface is shown in Figure 2/G.703 and Figure 3/G.703.
[b] Six consecutive zeros are replaced with 0+–0–+ if the preceding pulse was + ; 0–+0+– if the preceding pulse was –.
[c] An AMI code is scrambled by a five-stage reset-type scrambler with the primitive polynominal of $x^5 + x^3 + 1$.

2.6 The shape for an isolated pulse measured at either the out- or in-jack shall fall within the mask either of Figure 2/G.703 or of Figure 3/G.703 and meet the other requirements of Table 2/G.703.

2.7 The voltage within a time slot containing a zero (space) shall be no greater than either the value produced in that time slot by other pulses (marks) within the template of Figure 2/G.703, or ± 0.1 of the peak pulse (mark) amplitude, whichever is greater in magnitude.

	T	Value of curve
Lower curve	$T \leqslant -0.41$ $-0.41 \leqslant T \leqslant 0.24$ $0.24 \leqslant T$	0 $0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.205}\right)\right]$ $0.331\, e^{-1.9\,(T-0.3)}$
Upper curve	$T \leqslant -0.72$ $-0.72 \leqslant T \leqslant 0.2$ $0.2 \leqslant T$	0 $0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.36}\right)\right]$ $0.1 + 0.72\, e^{-2.13\,(T-0.2)}$

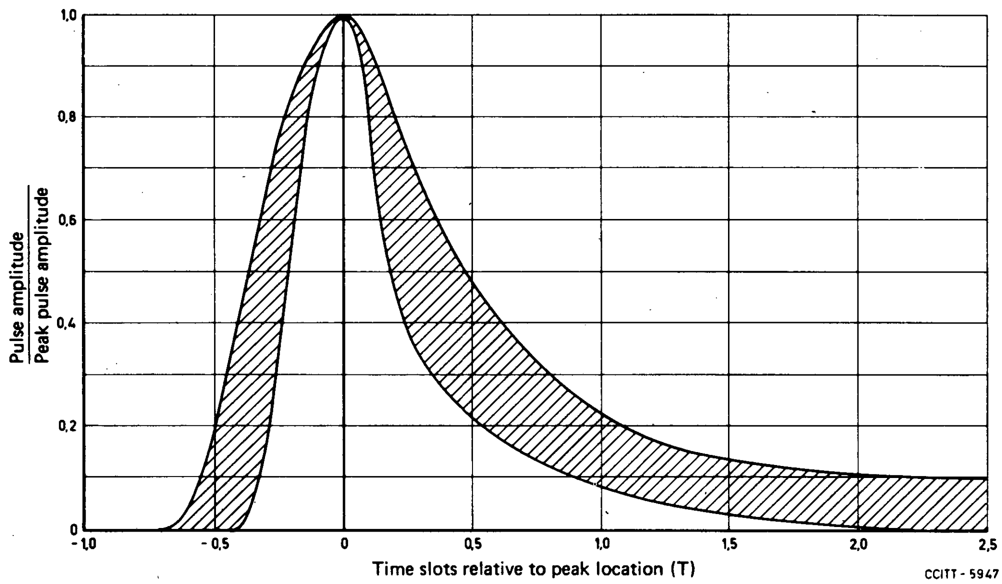
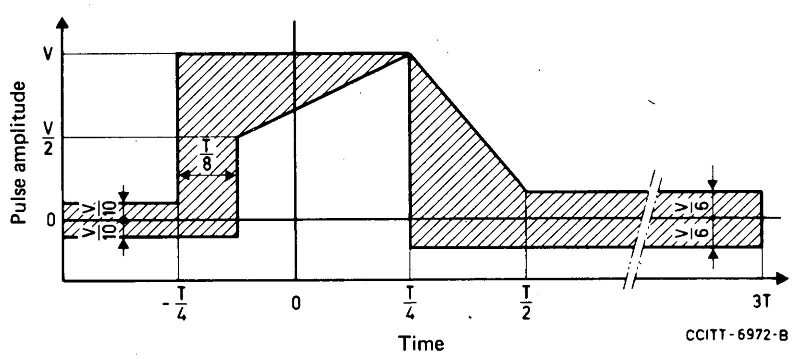


FIGURE 2/G.703 – **Pulse mask for the symmetric pair interface**



T = time-slot width

FIGURE 3/G.703 – **Pulse mask for the coaxial pair interface at 6312 kbit/s**

3. *Interface at 32 064 kbit/s*

3.1 Interconnection of 32 064 kbit/s signals for transmission purposes is accomplished at a digital distribution frame.

3.2 The signal shall have a bit rate of 32 064 kbit/s ± 10 ppm.

3.3 One coaxial pair shall be used for each direction of transmission. The distribution frame jack connected to a coaxial pair bringing signals to the distribution frame is termed the in-jack. The distribution frame jack connected to a coaxial pair carrying signals away from the distribution frame is termed the out-jack.

3.4 The test load impedance shall be 75 ohms ± 5 per cent resistive and the test method shall be direct.

3.5 A scrambled AMI code shall be used.

3.6 The shape for an isolated pulse measured at the in-jack shall fall within the mask in the Figure 4/G.703.

	T	Value of curve
Lower curve	$-0.36 \leqslant T < -0.30$	$5.76\,T + 2.07$
	$-0.30 \leqslant T < 0$	$0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.25}\right)\right]$
	$0 \leqslant T < 0.22$	$0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.16}\right)\right]$
	$0.22 \leqslant T$	$0.11\, e^{-3.42\,(T - 0.3)}$
Upper curve	$-0.65 \leqslant T < 0$	$1.05\left[1 - e^{-4.6\,(T + 0.65)}\right]$
	$0 \leqslant T < 0.25$	$0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.28}\right)\right]$
	$0.25 \leqslant T$	$0.11 + 0.407\, e^{-2.1\,(T - 0.29)}$

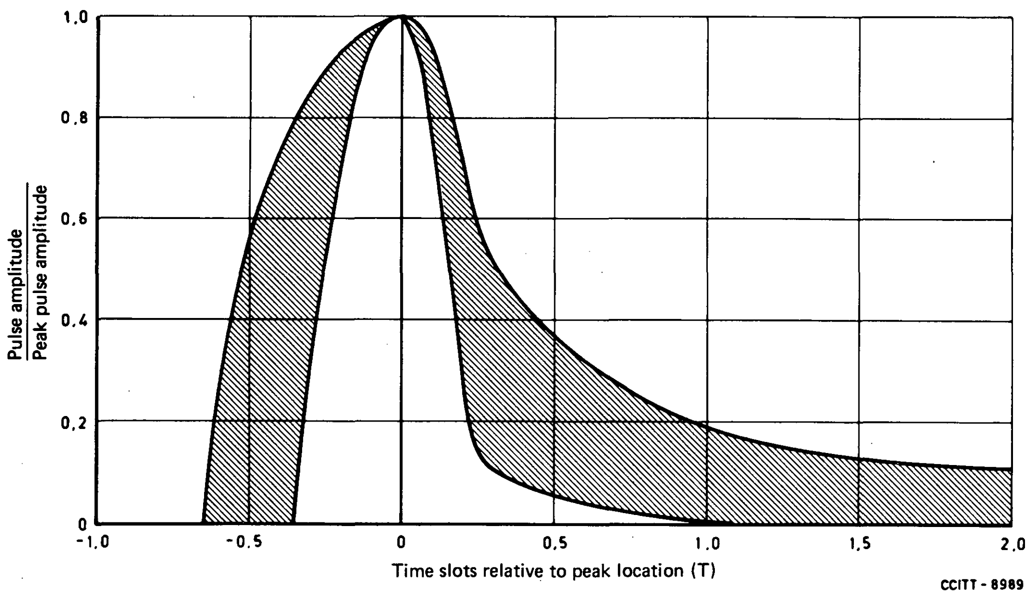


FIGURE 4/G.703 – **Pulse mask for the coaxial pair interface at 32 064 kbit/s**

3.7 The voltage within a time slot containing a zero (space) shall be no greater than either the value produced in that time slot by other pulses (marks) within the mask of Figure 4/G.703 or ± 0.1 of the peak pulse (mark) amplitude, whichever is greater in magnitude.

3.8 For an "all ones" pattern transmitted, the power measured in a 3-kHz bandwidth at the in-jack shall be as follows:

16 032 kHz: +5 dBm to +12 dBm
32 064 kHz: at least 20 dB below the power at 16 032 kHz

3.9 The connectors and coaxial cable pairs in the distribution frame shall be 75 ohms ± 5 per cent.

4. *Interface at 44 736 kbit/s*

4.1 Interconnection of 44 736 kbit/s signals for transmission purposes is accomplished at a digital distribution frame.

4.2 The signal shall have a bit rate of 44 736 kbit/s ± 20 ppm.

4.3 One coaxial pair shall be used for each direction of transmission. The distribution frame jack connected to a coaxial pair bringing signals to the distribution frame is termed the in-jack. The distribution frame jack connected to a coaxial pair carrying signals away from the distribution frame is termed the out-jack.

4.4 Test load impedance shall be 75 ohms ± 5 per cent resistive, and the test method shall be direct.

4.5 A bipolar code shall be used. The code shall be as specified in 4.5.1 below.

4.5.1 *B3ZS code*

The B3ZS bipolar with three-zero substitution code is a modified bipolar pulse format. Logical 1 bits are 50 per cent duty cycle and are generally alternately positive and negative with respect to the logical 0 level. Exceptions are cases where three logical 0s appear together in the bitstream. In the B3ZS format, each block of three consecutive zeros is removed and replaced by B0V or 00V where B represents a pulse conforming with the bipolar rule and V represents a pulse violating the bipolar rule. The choice of B0V or 00V is made so that the number of B pulses between consecutive V pulses is odd. Frame alignment bits per G.752 shall be included.

4.6 The shape for an isolated pulse measured at the in-jack shall fall within the mask in Figure 5/G.703.

4.7 The voltage within a time slot containing a zero (space) shall be no greater than either the value produced in that time slot by other pulses (marks) within the mask of Figure 5/G.703, or ± 0.05 of the peak pulse (mark) amplitude, whichever is greater in magnitude.

4.8 For an "all ones" pattern transmitted, the power measured in a 3-kHz bandwidth at the transmit jack shall be as follows:

22 368 kHz: −1.8 to +5.7 dBm
44 736 kHz: at least 20 dB below the power at 22 368 kHz

4.9 The digital distribution frame for 44 736 kbit/s signals shall have the characteristics specified in 4.9.1 and 4.9.2 below.

4.9.1 The loss between the in- and out-jacks on the distribution frame shall be as follows:

0.60 ± 0.55 dB at 22 368 kHz
(comprised of any combination of flat and shaped losses)

4.9.2 The connectors and coaxial pair cables in the distribution frame shall be 75 ohms ± 5 per cent.

	T	Value of curve
Lower curve	$T \leq -0.36$	0
	$-0.36 \leq T \leq 0.28$	$0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.18}\right)\right]$
	$0.28 \leq T$	$0.11\ e^{-3.42\ (T - 0.3)}$
Upper curve	$T \leq -0.65$	0
	$-0.65 \leq T \leq 0$	$1.05\ [1 - e^{-4.6\ (T + 0.65)}]$
	$0 \leq T \leq 0.36$	$0.5\left[1 + \sin\frac{\pi}{2}\left(1 + \frac{T}{0.34}\right)\right]$
	$0.36 \leq T$	$0.5 + 0.407\ e^{-1.84\ (T - 0.36)}$

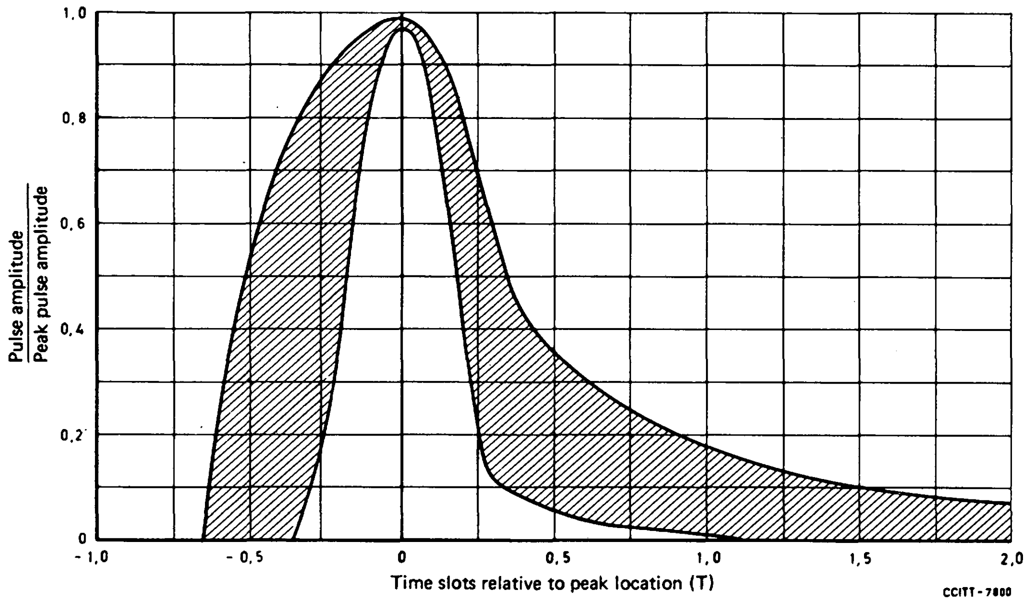


FIGURE 5/G.703 – **Pulse mask for the coaxial pair interface at 44 736 kbit/s**

5. *Interface at 2048 kbit/s*

The interface described here is the preferred solution. In particular cases, such as with connections between equipment parts close to each other, interfaces as described in the Appendix below to 5. may be used.

5.1 *General characteristics*

Bit rate: 2048 kbit/s ± 50 ppm
Code: HDB3 (a description of this code can be found in the Annex)

5.2 *Specifications at the output ports* (see Table 3/G.703)

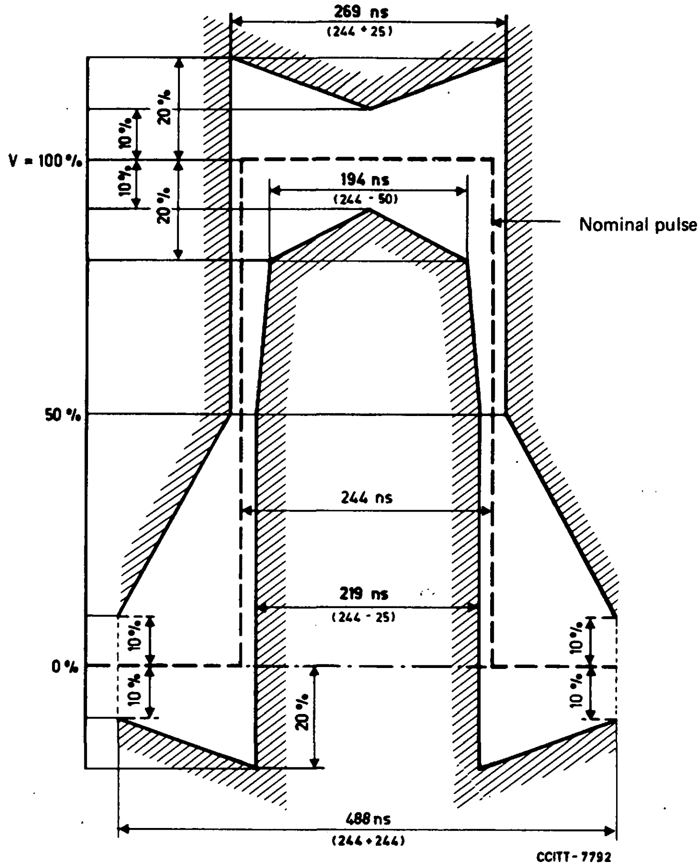
TABLE 3/G.703

Pulse shape (nominally rectangular)	All marks of a valid signal must conform with the mask (Figure 6/G.703), irrespective of the sign. The value V corresponds to the nominal peak value	
Pair(s) in each direction	One coaxial pair (see Note below)	One symmetrical pair (see Note below)
Test load impedance	75 ohms resistive	120 ohms resistive
Nominal peak voltage of a mark (pulse)	2.37 V	3 V
Peak voltage of a space (no pulse)	0 ± 0.237 V	0 ± 0.3 V
Nominal pulse width	244 ns	
Ratio of the amplitudes of positive and negative pulses at the midpoint of a pulse width	0.95 to 1.05	
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05	
Maximum jitter to be accepted by the equipments connected to the interface	Under study	

5.3 *Specifications at the input ports*

The digital signal presented at the input port shall be as defined above but modified by the characteristic of the interconnecting pairs. The attenuation of this pair shall be assumed to follow a $\sqrt{f}$ law and the loss at a frequency of 1024 kHz shall be in the range 0 to 6 dB. This attenuation should take into account any losses incurred by the presence of a digital distribution frame between the equipments.

Note. – The outer conductor of the coaxial pair or the screen of the symmetrical pair shall be connected to the earth at the output port, and provision shall be made for connecting the outer conductor of the coaxial pair or the screen of the symmetrical pair to earth if required, at the input port.



Note. – V corresponds to the nominal peak value.

FIGURE 6/G.703 – **Mask of the pulse at the 2048 kbit/s interface**

APPENDIX

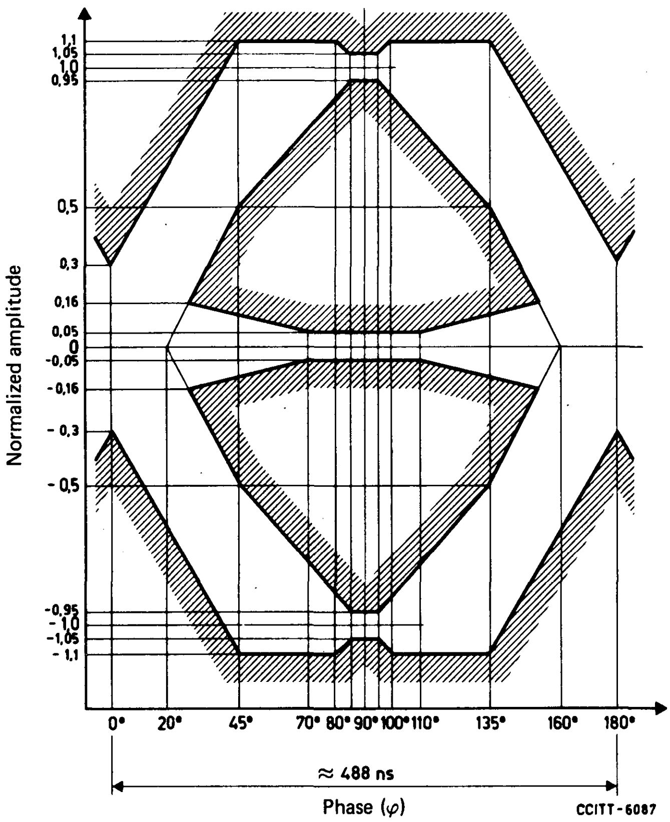
(to 5. of the text)

Other solutions concerning the 2048 kbit/s interface

TABLE 4/G.703

Primary digital interfaces		
Bit rate	2048 kbit/s	
Location	Equipment output port	
Code	AMI + Timing	
Pulse shape	Signal : half sinusoidal Timing : sinusoidal	Rectangular [a]
Ratio of amplitude of positive pulses to amplitude of negative pulses at the centre of the pulse interval	0.95 to 1.05	0.95 to 1.05
Rise time and decay time between 10 % and 90 % of the pulse amplitude	[b]	⩽ 80 ns
Overshoot relative to the pulse amplitude	[b]	⩽ 10 %
Pulse width	[b]	244 ± 30 ns at half amplitude
Peak voltage of pulse	2.37 V ± 10 %	3 V ± 10 %
Peak voltage of space	0 ± 0.118 V	Under study
Test load impedance	130 ohms resistive	120 ohms resistive
Pairs in each direction of transmission	Two symmetric pairs	Two symmetric pairs

[a] Pulse mask is under study.
[b] Relates to Figure 7/G.703.



Note. – For 2048 kbit/s.

FIGURE 7/G.703 – Tolerance mask for the shape of the half-sinusoidal pulse.
A phase angle of 180° corresponds to the duration of a bit time slot

6. *Interface at 8448 kbit/s*

The interface described here is the preferred solution. In particular cases, such as with connections between equipment parts close to each other, an interface as described in the Appendix below to 6. may be used.

6.1 *General characteristics*

Bit rate: 8448 kbit/s ± 30 ppm
Code: HDB3 (a description of this code can be found in the Annex)

6.2 *Specification at the output ports* (see Table 5/G.703)

6.3 *Specifications at the input ports*

The digital signal presented at the input port shall be as defined above, but modified by the characteristics of the interconnecting pairs. The attenuation of this pair shall be assumed to follow a $\sqrt{f}$ law and the loss at a frequency of 4224 kHz shall be in the range 0 to 6 dB. This attenuation should take into account any losses incurred by the presence of a digital distribution frame between the equipments.

Note. – The outer conductor of the coaxial pair shall be connected to the earth at the output port, and provision shall be made for connecting this conductor to earth, if required, at the input port.

TABLE 5/G.703

Pulse shape (nominally rectangular)	All marks of a valid signal must conform with the mask (Figure 8/G.703), irrespective of the sign
Pair(s) in each direction	One coaxial pair (see Note below)
Test load impedance	75 ohms resistive
Nominal peak voltage of a mark (pulse)	2.37 V
Peak voltage of a space (no pulse)	0 ± 0.237 V
Nominal pulse width	59 ns
Ratio of the amplitudes of positive and negative pulses at the midpoint of pulse width	0.95 to 1.05
Ratio of widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05
Maximum jitter to be accepted by the equipments connected to the interface	Under study

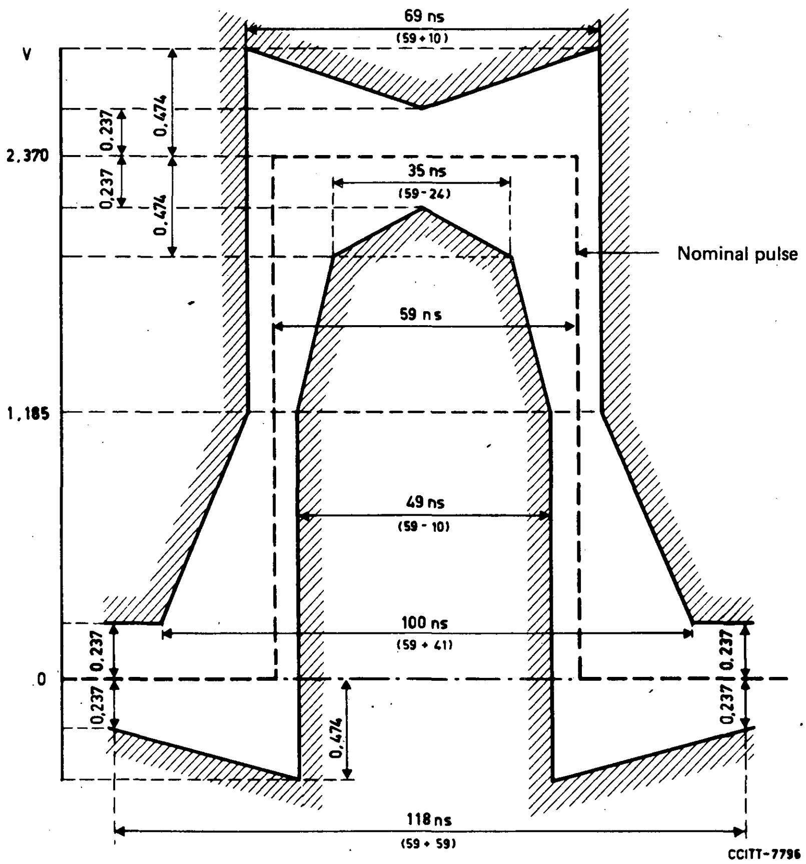


FIGURE 8/G.703 – **Pulse mask at the 8448 kbit/s interface**

APPENDIX

(to 6. of the text)

Other solutions concerning the 8448 kbit/s interface

TABLE 6/G.703

Second order digital interface	
Bit rate	8448 kbit/s
Location	Equipment output port
Code	AMI + Timing
Nominal pulse shape	Rectangular
Peak voltage	Under study
Peak voltage of space	Under study
Test load impedance	150 ohms resistive
Ratio of amplitude of positive pulses to amplitude of negative pulses	0.95 to 1.05
Rise time and decay time between 10% and 90% of the pulse amplitude	≤ 20 ns
Overshoot relative to the pulse amplitude	Under study
Pulse width	59 ± 6 ns
Pairs in each direction of transmission	Two symmetric pairs

7. *Interface at 34 368 kbit/s*

7.1 *General characteristics*

Bit rate: 34 368 kbit/s ± 20 ppm
Code: HDB3 (a description of this code can be found in the Annex)

7.2 *Specification at the output ports* (see Table 7/G.703)

TABLE 7/G.703

Pulse shape (nominally rectangular)	All marks of a valid signal must conform with the mask (Figure 9/G.703), irrespective of the sign
Pair(s) in each direction	One coaxial pair (see Note below)
Test load impedance	75 ohms resistive
Nominal peak voltage of a mark (pulse)	1.0 V
Peak voltage of a space (no pulse)	0 ± 0.1 V
Nominal pulse width	14.55 ns
Ratio of the amplitudes of positive and negative pulses at the centre of a pulse interval	0.95 to 1.05
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05
Maximum jitter to be accepted by the equipments connected to the interface	Under study

7.3 *Specifications at the input ports*

The digital signal presented at the input port shall be as defined above but modified by the characteristic of the interconnecting cable. The attenuation of this cable shall be assumed to follow approximately a $\sqrt{f}$ law and the loss at a frequency of 17 184 kHz shall be in the range 0 to 12 dB.

Note. – The outer conductor of the coaxial pair shall be connected to the earth at the output port, and provision shall be made for connecting this conductor to earth, if required, at the input port.

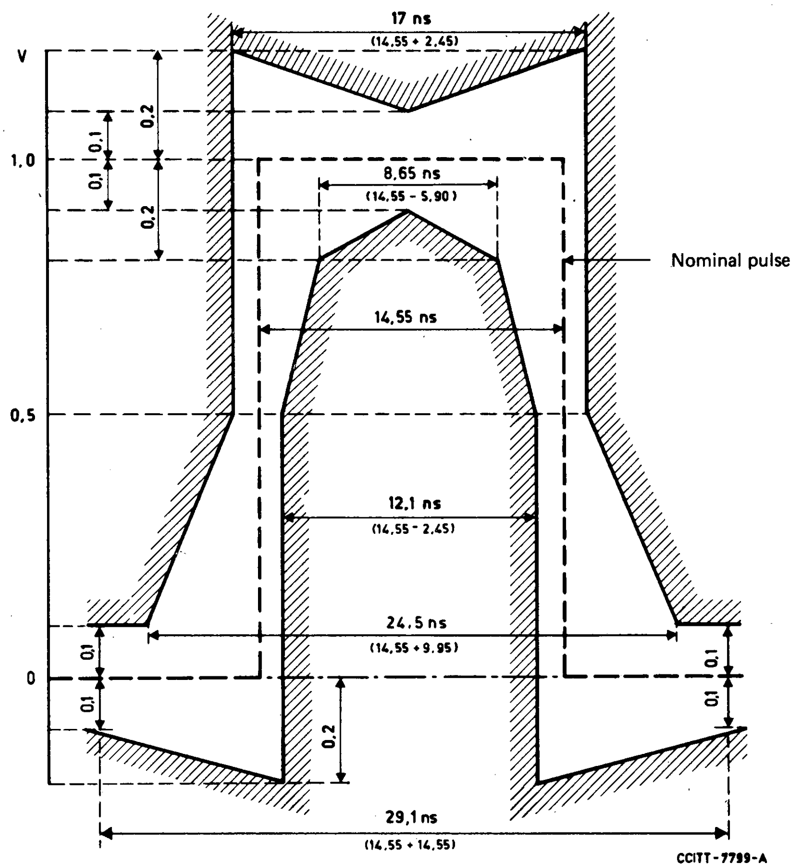


FIGURE 9/G.703 – Pulse mask at the 34 368 kbit/s interface

8. *Interface at 139 264 kbit/s*

8.1 *General characteristics*

Bit rate: 139 264 kbit/s ± 15 ppm
Code: Coded Mark Inversion (CMI)

CMI is a 2-level non-return-to-zero code in which binary zero is coded so that both amplitude levels, A_1 and A_2 are attained consecutively, each for half a unit time interval ($T/2$).

Binary one is coded so that either of the amplitude levels, A_1 and A_2, are attained alternately for one full unit time interval (T).

An example is given in Figure 10/G.703.

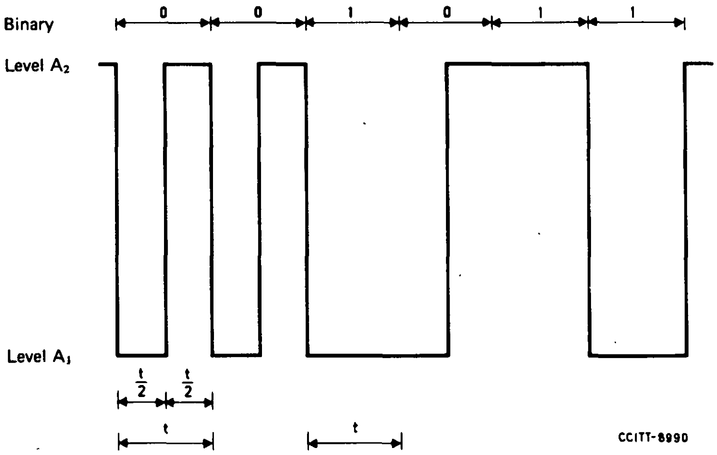


FIGURE 10/G.703 – **Example of CMI coded binary signal**

8.2 *Specifications at the output ports* (see Table 8/G.703)

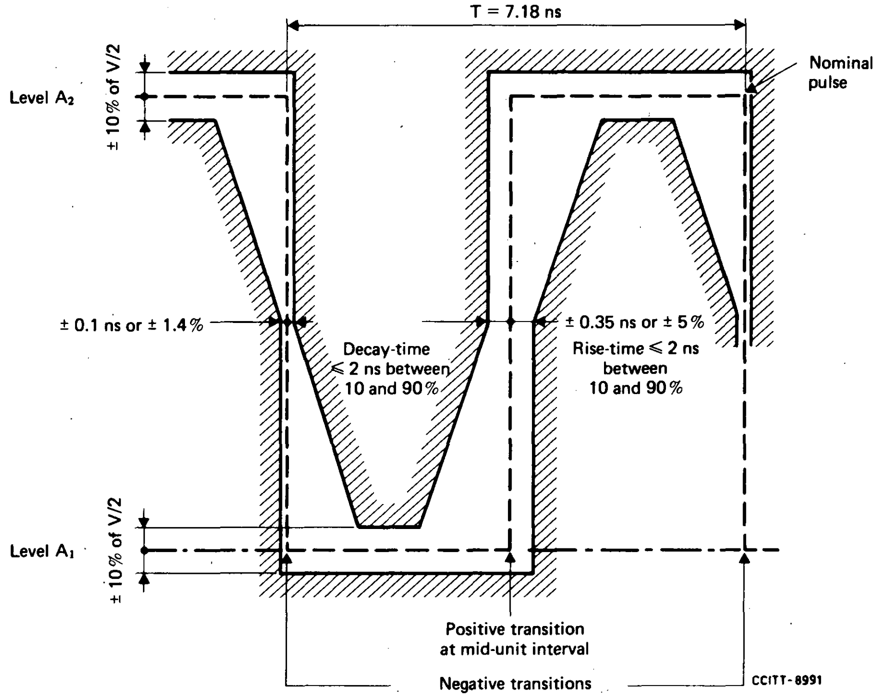
TABLE 8/G.703

Nominal pulse shape	Rectangular
Pair(s) in each direction	One coaxial pair
Test load impedance	75 ohms resistive
Nominal peak-to-peak voltage	1 volt ± 0.1 volt
Overshoot	⩽ 5 % of measured peak-to-peak voltage
Rise time between 10 % and 90 % amplitudes of the measured amplitude	⩽ 2 ns
Transition timing tolerance. (Referred to the mean value of the 50 % amplitude points of negative transitions)	Negative transitions: ± 0.1 ns Positive transitions at unit interval boundaries: ± 0.5 ns Positive transitions at mid-unit interval: ± 0.35 ns
Return-loss	⩾ 15 dB over the frequency range 7 MHz to 210 MHz

Note 1. – A method based on the measurement of the levels of the fundamental frequency component, the second (and possibly the third) harmonic of a signal corresponding to binary-all-zeros and binary-all-ones, is considered to be a perfectly adequate method of checking that the requirements of Table 8/G.703 have been met.

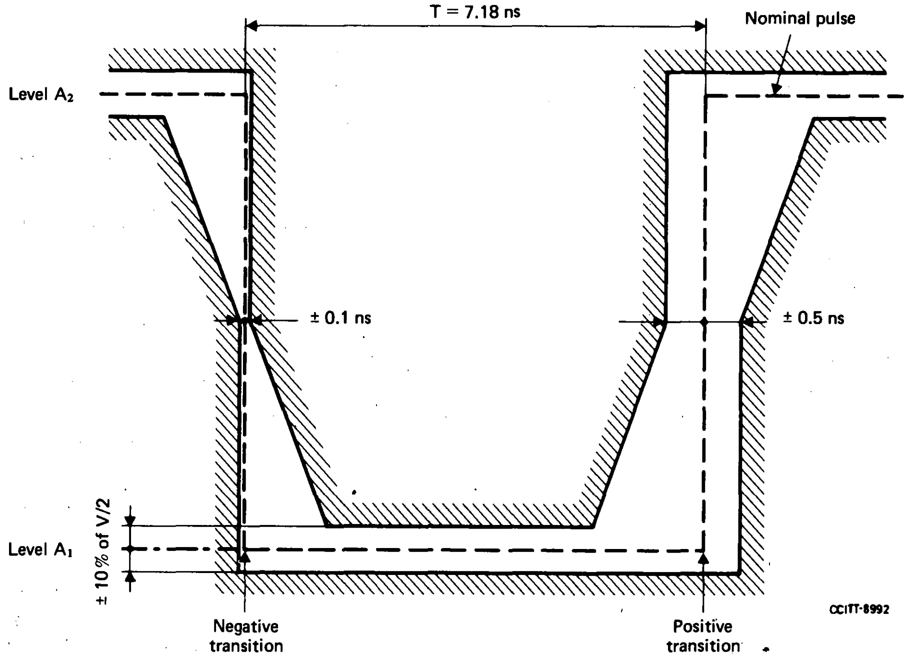
The relevant values are under study.

Note 2. – The masks given in Figure 11/G.703 and Figure 12/G.703, are for guidance purposes only and should not necessarily be used for measurement.



Note 1. – V is the nominal peak-to-peak amplitude.
Note 2. – The mask does not include the overshoot tolerance, see Table 8/G.703.

FIGURE 11/G.703 – **Mask of a pulse corresponding to a binary zero**



Note 1. – The inverse pulse will have the same characteristics.
Note 2. – V is the nominal peak-to-peak amplitude.
Note 3. – The mask does not include the overshoot tolerance, see Table 8/G.703.

FIGURE 12/G.703 – **Mask of a pulse corresponding to a binary one**

8.3 *Specifications at the input ports*

The digital signal presented at the input port should conform to Table 8/G.703, modified by the characteristics of the interconnecting coaxial pair.

The attenuation of the coaxial pair should be assumed to follow an approximate $\sqrt{f}$ law and to have a maximum insertion loss of 12 dB at a frequency of 70 MHz.

The return loss characteristic should be the same as that specified for the output port.

Note. – The outer conductor of the coaxial pair should be connected to earth at the output port and provision should be made for connecting this conductor to earth, if required, at the input port.

9. *Interface at 64 kbit/s*

9.1 *Functional requirements*

9.1.1 The following basic requirements for the design of the interface are recommended.

9.1.2 Both in transmit and receive directions, three signals are carried across the interface:

– 64-kbit/s information signal,
– 64-kHz timing signal,
– 8-kHz timing signal.

Note 1. – An 8-kHz timing signal must be generated but it should not be mandatory for the equipment on the service side of the interface, e.g. data signals or signalling, to either utilize the 8-kHz timing signal from the PCM multiplex or time slot access equipment or supply an 8-kHz timing signal to the PCM equipment.

Note 2. – The detection of an upstream fault can be transmitted across a 64 kbit/s interface either by transmitting an AIS signal and/or by interruption of the 8-kHz timing signal in the receive direction.

9.1.3 The interface should be bit sequence independent at 64 kbit/s.

Note 1. – An unrestricted 64 kbit/s signal can be transmitted across the interface. However, this does not imply that unrestricted 64 kbit/s paths are relizable on a global basis. This is because some Administrations presently have or are continuing to install extensive networks composed of digital line sections whose characteristics do not permit the transmission of long sequences of zeros. (Recommendation G.733 provides for PCM multiplexes with characteristics appropriate for such digital line sections.) Specifically for byte timed sources, a large portion of the 1544 kbit/s digital networks require that at least one binary "one" should be contained in any 8-bit byte of a 64 kbit/s digital signal. For a 64 kbit/s stream which is not byte timed no more than 7 consecutive zeros should appear in the signal.

Note 2. – Although the interface is bit sequence independent, the use of the AIS ("all ones" bit pattern) may result in some minor restrictions for the 64 kbit/s source. For example, an "all ones" alignment signal could result in problems.

9.1.4 *Three types of envisaged interfaces*

– *Codirectional interfaces*

The term codirectional is used to describe an interface across which the information and its associated timing signal are transmitted in the same direction (See Figure 13/G.703.

– *Centralized clock interface*

For both directions of transmission of the information signal, the associated timing signals for both the PCM equipment and the office terminal on the service side are supplied from a centralized clock, which may be derived for example from certain incoming line signals. (See Figure 14/G.703.).

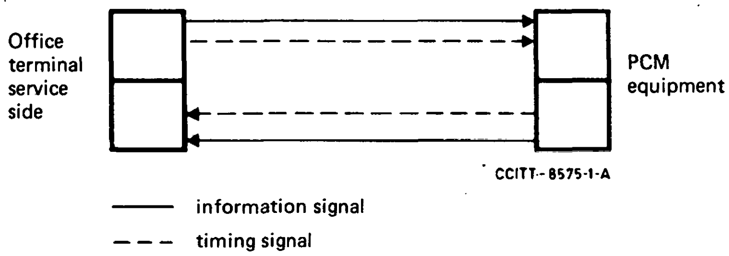


FIGURE 13/G.703 – **Codirectional interface**

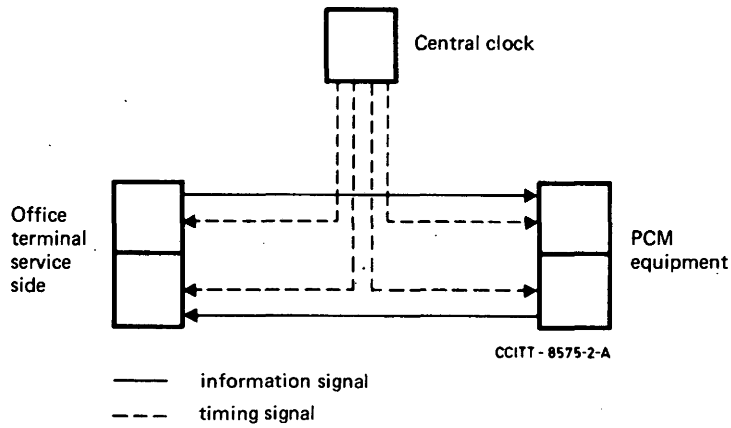


FIGURE 14/G.703 – **Centralized clock interface**

Note. – The codirectional interface or centralized clock interface should be used for synchronous networks and for plesiochronous networks having clocks of the stability required, to ensure an adequate interval between the occurrence of slips.

– *Contradirectional interface*

The term contradirectional is used to describe an interface across which the information in one direction only and its associated timing signal are transmitted in opposite sense. (See Figure 15/G.703.)

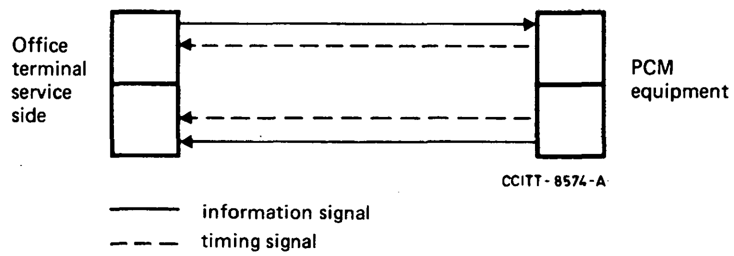


FIGURE 15/G.703 – **Contradirectional interface**

ANNEX

(to Recommendation G.703)

Definition of the HDB3 code

Coding of a binary signal into an HDB3 signal is done according to the following rules:

1. The HDB3 signal is pseudo-ternary; the three states are denoted B_+, B_- and 0.
2. Spaces in the binary signal are coded as spaces in the HDB3 signal. For strings of four spaces however, special rules apply (see 4. below).
3. Marks in the binary signal are coded alternately as B_+ and B_- in the HDB3 signal (alternate mark inversion). Violations of the rule of alternate mark inversion are introduced when coding strings of four spaces (see 4. below).
4. Strings of four spaces in the binary signal are coded according to the following rules:
   a) The first space of a string is coded as a space if the preceding mark of the HDB3 signal has a polarity opposite to the polarity of the preceding violation and is not a violation by itself; it is coded as a mark, i.e. not a violation (i.e. B_+ or B_-), if the preceding mark of the HDB3 signal has the same polarity as that of the preceding violation or is by itself a violation.

   This rule ensures that successive violations are of alternate polarity so that no d.c. component is introduced;

   b) The second and third spaces of a string are always coded as spaces;

   c) The last space of a string of four is always coded as a mark, the polarity of which is such that it violates the rule of alternate mark inversion. Such violations are denoted V_+ or V_- according to their polarity.

7.1 Coding of analogue signals

Recommendation G.711

PULSE CODE MODULATION (PCM) OF VOICE FREQUENCIES

(Geneva, 1972; amended at Geneva, 1976)

1. *General*

The characteristics given below are recommended for encoding voice-frequency signals.

2. *Sampling rate*

The nominal value recommended for the sampling rate is 8000 samples per second. The tolerance on that rate should be ± 50 parts per million (ppm).

3. *Encoding law*

3.1 Eight binary digits per sample should be used for international circuits.

3.2 Two encoding laws are recommended and these are commonly referred to as the A-law and the μ-law. The definition of these laws is given in Tables 1a/G.711 and 1b/G.711 and Tables 2a/G.711 and 2b/G.711 respectively.

When using the μ-law in networks where suppression of the "all zero" character signal is required, the character signal corresponding to negative input values between decision values numbers 127 and 128 should be 00000010 and the value at the decoder output is −7519. The corresponding decoder output value number is 125.

3.3 The number of quantized values results from the encoding law.

3.4 Digital paths between countries which have adopted different encoding laws should carry signals encoded in accordance with the A-law. Where both countries have adopted the same law, that law should be used on digital paths between them. Any necessary conversion will be done by the countries using the μ-law.

3.5 The rules for conversion are given in Tables 3/G.711 and 4/G.711.

4. *Relationship between the encoding laws and the audio level*

The relationship between the encoding laws of Tables 1/G.711 and 2/G.711 and the audio signal level is defined as follows:

A sine-wave signal of 1 kHz at a nominal level of 0 dBm0 should be present at any voice frequency output of the PCM multiplex when the periodic sequence of character signals of Table 5/G.711 for the A-law and of Table 6/G.711 for the μ-law is applied to the decoder input.

The resulting theoretical load capacity (T_{max}) is +3.14 dBm0 for the A-law, and +3.17 dBm0 for the μ-law.

TABLE 1*a*/G.711 – **A-law, positive input values**

1	2	3	4	5	6	7	8
Segment number	Number of intervals × interval size	Value at segment end points	Decision value number n	Decision value x_n (see Note 1)	Character signal before inversion of the even bits Bit number 1 2 3 4 5 6 7 8	Value at decoder output y_n (see Note 3)	Decoder output value number
		4096	(128)	(4096)			
					1 1 1 1 1 1 1 1	4032	128
			127	3968			
7	16 × 128				(see Note 2)		
			113	2176			
					1 1 1 1 0 0 0 0	2112	113
		2048	112	2048			
6	16 × 64				(see Note 2)		
			97	1086			
					1 1 1 0 0 0 0 0	1056	97
		1024	96	1024			
5	16 × 32				(see Note 2)		
			81	544			
					1 1 0 1 0 0 0 0	528	81
		512	80	512			
4	16 × 16				(see Note 2)		
			65	272			
					1 1 0 0 0 0 0 0	264	65
		256	64	256			
3	16 × 8				(see Note 2)		
			49	136			
					1 0 1 1 0 0 0 0	132	49
		128	48	128			
2	16 × 4				(see Note 2)		
			33	68			
					1 0 1 0 0 0 0 0	66	33
		64	32	64			
1 ↓	32 × 2				(see Note 2)		
			1	2			
					1 0 0 0 0 0 0 0	1	1
			0	0			

Note 1. – 4096 normalized value units correspond to T_{max} = 3.14 dBm0.

Note 2. – The character signals are obtained by inverting the even bits of the signals of column 6. Before this inversion, the character signal corresponding to positive input values between two successive decision values numbered n and $n + 1$ (see column 4) is $(128 + n)$ expressed as a binary number.

Note 3. – The value at the decoder output is $y_n = \frac{x_{n-1} + x_n}{2}$ for $n = 1, ..., 127, 128$.

Note 4. – x_{128} is a virtual decision value.

TABLE 1*b*/G.711 – **A-law, negative input values**

1	2	3	4	5	6	7	8
Segment number	Number of intervals × interval size	Value at segment end points	Decision value number n	Decision value x_n (see Note 1)	Character signal before inversion of the even bits Bit number 1 2 3 4 5 6 7 8	Value at decoder output y_n (see Note 3)	Decoder output value number
			0	0			
					0 0 0 0 0 0 0 0	−1	1
			1	−2			
1	32 × 2				(see Note 2)		
		−64	32	−64			
					0 0 1 0 0 0 0 0	−66	33
			33	−68			
2	16 × 4				(see Note 2)		
		−128	48	−128			
					0 0 1 1 0 0 0 0	−132	49
			49	−136			
3	16 × 8				(see Note 2)		
		−256	64	−256			
					0 1 0 0 0 0 0 0	−264	65
			65	−272			
4	16 × 16				(see Note 2)		
		−512	80	−512			
					0 1 0 1 0 0 0 0	−528	81
			81	−544			
5	16 × 32				(see Note 2)		
		−1024	96	−1024			
					0 1 1 0 0 0 0 0	−1056	97
			97	−1088			
6	16 × 64				(see Note 2)		
		−2048	112	−2048			
					0 1 1 1 0 0 0 0	−2112	113
			113	−2176			
7	16 × 128				(see Note 2)		
			127	−3968			
					0 1 1 1 1 1 1 1	−4032	128
		−4096	(128)	(−4096)			

Note 1. – 4096 normalized value units correspond to $T_{max} = 3.14$ dBm0.

Note 2. – The character signals are obtained by inverting the even bits of the signals of column 6. Before this inversion, the character signal corresponding to negative input values between two successive decision values numbered n and $n + 1$ (see column 4) is n expressed as a binary number.

Note 3. – The value at the decoder output is $y_n = \frac{x_{n-1} + x_n}{2}$ for $n = 1, ..., 127, 128$.

Note 4. – x_{128} is a virtual decision value.

TABLE 2*a*/G.711 – **µ-law, positive input values**

1	2	3	4	5	6	7	8
Segment number	Number of intervals × interval size	Value at segment end points	Decision value number n	Decision value x_n (see Note 1)	Character signal Bit number 1 2 3 4 5 6 7 8	Value at decoder output y_n (see Note 3)	Decoder output value number
		8159	(128)	(8159)			
					1 0 0 0 0 0 0 0	8031	127
8	16 × 256		127	7903	(see Note 2)		
			113	4319			
					1 0 0 0 1 1 1 1	4191	112
		4063	112	4063			
7	16 × 128				(see Note 2)		
			97	2143			
					1 0 0 1 1 1 1 1	2079	96
		2015	96	2015			
6	16 × 64				(see Note 2)		
			81	1055			
					1 0 1 0 1 1 1 1	1023	80
		991	80	991			
5	16 × 32				(see Note 2)		
			65	511			
					1 0 1 1 1 1 1 1	495	64
		479	64	479			
4	16 × 16				(see Note 2)		
			49	239			
					1 1 0 0 1 1 1 1	231	48
		223	48	223			
3	16 × 8				(see Note 2)		
			33	103			
					1 1 0 1 1 1 1 1	99	32
		95	32	95			
2	16 × 4				(see Note 2)		
			17	35			
					1 1 1 0 1 1 1 1	33	16
		31	16	31			
	15 × 2				(see Note 2)		
			2	3			
1					1 1 1 1 1 1 1 0	2	1
	1 × 1		1	1	1 1 1 1 1 1 1 1		
↓			0	0		0	0

Note 1. – 8159 normalized value units correspond to T_{max} = 3.17 dBm0.

Note 2. – The character signal corresponding to positive input values between two successive decision values numbered n and $n + 1$ (see column 4) is $(255 - n)$ expressed as a binary number.

Note 3. – The value at the decoder output is $y_0 = x_0 = 0$ for $n = 0$, and $y_n = \frac{x_n + x_{n+1}}{2}$ for $n = 1, 2, ..., 127$.

Note 4. – x_{128} is a virtual decision value.

TABLE 2*b*/G.711 – **μ-law, negative input values**

1	2	3	4	5	6	7	8
Segment number	Number of intervals × interval size	Value at segment end points	Decision value number n	Decision value x_n (see Note 1)	Character signal Bit number 1 2 3 4 5 6 7 8	Value at decoder output y_n (see Note 3)	Decoder output value number
			0	0		0	0
	1 × 1				0 1 1 1 1 1 1 1		
1			1	–1			
					0 1 1 1 1 1 1 0	–2	1
			2	–3			
	15 × 2				(see Note 2)		
		–31	16	–31			
					0 1 1 0 1 1 1 1	–33	16
			17	–35			
2	16 × 4				(see Note 2)		
		–95	32	–95			
					0 1 0 1 1 1 1 1	–99	32
			33	–103			
3	16 × 8				(see Note 2)		
		–223	48	–223			
					0 1 0 0 1 1 1 1	–231	48
			49	–239			
4	16 × 16				(see Note 2)		
		–479	64	–479			
					0 0 1 1 1 1 1 1	–495	64
			65	–511			
5	16 × 32				(see Note 2)		
		–991	80	–991			
					0 0 1 0 1 1 1 1	–1023	80
			81	–1055			
6	16 × 64				(see Note 2)		
		–2015	96	–2015			
					0 0 0 1 1 1 1 1	–2079	96
			97	–2143			
7	16 × 128				(see Note 2)		
		–4063	112	–4063			
					0 0 0 0 1 1 1 1	–4191	112
			113	–4319			
					(see Note 2)		
8	16 × 256		126	–7647			
					0 0 0 0 0 0 0 1	–7775	126
			127	–7903			
					0 0 0 0 0 0 0 0	–8031	127
		–8159	(128)	(–8159)			

Note 1. – 8159 normalized value units correspond to $T_{max} = 3.17$ dBm0.

Note 2. – The character signal corresponding to negative input values between two successive decision values numbered n and $n + 1$ (see column 4) is $(127 - n)$ expressed as a binary number for $n = 0, 1, ..., 127$.

Note 3. – The value at the decoder output is $y_0 = x_0 = 0$ for $n = 0$, and $y_n = \frac{x_n + x_{n+1}}{2}$ for $n = 1, 2, ..., 127$.

Note 4. – x_{128} is a virtual decision value.

TABLE 3/G.711 – μ-A conversion

μ-law Decoder output value number	*A-law* Decoder ouptut value number	*μ-law* Decoder output value number	*A-law* Decoder output value number
0	1	44	41
1	1	45	42
2	2	46	43
3	2	47	44
4	3	48	46
5	3	49	48
6	4	50	49
7	4	51	50
8	5	52	51
9	5	53	52
10	6	54	53
11	6	55	54
12	7	56	55
13	7	57	56
14	8	58	57
15	8	59	58
16	9	60	59
17	10	61	60
18	11	62	61
19	12	63	62
20	13	64	64
21	14	65	65
22	15	66	66
23	16	67	67
24	17	68	68
25	18	69	69
26	19	70	70
27	20	71	71
28	21	72	72
29	22	73	73
30	23	74	74
31	24	75	75
32	25	76	76
33	27	77	77
34	29	78	78
35	31	79	79
36	33	80	80
37	34	81	82
38	35	82	83
39	36	83	84
40	37	84	85
41	38	85	86
42	39	86	87
43	40	87	88
		.	.
		.	.
		.	.
		127	128

TABLE 4/G.711 – A-μ conversion

A-law Decoder output value number	*μ-law* Decoder output value number	*A-law* Decoder output value number	*μ-law* Decoder output value number
1	1	51	52
2	3	52	53
3	5	53	54
4	7	54	55
5	9	55	56
6	11	56	57
7	13	57	58
8	15	58	59
9	16	59	60
10	17	60	61
11	18	61	62
12	19	62	63
13	20	63	64
14	21	64	64
15	22	65	65
16	23	66	66
17	24	67	67
18	25	68	68
19	26	69	69
20	27	70	70
21	28	71	71
22	29	72	72
23	30	73	73
24	31	74	74
25	32	75	75
26	32	76	76
27	33	77	77
28	33	78	78
29	34	79	79
30	34	80	80
31	35	81	80
32	35	82	81
33	36	83	82
34	37	84	83
35	38	85	84
36	39	86	85
37	40	87	86
38	41	88	87
39	42	89	88
40	43	90	89
41	44	91	90
42	45	92	91
43	46	93	92
44	47	94	93
45	48	95	94
46	48	96	95
47	49	97	96
48	49	98	97
49	50	.	.
50	51	.	.
		.	.
		128	127

TABLE 5/G.711

A-law							
1	*2*	*3*	*4*	*5*	*6*	*7*	*8*
0	0	1	1	0	1	0	0
0	0	1	0	0	0	0	1
0	0	1	0	0	0	0	1
0	0	1	1	0	1	0	0
1	0	1	1	0	1	0	0
1	0	1	0	0	0	0	1
1	0	1	0	0	0	0	1
1	0	1	1	0	1	0	0

TABLE 6/G.711

μ-law							
1	*2*	*3*	*4*	*5*	*6*	*7*	*8*
0	0	0	1	1	1	1	0
0	0	0	0	1	0	1	1
0	0	0	0	1	0	1	1
0	0	0	1	1	1	1	0
1	0	0	1	1	1	1	0
1	0	0	0	1	0	1	1
1	0	0	0	1	0	1	1
1	0	0	1	1	1	1	0

Recommendation G.712

PERFORMANCE CHARACTERISTICS OF PCM CHANNELS AT AUDIO FREQUENCIES

(Geneva, 1972; amended at Geneva, 1976)

1. *General*

The CCITT

recommends

that the performance characteristics which follow should be met between the audio-frequency ports of PCM channels coded in accordance with Recommendation G.711.

The performance limits quoted are to be considered as recommendations to be met in all cases, making due allowance for any inaccuracy in the testing techniques.

The values and limits specified are those which should be obtained in 4-wire measurements using two PCM multiplex terminal equipments connected back-to-back (except for 5.3 below) and with the input and output ports of the channels terminated with their nominal impedance.

Further study is required for the method of measurement of the send and the receive sides separately.

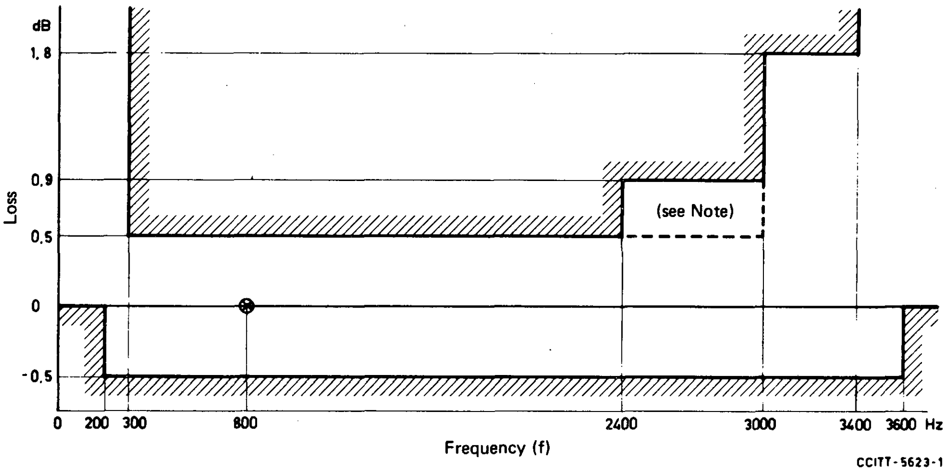
2. *Attenuation/frequency distortion*

The variations with frequency of the attenuation of any channel should lie within the limits shown in the mask of Figure 1/G.712.

The reference frequency is 800 Hz.

The input power level should be 0 dBm0.

The distortion contributed by the separate send and receive sides of the equipment should be nominally equal.



Note. – In some applications in which several PCM channels may be connected in tandem, it may be necessary to extend the +0.5 dB-limit from 2400 Hz to 3000 Hz.

FIGURE 1/G.712 – **Attenuation/frequency distortion**

3. *Envelope delay distortion with frequency*

The envelope delay distortion should lie within the limits shown in the template of Figure 2/G.712.

The minimum value of the group propagation delay is taken as a reference.

The input power level should be 0 dBm0.

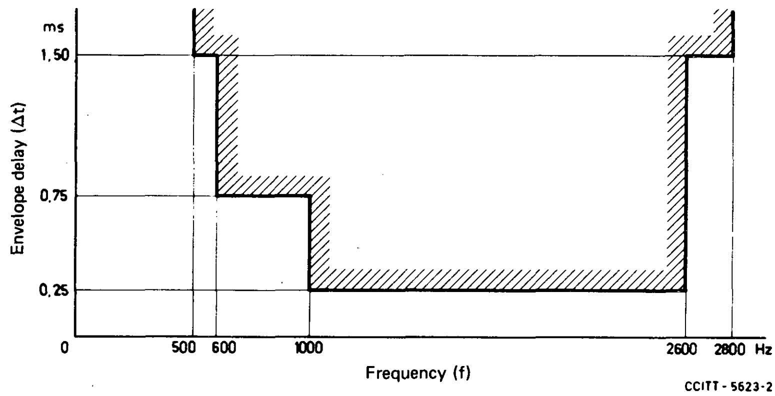


FIGURE 2/G.712 – **Envelope delay distortion with frequency**

4. *Impedance of audio-frequency ports*

4.1 *Nominal impedance*

The nominal impedance at the 4-wire audio input and output ports should be 600 ohms, balanced.

4.2 *Return loss* (provisional recommendation)

The departure from the nominal value, measured as return loss against the nominal value, should not be less than 20 dB over the frequency range 300 to 3400 Hz.

Note. – The return loss is under study (see Question 9/XVI).

4.3 *Longitudinal balance*

Under study.

5. *Idle channel noise*

5.1 *Weighted noise*

With the input and output ports of the channel terminated in the nominal impedance, the idle channel noise should not exceed −65 dBm0p.

5.2 *Single frequency noise*

The level of any single frequency (in particular the sampling frequency and its multiples), measured selectively, should not exceed −50 dBm0.

5.3 *Receiving equipment noise*

Noise contributed by the receiving equipment alone should be less than −75 dBm0p when its input is driven by a PCM signal corresponding to the decoder output value number 0 for the μ-law or decoder output value number 1 for the A-law.

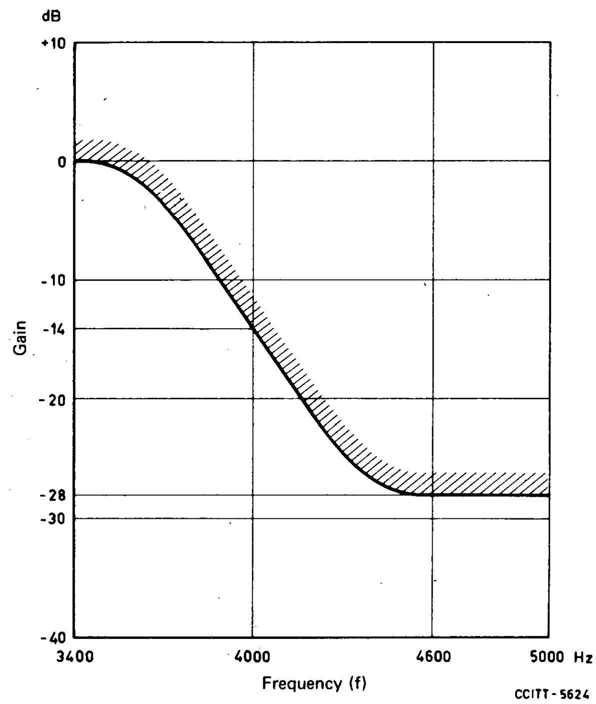
6. *Discrimination against out-of-band input signals*

6.1 With any sine-wave signal in the range 4.6-72 kHz applied to the input port of the channel at a suitable level, the level of any image frequency produced at the output port of the channel should, as a minimum requirement, be at least 25 dB below the level of the test signal.

6.2 Under the most adverse conditions encountered in a national network, the PCM channel should not contribute more than 100 pW0p of additional noise in the band 0-4 kHz at the channel output, as a result of the presence of out-of-band signals at the channel input.

Note 1. – The discrimination required depends on the performance of FDM channel equipments and telephone instruments in national networks and individual Administrations should carefully consider the requirements they should specify taking into account the comments above and the requirement of 6.2 above. In all cases at least the minimum requirement of 6.1 above should be met.

Note 2. – Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the requirements 6.1 and 6.2 above, one Administration has found that the filter template of Figure 3/G.712 (which relates to a stopband attenuation of 28 dB) gives adequate protection against the out-of-band signals found in their national network.



Note. – The curved portion of the graph conforms to the equation $G = 14\left[\sin\frac{\pi\,(4000-f)}{1200} - 1\right]$ dB for the range $3400 \leqslant f \leqslant 4600$.

FIGURE 3/G.712 – **Gain relative to gain at 800 Hz**

7. *Spurious out-of-band signals at the channel output*

7.1 With any sine-wave signal in the range 300-3400 Hz at a level of 0 dBm0 applied to the input port of a channel, the level of spurious out-of-band image signals measured selectively at the output port should be lower than −25 dBm0.

7.2 The spurious out-of-band signals should not give rise to unacceptable interference in equipment connected to the PCM channel. In particular, the intelligible or unintelligible crosstalk, in a connected FDM channel should not exceed a level of −65 dBm0 as a consequence of the spurious out-of-band signals at the PCM channel output.

Note 1. – The discrimination required depends on the performance of FDM channel equipment and telephone instruments in national networks and individual Administrations should carefully consider the requirements they should specify taking into account the comments above and the requirement of 7.2 above. In all cases at least the minimum requirement of 7.1 above should be met.

Note 2. – Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the requirements 7.1 and 7.2 above, one Administration has found that the filter template of Figure 3/G.712 (which relates to a stopband attenuation of 28 dB) gives adequate protection against the out-of-band signals found in their national network.

8. *Intermodulation*

8.1 Two sine-wave signals of different frequencies f_1 and f_2 not harmonically related, in the range 300-3400 Hz and of equal levels in the range −4 to −21 dBm0, applied simultaneously to the input port of a channel should not produce any $2f_1 - f_2$ intermodulation product having a level greater than −35 dB relative to the level of one of the two input signals.

8.2 A signal having a level of −9 dBm0 at any frequency in the range 300-3400 Hz and a signal of 50 Hz at a level of −23 dBm0 applied simultaneously to the input port should not produce any intermodulation product of a level exceeding −49 dBm0.

9. *Total distortion, including quantizing distortion*

Two alternative methods are recommended.

Method 1

With a suitable noise signal applied to the input port of a channel, the ratio of signal-to-total distortion power measured at the output port should lie above the limits shown in Figure 4/G.712.

Note 1. – These limits are based on a noise signal having a gaussian distribution of amplitudes and the derivation of the limits is given in Annex 1.

Note 2. – Appropriate corrections must be made depending on the characteristics of the test apparatus in order that the results of measurements may be related correctly to the specified limits (see Recommendation O.131 concerning the basic specification clauses for a quantizing distortion measuring apparatus giving a pseudo-random noise stimulus).

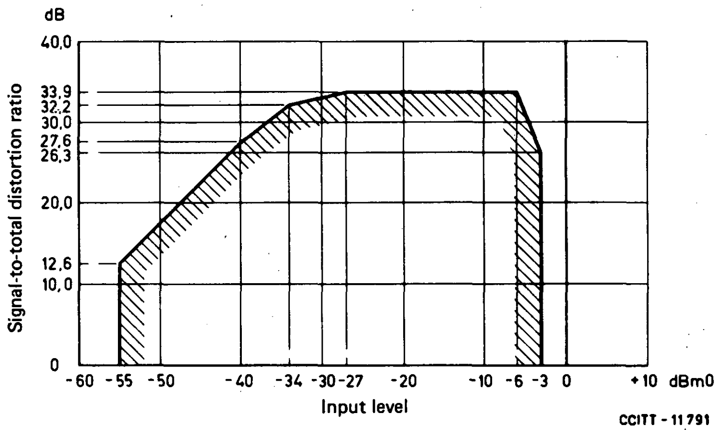


FIGURE 4/G.712 – **Signal-to-total distortion ratio as a function of input level (Method 1)**

Method 2

With a sine-wave signal in the frequency range 700-1100 Hz (excluding submultiples of 8 kHz), applied to the input port of a channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see 7. of Recommendation G.223) should lie above the limits shown in Figure 5/G.712.

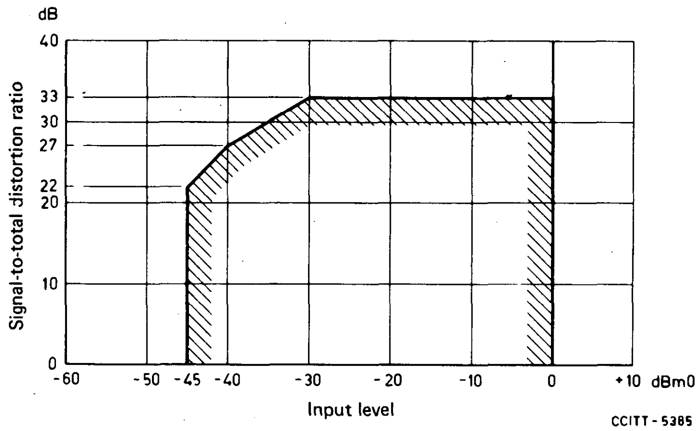


FIGURE 5/G.712 – **Signal-to-total distortion ratio as a function of input level (Method 2)**

10. *Spurious in-band signals at the channel output port*

With a sine-wave signal in the frequency range 700-1100 Hz (excluding submultiples of 8 kHz) and at a level of 0 dBm0 applied to the input port of a channel, the output level at any frequency other than the frequency of the applied signal, measured selectively in the frequency band 300-3400 Hz, should be less than –40 dBm0.

11. *Variation of gain with input level*

Two alternative methods are recommended.

Method 1

With a suitable noise signal applied to the input of any channel at a level between –60 dBm0 and –10 dBm0, the gain variation of that channel, relative to the gain at an input level of –10 dBm0, should lie within the limits of the mask of Figure 6a/G.712.

Furthermore, with a sine-wave signal in the frequency range 700-1100 Hz (excluding submultiples of 8 kHz) applied to the input port of any channel at a level between –10 dBm0 and +3 dBm0, the gain variation of that channel relative to the gain at an input level of –10 dBm0, should lie within the limits of the mask of Figure 6b/G.712.

Method 2

With a sine-wave signal in the frequency range 700-1100 Hz (excluding submultiples of 8 kHz) applied to the input port of any channel at a level between –55 dBm0 and +3 dBm0, the gain variation of that channel relative to the gain at an input level of –10 dBm0, should lie within the limits of the mask of Figure 6c/G.712.

12. *Interchannel crosstalk*

12.1 The crosstalk between individual channels of a multiplex should be such that with a sine-wave signal in the frequency range 700-1100 Hz (excluding submultiples of 8 kHz) and a level of 0 dBm0 applied to an input port, the crosstalk level received in any other channel should not exceed –65 dBm0.

12.2 When a white noise signal shaped in accordance with Recommendation G.227, at a level of 0 dBm0 is applied to the input port of up to four channels, the level of the crosstalk received in any other channel should not exceed –60 dBm0p. Uncorrelated noise should be used when more than one input channel is energized.

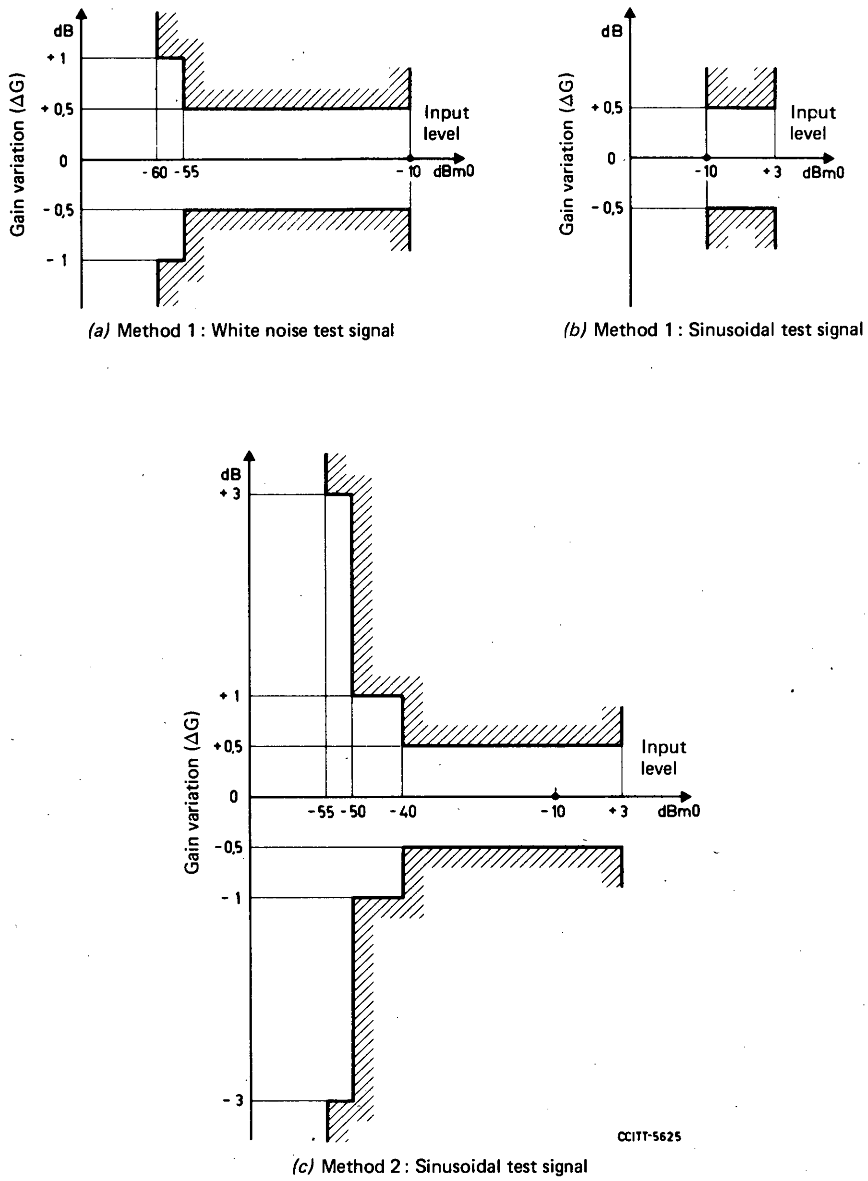


FIGURE 6/G.712 – **Variation of gain (ΔG) with input level**

13. *Go-to-return crosstalk*

The near-end signal to crosstalk ratio between one channel and its associated return channel should be better than 60 dB when using a sine-wave signal at 0 dBm0 at any frequency in the range 300-3400 Hz.

14. *Interference from signalling*

The maximum level of any interference into a channel should not exceed –60 dBm0 when signalling is active simultaneously on all channels.

15. *Relative levels at input and output*

The specifications should conform to Recommendation G.232, L.

16. *Short-term and long-term stability*

When a sine-wave signal at a level of 0 dBm0 is applied to any audio-frequency input, the level measured at the corresponding audio frequency output should not vary by more than ± 0.2 dB during any 10-minute interval of typical operation, nor by more than ± 0.5 dB during any 30-day interval, nor by more than ± 1 dB during any one year under the permitted variations in the power supply voltage and temperature.

17. *Adjustment of relationship between encoding law and audio level*

The decoder should be adjusted to conform with 4. of Recommendation G.711 within a tolerance of ± 0.3 dB in practice.

The encoder is adjusted by connecting its output to the input of a properly calibrated decoder, and applying a sine-wave signal of 1 kHz at a level of 0 dBm0 to the audio input of the encoder. The encoder is then adjusted so that the resulting 1 kHz sine-wave signal at the audio output of the decoder is at a level of 0 dBm0. In practice the adjustment should be made with a tolerance of ± 0.3 dB.

The load capacity of the encoder may be checked by applying a sine-wave signal at its audio input. The level of this signal is initially well below T_{max} and is then slowly increased. The input level is measured at which the first occurrence is observed of the character signal corresponding to the extreme quantizing interval for both positive and negative values. T_{max} is then taken as being 0.3 dB greater than the measured input level.

This method allows T_{max} to be checked for both positive and negative amplitudes and the values thus obtained should be within ± 0.3 dB of the theoretical load capacity (i.e. +3.14 dB for the A-law or +3.17 dB for the μ-law).

As an alternative, the occurrence of the largest pulse amplitude at the decoder output may be used as a means of identifying T_{max}.

ANNEX 1

(to Recommendation G.712)

Method of derivation of the signal-to-total distortion ratio for the A-law

The signal-to-quantizing distortion ratio produced by PCM systems can be obtained analytically in a number of different ways. The method adopted here is a special case of a more general analysis which enables the calculated results to be compared directly with those obtained by practical measurements of the systems.

The compression characteristic of the system is assumed to be "ideal" – i.e. to meet precisely the theoretical segmented law, with the system a.c. zero coincident with the centre decision amplitude. The input signal is assumed to be symmetrical about a.c. zero, and to have a gaussian distribution of instantaneous amplitudes. For a given input, of variance σ_v^2, the total output variance may be determined as σ_u^2, and the variance of the signal content in the output, by linear regression, as $m^2 \sigma_v^2$ where m is the slope of the regression line of output on input.

The variance of the distortion components is then $\sigma_\varepsilon^2 = \sigma_u^2 - m^2\sigma_v^2$, and the signal-to-quantizing distortion ratio in dB is:

$$10 \log_{10} \frac{m^2 \sigma_v^2}{\sigma_\epsilon^2}$$

For input signals within the normal working range of the system – i.e. excluding the gross overload and dichotomized regions – the distortion power [8] can be assumed to have a uniform spectrum in the band 0 to 4000 Hz. The distortion power present in the filtered band 300 to 3400 Hz is therefore less than the power calculated from the above expression by

$$10 \log_{10} \frac{4000}{3100} = 1.1 \text{ dB}$$

The values of signal-to-total distortion ratios quoted in Figure 4/G.712 have been derived by subtracting 4.5 dB from the calculated signal to quantizing distortion ratio relating to the band 300-3400 Hz. Flat and not psophometric weighting is used throughout.

7.2 General Recommendations on digital systems and paths

Recommendation G.721

HYPOTHETICAL REFERENCE DIGITAL PATHS

(Geneva, 1976)

General definitions

hypothetical reference digital path

This is a hypothetical digital path of defined length and with a specified number of terminal and intermediate equipments, this number being sufficient but not excessive.

It forms a basis for the study of certain characteristics of long-distance digital paths (errors, jitter for example).

The design objectives recommended by the CCITT for transmission equipments are commonly expressed in terms of a maximum tolerable level of impairment arising in a hypothetical reference digital path.

As far as possible a design objective so expressed takes into account all possible usages of the system, e.g., for telephony, telegraphy, data, etc.

hypothetical reference digital path at 64 kbit/s

This is a complete digital path (between 64-kbit/s interfaces) established on a hypothetical international digital system and having a specified length and a specified number of multiplexing and demultiplexing equipments, these numbers being reasonably great but not having their maximum possible values.

Various "hypothetical reference digital paths" have been defined to allow the coordination of the different specifications concerning the constituent parts of the digital systems so that the complete connections set up on these systems can meet CCITT standards.

The CCITT has defined the following hypothetical reference digital paths:

- 2-Mbit/s system (see Figure 1/G.721),
- 8-Mbit/s system (see Figure 2/G.721),
- 34-Mbit/s system (see Figure 3/G.721),
- 149-Mbit/s system (see Figure 4/G.721).

(Other hypothetical reference digital paths are under study.)

[8] Signal and distortion powers are, of course, proportional to the respective variances mentioned above; the coefficient of proportionality involves only the impedance of the system.

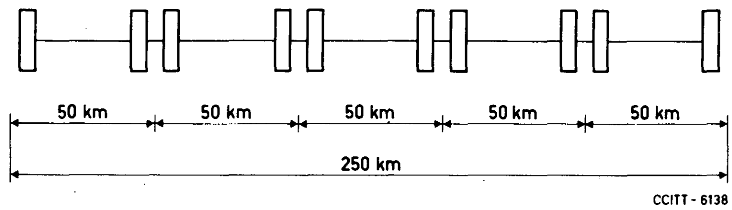


FIGURE 1/G.721 – **Hypothetical reference digital path for 2-Mbit/s systems**

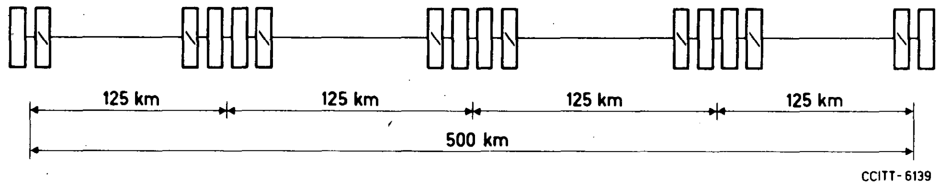


FIGURE 2/G.721 – **Hypothetical reference digital path for 8-Mbit/s system**

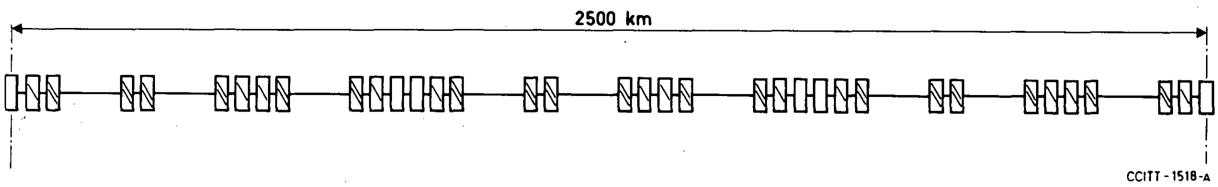


FIGURE 3/G.721 – **Hypothetical reference digital path for 4-MHz systems which could also be used for 34-Mbit/s systems**

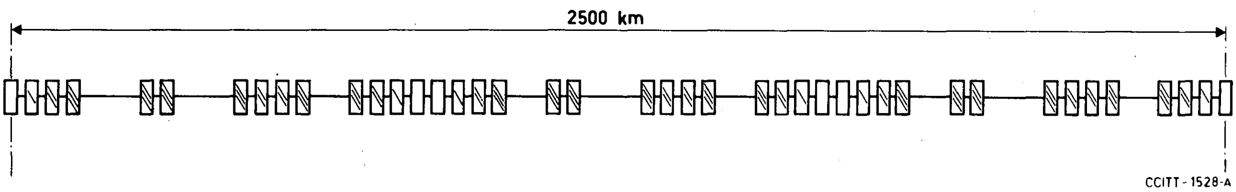


FIGURE 4/G.721 – **Hypothetical reference digital path for 140-Mbit/s systems**

Each of these various hypothetical reference digital paths has a specified length and they are all used in the same way. They are a basis for the design of the transmission systems to which they relate, respectively, and of other transmission equipments which may be used in association with them.

In addition, because of the constitution of these hypothetical reference digital paths, they can be used not only to study the case of a path of the specified length, set up on a digital system or systems, but also that of an international connection having the same total length and made up of digital paths set up on different digital systems.

A homogeneous section is a section without diversion of multiplexing or demultiplexing of any one of the digital signals established on the system which is being considered, except for those multiplexing equipments defined at the ends of the section.

It is assumed that at the end of each homogeneous section, the digital paths, as appropriate, are connected through at random.

7.3 Principal characteristics of primary multiplex equipment

Recommendation G.731

PRIMARY PCM MULTIPLEX EQUIPMENT FOR VOICE FREQUENCIES

(Geneva, 1972)

The CCITT

considering,

that pulse modulation (PCM) multiplex equipments are already used in various countries, in particular to provide a large number of short-distance telephone circuits on certain pairs in existing cables, and in order to minimize the number of different PCM multiplex equipments providing circuits which may be used in international connections,

recommends

that Administrations concerned should make their choice between the two primary PCM multiplex equipments described in Recommendations G.732 and G.733, until the appropriate studies, undertaken with a view to the establishment of a single standard, are completed.

Recommendation G.732

CHARACTERISTICS OF PRIMARY PCM MULTIPLEX EQUIPMENT OPERATING AT 2048 kbit/s

(Geneva, 1972; amended at Geneva, 1976)

1. *General characteristics*

1.1 *Fundamental characteristics*

The encoding law used in the A-law as specified in Recommendation G.711. The sampling rate, load capacity and the code are also specified in that Recommendation.

The number of quantized values is 256.

Note. – The inversion of bits 2, 4, 6, and 8 is covered by the encoding law and is applicable only to voice-channel time slots.

1.2 *Bit rate*

The nominal bit rate is 2048 kbit/s. The tolerance on this rate is ± 50 parts per million (ppm).

1.3 *Timing signal*

It should be possible to derive the transmitting timing signal of a PCM multiplex equipment from an internal source, from the incoming digital signal and also from an external source.

Note. – Further study is required on the effect of jitter of the incoming signal on the timing signal, and on the measures to be taken in case of loss of the incoming signal or the external source.

2. *Frame structure*

2.1 *Number of bits per channel time slot*

Eight, numbered from 1 to 8.

2.2 *Number of channel time-slots per frame*

Thirty-two, numbered from 0 to 31. The number of bits per frame is 256, and the frame repetition rate is 8000 Hz.

2.3 *Channel time-slot assignment*

2.3.1 Channel time-slots 1 to 15 and 17 to 31 are assigned to 30 telephone channels numbered from 1 to 30.

2.3.2 The allocation of the bits of channel time slot 0 is given in Table 1/G.732.

TABLE 1/G.732 – **Allocation of bits in channel time slot 0**

<table>
<tr><td rowspan="2"></td><td colspan="8">Bit number</td></tr>
<tr><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td></tr>
<tr><td rowspan="2">Time-slot 0 containing the frame alignment signal</td><td rowspan="2">Reserved for international use (see Note 1)</td><td>0</td><td>0</td><td>1</td><td>1</td><td>0</td><td>1</td><td>1</td></tr>
<tr><td colspan="7">Frame alignment signal (see 2.4)</td></tr>
<tr><td>Time-slot 0 not containing the frame alignment signal</td><td>Reserved for international use (see Note 1)</td><td>1 (see 2.4)</td><td>Alarm indication to the remote PCM multiplex equipment (see 3.2.3)</td><td colspan="5">Reserved for national use (see Note 2)</td></tr>
</table>

Note 1. – The use will be defined at a later stage. For the moment these bits are fixed at 1.

Note 2. – The bits allocated for national use may not be used internationally. On the digital path crossing the border, they are fixed at 1.

2.3.3 Channel time slot 16 is assigned to signalling as covered in 4. below. If channel time slot 16 is not needed for signalling it may be used for purposes other than a voice channel encoded within the PCM multiplex equipment.

2.4 *Frame alignment signal*

As shown in Table 1/G.732 the frame alignment signal occupies positions 2 to 8 in channel time-slot 0 of every other frame.

The frame alignment signal is:

0 0 1 1 0 1 1.

In order to avoid simulation of the frame alignment signal by bits 2 to 8 of channel time slot 0 in frame not containing the frame alignment signal, bit 2 in those channel time slots is fixed at 1.

2.5 *Loss and recovery of frame alignment*

Frame alignment will be assumed to have been lost when three or four consecutive frame alignment signals have been received with an error.

Frame alignment will be assumed to have been recovered when the following sequence is detected:

- for the first time, the presence of the correct frame alignment signal;
- the absence of the frame alignment signal in the following frame detected by verifying that bit 2 in channel time-slot 0 is a 1;
- for the second time, the presence of the correct frame alignment signal in the next frame.

Note. – To avoid the possibility of a state in which no frame alignment can be achieved due to the presence of an imitative frame alignment signal the following procedure may be used:

When a valid frame alignment signal is detected in frame n, a check should be made to ensure that a frame alignment signal does not exist in frame $n + 1$, and also that a frame alignment signal exists in frame $n + 2$. Failure to meet one or both of these requirements should cause a new search to be initiated in frame $n + 2$.

3. *Fault conditions and consequent actions*

3.1 *Fault conditions*

The PCM multiplex equipment should detect the following fault conditions:

3.1.1 Failure of power supply.

3.1.2 Failure of codec (except when using single-channel codecs).

Note. – As a minimum requirement, this fault condition should be recognized when, for at least one signal level in the range -21 to -6 dBm0, the signal-to-quantizing noise ratio performance of the local codec is 18 dB or more below the level recommended in Recommendation G.712. One Administration noted that this requirement allows for adequate monitoring of the speech capability of the codec. However, transmission of Signalling System R2 information through the codec imposes more stringent requirements on the codec which may not be covered by the above requirement. Further study is required.

3.1.3 Loss of incoming signal at the 64-kbit/s input port (time-slot 16).

Note. – The detection of this fault condition is not mandatory when channel-associated signalling is used and the signalling multiplex equipment is situated within a few metres of the PCM multiplex equipment.

3.1.4 Loss of the incoming signal at 2048 kbit/s.

Note 1. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

Note 2. – Where separate circuits are used for the digital signal and the timing signal, the loss of either or both should constitute loss of the incoming signal.

3.1.5 Loss of frame alignment

3.1.6 Excessive error rate, detected in frame alignment signal.

3.1.6.1 Criteria for activating the indication of fault condition:

- Error rate $\leqslant$ 1 in 10^4

  The probability of activating the indication of fault condition in a few seconds should be less than 10^{-6}.

- Error rate $\geqslant$ 1 in 10^3

  The probability of activating the indication of fault condition in a few seconds should be higher than 0.95.

3.1.6.2 Criteria for deactivating the indication of fault condition:

– Error rate ⩾ 1 in 10^3

The probability for deactivating the indication of fault condition in a few seconds should be almost 0.

– Error rate ⩽ 1 in 10^4

The probability for deactivating the indication of fault condition in a few seconds should be higher than 0.95.

Note. – The activating and the deactivating period specified as "a few seconds" is intended to be in the order of 4 to 5 seconds.

3.1.7 Alarm indication received from the remote PCM multiplex equipment (see 3.2.3 below).

3.2 *Consequent actions*

Further to the detection of a fault condition, appropriate actions should be taken as specified in Table 2/G.732. The consequent actions are as follows:

TABLE 2/G.732 – **Fault conditions and consequent actions for the PCM multiplex equipment**

Equipment part	Fault conditions (see 3.1)	Consequent actions (see 3.2)					
		Service alarm indication generated	Prompt maintenance alarm indication generated	Alarm indication to the remote end transmitted	Transmission suppressed at the analogue outputs	AIS applied to 64-kbit/s output (TS16)	AIS applied to TS16 of the 2048 kbit/s composite signal
Multiplexer and demultiplexer	Failure of power supply	Yes	Yes	Yes, if practicable	Yes, if practicable	Yes, if practicable	Yes, if practicable
	Failure of codec	Yes	Yes	Yes	Yes		
Multiplexer only	Loss of incoming signal at 64-kbit/s input (TS16)		Yes				Yes
Demultiplexer only	Loss of incoming signal at 2048 kbit/s	Yes	Yes	Yes	Yes	Yes	
	Loss of frame alignment	Yes	Yes	Yes	Yes	Yes	
	Error rate 1 in 10^3 for the alignment signal	Yes	Yes	Yes	Yes	Yes	
	Alarm indication received from the remote end	Yes					

Note. – A *Yes* in the table signifies that an action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

3.2.1 Service alarm indication generated to signify that the service provided by the PCM multiplex is no longer available. This indication should be forwarded at least to the switching and/or signalling multiplex equipment depending upon the arrangements provided. The indication should be given as soon as possible and not later than 2 ms after detection of the relevant fault condition.

This specification, taking into account the specification given in 2.5 above, is equivalent to recommend that the average time to detect a loss of frame alignment and to give the relevant indication should not be greater than 3 ms.

When using common-channel signalling, the indication should be forwarded to the switching equipment by means of a separate interface on the PCM multiplex equipment.

3.2.2 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 1 in 3.2.6) is detected, the prompt maintenance alarm indication associated with loss of frame alignment (see 3.1.5 above) and excessive error rate (see 3.1.6 above) should be inhibited.

Note. – The location and provision of any visual and/or audible alarm activated by the alarm indications given in 3.2.1 and 3.2.2 above, is left to the discretion of each Admintration.

3.2.3 Alarm indication to the remote end, transmitted by changing bit 3 of channel time-slot 0 from the state 0 to the state 1 in those frames not containing the frame alignment signal. This should be effected as soon as possible.

3.2.4 Transmission suppressed at the analogue outputs.

3.2.5 AIS applied to time-slot 16 64-kbit/s output (see Note 1). This action should be taken as soon as possible and not later than 2 ms after the detection of the fault condition.

3.2.6 AIS applied to time-slot 16 of the output 2048-kbit/s composite signal.

Note 1. – The equivalent binary content of the Alarm Indication Signal (AIS) is a continuous stream of 1s.

Note 2. – All timing requirements quoted apply equally to restoration, subsequent to the fault condition clearing.

4. *Signalling*

The use of channel time-slot 16 is recommended for either common channel or channel associated signalling as required. The actions described in 3.2.1 above, consequent to the corresponding fault conditions according to Table 2/G.732 should be applied.

Channel time-slot 16 may be used to provide an interface at 64 kbit/s which shall be suitable for use with either common channel or channel associated signalling. The action described in 3.2.5 above, consequent to the corresponding fault conditions according to Table 2/G.732 should be applied.

The detailed requirements for the organization of particular signalling systems will be included in the specifications for those signalling systems.

4.1 *Common channel signalling*

Channel time-slot 16 may be used for common channel signalling up to a rate of 64 kbit/s. The method of obtaining signal alignment will form part of the particular common channel signalling specification. In this case the 64-kbit/s interface to be used for TS16 should be in accordance with 5. below and Recomendation G.703.

4.2 *Channel associated signalling*

This paragraph contains the recommended arrangement for use of the 64-kbit/s capability of channel time-slot 16 for channel associated signalling.

4.2.1 *Multiframe structure*

A multiframe comprises 16 consecutive frames (whose structure is given in 2.2 above) and these are numbered from 0 to 15.

The multiframe alignment signal is 0000 and occupies digit time-slots 1 to 4 of channel time-slot 16 in frame 0.

4.2.2 *Allocation of channel time-slot 16*

When channel time-slot 16 is used for channel associated signalling it provides a 64-kbit/s digital path which is subdivided into lower-rate paths using the multiframe alignment signal as a reference.

Details of the bit allocation are given in Table 3/G.732.

TABLE 3/G.732

Channel time-slot 16 of frame 0	Channel time-slot 16 of frame 1		Channel time-slot 16 of frame 2		Channel time-slot 16 of frame 15	
0000 *xyxx*	*abcd* channel 1	*abcd* channel 16	*abcd* channel 2	*abcd* channel 17	*abcd* channel 15	*abcd* channel 30

Note. – x = spare bit to be made 1 if not used.
y = bit used to indicate loss of multiframe alignment (see 4.2.4.2.3 below).
When bits b, c or d are not used they should have the value:
$b = 1$
$c = 0$
$d = 1$
It is recommended that the combination 0000 of bits a, b, c and d should not be used for signalling purposes for channels 1-15.

This bit allocation provides four 500-bit/s signalling channels designated a, b, c and d, for each telephone channel. With this arrangement, the signalling distortion of each signalling channel introduced by the PCM transmission system, will not exceed ± 2 ms.

4.2.3 *Loss and recovery of multiframe alignment*

Multiframe alignment should be assumed to have been lost when two consecutive multiframe alignment signals have been received with an error.

Multiframe alignment should be assumed to have been recovered as soon as the first correct multiframe alignment signal is detected.

Note. – To avoid a condition of spurious multiframe alignment, the following procedure may be used in addition to the above:

- Multiframe alignment should be assumed to have been lost when, for a period of one or two multiframes, all the bits in TS16 are in state 0.
- Multiframe alignment should be assumed to have been recovered only when at least one bit in state 1 is present in the time-slot 16 preceding the multiframe alignment signal first detected.

4.2.4 *Fault conditions and consequent actions*

4.2.4.1 *Fault conditions*

The signalling multiplex equipment should detect the following fault conditions:

4.2.4.1.1 Failure of power supply.

4.2.4.1.2 Loss of 64-kbit/s incoming signal at the input of the signalling demultiplexer.

Note 1. – The detection of this fault condition is not mandatory when the signalling multiplex equipment is situated within a few metres of the PCM multiplex equipment or when this fault condition results in an indication of loss of multiframe alignment.

Note 2. – Where separate circuits are used for the digital signal and the timing signal then loss of either or both should constitute loss of the incoming signal.

4.2.4.1.3 Loss of multiframe alignment.

4.2.4.1.4 Alarm indication received from the remote signalling multiplex equipment (see 4.2.4.2.3 below).

4.2.4.1.5 Receipt of the service alarm indication from the PCM multiplex equipment (see 3.2.1 above).

4.2.4.2 *Consequent actions*

Further to the detection of a fault condition appropriate actions should be taken as specified in Table 4/G.732. The consequent actions are as follows:

TABLE 4/G.732 – **Fault conditions and consequent actions for channel-associated signalling multiplex equipment**

Equipment part	Fault conditions (see 4.2.4.1)	Consequent actions (see 4.2.4.2)			
		Service alarm indication generated	Prompt maintenance alarm indication generated	Alarm indication to the remote end transmitted	Application of state, equivalent to state 1, on line to all receive signalling channels
Multiplexer and demultiplexer	Failure of power supply	Yes	Yes	Yes, if practicable	Yes, if practicable
Demultiplexer only	Loss of incoming signal	Yes	Yes	Yes	Yes
	Loss of multiframe alignment	Yes	Yes	Yes	Yes
	Alarm indication received from the remote signalling multiplex equipment	Yes			Yes
	Receipt of the service alarm indication from PCM mux	Yes			Yes

Note. – A *Yes* in the table signifies that an action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

4.2.4.2.1 Service alarm indication to be forwarded to the switching equipment depending upon the switching and signalling arrangements provided.

4.2.4.2.2 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenace attention is required locally. If provision is made for detecting the AIS, then on the reception of the AIS, the prompt maintenance alarm indication should be inhibited in the case of loss of multiframe (see 4.2.4.1.3 above).

Note. – The location and provision of any visual and or audible alarms activated by the alarm indications given in 4.2.4.2.1 and 4.2.4.2.2 above is left to the discretion of each Administration.

4.2.4.2.3 Alarm indication to the remote signalling multiplex equipment, generated by changing from the state 0 to the state 1 bit 6 of channel time-slot 16 of frame 0 of the multiframe (see Table 3/G.732); this should be effected as soon as possible.

4.2.4.2.4 Application of the condition corresponding to state 1 on the line to all receive signalling channels. This condition should be forwarded as soon as possible and not later than 3 ms after the detection of the fault condition.

Note. – All timing requirements quoted apply equally to restoration, subsequent to the fault condition clearing.

5. *Interfaces*

The analogue interfaces should comply with Recommendation G.712. The digital interfaces should comply with Recommendation G.703. In addition, the electrical characteristics of the 64-kbit/s interface of the codirectional and the contradirectional type are given in 5.1 and 5.2 respectively. These latter specifications are not mandatory for channel-associated signalling.

5.1 *Electrical characteristics of 64-kbit/s codirectional interface*

5.1.1 *General*

5.1.1.1 Nominal bit rate: 64 kbit/s.

5.1.1.2 Maximum tolerance of signals to be transmitted through the interface: ± 100 ppm.

5.1.1.3 64-kHz and 8-kHz timing signal to be transmitted in a codirectional way with the information signal.

5.1.1.4 One balanced pair for each direction of transmission; the use of transformers is recommended.

5.1.1.5 *Code conversion rules*

Step 1: A 64-kbit/s bit period is divided into four unit intervals.

Step 2: A binary one is coded as a block of the following four bits:

1 1 0 0

Step 3: A binary zero is coded as a block of the following four bits:

1 0 1 0

Step 4: The binary signal is converted into a three-level signal by alternating the polarity of consecutive blocks.

Step 5: The alternation in polarity of the blocks is violated every 8th block. The violation block marks the last bit in an octet.

These conversion rules are illustrated in Figure 1/G.732.

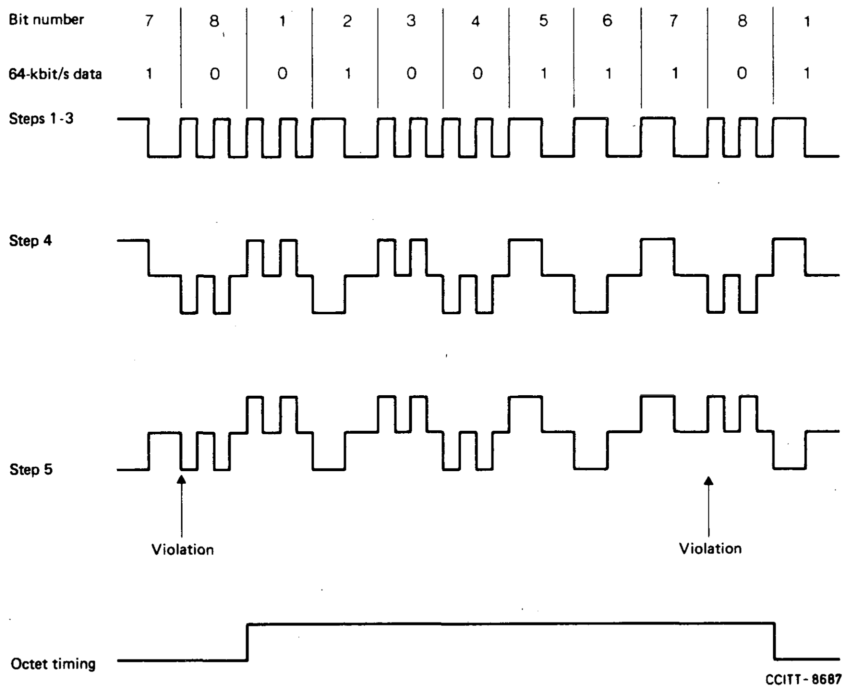
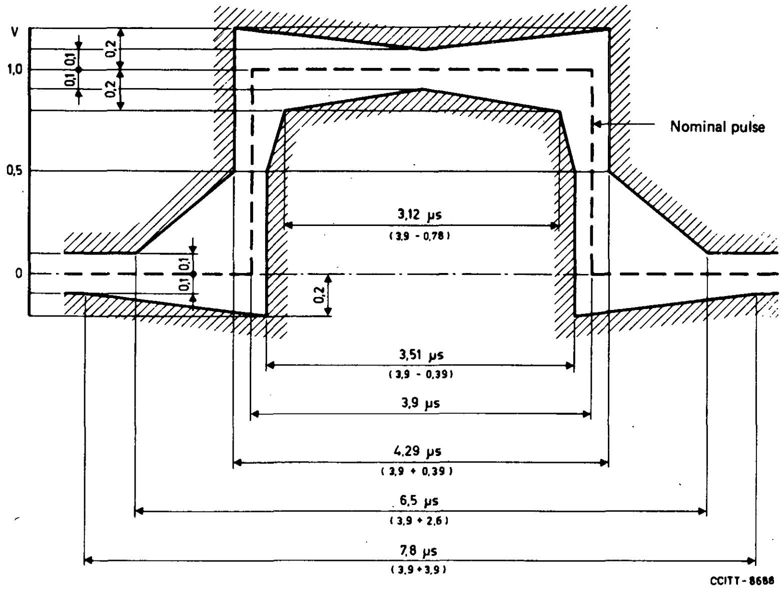


FIGURE 1/G.732 – **Steps in framing a codirectional 64-kbit/s signal**

5.1.2 *Specifications at the output ports* (see Table 5/G.732)

TABLE 5/G.732

Symbol rate	256 kbauds
Pulse shape (nominally rectangular)	All pulses of a valid signal must conform to the mask in Figure 2/G.732, irrespective of the polarity
Pair for each direction	One symmetric pair
Test load impedance	120 ohms resistive
Nominal peak voltage of a "mark" (pulse)	1.0 V
Peak voltage of a "space" (no pulse)	0 V ± 0.10 V
Nominal pulse width	3.9 μs
Ratio of the amplitudes of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05



Note 1. – The limits apply to pulses of either polarity.
Note 2. – For the double-width pulse the time-scale values are multiplied by 2.

FIGURE 2/G.732 – **Mask of the pulse of the 64-kbit/s codirectional interface**

5.1.3 *Specifications at the input ports*

The digital signal presented at the input port shall be as defined above but modified by the characteristics of the interconnecting pairs. The attenuation of these pairs at a frequency of 128 kHz should be in the range 0 to 3 dB. This attenuation should take into account any losses incurred by the presence of a digital distribution frame between the equipments.

Note. – If the symmetrical pair is screened, the screen shall be connected to the earth at the output port, and provision shall be made for connecting the screen of the symmetrical pair to earth, if required, at the input port.

5.2 *Electrical characteristics of 64-kbit/s contradirectional interface*

5.2.1 *General*

5.2.1.1 Bit rate: 64 kbit/s.

5.2.1.2 Maximum tolerance for signals to be transmitted through the interface: ± 100 ppm.

5.2.1.3 For each direction of transmission there should be two symmetrical pairs of wires, one pair carrying the data signal and the other carrying a composite timing signal (64 kHz and 8 kHz). The interface configuration including the directions of the signals involved is shown in Figure 3/G.732. The use of transformers is recommended.

Note. – If there is a national requirement to provide a separate alarm signal across the interface, this can be done by cutting the 8-kHz timing signal for the transmission direction concerned, that is, by inhibiting the code violations introduced in the corresponding composite timing signal (see below).

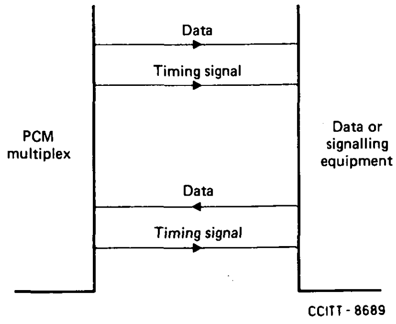


FIGURE 3/G.732 – **64-kbit/s contradirectional interface**

5.2.1.4. *Code conversion rules*

The data signals are coded in AMI code with a 100% duty ratio. The composite timing signals convey the 64-kHz bit-timing information using AMI code with a 50% duty ratio and the 8-kHz octet-phase information by introducing violations of the code rule. The structures of the signals and their phase relationships at data output ports are shown in Figure 4/G.732.

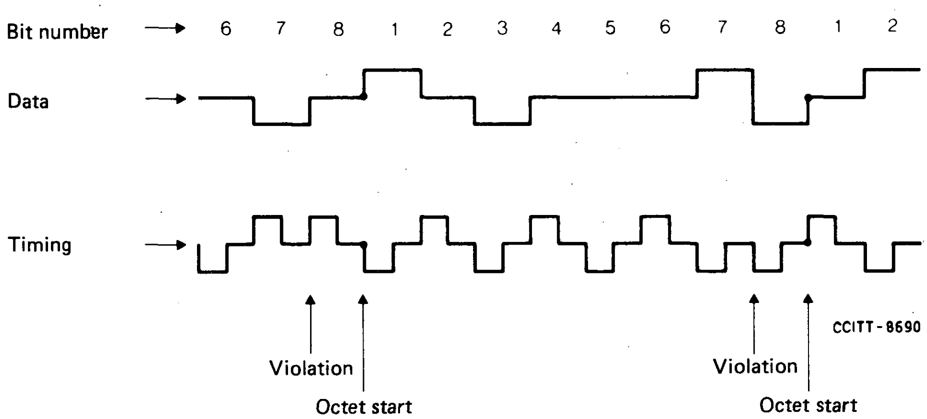


FIGURE 4/G.732 – **Signal structures of the 64-kbit/s contradirectional interface at data output ports**

The data pulses received by the PCM equipment will be somewhat delayed in relation to the corresponding timing pulses. The detection instant for a received data pulse on the PCM side of the interface should therefore be at the leading edge of the next timing pulse.

5.2.2 *Specifications at the output ports* (see Table 6/G.732)

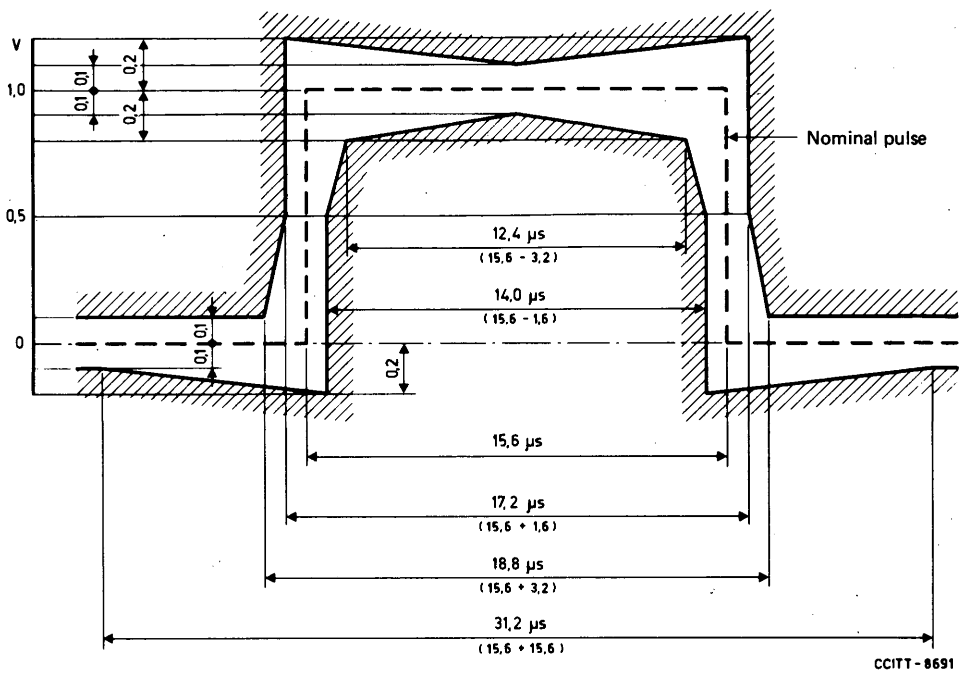
TABLE 6/G.732

Parameters	Data	Timing
Pulse shape (nominally rectangular)	All pulses of a valid signal must conform to the mask in Figure 5/G.732, irrespective of the polarity	All pulses of a valid signal must conform to the mask in Figure 6/G.732, irrespective of the polarity
Pairs in each direction of transmission	One symmetric pair	One symmetric pair
Test load impedance	120 ohm resistive	120 ohm resistive
Nominal peak voltage of a "mark" (pulse)	1.0 V	1.0 V
Peak voltage of a "space" (no pulse)	0 V ± 0.1 V	0 V ± 0.1 V
Nominal pulse width	15.6 μs	7.8 μs
Ratio of the amplitudes of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05	0.95 to 1.05
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05	0.95 to 1.05

5.2.3 *Specifications at the input ports*

The digital signals presented at the input ports should be as defined above but modified by the characteristics of the interconnecting pairs. The attenuation of these pairs at a frequency of 32 kHz should be in the range 0 to 3 dB. This attenuation should take into account any losses incurred by the presence of a digital distribution frame between the equipments.

Note. — If the symmetrical pairs are screened, the screens shall be connected to the earth at the output port, and provision shall be made for connecting the screens of the symmetrical pairs to earth, if required, at the input port.



Note 1. – When one pulse is immediately followed by another pulse of the opposite polarity, the time limits at the zero-crossing between the pulses should be ± 0.8 µs.

Note 2. – The time instants at which a transition from one state to another in the data signal may occur are determined by the timing signal. On the data or signalling side of the interface it is essential that these transitions are not initiated in advance of the timing instants given by the received timing signal.

FIGURE 5/G.732 – **Mask of the data pulse of the 64-kbit/s contradirectional interface**

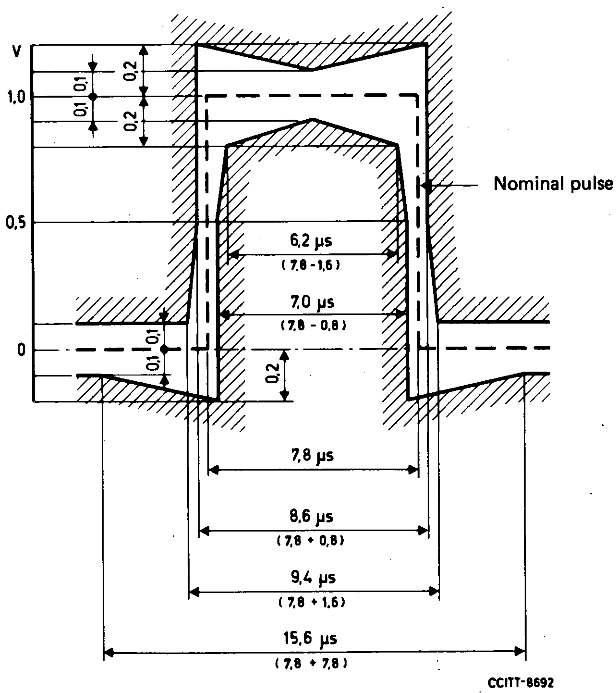


FIGURE 6/G.732 – **Mask of the timing pulse of the 64-kbit/s contradirectional interface**

Recommendation G.733

CHARACTERISTICS OF PRIMARY PCM MULTIPLEX EQUIPMENT OPERATING AT 1544 kbit/s

(Geneva, 1972; amended at Geneva, 1976)

1. *General characteristics*

1.1 *Fundamental characteristics*

The encoding law used is the μ-law as specified in Recommendation G.711. The sampling rate, load capacity and the code are also specified in that Recommendation.

The number of quantized values is 255. Two character signals are reserved for zero value (11111111 and 01111111).

In some networks the "all zero" character signal (00000000) is eliminated to avoid loss of timing information to the digital line, resulting in 254 quantized values.

1.2 *Bit rate*

The nominal bit rate is 1544 kbit/s. The tolerance on this rate is ± 50 parts per million (ppm).

1.3 *Timing signal*

It should be possible to derive the transmitting timing signal of a PCM multiplex equipment from an internal source, from the incoming digital signal and also from an external source.

2. *Frame structure*

2.1 *Number of bits per channel time-slot:* eight, numbered 1 to 8.

2.2 *Number of channel time-slots per frame:* twenty-four, numbered from 1 to 24.

One bit frame is added for a frame alignment signal and for a multiframe alignment signal or signalling. The number of bits per frame is 193, and the frame repetition rate is 8000 Hz.

2.3 *Channel time-slot assignment*

2.3.1 Channel time-slots 1 to 24 are assigned to 24 telephone channels numbered 1 to 24.

2.3.2 Allocation of the frame alignment signal and *S*-bit (for multiframe alignment or signalling) is given in Table 1/G.733.

TABLE 1/G.733

Frame number	Frame alignment signal (see 2.4)	Multiframe alignment signal or signalling
1	1	–
2	–	*S*
3	0	–
4	–	*S*

2.3.3 The assignment of the *S*-bit is covered in 4. below.

2.4 *Frame alignment signal*

The frame alignment signal occupies the first bit position of every other frame.

This signal is: 101010 . . .

2.5 *Loss and recovery of frame alignment*

The frame alignment signal should be monitored to determine if frame alignment has been lost. Frame alignment should be recovered after a valid frame alignment signal is available at the receiving terminal equipment.

3. *Fault conditions and consequent action*

3.1 *The PCM multiplex equipment should detect the following fault conditions:*

- loss of frame alignment;
- failure of the codec, if possible;
- failure of the main power supply.

When provided, monitoring of the codec should be arranged locally.

3.2 *Action following detection of a fault condition*

When a fault condition has been detected at local end A, the following action should be taken:

3.2.1 *At local end A*

An alarm should be given by PCM multiplex equipment A after the appropriate period required to assure that channels will be disconnected only due to a true interruption of the received digital line signal.

3.2.2 *At distant end B*

When a fault condition is detected at local end A, the following measures may be instituted:

PCM multiplex equipment A may transmit a signal in its digital line signal by forcing bit 2 in every channel time slot to the value 0 in the direction of transmission A to B.

In some networks where channel-associated signalling is employed, local end A may transmit a distant end alarm signal by modifying the *S*-bit as described in 4.2.1 below.

When the alarm signal from A is detected in PCM multiplex equipment B, an alarm should be produced at B, in order to indicate the loss of frame alignment at the distant end A, except in the case where the PCM multiplex equipment at end B has itself lost frame alignment.

3.2.3 *Use of the alarm to automatically remove ciruits from service*

The alarm described in 3.2.1 and 3.2.2 above should be used both at end A and end B to automatically remove the associated circuits from service and to restore them to service when frame alignment has been recovered.

3.2.4 *Rapid indication of loss of frame alignment*

An indication should be given to the Signalling System No. 6 equipment (digital version) when the PCM multiplex equipment (Equipment A only) detects a loss of frame alignment. The average time to detect and give an indication of random bits in the frame alignment signal bit position should not be greater than 3 ms. This indication will serve the same function as that provided by the data carrier failure alarm in the analogue version (see Recommendation Q.275).

4. *Signalling*

4.1 *Common channel signalling*

The pattern of the *S*-bit may be arranged to carry common channel signalling at a rate of 4 kbit/s or a submultiple of this rate.

4.2 *Channel associated signalling*

Based on agreement the Administrations involved, channel between associated-signalling is provided for intraregional circuits according to the following arrangement which provides two independent signalling channels designated A and B.

4.2.1 *Multiframe structure*

A multiframe comprises 12 frames as shown in Table 2/G.733. The multiframe alignment is carried on the *S*-bit, as shown in that table.

TABLE 2/G.733 – **Multiframe structure**

Frame number	Frame alignment signal	Multiframe alignment signal (*S*-bit)	Bit number(s) in each channel time slot		Signalling channel
			For character signal	For signalling	
1	1	–	1-8	–	
2	–	0	1-8	–	
3	0	–	1-8	–	
4	–	0	1-8	–	
5	1	–	1-8	–	
6	–	1	1-7	8	A
7	0	–	1-8	–	
8	–	1	1-8	–	
9	1	–	1-8	–	
10	–	1	1-8	–	
11	0	–	1-8	–	
12	–	0	1-7	8	B

When the *S*-bit is modified to signal the loss of frame alignment alarm as indicated in 3.2.2 above, the *S*-bit in frame 12 is changed from 0 to 1.

4.2.2 *Loss of multiframe alignment*

Loss of multiframe alignment is assumed to have taken place when loss of frame alignment occurs.

4.2.3 *Allocation of signalling bits*

Frames 6 and 12 are designated as signalling frames. The eighth bit in each channel time-slot is used in every signalling frame to carry the signalling associated with that channel.

4.2.4 *Minimization of quantizing distortion*

In the signalling frame only seven bits are available for encoding of voice frequencies. In order to minimize the quantizing distortion, the decoder output values are shifted slightly. All even numbered decoder output values y_n, are changed to be equal to the next higher decision value, i.e. x_{n+1}. All odd numbered decoder output values y_{n+1} are changed to be equal to the same numbered decision value, i.e. x_{n+1}, as shown on Figure 1/G.733.

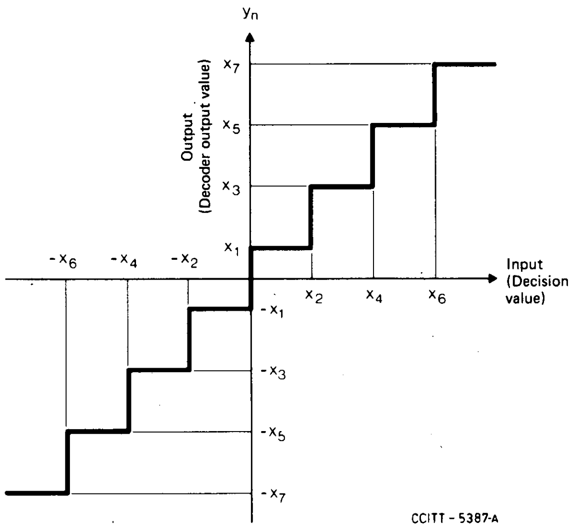


FIGURE 1/G.733 – **Seven-bit codec transfer characteristic**

When suppression of the "all zero" character signal is required, the value of the seventh bit is forced to be 1 when all the other bits of the character signal have the value 0.

5. *Interfaces*

Analogue: See Recommendation G.712.

Digital: See Recommendation G.703.

Recommendation G.734

CHARACTERISTICS OF 2048-kbit/s FRAME STRUCTURE FOR USE WITH DIGITAL EXCHANGES

(Geneva, 1976)

1. *General characteristics*

The multiplex structure described in this Recommendation is suitable for use on 2048-kbit/s digital paths which terminate at digital exchanges. The structure is compatible with that of the PCM primary multiplex described in Recommendation G.732, and is applicable to digital paths which connect such PCM multiplex equipments to exchanges and to digital paths which interconnect digital exchanges.

Some of the characteristics of this multiplex structure are identical to those in Recommendation G.732 and are covered by cross references to that Recommendation.

1.1 *Fundamental characteristics*

The multiplex structure contains 32 time slots, each of 64 kbit/s, which are all switchable. In time slots allocated to telephony, speech will be encoded according to Recommendation G.711. Time slots allocated to other services may need to be utilized in an agreed manner (e.g. Recommendation X.50 for synchronous data services).

1.2 *Bit rate*

The nominal bit rate is 2048 kbit/s. This rate will be controlled to within at least ± 50 parts per million (ppm) at the transmitting end for each direction of transmission.

1.3 *Timing signal*

The timing signal is a 2048-kHz signal from which the bit rate is derived.

1.3.1 *Timing in a non-synchronous network*

For a PCM multiplex equipment, the timing signal will be derived from the incoming timing signal at the receive side. For a digital exchange, the transmitting timing signal will be derived from a clock within the digital exchange.

1.3.2 *Timing in a synchronous network*

In case of synchronous operation of the network, a network synchronization system will maintain the timing signal or clocks within agreed timing limits.

1.4 *Interfaces*

Refer to 5. of Recommendation G.732 and Recommendation G.703. No interface, internal to the switch, will be recommended.

1.5 *Transmission performance*

The transmission performance of the digital path will be the same as that for 2048-kbit/s digital paths between primary PCM multiplex equipments.

2. *Frame structure*

The frame structure, frame alignment procedures, and normally the time-slot assignment will be as defined in Recommendation G.732.

Where more signalling capacity is required between exchanges, additional time slots may be utilized for common channel signalling. They should be selected from the slots allocated in PCM multiplexes for data purposes. On routes between exchanges comprising more than one 2048-kbit/s digital path, it may be possible to provide an adequate signalling capacity without using time-slot 16 of all systems on the route. In these circumstances time-slot 16 in those systems not carrying signalling can be allocated to speech or other services. Time-slot 0 is reserved for frame alignment, alarms and network synchronization information and should not be used for signalling or speech purposes.

3. *Fault conditions and consequent actions*

3.1 *Fault conditions*

The PCM multiplex equipment should detect the fault conditions mentioned in Recommendation G.732, 3.1.

The digital exchange terminal equipment should detect the following fault conditions.

3.1.1 Failure of power supply.

3.1.2 Loss of the incoming signal at 2048 kbit/s.

Note 1. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

Note 2. – Where separate circuits are used for the digital signal and the timing signal, then loss of either or both should constitute loss of the incoming signal.

3.1.3 Loss of frame alignment.

3.1.4 Excessive error rate detected in the frame alignment signal. The criteria for activating and deactivating the indication of this fault condition are given in Recommendation G.732, 3.1.6.

3.1.5 Alarm indication received from the remote end (see 3.2.3 below).

3.2 *Consequent actions*

Further to the detection of a fault condition, for PCM multiplexing equipment appropriate actions should be taken as specified by Table 2/G.732 and Recommendation G.732, 3.2.

The consequent actions for the digital exchange are specified by Table 1/G.734 and are as follows:

3.2.1 Service alarm indication generated to signify that the service provided by the exchange terminal (ET) is no longer available. This indication should be given by the ET as soon as possible and not later than 2 ms after the detection of the relevant fault condition.

This specification, taking into account the specification given in Recommendation G.732, 2.5 is equivalent to recommending that the average time to detect a loss of frame alignment and to give the relevant indication should not be greater than 3 ms.

3.2.2 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 1 below) is detected, the prompt maintenance alarm indication, associated with loss of frame alignment and excessive error rate in the frame alignment pattern, should be inhibited.

3.2.3 Alarm indication to the remote end generated by changing bit 3 of channel time-slot 0 from the state 0 to the state 1 in those frames not containing the frame alignment signal. This should be effected as soon as possible.

3.2.4 Alarm Indication Signal (see Note 1 below) applied in all received time-slots containing speech, data and/or signalling. This action should be taken as soon as possible and not later than 2 ms after detection of the fault conditions mentioned in 3.1.1, 3.1.2, 3.1.3 and 3.1.4 above.

Note 1. – The equivalent binary content of the Alarm Indication Signal (AIS) is a continuous stream of 1s.

Note 2. – All timing requirements quoted apply equally to restoration, subsequent to the fault condition clearing.

Note 3. – The utilization of these indications will depend upon the switching and signalling arrangements provided nationally. Separate indications for some of the fault conditions listed may be provided nationally if required.

The reaction of the processing equipment on the fault indication and the times within which the service and maintenance alarms should be provided need further study.

TABLE 1/G.734 – **Fault conditions and consequent actions for the digital exchange**

Fault conditions (see 3.1)	Consequent actions (see 3.2)			
	Service alarm indication generated	Prompt maintenance alarm indication generated	Alarm indication to the remote end generated	AIS applied in the exchange terminal
Failure of power supply	Yes	Yes	Yes, if practicable	Yes, if practicable
Loss of incoming signal at 2048 kbit/s	Yes	Yes	Yes	Yes
Loss of frame alignment	Yes	Yes	Yes	Yes
Error rate 1 in 10^3 for the alignment signal	Yes	Yes	Yes	Yes
Alarm indication received from the remote end	Yes			

Note. – A *Yes* in the table signifies that a certain action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

7.4 Principal characteristics of second order multiplex equipments

Recommendation G.741

GENERAL CONSIDERATIONS ON SECOND ORDER MULTIPLEX EQUIPMENTS

(Geneva, 1972; amended at Geneva, 1976)

Characteristics of second order PCM and digital multiplex equipments are under study.

The CCITT,

considering

a) that different primary and second order multiplex equipments exist, depending upon the characteristics of different networks and the various types of signals to be transmitted in those networks;

b) that, although studies will continue with the aim of reducing the differences between various systems, the existing situation cannot be changed in the near future;

recommends the following:

1. When two countries, both using 2048-kbit/s primary multiplex equipments such as the PCM multiplex equipment according to Recommendation G.732, have to be connected by a digital path at the second order bit rate, that bit rate should be 8448 kbit/s.

2. When two countries, both using 1544-kbit/s primary multiplex equipments such as the PCM multiplex equipment according to Recommendation G.733, have to be connected by a digital path at the second order bit rate, that bit rate should be 6312 kbit/s.

In the meantime, it is extremely desirable to define a preferred method of interconnecting different systems (see Question 14/XVIII).

Recommendations G.742 and G.743 give the characteristics of second order digital multiplex equipments using positive justification, and Recommendation G.745 gives the characteristics of second order multiplex equipment using positive-negative justification. Recommendation G.744 gives the characteristics of 8448-kbit/s PCM multiplex equipment.

It is recognized however that in the evolving digital networks a need may arise for synchronous digital multiplex equipments, especially with the introduction of digital switching. Proposals for such equipment are given in the Annex.

In view of the fact that the frame structure for synchronous digital multiplex equipment (see Annex) is almost the same as the frame structures contained in Recommendations G.744 and G.746, the CCITT is committed to the study of the possibility of preparing a single Recommendation for these types of equipment.

ANNEX

(to Recommendation G.741)

Characteristics of synchronous digital multiplex equipment operating at 8448 kbit/s

1. *Bit rate*

The nominal bit rate should be 8448 kbit/s.

The tolerance on that rate should be ± 30 parts per million (ppm).

2. *Frame structure*

Table 1 gives:

- the tributary bit rate and the number of tributaries;
- the number of time slots per frame;
- the time-slot numbering scheme;
- the time-slot assignment;
- the frame-alignment signal, 14 bits long, distributed into time-slots Nos. 0 and 66.

3. *Loss and recovery of frame alignment and consequent action*

Loss of frame alignment should be assumed to have taken place when four consecutive frame alignment signals have been incorrectly received in their predicted positions.

When frame alignment is assumed to be lost, the frame alignment device should decide that such alignment has effectively been recovered when it detects the presence of three consecutive frame alignment signals.

The frame alignment device having detected the appearance of a single correct frame alignment signal, should begin a new search for the frame alignment signal when it detects the absence of the frame alignment signal in one of the two following frames.

As soon as frame alignment has been lost and until it has been recovered a definite pattern should be sent on all tributaries at the output of the demultiplexer. The equivalent binary content of this alarm indication signal (AIS) pattern at 2048 kbit/s is a continuous stream of 1s.

TABLE 1 – **8448-kbit/s synchronous digital multiplexing frame structure**

Tributary bit rate (kbit/s)[a]	2048
Number of tributaries	4
Frame structure	Time-slot number
Frame alignment signal (the first 8 bits out of 14 are 11100110)	0
Frame alignment time-slots of the tributaries	1 to 4
Time slots from tributaries	5 to 32
Spare time-slot	33
Time slots from tributaries	34 to 65
Frame alignment signal (the last 6 bits out of 14 are 100000)	66
Service bits (bits Nos. 7 and 8)	66
Signalling time-slots of the tributaries	67 to 70
Time slots from tributaries	71 to 98
Spare time-slot	99
Time slots from tributaries	100 to 131
Frame length	132 time-slots
Number of telephone channels	120

[a] The tributary frame structure should be that recommended for the 2048 kbit/s PCM multiplex equipments.

4. ***Multiplexing method***

Cyclic time-slot interleaving in the tributary numbering order and synchronous multiplexing is recommended. The tributary frame structure should be that recommended for the 2048 kbit/s PCM multiplex equipments. The time slots used for the frame alignment signals of the tributaries should be identified at the multiplexer input and multiplexed into the preassigned time slots, positions Nos. 1 to 4, of the 8448-kbit/s frame.

5. ***Jitter***

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

6. ***Digital interface***

The digital interfaces at 2048 kbit/s and 8448 kbit/s should be in accordance with Recommendation G.703.

7. ***Timing signal***

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

8. *Service digits*

Two digits per frame are available for service functions. Bit 7 of time-slot No. 66 is used to transmit an alarm indication to the remote multiplex equipment when specific fault conditions are detected in the multiplex equipment.

9. *Spare time-slots*

Time-slots Nos. 33 and 99 are left for national use. On a digital path crossing an international border the bits of these time slots are fixed at state 1.

Recommendation G.742

SECOND ORDER DIGITAL MULTIPLEX EQUIPMENT OPERATING AT 8448 KBIT/S AND USING POSITIVE JUSTIFICATION

(Geneva, 1972; amended at Geneva, 1976)

1. *General*

The second order digital multiplex equipment using positive justification, described below, is intended for use on digital paths between countries using 2048 kbit/s primary multiplex equipments.

2. *Bit rate*

The nominal bit rate should be 8448 kbit/s.

The tolerance on that rate should be ± 30 parts per million (ppm).

3. *Frame structure*

Table 1/G.742 gives:

- the tributary bit rate and the number of tributaries;
- the number of bits per frame;
- the bit numbering scheme;
- the bit assignment;
- the bunched frame alignment signal.

4. *Loss and recovery of frame alignment and consequent action*

Loss of frame alignment should be assumed to have taken place when four consecutive frame alignment signals have been incorrectly received in their predicted positions.

When frame alignment is assumed to be lost, the frame alignment device should decide that such alignment has effectively been recovered when it detects the presence of three consecutive frame alignment signals.

The frame alignment device having detected the appearance of a single correct frame alignment signal, should begin a new search for the frame alignment signal when it detects the absence of the frame alignment signal in one of the two following frames.

Note. – As it is not strictly necessary to specify the detailed frame alignment strategy, any suitable frame alignment strategy may be used provided the performance achieved is at least as efficient in all respects as that obtained by the above frame alignment strategy.

TABLE 1/G.742 – **8448 kbit/s multiplexing frame structure**

Tributary bit rate (kbit/s)	2048
Number of tributaries	4
Frame structure	Bit number
	Set I
Frame alignment signal (1111010000)	1 to 10
Alarm indication to the remote digital multiplex equipment	11
Bit reserved for national use	12
Bits from tributaries	13 to 212
	Set II
Justification control bits C_{j1} (see Note)	1 to 4
Bits from tributaries	5 to 212
	Set III
Justification control bits C_{j2} (see Note)	1 to 4
Bits from tributaries	5 to 212
	Set IV
Justification control bits C_{j3} (see Note)	1 to 4
Bits from tributaries available for justification	5 to 8
Bits from tributaries	9 to 212
Frame length	848 bits
Bits per tributary	206 bits
Maximum justification rate per tributary	10 kbit/s
Nominal justification ratio	0.424

Note. – C_{ji} indicates the ith justification control bit of the jth tributary.

5. *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive justification is recommended.

The justification control signal should be distributed and use the C_{jn}-bits (n = 1, 2, 3, see Table 1/G.742).

Positive justification should be indicated by the signal 111, no justification by the signal 000. Majority decision is recommended.

Table 1/G.742 gives the maximum justification rate per tributary and the nominal justification ratio.

6. *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

7. *Digital interfaces*

The digital interfaces at 2048 kbit/s and 8448 kbit/s should be in accordance with Recommendation G.703.

8. *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

9. *Service digits*

Two bits per frame are available for service functions. Bit 11 of Set I is used to transmit an alarm indication to the remote multiplex equipment when specific fault conditions are detected in the multiplex equipment (see 10. below). Bit 12 of Set I is reserved for national use. On the digital path crossing the border, this bit is fixed at 1.

10. *Fault conditions and consequent conditions*

10.1 *Fault conditions*

The digital multiplex equipment should detect the following fault conditions:

10.1.1 Failure of power supply.

10.1.2 Loss of an incoming signal at 2048 kbit/s at the input of the multiplexer.

Note. – Where separate circuits are used for the digital signal and the timing signal then loss of either or both should constitute loss of the incoming signal.

10.1.3 Loss of the incoming signal at 8448 kbit/s at the input of the demultiplexer.

Note 1. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

Note 2. – Where separate circuits are used for the digital signal and the timing signal, then loss of either or both should constitute loss of the incoming signal.

10.1.4 Loss of frame alignment.

10.1.5 Alarm indication received from the remote multiplex equipment at the 8448 kbit/s input of the demultiplexer (see 10.2.2 below).

10.2 *Consequent actions*

Further to the detection of a fault condition, appropriate actions should be taken as specified by Table 2/G.742. The consequent actions are as follows:

10.2.1 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 2 under 10.2.5 below) at 8448 kbit/s is detected at the input of the demultiplexer, the prompt maintenance alarm indication associated with loss of frame alignment should be inhibited.

Note. – The location and provision of any visual and/or audible alarm activated by this maintenance alarm indication is left to the discretion of each Administration.

10.2.2 Alarm indication to the remote multiplex equipment generated by changing from the state 0 to the state 1 bit 11 of Set I at the 8448 kbit/s output of the multiplexer.

10.2.3 AIS (see Note 2 below) applied to all the four 2048-kbit/s tributary outputs from the demultiplexer.

10.2.4 AIS (see Note 2 below) applied to the 8448-kbit/s output of the multiplexer.

10.2.5 AIS (see Note 2 below) applite to the time slots of the 8448-kbit/s signal at the output of the multiplexer, corresponding to the relevant 2048-kbit/s tributary.

Note 1. – The bit rate of the AIS at the output of the corresponding demultiplexer equipment should be as specified for the tributaries. The method how to achieve this is under study.

Note 2. – The equivalent binary content of the Alarm Indication Signal (AIS) at 2048 kbit/s and 8448 kbit/s is a continuous stream of 1s.

TABLE 2/G.742 – **Fault conditions and consequent actions**

Equipment part	Fault conditions (see 10.1)	Consequent actions (see 10.2)				
		Prompt maintenance alarm indication generated	Alarm indication to the remote multiplex equipment generated	AIS applied		
				to all the tributaries	to the composite signal	to the relevant time-slots of the composite signal
Multiplexer and demultiplexer	Failure of power supply	Yes		Yes, if practicable	Yes, if practicable	
Multiplexer only	Loss of incoming signal on a tributary	Yes				Yes
Demultiplexer only	Loss of incoming signal at 8448 kbit/s	Yes	Yes	Yes		
	Loss of frame alignment	Yes	Yes	Yes		
	Alarm indication received from the remote multiplex equipment					

Note. – A *Yes* in the table signifies that a certain action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

Recommendation G.743

SECOND ORDER DIGITAL MULTIPLEX EQUIPMENT OPERATING AT 6312 KBIT/S AND USING POSITIVE JUSTIFICATION

(Geneva, 1972; amended at Geneva, 1976)

1. *General*

The second order digital multiplex equipment using positive justification described below, is intended for use on digital paths between countries using 1544-kbit/s primary multiplex equipments.

2. *Bit rate*

The nominal bit rate should be 6312 kbit/s.

The tolerance on that rate should be ± 30 parts per million (ppm).

3. *Frame structure*

Table 1/G.743 gives:

- the tributary bit rate and the number of tributaries;
- the number of bits per frame;
- the bit numbering scheme;
- the bit assignment;
- the distributed frame and multiframe alignment signals.

TABLE 1/G.743 – 6312-kbit/s multiplexing frame structure

Tributary bit rate (kbit/s)	1544
Number of tributaries	4
Frame structure (see Notes 1 and 2)	Bit number
	Set I
Bit for multiframe alignment signal (M_j) (see Note 1)	1
Bits from tributaries	2 to 49
	Set II
1st bit for justification control signal (C_{j_1})	1
Bits from tributaries	2 to 49
	Set III
1st bit for frame alignment signal (F_0) (see Note 3)	1
Bits from tributaries	2 to 49
	Set IV
2nd bit for justification control signal (C_{j_2})	1
Bits from tributaries	2 to 49
	Set V
3rd bit for justification control signal (C_{j_3})	1
Bits from tributaries	2 to 49
	Set VI
2nd bit for frame alignment signal (F_1) (see Note 3)	1
Bits from tributaries (see Note 4)	2 to 49
Frame length	249 bits
Multiframe length	1176 bits
Bits per tributary per multiframe (including justification)	288 bits
Maximum justification rate per tributary	5367 bit/s
Nominal justification ratio	0.334

Note 1. – This frame is repeated 4 times to form a multiframe with frames designated j = 1, 2, 3, 4. The multiframe alignment signal is a 011x-pattern. x may be used as an alarm service digit.

Note 2. – The bits from the second and fourth tributaries are inverted logically before multiplexing with the bits from the first and third tributaries.

Note 3. – The frame alignment signal is $F_0 = 0$ and $F_1 = 1$.

Note 4. – The bit available for the justification of tributary j is the first time slot of tributary j following F_1 in the jth frame.

4. *Loss and recovery of frame and multiframe alignment and consequent action*

The frame alignment recovery time should not exceed 16 ms. The signal to be applied to the tributaries during the out-of-frame-alignment time should be studied.

Once frame alignment is established, multiframe alignment should be recovered in less than 420 microseconds.

5. *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive justification is recommended.

The justification control signal should be distributed and use the C_{jn}-bits (n = 1, 2, 3, see Table 1/G.743).

Positive justification should be indicated by the signal 111, no justification by the signal 000. Majority decision is recommended.

Table 1/G.743 gives the maximum justification rate per tributary and the nominal justification ratio.

6. *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

7. *Digital interfaces*

The digital interfaces at 1544 kbit/s and 6312 kbit/s should be in accordance with Recommendation G.703.

8. *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

9. *Service digits*

The service digits are reserved for national use.

Recommendation G.744

SECOND ORDER PCM MULTIPLEX EQUIPMENT OPERATING AT 8448 KBIT/S

(Geneva, 1976)

1. *General characteristics*

1.1 *Fundamental characteristics*

The encoding law used is the A-law as specified in Recommendation G.711. The sampling rate, load capacity and the code are also specified in that Recommendation.

The number of quantized values is 256.

Note. – The inversion of bits 2, 4, 6 and 8 is covered by the encoding law and is applicable only to voice-channel time-slots.

1.2 *Bit rate*

The nominal bit rate is 8448 kbit/s. The tolerance on this rate is ± 30 parts per million (ppm).

1.3 *Timing signal*

It should be possible to derive the transmitting timing signal of a PCM multiplex equipment from an internal source, from the incoming digital signal and also from an external source.

Note. – Further study is required on the effect of jitter of the incoming signal on the timing signal, and on the measures to be taken in case of loss of the incoming signal or the external source.

2. *Frame structure*

2.1 *Number of bits per channel time slot*

Eight, numbered from 1 to 8.

2.2 *Number of channel time-slots per frame*

One hundred and thirty-two, numbered from 0 to 131. The number of bits per frame is 1056, and the frame repetition rate is 8000 Hz.

2.3 *Channel time-slot assignment in case of channel-associated signalling*

2.3.1 Channel time-slots 5 to 32, 34 to 65, 71 to 98 and 100 to 131 are assigned to 120 telephone channels numbered from 1 to 120.

2.3.2 Channel time-slot 0 and the first 6 bits in channel time-slot 66 are assigned to framing; the remaining 2 bits in time-slot 66 are devoted to services.

2.3.3 Channel time-slots 67 to 70 are assigned to channel-associated signalling as covered in 4. below.

2.3.4 Channel time-slots 1 to 4, 33 and 99 are left free for national use.

2.4 *Channel time-slot assignment in case of common channel signalling*

2.4.1 Channel time-slots 1 (2) [9] to 32, 34 to 65, 67 to 98 and 100 to 131 are available for 128 (127) [9] telephone, signalling or other service channels numbered from 1 to 128 (127) [9].

2.4.2 Channel time-slot 0 and the first 6 bits in channel time-slot 66 are assigned to framing; the remaining 2 bits in time-slot 66 are assigned to services.

2.4.3 Channel time-slots 67 to 70 are in descending order of priority available for common channel signalling as covered in 4. below.

2.4.4 Channel time-slots 33 to 99 are left free for national use.

2.5 *Frame alignment signal*

The frame alignment signal is: 11100110 100000 and occupies the 8 bit-positions in channel time-slot 0 and the first 6 bit-positions in channel time-slot 66.

---

[9] The total number (127 or 128) of time slots available to speech, signalling or other services is under study.

2.6 *Loss and recovery of frame alignment*

Loss of frame alignment should be assumed to have taken place when four consecutive frame alignment signals have been incorrectly received in their predicted positions.

When frame alignment is assumed to be lost, the frame alignment device should decide that such alignment has effectively been recovered when it detects the presence of three consecutive frame alignment signals.

The frame alignment device having detected the appearance of a single correct frame alignment signal, should begin a new search for the frame alignment signal when it detects the absence of the frame alignment signal in one of the two following frames.

2.7 *Service digits*

Bit 7 of time-slot 66 is used to convey alarm indication as covered in 3. below. Bit 8 of time-slot 66 and all bits of time-slots 33 and 99 are left free for national use and should be fixed at 1 on paths crossing the international border. The same applies to all bits of time-slots 1 to 4 in the case of channel-associated signalling.

3. *Fault conditions and consequent actions*

3.1 *Fault conditions*

The PCM multiplex equipment should detect the following fault conditions.

3.1.1 Failure of power supply.

3.1.2 Failure of codec (except when using single channel codecs).

Note. – As a minimum requirement, this fault condition should be recognized when, for at least one signal level in the range −21 to −6 dBm0, the signal-to-quantizing noise ratio performance of the local codec is 18 dB or more below the level recommended in Recommendation G.712. One Administration noted that this requirement allows for adequate monitoring of the speech capability of the codec. However, transmission of Signalling System R2 information through the codec imposes more stringent requirements on the codec which may not be covered by the above requirement. Further study is required.

3.1.3 Loss of incoming signal at 64-kbit/s inputs (time-slots 67 to 70).

Note. – The detection of this fault condition is not mandatory when channel-associated signalling is used and the signalling multiplex equipment is situated within a few metres of the PCM multiplex equipment.

3.1.4 Loss of the incoming signal at 8448 kbit/s.

Note 1. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

Note 2. – Where separate circuits are used for the digital signal and the timing signal, then loss of either or both should constitute loss of the incoming signal.

3.1.5 Loss of frame alignment.

3.1.6 Excessive error rate, detected in frame alignment signal

3.1.6.1 Criteria for activating the indication of fault condition:

– Error rate $\leqslant$ 1 in 10^4

The probability of activating the indication of fault condition in a few seconds should be less than 10^{-6}.

– Error rate $\geqslant$ 1 in 10^3

The probability of activating the indication of fault condition in a few seconds should be higher than 0.95.

3.1.6.2 Criteria for deactivating the indication of fault condition:

– Error rate $\geqslant$ 1 in 10^3

The probability for deactivating the indication of fault condition in a few seconds should be almost 0.

– Error rate $\leqslant$ 1 in 10^4

The probability for deactivating the indication of fault condition in a few seconds should be higher than 0.95.

Note. – The activating and the deactivating period specified as "a few seconds" is intended to be in the order of 4 to 5 seconds.

3.1.7 Alarm indication received from the remote end (see 3.2.3 below).

3.2 *Consequent actions*

Further to the detection of a fault condition, appropriate actions should be taken as specified in Table 1/G.744. The consequent actions are as follows:

3.2.1 Service alarm indication generated to signify that the service provided by the PCM multiplex is no longer available. This indication should be forwarded at least to the switching and/or signalling multiplex equipment depending upon the arrangements provided. The indication should be given as soon as possible and not later than 2 ms after detection of the relevant fault condition.

This specification, taking into account the specification given in 2.6 above, is equivalent to recommend that the average time to detect the loss of frame alignment and to give the relevant indication should not be greater than 3 ms.

When using common channel signalling, the indication should be forwarded to the switching equipment by means of a separate interface on the PCM multiplex equipment.

3.2.2 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 1 under 3.2.6 below) is detected the prompt maintenance alarm indication, associated with loss of frame alignment (see 3.1.5 above) and excessive error rate (see 3.1.6 above), should be inhibited.

Note. – The location and provision of any visual and/or audible alarms activated by the alarm indications given in 3.2.1 and 3.2.2, is left to the discretion of each Administration.

3.2.3 Alarm indication to the remote end generated by changing bit 7 of channel time-slot 66 from the state 0 to the state 1. This should be effected as soon as possible.

3.2.4 Transmission suppressed at the analogue outputs.

3.2.5 AIS applied to time-slots 67 to 70, 64-kbit/s outputs when not used for speech (see Note 1 below). This action should be taken as soon as possible and not later than 2 ms after the detection of the fault condition.

3.2.6 AIS applied to time-slots 67 to 70 of the output 8448 kbit/s composite signal when these time slots are not used for speech.

Note 1. – The equivalent binary content of the Alarm Indication Signal (AIS) is a continuous stream of 1s.

Note 2. – All timing requirements quoted apply equally to restoration, subsequent to the fault condition clearing.

4. *Signalling*

The use of channel time-slots 67 to 70 is recommended for either common channel or channel-associated signalling as required. The actions described under 3.2.1 above, consequent to the corresponding fault conditions according to Table 1/G.744, should be applied.

TABLE 1/G.744 – **Fault conditions and consequent actions for the PCM multiplex equipment**

Equipment part	Fault conditions (see 3.1)	Consequent action (see 3.2)					
		Service alarm indication generated	Prompt maintenance alarm indication generated	Alarm indication to the remote end generated	Transmission suppressed at the analogue outputs	AIS applied to 64-kbit/s outputs (TS 67 to 70)	AIS applied to TS 67 to 70 of the 8448-kbit/s composite signal
Multiplexer and demultiplexer	Failure of power supply	Yes	Yes	Yes, if practicable	Yes, if practicable	Yes, if practicable	Yes, if practicable
	Failure of codec	Yes	Yes	Yes	Yes		
Multiplexer only	Loss of incoming signal at 64-kbit/s inputs (TS 67 to 70)		Yes				Yes
Demultiplexer only	Loss of incoming signal at 8448 kbit/s	Yes	Yes	Yes	Yes	Yes	
	Loss of frame alignment	Yes	Yes	Yes	Yes	Yes	
	Error rate 1 in 10^3 for the alignment signal	Yes	Yes	Yes	Yes	Yes	
	Alarm indication received from the remote end	Yes					

Note. – A *Yes* in the table signifies that an action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

Channel time-slots 67 to 70 may be used to provide an interface at 64 kbit/s which shall be suitable for use with either common channel or channel-associated signalling or other services as required. The action described under 3.2.5 above, consequent to the corresponding fault conditions according to Table 1/G.744, should be applied.

The detailed requirements for the organization of particular signalling systems will be included in the specifications for those signalling systems.

4.1 *Common channel signalling*

Channel time-slots 67 to 70 may be used for common channel signalling in a descending order of priority up to a rate of 64 kbit/s. The method of obtaining signal alignment will form part of the particular common channel signalling specification. In this case the 64-kbit/s interface to be used for time-slots 67 to 70 should be in accordance with 5. below and Recommendation G.703.

4.2 *Channel associated signalling*

The recommended arrangement for use of the 64-kbit/s capability of each channel time-slot 67 to 70 for channel-associated signalling is given in the following.

4.2.1 *Multiframe structure*

A multiframe for each 64-kbit/s bit-stream comprises 16 consecutive frames (whose structure is given in 2. above) and these are numbered from 0 to 15.

The multiframe alignment signal is 0000 and occupies digit time-slots 1 to 4 of channel time-slots 67 to 70 in frame 0.

4.2.2 *Allocation of channel time-slots 67 to 70*

When channel time-slots 67 to 70 are used for channel associated signalling, they provide four 64-kbit/s digital paths which are subdivided into lower rate paths using the multiframe alignment signal as a reference. Details of the bit allocation are given in Table 2/G.744.

TABLE 2/G.744

Frame \ Channel time-slot	67		68		69		70	
0	0000*xyxx*		0000*xyxx*		0000*xyxx*		0000*xyxx*	
1	*abcd* channel 1	*abcd* channel 16	*abcd* channel 31	*abcd* channel 46	*abcd* channel 61	*abcd* channel 76	*abcd* channel 91	*abcd* channel 106
15	*abcd* channel 15	*abcd* channel 30	*abcd* channel 45	*abcd* channel 60	*abcd* channel 75	*abcd* channel 90	*abcd* channel 105	*abcd* channel 120

$x =$ spare bit to be made 1 if not used.
$y =$ bit used to indicate loss of multiframe alignment (see 4.2.4.2.3 of Recommendation G.732).

When bits b, c or d are not used they should have the value:

$b = 1$
$c = 0$
$d = 1$

It is recommended that the combination 0000 of bits a, b, c and d should not be used for signalling purposes for channels 1-15, 31-45, 61-75 and 91-125.

This bit allocation provides four 500-bit/s signalling channels designated *a*, *b*, *c* and *d* for each telephone channel. With this arrangement the signalling distortion of each signalling channel introduced by the PCM transmission system, will not exceed ± 2 ms.

4.2.3 *Loss and recovery of multiframe alignment*

For multiframe alignment each 64-kbit/s channel should be treated separately. For each channel, multiframe alignment should be assumed to have been lost when two consecutive multiframe alignment signals have been received with an error.

Multiframe alignment should be assumed to have been recovered as soon as the first correct multiframe signal is detected.

Note. – To avoid a condition of spurious multiframe alignment, the following procedure may be used, in addition to the above:

- Multiframe alignment should be assumed to have been lost when, for a period of one or two multiframes, all the bits in the relevant channel time-slots 67, 68, 69 or 70 are at the state 0.
- Multiframe alignment should be assumed to have been recovered, only when at least one bit in the state 1 is present in the relevant time-slots 67, 68, 69 or 70 preceding the multiframe alignment signal first detected.

4.2.4 *Fault conditions and consequent actions*

The fault conditions and consequent actions for each 64-kbit/s signalling channel and for each signalling multiplex equipment are the same as recommended in Recommendation G.732, 4.2.4.

5. *Interfaces*

The analogue interfaces should comply with Recommendation G.712. The digital interfaces should comply with Recommendation G.703. In addition, the electrical characteristics of the 64-kbit/s interface of the codirectional and the contradirectional type are given in 5.1 and 5.2 below, respectively. These latter specifications are not mandatory for channel-associated signalling.

5.1 *Electrical characteristics of 64-kbit/s codirectional interface*

5.1.1 *General*

5.1.1.1 Nominal bit rate: 64 kbit/s.

5.1.1.2 Maximum tolerance for signals to be transmitted through the interface: ± 100 ppm.

5.1.1.3 64-kHz and 8-kHz timing signal to be transmitted in a codirectional way with the information signal.

5.1.1.4 On balanced pair for each direction of transmission, the use of transformers is recommended.

5.1.1.5 *Code conversion rules:*

Step 1: A 64-kbit/s period is divided into four unit intervals.

Step 2: A binary one is coded as a block of the following four bits:

1 1 0 0

Step 3: A binary zero is coded as a block of the following four bits:

1 0 1 0

Step 4: The binary signal is converted into a three-level signal alternating the polarity of consecutive blocks.

Step 5: The alternation in polarity of the blocks is violated every 8th block. The violation block marks the last bit in an octet.

These conversion rules are illustrated in Figure 1/G.744.

5.1.2 *Specifications at the output ports* (see Table 3/G.744)

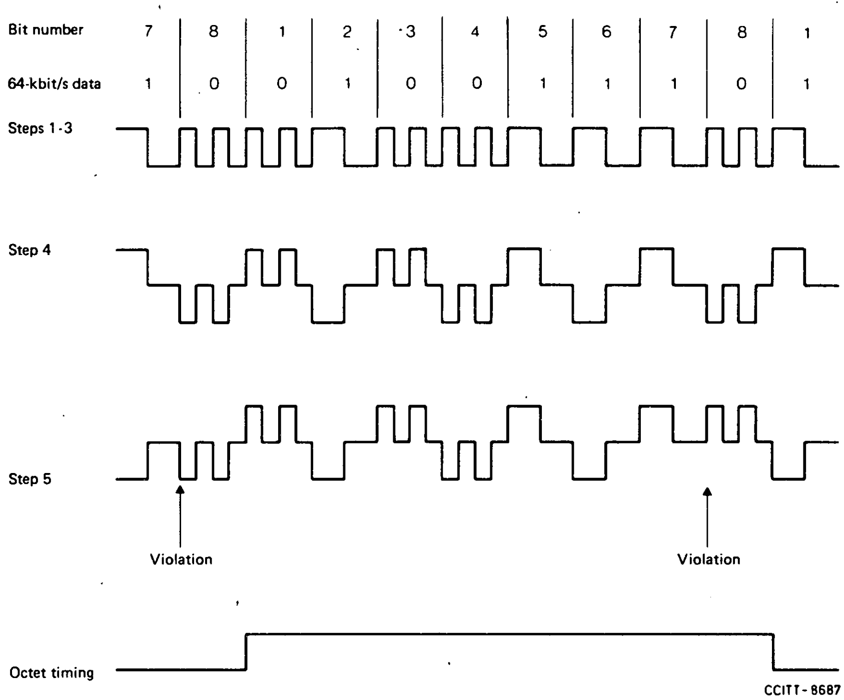
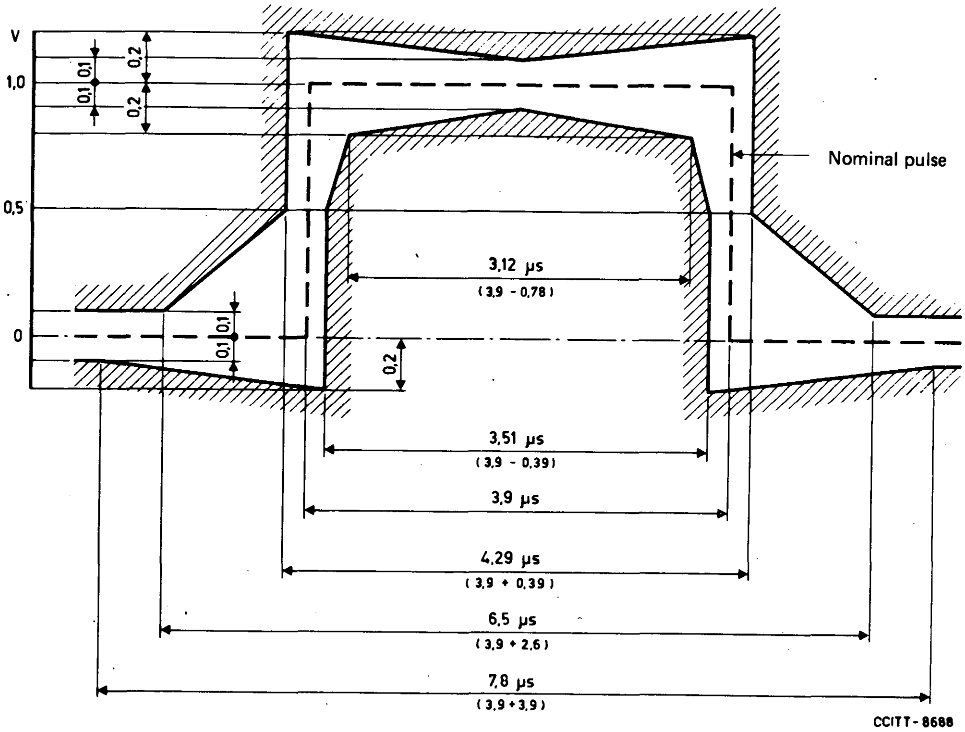


FIGURE 1/G.744 – **Steps in framing a codirectional 64-kbit/s signal**

TABLE 3/G.744

Symbol rate	256 kbauds
Pulse shape (nominally rectangular)	All pulses of a valid signal must conform to the mask in Figure 2/G.744, irrespective of the polarity
Pair for each direction	One symmetric pair
Test load impedance	120 ohms resistive
Nominal peak voltage of a "mark" (pulse)	1.0 V
Peak voltage of a "space" (no pulse)	0 V ± 0.10 V
Nominal unit interval	3.9 μs
Ratio of the amplitudes of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05



Note 1. – The limits apply to pulses of either polarity.
Note 2. – For the double-width pulse the time-scale values are multiplied by 2.

FIGURE 2/G.744 – **Mask of the pulse of the 64-kbit/s codirectional interface**

5.1.3 *Specifications at the input ports*

The digital signal presented at the input port shall be as defined above but modified by the characteristics of the interconnecting pairs. The attenuation of these pairs at the frequency of 128 kHz should be in the range 0 to 3 dB. This attenuation should take into account any losses incurred by the presence of a digital distribution frame between the equipments.

Note. – If the symmetrical pair is screened, the screen shall be connected to the earth at the output port, and provision shall be made for connecting the screen of the symmetrical pair to earth, if required, at the input port.

5.2 *Electrical characteristics of 64-kbit/s contradirectional interface*

5.2.1 *General*

5.2.1.1 Bit rate: 64 kbit/s.

5.2.1.2 Maximum tolerance for signals to be transmitted through the interface: ± 100 ppm.

5.2.1.3 For each direction of transmission there should be two symmetrical pairs of wires, one pair carrying the data signal and the other carrying a composite timing signal (64 kHz and 8 kHz). The interface configuration including the directions of the signals involved is shown in Figure 3/G.744. The use of transformers is recommended.

Note. – If there is a national requirement to provide a separate alarm signal across the interface, this can be done by cutting the 8-kHz timing signal for the transmission direction concerned, that is, by inhibiting the code violations introduced in the corresponding composite timing signal (see below).

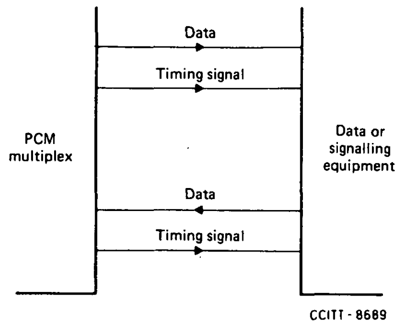


FIGURE 3/G.744 – **64-kbit/s contradirectional interface**

5.2.1.4 *Code conversion rules*

The data signals are coded in AMI code with a 100% duty ratio. The composite timing signals convey the 64-kHz bit-timing information using AMI code with a 50% duty ratio and the 8-kHz octet-phase information by introducing violations of the code rule. The structures of the signals and their phase relationships at data output ports are shown in Figure 4/G.744.

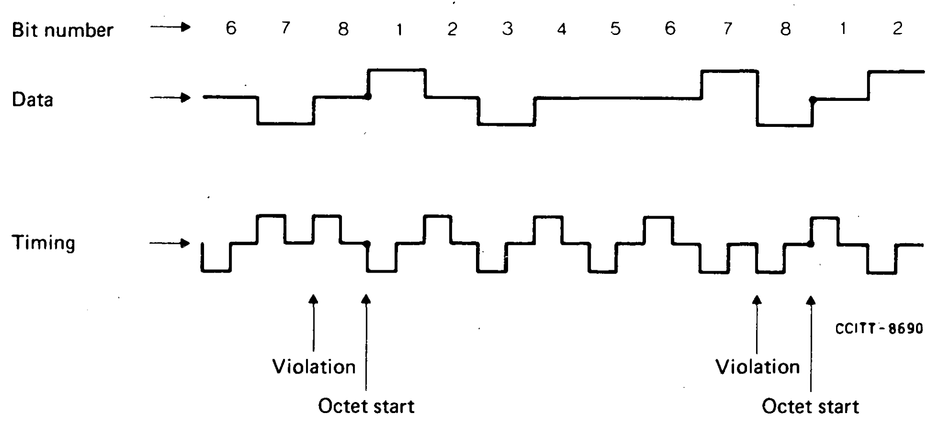


FIGURE 4/G.744 – **Signal structures of the 64-kbit/s contradirectional interface at data output ports**

The data pulses received by the PCM equipment will be somewhat delayed in relation to the corresponding timing pulses. The detection instant for a received data pulse on the PCM side of the interface should therefore be at the leading edge of the next timing pulse.

5.2.2 *Specifications at the output ports* (see Table 4/G.744)

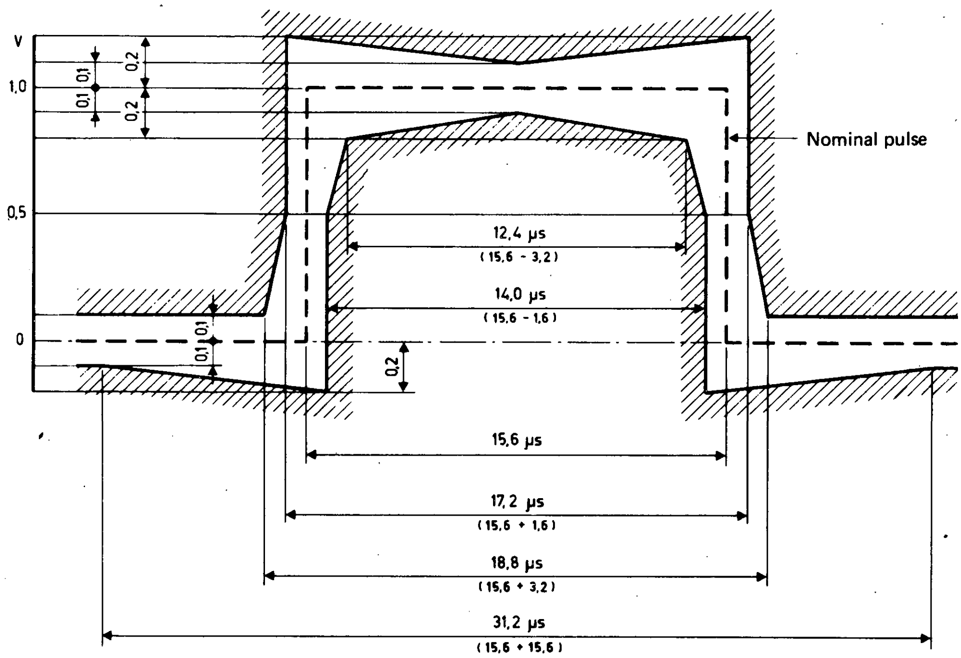
TABLE 4/G.744

Parameters	Data	Timing
Pulse shape (nominally rectangular)	All pulses of a valid signal must conform to the mask in Figure 5/G.744, irrespective of the polarity	All pulses of a valid signal must conform to the mask in Figure 6/G.744, irrespective of the polarity
Pairs in each direction of transmission	One symmetric pair	One symmetric pair
Test load impedance	120 ohm resistive	120 ohm resistive
Nominal peak voltage of a "mark" (pulse)	1.0 V	1.0 V
Peak voltage of a "space" (no pulse)	0 V ± 0.1 V	0 V ± 0.1 V
Nominal pulse width	15.6 μs	7.8 μs
Ratio of the amplitudes of positive and negative pulses at the centre of the pulse interval	0.95 to 1.05	0.95 to 1.05
Ratio of the widths of positive and negative pulses at the nominal half amplitude	0.95 to 1.05	0.95 to 1.05

5.2.3 *Specifications at the input ports*

The digital signals presented at the input ports should be as defined above but modified by the characteristics of the interconnecting pairs. The attenuation of these pairs at a frequency of 32 kHz should be in the range 0 to 3 dB. This attenuation should take into account any losses incurred by the presence of a digital distribution frame between the equipments.

Note. – If the symmetrical pairs are screened, the screens shall be connected to the earth at the output port, and provision shall be made for connecting the screens of the symmetrical pairs to earth, if required, at the input port.



Note 1. – When one pulse is immediately followed by another pulse of the opposite polarity, the time limits at the zero-crossing between the pulses should be ± 0.8 μs.

Note 2. – The time instants at which a transition from one state to another in the data signal may occur are determined by the timing signal. On the data or signalling side of the interface it is essential that these transitions are not initiated in advance of the timing instants given by the received timing signal.

FIGURE 5/G.744 – **Mask of the data pulse of the 64-kbit/s contradirectional interface**

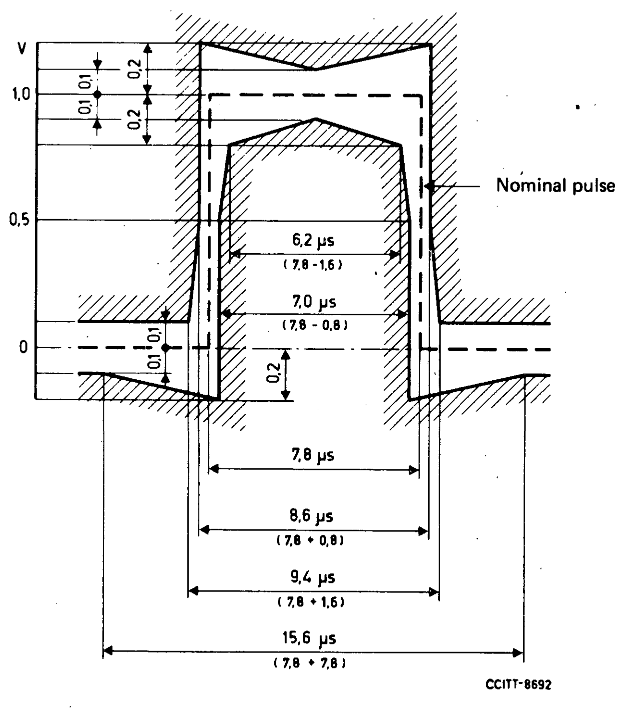


FIGURE 6/G.744 – **Mask of the timing pulse of the 64-kbit/s contradirectional interface**

Recommendation G.745

SECOND ORDER DIGITAL MULTIPLEX EQUIPMENT OPERATING AT 8448 KBIT/S AND USING POSITIVE/ZERO/NEGATIVE JUSTIFICATION

(Geneva, 1976)

1. *General*

The second order digital multiplex equipment using positive/zero/negative justification, considered below, is intended for use on digital paths between countries using 2048 kbit/s primary multiplex equipments, such as the PCM multiplex equipment described in Recommendation G.732 or any identical equipment.

2. *Bit rate*

The nominal bit rate should be 8448 kbit/s. The tolerance on that rate should be ± 30 parts per million (ppm).

3. *Frame structure*

Table 1/G.745 gives:

- the tributary bit rate and the number of tributaries;
- the number of bits per frame;
- the bit numbering scheme;
- the bit assignment;
- the bunched frame alignment signal.

4. *Loss and recovery of frame alignment and consequent action*

Loss of frame alignment should be assumed to have taken place when five consecutive frame alignment signals have been incorrectly received in their predicted positions.

Recovery of frame alignment should take place in the case of receiving without errors at least two consecutive frame signals in their predicted positions.

As soon as frame alignment has been lost and until it has been recovered, a definite pattern should be sent to all tributaries from the output of the demultiplexer. The equivalent binary content of this pattern, called the Alarm Indication Signal (AIS), at 2048 kbit/s is a continuous stream of 1s.

5. *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive/zero/negative justification with two-command control are recommended.

The justification control signal should be distributed and use the C_{jn}-bits ($n = 1, 2, 3$, see Table 1/G.745. Corection of one error in command is possible.

Positive justification should be indicated by the signal 111, transmitted in each of two consecutive frames; negative justification should be indicated by the signal 000 transmitted in each of two consecutive frames, and no justification by the signal 111 in one frame followed by 000 in the next frame. Bits 5, 6, 7 and 8 in Set IV (see Table 1/G.745) are used for negative justification of tributaries 1, 2, 3 and 4 respectively, and bits 9 to 12 for positive justification of the same tributaries.

Table 1/G.745 gives the maximum justification rate per tributary.

TABLE 1/G.745 – **8448-kbit/s digital multiplexing frame structure using positive/zero/negative justification**

Tributary bit rate (kbit/s)	2048
Number of tributaries	4
Frame structure	Bit number
	Set I
Frame alignment signal (10111000)	1 to 8
Bits from tributaries	9 to 264
	Set II
Justification control bits C_{j1} (see Note)	1 to 4
Bits for service functions	5 to 8
Bits from tributaries	9 to 264
	Set III
Justification control bits C_{j2} (see Note)	1 to 4
Spare bits	5 to 8
Bits from tributaries	9 to 264
	Set IV
Justification control bits C_{j3} (see Note)	1 to 4
Bits from tributaries available for negative justification	5 to 8
Bits from tributaries available for positive justification	9 to 12
Bits from tributaries	13 to 264
Frame length	1056 bits
Frame duration	125 μs
Bits per tributary	256
Maximum justification rate per tributary	8 kbit/s

Note. – C_{jn} indicates *n*th justification control bit of the *j*th tributary.

6. *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer should be studied and specified.

7. *Digital interface*

The digital interfaces at 2048 kbit/s and 8448 kbit/s should be in accordance with Recommendation G.703.

8. *Timing signal*

It might be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal one.

9. *Service digits*

Some spare bits per frame are available for service functions (bits from 5 to 8 in Set II) for national and international use. Utilisation of these bits is under study.

Recommendation G.746

CHARACTERISTICS OF 8448-KBIT/S FRAME STRUCTURE FOR USE WITH DIGITAL EXCHANGES

(Geneva, 1976)

1. *General characteristics*

The multiplex structure described in this Recommendation is suitable for use on 8448-kbit/s digital paths which terminate at digital exchanges. The structure is compatible with that of the PCM secondary multiplex described in Recommendation G.744 and is applicable to digital paths which connect such PCM multiplex equipments to exchanges and to digital paths which interconnect digital exchanges.

Some of the characteristics of this multiplex structure are identical to those in Recommendation G.744 and are covered by cross references to that Recommendation.

1.1 *Fundamental characteristics*

The multiplex structure contains 132 time-slots, each of 64 kbit/s, of which 128 are switchable. In time slots allocated to telephony, speech will be encoded according to Recommendation G.711. Time slots allocated to other services may need to be utilized in a agreed manner (e.g Recommendation X.50 for synchronous date services).

1.2 *Bit rate*

The nominal bit rate is 8448 kbit/s. This rate will be controlled to within at least ± 30 parts per million at the transmitting end for each direction of transmission.

1.3 *Timing signal*

The timing signal is an 8448-kHz signal from which the bit rate is derived.

1.3.1 *Timing in a non-synchronous network*

For a PCM multiplex equipment, the timing signal will be derived from the incoming timing signal at the receive side. For a digital exchange, the transmitting timing will be derived from a clock within the digital exchange.

1.3.2 *Timing in a synchronous network*

In case of synchronous operation of the network, a network synchronization system will maintain the timing signal or clocks within agreed timing limits.

1.4 *Interfaces*

Refer to Recommendations G.744, 5 and G.703. No interface, internal to the switch, will be recommended.

1.5 *Transmission performance*

The transmission performance of the digital path will be the same as that for 8448-kbit/s digital paths between secondary PCM and/or digital multiplex equipments.

2. *Frame structure*

The frame structure, frame alignment procedures, and normally the time-slot assignment will be as defined in Recommendation G.744.

Where signalling capacity is required between exchanges, time-slots 67, 68, 69 and 70 may be utilized for common channel signalling in this order of descending priority. Those channels not used for common channel signalling can be used for speech or other purposes.

Note. – If a time slot will be dedicated to service purposes internal to the switch, it will be time-slot 1.

3. *Fault conditions and consequent actions*

3.1 *Fault conditions*

The PCM multiplex equipment should detect the fault conditions mentioned in Recommendation G.744, 3.1.

The digital exchange terminal equipment, should detect the following fault conditions.

3.1.1 Failure of power supply.

3.1.2 Loss of the incoming signal at 8448 kbit/s.

Note 1. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

Note 2. – Where separate circuits are used for the digital signal and the timing signal, then loss of either or both should constitute loss of the incoming signal.

3.1.3 Loss of frame alignment.

3.1.4 Excessive error rate detected in the frame alignment signal. The criteria for activating and deactivating this fault condition are given in Recommendation G.744, 3.1.6.

3.1.5 Alarm indication received from the remote end (See 3.2.3 below).

3.2 *Consequent actions*

Further to the detection of a fault condition, for PCM multiplexing equipment appropriate actions should be taken as specified by Table 1/G.746 and Recommendation G.744, 3.2. The consequent actions for the digital exchange are specified by Table 1/G.746 and are as follows:

3.2.1 Service alarm indication generated to signify that the service provided by the exchange terminal (ET) is no longer available. This indication should be given by the ET as soon as possible and not later than 2 ms after the detection of the relevant fault condition.

This specification, taking into account the specification given in Recommendation G.744, 2.6 is equivalent to recommend that the average time to detect a loss of frame alignment and to give the relevant fault indication should not be greater than 3 ms.

3.2.2 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 1 below) is detected, the prompt maintenance alarm indication, associated with loss of frame alignment and excessive error rate in the frame alignment pattern, should be inhibited.

3.2.3 Alarm indication to the remote end generated by changing bit 7 of channel time-slot 66 from the state 0 to the state 1. This should be effected as soon as possible.

3.2.4 Alarm Indication Signal (see Note 1) applied in all received time-slots containing speech, data and/or signalling. This action should be taken as soon as possible and not later than 2 ms after detection of the fault conditions mentioned in 3.1.1, 3.1.2, 3.1.3 and 3.1.4 above.

Note 1. – The equivalent binary content of the Alarm Indication Signal (AIS) is a continous stream of 1 s.

Note 2. – All timing requirements quoted apply equally to restoration, subsequent to the fault condition clearing.

Note 3. – The utilization of these indications will depend upon the switching and signalling arrangements provided nationally. Separate indications for some of the fault conditions listed may be provided nationally if required.

The reaction of the processing equipment on the fault indication and the times within which the service and maintenance alarms should be provided need further study.

TABLE 1/G.746 – **Fault conditions and consequent actions for the digital exchange**

Fault conditions (see 3.1)	Consequent actions (see 3.2)			
	Service alarm indication generated	Prompt maintenance alarm indication generated	Alarm indication to the remote end generated	AIS applied in the ET
Failure of power supply	Yes	Yes	Yes, if practicable	Yes, if practicable
Loss of incoming signal at 8448 kbit/s	Yes	Yes	Yes	Yes
Loss of frame alignment	Yes	Yes	Yes	Yes
Error rate 1 in 10^3 for the alignment signal	Yes	Yes	Yes	Yes
Alarm indication received from the remote end	Yes			

Note. – A *Yes* in the table signifies that an action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

7.5 Principal characteristics of higher order multiplex equipments

Recommendation G.751

DIGITAL MULTIPLEX EQUIPMENTS OPERATING AT THE THIRD ORDER BIT RATE OF 34 368 KBIT/S AND THE FOURTH ORDER BIT RATE OF 139 264 KBIT/S AND USING POSITIVE JUSTIFICATION

(Geneva, 1976)

1. *General characteristics*

1.1 There should be a 4th-order bit rate of 139 264 kbit/s in the digital hierarchy which is based on the 2nd-order bit rate of 8448 kbit/s.

There should be two methods of achieving the 4th-order bit rate:

Method 1. – by using a 3rd-order bit rate of 34 368 kbit/s in the digital hierarchy.

Method 2. – by directly multiplexing sixteen digital signals at 8448 kbit/s.

The digital signals at the bit rate of 139 264 kbit/s obtained by these two methods should be identical.

1.2 The existence of the above two methods implies that the use of the bit rate of 34 368 kbit/s should not be imposed on an Administration that does not wish to realize the corresponding equipment.

1.3 In accordance with the above two methods, the following realizations of digital multiplex equipments using positive justification are recommended:

Method 1. – Realization by separate digital multiplex equipments: one type which operates at 34 368 kbit/s and multiplexes four digital signals at 8448 kbit/s; the other type which operates at 139 264 kbit/s and multiplexes four digital signals at 34 368 kbit/s.

The multiplexing for the 34 368-kbit/s digital multiplex equipment is recommended in 1.4 below, while further specification of this equipment is given in 2. below.

The multiplexing for the 139 264-kbit/s digital multiplex equipment is recommended in 1.5 below, while further specification of this equipment is given in 3. below.

Method 2. – Realization by a single digital multiplex equipment which operates at 139 264 kbit/s and multiplexes sixteen digital signals at 8448 kbit/s.

The digital multiplexing for the 139 264-kbit/s bit rate should be achieved by multiplexing, in accordance with 1.5 below, four digital signals at 34 368 kbit/s, each of which is obtained by multiplexing, in accordance with 1.4 below, four digital signals at 8448 kbit/s. Further specification of this equipment is given in 4. below.

1.4 *Multiplexing four digital signals at 8448 kbit/s*

1.4.1 *Bit rate*

The nominal bit rate should be 34 368 kbit/s.

The tolerance on that rate should be ± 20 parts per million (ppm).

1.4.2 *Frame structure*

Table 1/G.751 gives:

- the tributary bit rate and the number of tributaries;
- the number of bits per frame;
- the bit numbering scheme;
- the bit assignment;
- the bunched frame alignment signal.

TABLE 1/G.751 – 34 368-kbit/s multiplexing frame structure

Tributary bit rate (kbit/s)	8448
Number of tributaries	4
Frame structure	Bit number
	Set I
Frame alignment signal (1111010000)	1 to 10
Alarm indication to the remote digital multiplex equipment	11
Bit reserved for national use	12
Bits from tributaries	13 to 384
	Set II
Justification service bits C_{j1} (see Note)	1 to 4
Bits from tributaries	5 to 384
	Set III
Justification service bits C_{j2} (see Note)	1 to 4
Bits from tributaries	5 to 384
	Set IV
Justification service bits C_{j3} (see Note)	1 to 4
Bits from tributaries available for justification	5 to 8
Bits from tributaries	9 to 384
Frame length	1536 bits
Bits per tributary	378 bits
Maximum justification rate per tributary	22 375 bit/s
Nominal justification ratio	0.436

Note. – C_{jn} indicates the nth justification service bit of the jth tributary.

1.4.3 *Loss and recovery of frame alignment*

Loss of frame alignment should be assumed to have taken place when four consecutive frame alignment signals have been incorrectly received in their predicted positions.

When frame alignment is assumed to be lost, the frame alignment device should decide that such alignment has effectively been recovered when it detects the presence of three consecutive frame alignment signals.

The frame alignment device having detected the appearance of a single correct frame alignment signal, should begin a new search for the frame alignment signal when it detects the absence of the frame alignment signal in one of the two following frames.

Note. – As it is not strictly necessary to specify the detailed frame alignment strategy, any suitable frame alignment strategy may be used provided the performance achieved is at least as efficient in all respects as that obtained by the above frame alignment strategy.

1.4.4 *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive justification is recommended. The justification control signal should be distributed and use the C_{jn}-bits (n = 1, 2, 3, see Table 1/G.751). Positive justification should be indicated by the signal 111, no justification by the signal 000. Majority decision is recommended.

Table 1/G.751 gives the maximum justification rate per tributary and the nominal justification ratio.

1.4.5 *Service digits*

Two bits per frame are available for service functions. Bit 11 of Set I is used to transmit an alarm indication to the remote multiplex equipment when specific fault conditions are detected in the multiplex equipment (see 2.5 and 4.5 below). Bit 12 of Set I is reserved for national use. On a digital path crossing the border, this bit is fixed at 1.

1.5 *Multiplexing four digital signals at 34 368 kbit/s*

1.5.1 *Bit rate*

The nominal bit rate should be 139 264 kbit/s. The tolerance on that rate should be ± 15 parts per million (ppm).

1.5.2 *Frame structure*

Table 2/G.751 gives:

- the tributary bit rate and the number of tributaries:
- the number of bits per frame;
- the bits numbering scheme;
- the bit assignment;
- the bunched frame alignment signal.

TABLE 2/G.751 – 139 264-kbit/s multiplexing frame structure

Tributary bit rate (kbit/s)	34 368
Number of tributaries	4
Frame structure	Bit number
	Set I
Frame alignment signal (111110100000)	1 to 12
Alarm indication to the remote digital multiplex equipment	13
Bits reserved for national use	14 to 16
Bits from tributaries	17 to 488
	Sets II to V
Justification service bits C_{jn} (n = 1 to 4) (see Note)	1 to 4
Bits from tributaries	5 to 488
	Set VI
Justification service bits C_{j5} (see Note)	1 to 4
Bits from tributaries available for justification	5 to 8
Bits from tributaries	9 to 488
Frame length	2928 bits
Bits per tributary	723 bits
Maximum justification rate per tributary	47 560 bit/s
Nominal justification ratio	0.419

Note. – C_{jn} indicates the nth justification service bit of the jth tributary.

1.5.3 *Loss and recovery of frame alignment*

Loss of frame alignment should be assumed to have taken place when four consecutive frame alignment signals have been incorrectly received in their predicted positions.

When frame alignment is assumed to be lost, the frame alignment device should decide that such alignment has effectively been recovered when it detects the presence of three consecutive frame alignment signals.

The frame alignment device having detected the appearance of a single correct frame alignment signal, should begin a new search for the frame alignment signal when it detects the absence of the frame alignment signal in one of the two following frames.

Note. – As it is not strictly necessary to specify the detailed frame alignment strategy, any suitable frame alignment strategy may be used provided the performance achieved is at least as efficient in all respects as that obtained by the above frame alignment strategy.

1.5.4 *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive justification is recommended. The justification control signal should be distributed and use the C_{jn}-bits (n = 1, 2, 3, 4, 5, see Table 2/G.751). Positive justification should be indicated by the signal 11111, no justification by the signal 00000. Majority decision is recommended.

Table 2/G.751 gives the minimum justification rate per tributary and the nominal justification ratio.

1.5.5 *Service digits*

Four bits per frame are available for service functions. Bit 13 of Set I is used to transmit an alarm indication to the remote multiplex equipment when specific fault conditions are detected in the multiplex equipment (see 3.5 and 4.5 below). Bits 14 to 16 of Set I are reserved for national use. On a digital path crossing the border, these bits are fixed at 1.

2. *Digital multiplex equipment operating at 34 368 kbit/s and multiplexing four tributaries at 8448 kbit/s*

2.1 *Multiplexing*

The multiplexing for the 34 368-kbit/s bit rate should be in accordance with 1.4 below.

2.2 *Digital interfaces*

The digital interfaces at 8448 kbit/s and 34 368 kbit/s should be in accordance with Recommendation G.703.

2.3 *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

2.4 *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

2.5 *Fault conditions, and consequent actions*

2.5.1 *Fault conditions*

The digital multiplex equipment should detect the following fault conditions:

2.5.1.1 Failure of power supply

2.5.1.2 Loss of an incoming signal at 8448 kbit/s at the input of the multiplexer.

Note. – Where separate circuits are used for the digital signal and the timing signal then loss of either or both should constitute loss of the incoming signal.

2.5.1.3 Loss of the incoming signal at 34 368 kbit/s at the input of the demultiplexer.

Note. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

2.5.1.4 Loss of frame alignment.

2.5.1.5 Alarm indication received from the remote multiplex equipment at the 34 368-kbit/s input of the demultiplexer (see 2.5.2.2 below)

2.5.2 *Consequent actions*

Further to detection of a fault condition, actions should be taken as specified by Table 3/G.751. The consequent actions are as follows:

TABLE 3/G.751 – **Fault conditions and consequent actions**

Equipment part	Fault conditions (see 2.5.1 or 3.5.1)	Consequent actions (see 2.5.2 or 3.5.2)				
		Prompt maintenance alarm indication generated	Alarm indication to the remote multiplex equipment generated	AIS applied		
				to all the tributaries	to the composite signal	to the relevant time-slots of the composite signal
Multiplexer and demultiplexer	Failure of power supply	Yes		Yes, if practicable	Yes, if practicable	
Multiplexer only	Loss of incoming signal on a tributary	Yes				Yes
Demultiplexer only	Loss of incoming signal	Yes	Yes	Yes		
	Loss of frame alignment	Yes	Yes	Yes		
	Alarm indication received from the remote multiplex equipment					

Note. – A *Yes* in the table signifies that a certain action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

2.5.2.1 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 2 under 2.5.2.5 below) at 34 368 kbit/s is detected at the input of the demultiplexer, the prompt maintenance alarm indication associated with loss of frame alignment should be inhibited.

Note. – The location and provision of any visual and/or audible alarm activated by this maintenance alarm indication is left to the discretion of each Administration.

2.5.2.2 Alarm indication to the remote multiplex equipment generated by changing from the state 0 to the state 1 bit 11 of Set I at the 34 368-kbit/s output of the multiplexer.

2.5.2.3 AIS (see Note 2 below) applied to all the four 8448-kbit/s tributary outputs from the demultiplexer.

2.5.2.4 AIS (see Note 2 below) applied to the 34 368 kbit/s output of the multiplexer.

2.5.2.5 AIS (see Note 2 below) applied to time slots of the 34 368-kbit/s signal at the output of the multiplexer, corresponding to the relevant 8448-kbit/s tributary.

Note 1. – The bit rate of the AIS at the output of the corresponding demultiplexer equipment should be as specified for the tributaries. The method how to achieve this is under study.

Note 2. – The equivalent binary content of the Alarm Indication Signal (AIS) at 8448 kbit/s and 34 368 kbit/s is a continuous stream of 1s.

3. *Digital multiplex equipment operating at 139 264 kbit/s and multiplexing four tributaries at 34 368 kbit/s*

3.1 *Multiplexing*

The multiplexing for the 139 264-kbit/s bit rate should be in accordance with 1.5 above.

3.2 *Digital interfaces*

The digital interfaces at 34 368 kbit/s and 139 264 kbit/s should be in accordance with Recommendation G.703.

3.3 *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

3.4 *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from internal source.

3.5 *Fault conditions and consequent actions*

3.5.1 *Fault conditions*

The digital multiplex equipment should detect the following fault conditions:

3.5.1.1 Failure of power supply.

3.5.1.2 Loss of an incoming signal at 34 368 kbit/s at the input of the multiplexer.

3.5.1.3 Loss of the incoming signal at 139 264 kbit/s at the input of the demultiplexer.

Note. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

3.5.1.4 Loss of frame alignment.

3.5.1.5 Alarm indication received from the remote multiplex equipment at the 139 264 kbit/s input of the demultiplexer (see 3.5.2.2 below).

3.5.2 *Consequent actions*

Further to detection of a fault condition actions should be taken as specified by Table 3/G.751. The consequent actions are as follows:

3.5.2.1 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 2 below) at 139 264 kbit/s is detected at the input to the demultiplexer, the prompt maintenance alarm indication associated with loss of frame alignment should be inhibited.

3.5.2.2 Alarm indication to the remote multiplex equipment generated by changing from the state 0 to the state 1 bit 13 of Set I at the 139 264 kbit/s output of the multiplexer.

3.5.2.3 AIS (see Note 2 below) applied to all the four 34 368-kbit/s tributary outputs from the demultiplexer.

3.5.2.4 AIS (see Note 2 below) applied to the 139 264-kbit/s output of the multiplexer.

3.5.2.5 AIS (see Note 2 below) applied to the time slots of the 139 264-kbit/s signal at the output of the multiplexer corresponding to the relevant 34 368-kbit/s tributary.

Note 1. – The bit rate of the AIS at the output of the corresponding demultiplexer equipment should be as specified for the tributaries. The method how to achieve this is under study.

Note 2. – The equivalent binary content of the Alarm Indication Signal (AIS) at 34 368 kbit/s and 139 264 kbit/s is a continuous stream of 1s.

4. *Digital multiplex equipment operating at 139 264 kbit/s and multiplexing sixteen tributaries at 8448 kbit/s*

4.1 *Multiplexing*

The multiplexing for the 139 264-kbit/s bit rate should be achieved by multiplexing, in accordance with 1.5 above, four digital signals at 34 368 kbit/s, each of which is obtained by multiplexing, in accordance with 1.4 above, four digital signals at 8448 kbit/s.

4.2 *Digital interfaces*

The digital interfaces at 8448 kbit/s and 139 264 kbit/s should be in accordance with Recommendation G.703.

4.3 *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

4.4 *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

4.5 *Fault conditions and consequent actions*

4.5.1 *Fault conditions*

The digital multiplex equipment should detect the following fault conditions:

4.5.1.1 Failure of power supply.

4.5.1.2 Loss of an incoming signal at 8448 kbit/s at the input of the multiplexer.

Note. – Where separate circuits are used for the digital signal and the timing signal then loss of either or both should constitute loss of the incoming signal.

4.5.1.3 Loss of the incoming signal at 139 264 kbit/s at the input of the demultiplexer.

Note. – The detection of this fault condition is required only when it does not result in an indication of loss of frame alignment.

4.5.1.4 Loss of frame alignment of the signal at 139 264 kbit/s at the input of the demultiplexer.

4.5.1.5 Loss of frame alignment of a signal at 34 368 kbit/s within the demultiplexer.

4.5.1.6 Alarm indication received from the remote multiplex equipment at the 139 264-kbit/s input of the demultiplexer (see 4.5.2.2 below).

4.5.1.7 Alarm indication received from the remote multiplex equipment on a signal at 34 368 kbit/s within the demultiplexer (see 4.5.2.3 below).

4.5.2 *Consequent actions*

Further to detection of a fault condition, actions should be taken as specified by Table 4/G.751. The consequent actions are as follows:

4.5.2.1 Prompt maintenance alarm indication generated to signify that performance is below acceptable standards and maintenance attention is required locally. When the Alarm Indication Signal (AIS) (see Note 2 under 4.5.2.7 below) at 139 264 kbit/s or 34 368 kbit/s is detected by the demultiplexer, the prompt maintenance alarm indication associated with the corresponding loss of frame alignment should be inhibited.

Note. – The location and provision of any visual and/or audible alarms activated by the maintenance alarm indication is left to the discretion of each Administration.

4.5.2.2 Alarm indication on the 139 264-kbit/s signal to the remote multiplex equipment generated by changing from the state 0 to the state 1 bit 13 of Set I at the 139 264 kbit/s output of the multiplexer.

4.5.2.3 Alarm indication on a 34 368-kbit/s signal to the remote multiplex equipment generated by changing from the state 0 to the state 1 bit 11 of Set I on the 34 368-kbit/s signal into the multiplexer.

4.5.2.4 AIS (see Note 2 below) applied to all the sixteen 8448-kbit/s tributary outputs from the demultiplexer.

4.5.2.5 AIS (see Note 2 below) applied to all the four 8448-kbit/s relevant tributary outputs from the demultiplexer.

4.5.2.6 AIS (see Note 2 below) applied to the 139 264-kbit/s output of the multiplexer.

4.5.2.7 AIS (see Note 2 below) applied to the time slot of the 139 264 kbit/s at the output of the multiplexer, corresponding to the relevant 8448-kbit/s tributary.

Note 1. – The bit rate of the AIS at the output of the corresponding demultiplexer equipment should be as specified for the tributaries. The method how to achieve this is under study.

Note 2. – The equivalent binary content of the Alarm Indication Signal (AIS) at 8448 kbit/s, 34 368 kbit/s and 139 264 kbit/s is a continuous stream of 1s.

TABLE 4/G.751 – **Fault conditions and consequent actions**

Equipment part	Fault conditions (see 4.5.1)	Consequent actions (see 4.5.2)						
		Prompt maintenance alarm indication generated	Alarm indication on the 139 264-kbit/s signal to the remote multiplex equipment generated	Alarm indication on a 34 368-kbit/s signal to the remote multiplex equipment generated	AIS applied: to all the 16 tributaries at 8448 kbit/s at the output of the DEMUX	AIS applied: to the 4 relevant tributaries at 8448 kbit/s at the output of the DEMUX	AIS applied: to the composite signal at 139 264 kbit/s at the output of the MUX	AIS applied: to the relevant time-slot of the composite signal
Multiplexer and demultiplexer	Loss of power supply	Yes			Yes, if practicable		Yes, if practicable	
Multiplexer only	Loss of incoming signal on a tributary	Yes						Yes
Demultiplexer only	Loss of incoming signal at 139 264 kbit/s	Yes	Yes		Yes			
	Loss of frame alignment on the 139 264-kbit/s signal	Yes	Yes		Yes			
	Alarm indication received from the remote multiplex equipment on the 139 264-kbit/s signal							
	Loss of frame alignment on a 34 368-kbit/s signal	Yes		Yes		Yes		
	Alarm indication received from the remote multiplex equipment on a 34 368-kbit/s signal							

Note. – A *Yes* in the table signifies that a certain action should be taken as a consequence of the relevant fault condition. An *open space* in the table signifies that the relevant action should *not* be taken as a consequence of the relevant fault condition, if this condition is the only one present. If more than one fault condition is simultaneously present the relevant action should be taken if, for at least one of the conditions, a *Yes* is defined in relation to this action.

Recommendation G.752

CHARACTERISTICS OF DIGITAL MULTIPLEX EQUIPMENTS BASED ON A SECOND ORDER BIT RATE OF 6312 KBIT/S AND USING POSITIVE JUSTIFICATION

(Geneva, 1976)

1. *General*

Characteristics of digital multiplex equipments above the third order are under study.

1.1 The CCITT,

considering

a) that various third- and higher-order multiplex equipments exist due to the differing characteristics of networks and signal sources in those networks;

b) that, although studies will continue with the aim of reducing the differences between various systems, the existing situation cannot be changed in the near future;

recommends the following:

When countries using 1544-kbit/s primary multiplex equipments, such as the PCM multiplex equipment according to Recommendation G.733 and second-order multiplex using 6312 kbit/s according to Recommendation G.743, are planning digital paths requiring interconnection at higher bit rates they should, when practical, utilize third-order bit rates of either 32 064 kbit/s or 44 736 kbit/s. The considerations behind the choices available to such Administrations are given in 1.3 below.

1.2 The characteristics of the third-order multiplex equipments using positive justification is given in 2. below.

1.3 In establishing recommendations for digital systems to facilitate the interconnection of digital networks, the requirements for directly encoding various signal sources must be considered. In addition, it is desirale that the recommendations for digital signals be established without unduly restricting the development of future national and international systems. It is also advisable that the rates recommended be limited to a reasonably small number. In cases where systems are operating according to different recommendations, interconnection will be feasible since digital techniques allow for efficient and economic conversion of signals from one bit rate to another.

Recognition of a basic set of bit rates and signal formats (a hierarchy) is therefore deemed necessary to facilitate interconnection and to provide guidance to the designers of future equipments. The choice of the basic signal structure should take into consideration the future connection of wideband analogue signals into the digital network. For these reasons recommendations for bit rates suitable for analogue source encoding (below approximately 100 Mbit/s) are fundamental and therefore should receive priority attention.

Figure 1/G.752 illustrates the basic multiplex arrangements recommended for Administrations using 1544-kbit/s primary multiplex equipment.

The bit rates of terrestrial systems should be a multiple of 1544 kbit/s. Whenever practicable, the bit rate should also be a multiple of 6312 kbit/s and either 32 064 or 44 736 kbit/s.

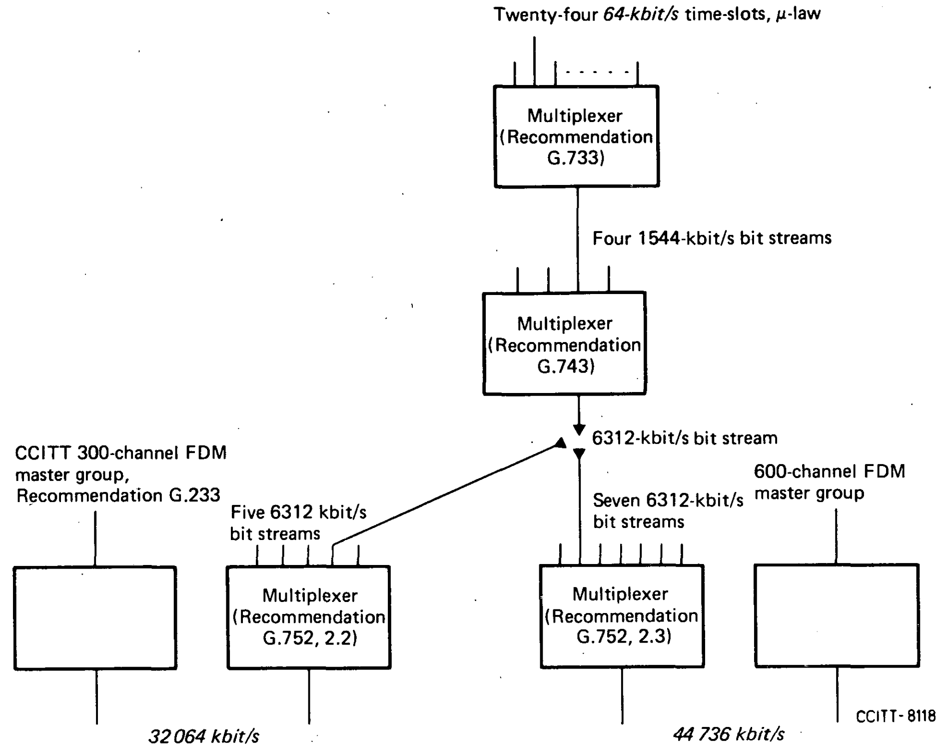


FIGURE 1/G.752 – **Basic multiplex arrangements for 1544-kbit/s derived networks**

2. *Third-order digital multiplex equipment based on second-order bit rate of 6312 kbit/s and using positive justification*

2.1 *General*

The third-order digital multiplex equipment using positive justification described below, is intended for use on digital paths and between countries using 1544-kbit/s and 6312-kbit/s primary and secondary multiplex equipments.

A bit rate of either 32 064 kbit/s or 44 736 kbit/s is recommended to allow for the efficient and economical coding of wideband signals in the networks of Administrations using primary systems according to Recommendations G.733 and G.743. For instance for a 300 voice-circuit mastergroup (Recommendation G.233) 32 064 kbit/s is appropriate, while for a 600 voice-circuit mastergroup 44 736 kbit/s coding is appropriate.

2.2 *Third-order digital multiplex equipment operating at 32 064 kbit/s*

2.2.1 *Bit rate*

The nominal bit rate should be 32 064 kbit/s. The tolerance on that rate should be ± 10 parts per million (ppm).

2.2.2 *Frame structure*

Table 1/G.752 gives:

– the tributary bit rate and the number of tributaries;
– the number of bits per frame;

- **the bit numbering scheme;**
- **the bit assignment;**
- **the frame alignment signal.**

TABLE 1/G.752 – **32 064-kbit/s multiplexing frame structure**

Tributary bit rate (kbit/s)	6312
Number of tributaries	5
Frame structure	Bit number
	Set I
Bits for frame alignment signal (see Note 1)	1 to 5
Bits from tributaries	6 to 320
	Set II
C_{jn} (n = 1, 2, 3) for justification control signal (see Note 2)	1 to 3
Auxiliary bits (bits for additional information transmission) H_{jn} (n = 1, 2)	4 to 5
Bits from tributaries	6 to 320
	Set III
C_{2n} (n = 1, 2, 3) for justification control signal	1 to 3
Auxiliary bits H_{2n} (n = 1, 2)	4 to 5
Bits from tributaries	6 to 320
	Set IV
C_{3n} (n = 1, 2, 3) for justification control signal	1 to 3
Auxiliary bits H_{3n} (n = 1, 2)	4 to 5
Bits from tributaries	6 to 320
	Set V
C_{4n} (n = 1, 2, 3) for justification control signal	1 to 3
Auxiliary bits H_{4n} (n = 1, 2)	4 to 5
Bits from tributaries	6 to 320
	Set VI
C_{5n} (n = 1, 2, 3) for justification control signal	1 to 3
Auxiliary bits H_{5n} (n = 1, 2) (see Note 3)	4 to 5
Bits from tributaries	6 to 320
Frame length	1920 bits
Bits per tributary (including justification)	378 bits
Maximum justification rate per tributary	16 700 bit/s
Nominal justification ratio	0.036

Note 1. – The frame alignment signal is a 11010 pattern for the odd frame and a 00101 pattern for the even frame.
Note 2. – C_{jn} indicates the nth justification control bit of the jth tributary (j = 1 to 5).
Note 3. – H_{52} is used for transmitting failure information from the receive end to the transmit end.
Note 4. – The bit available for the justification of tributary j is the first slot of tributary j in Set (j + 1).

2.2.3 *Loss and recovery of frame alignment and consequent action*

The frame alignment recovery time should not exceed 8 ms. The signal to be applied to the tributaries during the out-of-frame alignment time should be studied.

2.2.4 *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive justification is recommended.

The justification control signal should be distributed and use the C_{jn}-bits (n = 1, 2, 3, see Table 1/G.752).

Positive justification should be indicated by the signal 111, no justification by the signal 000. Majority decision is recommended.

Table 1/G.752 gives the maximum justification rate per tributary and the nominal justification ratio.

2.2.5 *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

2.2.6 *Digital interface*

The digital interfaces at 6312 kbit/s and 32 064 kbit/s should be in accordance with Recommendation G.703.

2.2.7 *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

2.2.8 *Service digits*

The service digits are reserved for national use.

2.3 *Third-order digital multiplex operating at 44 736 kbit/s*

2.3.1 *Bit rate*

The nominal bit rate should be 44 736 kbit/s.

The tolerance on that rate should be ± 20 parts per million (ppm).

2.3.2 *Frame structure*

Table 2/G.752 gives:

- the tributary bit rate and the number of tributaries;
- the number of bits per frame;
- the bit numbering scheme;
- the bit assignment;
- the distributed frame and multiframe alignment signals.

2.3.3 *Loss and recovery of frame and multiframe alignment and consequent action*

The frame alignment recovery time should not exceed 2.5 ms. The signal to be applied to the tributaries during the out-of-frame alignment time should be studied.

Once frame alignment is established, multiframe alignment should be recovered in less than 250 μs.

TABLE 2/G.752 – **44 736-kbit/s multiplexing frame structure**

Tributary bit rate (kbit/s)	6312
Number of tributaries	7
Frame structure (see Note 1)	Bit number
Bit for multiframe alignment signal (M_j) (see Note 1) Bits from tributaries	*Set I* 1 2 to 85
1st bit for frame alignment signal (F_{11}) (see Note 2) Bits from tributaries	*Set II* 1 2 to 85
1st bit for justification control signal (C_{j1}) Bits from tributaries	*Set III* 1 2 to 85
2nd bit for frame alignment signal (F_0) Bits from tributaries	*Set IV* 1 2 to 85
2nd bit for justification control signal (C_{j2}) Bits from tributaries	*Set V* 1 2 to 85
3rd bit for frame alignment signal (F_0) Bits from tributaries	*Set VI* 1 2 to 85
3rd bit for justification control signal (C_{j3}) Bits from tributaries	*Set VII* 1 2 to 85
4th bit for frame alignment signal (F_{12}) Bits from tributaries (see Note 3)	*Set VIII* 1 2 to 85
Frame length Multiframe length Bits per tributary per multiframe (including justification) Maximum justification rate per tributary Nominal justification ratio	680 bits 4760 bits 672 bits 9398 bits/s 0.390

Note 1. – This frame is repeated 7 times to form a multiframe with frames designated $j = 1, 2, 3, 4, 5, 6, 7$.
The multiframe alignment signal is a XXYY010 pattern where X an Y are bits assigned to service functions.
Note 2. – The frame alignment signal is $F_0 = 0$ and $F_{11} = F_{12} = 1$.
Note 3. – The bit available for the justification of tributary j is the first slot of tributary j following F_{12} in the jth frame.

2.3.4 *Multiplexing method*

Cyclic bit interleaving in the tributary numbering order and positive justification is recommended.

The justification control signal should be distributed and use the C_{jn}-bits ($n = 1, 2, 3$, see Table 2/G.752).

Positive justification should be indicated by the signal 111, no justification by the signal 000. Majority decision is recommended.

Table 2/G.752 gives the maximum justification rate per tributary and the nominal justification ratio.

2.3.5 *Jitter*

The amount of jitter that should be accepted at the input of the multiplexer and at the input of the demultiplexer, as well as the amount of jitter at the output of the multiplexer and at the output of the demultiplexer, should be studied and specified.

2.3.6 *Digital interfaces*

The digital interfaces at 6312 kbit/s and 44 736 kbit/s should be in accordance with Recommendation G.703.

2.3.7 *Timing signal*

If it is economically feasible, it may be desirable to be able to derive the multiplexer timing signal from an external source as well as from an internal source.

2.3.8 *Service digits*

The service digits are reserved for national use.

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 8

DIGITAL NETWORKS

8.1 Organization of digital networks

Recommendation G.811

PLESIOCHRONOUS OPERATION OF INTERNATIONAL DIGITAL LINKS

(Geneva, 1976)

1. *General*

This Recommendation has been prepared on the assumption that the national reference clock would be located at an international exchange and that other international exchanges within the same country might be operated by clocks which are synchronized to the reference clock by means of a national synchronization system. This necessitated the two specifications in 4. and 5. below.

Other structures within the national network for the purpose of timing may be possible. For example a central reference clock may be used whose location does not coincide with an international exchange. In this case, timing circuits would be extended from the central clock to all international exchanges within the country. The wording of this Recommendation is not adequate to deal with this structure. Further studies of alternative structures are necessary and this may necessitate changes in the Recommendation. It is hoped to produce a single Recommendation to cover all structures.

International digital links will be required to interconnect a variety of national networks. The national networks may be of the following forms:

a) a wholly synchronous network controlled by a single reference clock;

b) a set of synchronous subnetworks each controlled by a reference clock but with plesiochronous operation between the subnetworks;

c) a wholly synchronous network with two or more reference clocks which are mutually synchronized by national links;

d) a wholly plesiochronous national network.

All international links will be terminated on aligning equipments within which slips will occur.

The international plesiochronous network should be organized so that the rate of occurrence of slips in any 64-kbit/s channel is not greater than one in every 70 days per exchange. Aligning equipment may take the form of frame aligners located at the periphery of the exchange which introduce slips of one frame, or the equipment may be integrated into the switch structure and introduce slips of one channel time-slot, see 6. below.

Where a national network or subnetwork is controlled by a single reference clock the frequency of the clock must be unaffected by control signals generated within the national network or subnetwork.

Where more than one reference clock exists in a national network and where these reference clocks are mutually synchronized then, apart from controls from the mutual synchronization links, the frequencies of the clocks must be unaffected by control signals generated within the national network. Further, any disturbance introduced by the mutual synchronization link must not cause the inaccuracy of the reference clock to exceed those in the specification of 4. below.

Where a national network is wholly plesiochronous, each national clock controlling an exchange participating in an international connection should satisfy the specification of 5. below.

2. *Some requirements for exchange clocks*

All clocks controlling exchanges with international links must have a long term frequency inaccuracy of not greater than 1 in 10^{11}. Such clocks will need a very high reliability and are likely to include replication of equipment in order to ensure continuity of output. However, any phase discontinuity, due to internal operations within the clock or any other cause, should only result in a lengthening or shortening of the timing signal pulse and must not cause a phase discontinuity in excess of 1/8 of a unit interval on the outgoing digital stream from the exchange.

No agreement has yet been reached for the minimum figure for the mean time between failure (MTBF) for the catastrophic failure of a reference clock. Since such failures may render an international exchange inoperative, MTBFs of considerably better than 50 years may be necessary. Failures, which are not catastrophic in that the exchange continues to operate but at reduced frequency accuracy, will only result in an increased rate of slip. A shorter MTBF may be acceptable for such failures. Reductions in the frequency accuracy to 1 in 10^9 could be tolerated provided that the duration of the failure is limited. No figure for either form of failure has been agreed.

3. *Interaction between plesiochronous and synchronous operation*

Plesiochronous international operation will be introduced prior to international synchronization. It is important that the recommendations for plesiochronous operation should not preclude the possibility of the later introduction of synchronization. Synchronization systems will need to make adjustments to clock frequencies so that the long-term frequency of every clock is the same. Thus short-term departures greater than 1 in 10^{11} must be permitted for clocks during plesiochronous operation. These short-term departures must be sufficient so as not to present difficulties in the design of a synchronization system.

A period will occur when plesiochronous and synchronous links coexist within the international network and exchanges will be required to provide terminations for both types of link. It is therefore important that the synchronization controls do not cause short-term departures in the clocks accuracies which are unacceptable for plesiochronous operation. The magnitudes of the short-term departures are restricted by the specification recommended in 4. below.

4. *Specification for the output from an exchange with a reference clock*

The following applies to the output digit stream from an exchange directly controlled by a reference clock, but it is not anticipated that this will be significantly different from the output from the clock. The figures recommended are considered to be stringent but realizable, relaxation is desirable should further study show that this is feasible.

4.1 The time interval error (TIE) over any period up to 2^{11} unit intervals shall not exceed 1/8 of a unit interval. (Equivalent frequency inaccuracy of 60 in 10^6.)

4.2 The TIE over any period of 2 seconds shall not exceed 200 ns. (Equivalent frequency inaccuracy of 1 in 10^7.)

4.3 The TIE over any period of 50 seconds shall not exceed 500 ns. (Equivalent frequency inaccuracy of 1 in 10^8.)

4.4 The TIE over any period of 1000 seconds shall not exceed 1000 ns. (Equivalent frequency inaccuracy of 1 in 10^9.)

4.5 Over a period of S seconds the TIE shall not exceed $(10^{-2} S + 20\,000)$ ns. (Equivalent frequency inaccuracy of up to 1 in 10^{11} plus a maximum peak-to-peak jitter of 20 000 ns.)

Note 1. – 2^{11} unit intervals is approximately 1 ms at 2048 kbit/s and 1.37 ms at 1544 kbit/s.

Note 2. – 20 000 ns is approximately 40 unit-intervals at 2048 kbit/s and 30 unit-intervals at 1544 kbit/s.

Note 3. – These requirements are applicable only when the outputs are available for service.

The phase discontinuity of 4.1 above is limited by the requirements of line regenerators, but it would not be practical to design clocks with phase discontinuities of less than 10 ns.

Links between reference clocks conforming to the above specification may be used for synchronization purposes.

5. *Specification for the output from an exchange with a non-reference clock*

International links will be required between exchanges not having reference clocks but where the exchange clock is indirectly controlled from a reference clock via the national synchronization system.

The clocks of these exchanges will be dependent upon a reference clock situated at a remote exchange and phase excursions will occur relative to the reference clock. The exchange output will have the same long term frequency as the reference clock but will give rise to increased jitter. This jitter must be limited so that, when the reference clocks are mutually synchronized the jitter is not too large to be absorbed in the storage of the aligners.

The outputs from the exchange must conform to the following. Any relaxations as proposed in 4. above should be reflected in these requirements.

5.1 The TIE over any period up to 2^{11} unit-intervals shall not exceed 1/8 of a unit interval. (Equivalent frequency inaccuracy of 60 in 10^6.)

5.2 The TIE over any period of 2 seconds shall not exceed 200 ns. (Equivalent frequency inaccuracy of 1 in 10^7.)

5.3 Over a period of S seconds the TIE shall not exceed $(10^{-2} S - N)$ ns. (Equivalent frequency inaccuracy of up to 1 in 10^{11} plus a maximum peak-to-peak jitter of N ns.)

Note 1. – The maximum value of N is to be specified after further study. However the specified value cannot be less than that of the reference clock (see 4.5 above).

Note 2. – These requirements are applicable only when the outputs are available for service.

Links which terminate on one or two of these exchanges will not be used for synchronization purposes.

6. *Forms of aligning equipment*

The following two forms of aligning equipment are suitable for the termination of international digital sections.

– a frame aligner realized as a peripheral unit;

– a slot aligner which forms an integral part of the switch.

Where a frame aligner is used, a slip will consist of the insertion or removal of a consecutive set of digits amounting to a frame. In the case of frame structures as in Recommendations G.732, G.733 and G.734 the slip can consist of one complete frame. It is of importance that the maximum and mean delays introduced by the frame aligner should be as small as possible in order to minimize the delay through the exchange. The limits for these delays are under study taking into account the requirement for the delay through the exchange. It is also of importance that, after the frame aligner has produced a slip, it should be capable of absorbing changes in the arrival time of the frame alignment signals approaching the duration of a frame before a further slip is necessary. The requirement of 1 slip in 70 days per exchange in any 64-kbit/s channel must not be exceeded.

Where a slot aligner is used, a slip will consist of the insertion or removal of the eight digits of a channel time-slot in one or more 64-kbit/s channels. Because slips may occur on different channels at different times special control arrangements will be necessary within the switch if the sequence integrity of the multiple time-slot services is to be maintained. The control arrangements will need to be able to deal with the jitter on the inputs arising from clock phase variations (see 4.5 and 5.3 above) such that the requirement of 1 slip in 70 days per exchange is not exceeded.

SECTION 9

DIGITAL LINE SECTIONS

9.1 Transmission systems on cable

Recommendation G.911

CABLE SYSTEMS OPERATING AT 8448 KBIT/S

(Geneva, 1976)

For cable systems with a bit rate equal to 8448 kbit/s, it is recommended to use the HDB3 code described in Annex 1 to Recommendation G.703.

Note. – Studies are continuing on the code ADQ described in Supplement No. 15, which might offer some advantages and on which a recommendation might be issued at a later stage, if appropriate.

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

PART II

Series H Recommendations

LINES USED FOR THE TRANSMISSION OF SIGNALS OTHER THAN TELEPHONE SIGNALS, SUCH AS TELEGRAPH, FACSIMILE, DATA, ETC., SIGNALS

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

LINES USED FOR THE TRANSMISSION OF SIGNALS OTHER THAN TELEPHONE SIGNALS, SUCH AS TELEGRAPH, FACSIMILE, DATA, ETC., SIGNALS [1]

Part II contains two classes of Recommendations: those which define the characteristics of *transmission channels* (telephone-type, group, supergroup, etc., circuits) used only to transmit signals other than telephone signals, and those which define the characteristics of the *signals* used in such transmissions.

In this Part, "wideband" is used to qualify the transmission channels, and "wide-spectrum" the signals transmitted, so as to avoid any confusion between the transmission channels and the signals transmitted with regard to the frequency bands involved in transmission over group links, supergroup links, etc.

As far as possible, one should avoid specifying the characteristics of particular channels or signals in defining a new service and refer only to the characteristics of the channels mentioned in Section 1 of Part II.

[1] Excluding the transmission of programme and television signals, which is the subject of Series J Recommendations.

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 1

CHARACTERISTICS OF TRANSMISSION CHANNELS USED FOR OTHER THAN TELEPHONE PURPOSES

Recommendation H.11

CHARACTERISTICS OF CIRCUITS IN THE SWITCHED TELEPHONE NETWORK

(Mar del Plata, 1968)

The characteristics of these telephone circuits, when of modern type, are in conformity with Recommendations G.151, G.152 and G.153. Audio-frequency circuits, the characteristics of which are in accordance with Recommendations G.124, G.511 and G.543, may also be found.

Some information on the characteristics of communications established in the switched telephone network are given in Supplements No. 4.1 and 4.3 in Volume IV, *Green Book.*

Recommendation H.12

CHARACTERISTICS OF TELEPHONE-TYPE LEASED CIRCUITS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1976)

A. ORDINARY TELEPHONE-TYPE CIRCUITS [1), 2)]

1. *Scope of the Recommendation and constitution of a leased circuit*

This recommendation details the characteristics of international leased circuits for telephony and other purposes that do not require the use of special-quality leased circuits conforming to Recommendation H.12, B. These circuits are set up as described in Recommendation M.1050.

1) The application of this recommendation to multiterminal leased circuits is intended only for radial networks in which these specifications are to be met between a designated central station and each of the outlying stations. It does not apply to multiterminal conference networks between any two stations.

2) The contents of A. corresponds to part of Recommendation M.1040, Volume IV.1.

2. *Characteristics*

2.1 *Nominal overall loss*

Because of the differing nominal level at renters' premises due to the various national practices, it is not normally possible to predict the nominal overall loss of the circuit at the reference frequency. Only exceptionally can a predetermined specified nominal overall loss at the reference frequency between renters' installations be offered and then only after prior consultation among the Administrations concerned.

In the general case, however, for 4-wire circuits, the value of the receiving relative level at the renter's premises should, provisionally, not be less than −15 dBr. It may therefore be assumed that the maximum nominal overall loss will not normally exceed 28 dB.

It should be noted that the overall loss in each direction of transmission may not have the same value.

2.2 *Overall loss/frequency distortion*

The provisional limits for the overall loss relative to that at 800 Hz for the circuit between renter's installations are given in Figure 1/H.12.

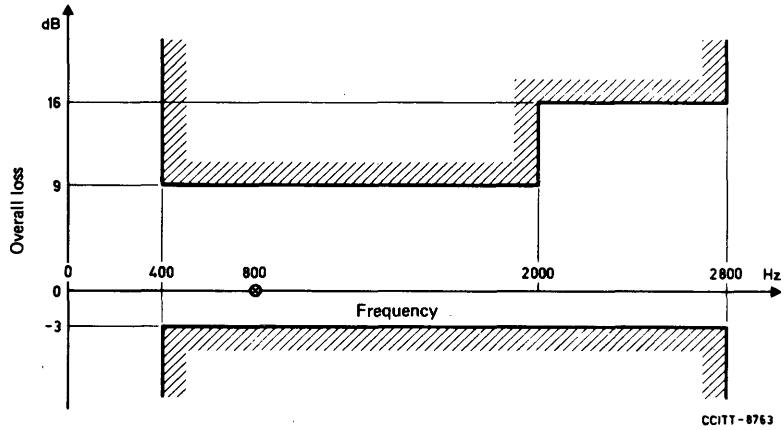


FIGURE 1/H.12 – Limits for overall circuit loss relative to that at 800 Hz

2.3 *Random circuit noise*

The nominal level of the psophometric noise power at a renter's premises depends upon the actual constitution of the circuit, in partiular upon the length of frequency division multiplex carrier systems in the circuit. The provisional limit for leased circuits of distances greater than 10 000 km is −38 dBm0. However, circuits of shorter length will have substantially less random noise (see also the Annex at the end of the Recommendation).

B. SPECIAL QUALITY LEASED CIRCUITS [3), 4)]

1. *Scope*

This recommendation deals with leased circuits for uses other than telephony – for example, data transmission.

3) The application of this recommendation to multiterminal leased circuits is intended only for radial networks in which these specifications are to be met between a designated central station and each of the outlying stations. It does not apply to multiterminal conference networks between any two stations.

4) The contents of B. corresponds to part of Recommendation M.1020, Volume IV.1.

The requirements of this recommendation are intended to ensure the provision of a circuit which will meet the requirements of digital transmission rates higher than those possible on a normal telephone-type circuit.

2. *Characteristics* [5)]

2.1 *Nominal overall loss*

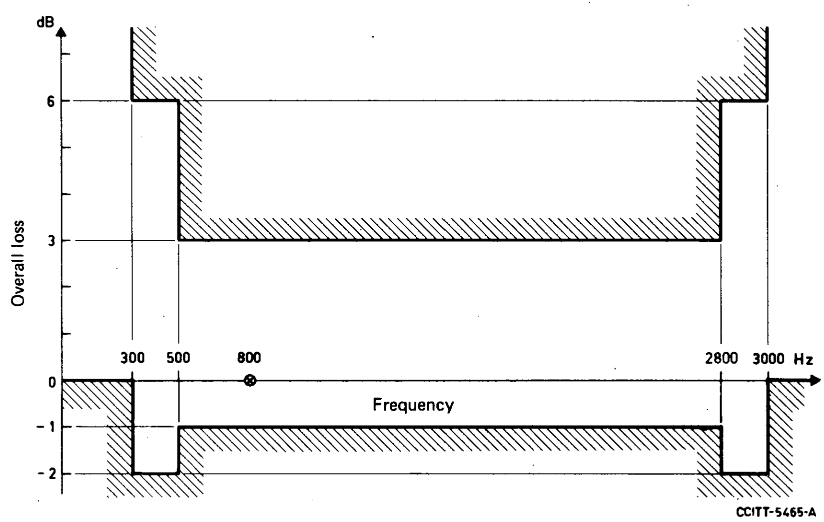
Because of the differing nominal level at renters' premises due to the various national practices, it is not normally possible to predict the nominal overall loss of the circuit at the reference frequency. Only exceptionally can a predetermined specified nominal overall loss at the reference frequency between renters' installations be offered and then only after prior consultation among the Administrations concerned.

In the general case, however, for 4-wire circuits, the value of the receiving relative level at the renters' premises should, provisionally, not be lower than -15 dBr. It may therefore, be assumed that the maximum nominal overall loss will not normally exceed 28 dB.

It should be noted that the overall loss in each direction of transmission may not have the same value.

2.2 *Loss/frequency distortion*

The limits for the overall loss relative to that at 800 Hz for the circuit between renters' installations are given in Figure 2/H.12.



Note. – At frequencies below 300 Hz and above 3000 Hz, the loss shall not be less than 0.0 dB but is otherwise unspecified.

FIGURE 2/H.12 – **Limits for the overall loss of the circuit relative to that at 800 Hz**

2.3 *Group-delay distortion*

The limits that apply to group-delay distortion [6)] are given in Figure 3/H.12 in which the limiting values over the frequency band are expressed as values relative to the minimum measured group delay.

[5)] Additionally, the characteristics for longitudinal balance and short breaks in transmission are under study.

[6)] The fact that national measuring instruments normally used may not be of the same nominal impedance is of negligible consequence. Impedances known to exist at present are 600, 800 and 900 ohms and using any pair of these introduces negligible error.

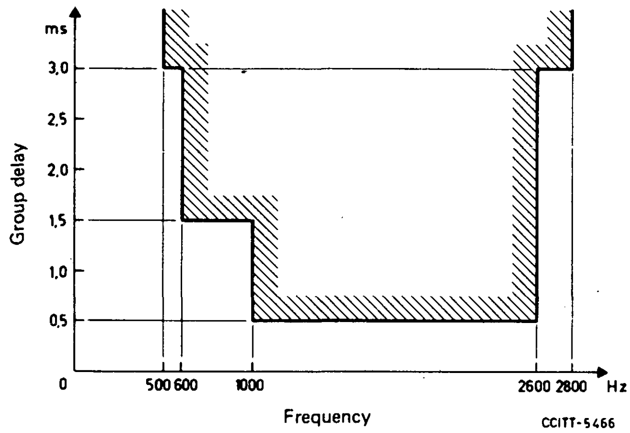


FIGURE 3/H.12 – Limits for group delay relative to the minimum measured group delay in the 500-2800 Hz band

2.4 *Variation with time of the overall loss at 800 Hz*

The variation with time of the overall loss at 800 Hz should be as small as possible and should not exceed the following limits:

- short-term variations
 (over a period of a few seconds) . . . . ± 3 dB
- long-term variations
 (over long periods including daily and seasonal variations . . . . ± 4 dB

2.5 *Random circuit noise*

The nominal level of the psophometric noise power at the renter's premises depends upon the actual constitution of the circuit, in particular upon the length of frequency division multiplex carrier systems in the circuit. The provisional limit for special-quality leased circuits of distances greater than 10 000 km is −38 dBm0p. However, circuits of shorter length will have substantially less random noise (see also the Annex at the end of the Recommendation).

2.6 *Impulsive noise*

Impulsive noise should be measured with an instrument complying with Recommendations O.71 and H.13.

As a provisional limit, the number of impulsive noise peaks exceeding −21 dBm0 should not be more than 18 in 15 minutes.

A method of measurement is described in the Annex to Recommendation H.13. Final values are under study.

2.7 *Phase jitter*

It is normally expected that any measurement of phase jitter using an instrument complying with Recommendation O.91 will not exceed 15° peak-to-peak. This is, however, a provisional guide and subject to further study.

2.8 *Quantizing noise (quantizing distortion)*

If any circuit section is routed over a PCM system, the signal will be accompanied by quantizing noise. The minimum ratio of signal-to-quantizing noise normally expected is 22 dB.

2.9 *Single tone interference*

The level of single tone interference in the band 300-3400 Hz shall not exceed a value which is 3 dB below the circuit noise objective indicated in Figure 4/H.12. This limit is provisional pending further study.

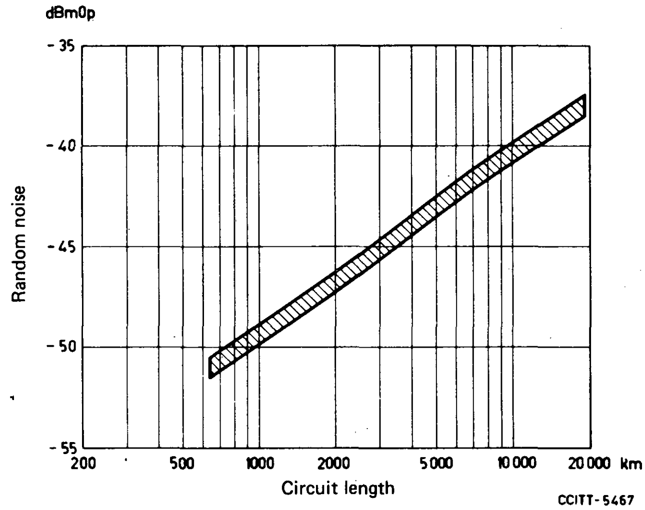


FIGURE 4/H.12 – Random noise circuit performance

2.10 *Frequency error*

The frequency error introduced by the circuit must not exceed ± 5 Hz. It is to be expected that in practice the error will be within closer limits than these.

2.11 *Harmonic distortion*

When a 700-Hz test frequency of – 13 dBm0 is injected at the transmit end of a point-to-point circuit, the level of any individual harmonic frequency at the receiving end shall provisionally be at least 25 dB below the received level of the fundamental frequency.

ANNEX

(to Recommendation H.12)

Random circuit noise

Figure 4/H.12 displays random noise versus length and is presented as a guide to the random noise performance which may be found on an international leased circuit.

Note. – At the present time the section of the circuit provided by satellite (between earth stations) contributes approximately 10 000 pW0p (– 50 dBm0p) of noise. Therefore, for the purpose of determining maintenance limits for noise measurements on leased circuits, the length of a circuit section routed on a communications satellite may be considered to be equivalent to 1000 km in Figure 4/H.12.

Recommendation H.13

CHARACTERISTICS OF AN IMPULSIVE-NOISE MEASURING INSTRUMENT FOR TELEPHONE-TYPE CIRCUITS [7)]

(Mar del Plata, 1968)

The CCITT,

considering,

that impulsive noise is of interest to data transmission and telegraphy and that first consideration should be given to a simple pulse counter suitable for field use,

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 midpoint 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 volts, 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 volts, and with an interval between pulses in excess of the operating time of the counter [dead-time, see f) below] shall be applied to the input of the instrument and shall 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 [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;

[7)] This instrument can be used to check whether a circuit is suitable for data transmission of VF telegraphy; it can also be used to assess the disturbing effect of impulsive noise on telephone conversations (Recommendation P.55, Volume V).

Response to fall-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 H.13)

Use of the impulsive noise counter 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 thresholds −18 dBm0 and −22 dBm0;

the threshold settings should be −18 dBm0 for the telephone-type circuit and −21 dBm0 for the special-quality circuits mentioned in Recommendation H.12, 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 h) of Recommendation O.71; the United Kingdom Post Office uses an impulse counter that also includes a filter having the following characteristics:

3-dB points at 300 Hz to 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 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.

4. The correlation between the bit error rate and the number of impulsive counts thus determined has not yet been established.

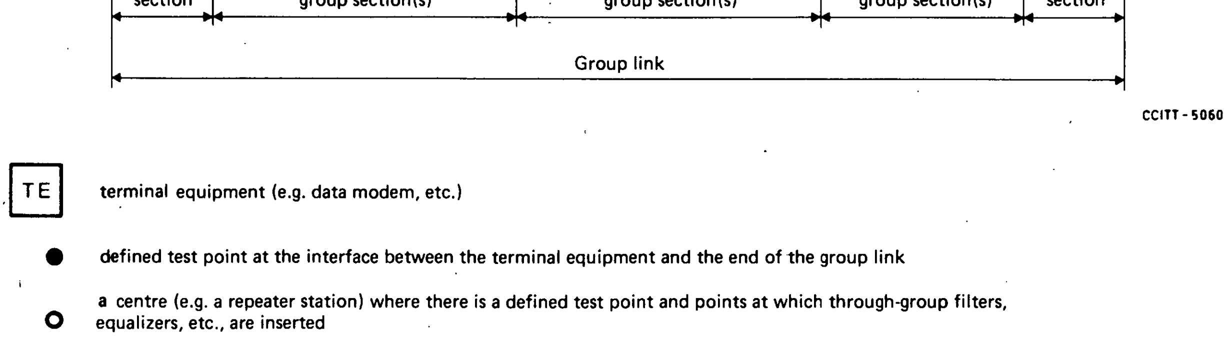
Recommendation H.14

CHARACTERISTICS OF GROUP LINKS FOR THE TRANSMISSION OF WIDE-SPECTRUM SIGNALS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1976)

A. CONSTITUTION OF A LINK, TERMINOLOGY, AND SCOPE OF THE RECOMMENDATION

A group link is composed of one or more group sections in tandem, generally prolonged at each end by "local lines" (denoted terminal group sections in Figure 1/H.14). These terminal group sections connect the group distribution frames of the terminal national centres with the equipment for sending and receiving wide-spectrum signals (modems, etc.) which may be situated either in the subscriber's premises or in any other place. In the latter case they are normally switched over the local telephone cable network, or sometimes over a special cable line or a radio-relay link. Only "local lines" carrying the 60-108 kHz wide-spectrum signal are



[a] These sections are composed of one or more group sections.

FIGURE 1/H.14 – Example of the constitution of a group link for wide-spectrum signal transmission termed terminal group sections and are included in the definition of a group link. The other case in which a baseband signal occupying a frequency band other than 60-108 kHz is transmitted over the "local lines", the frequency translation to the 60-108 kHz band being made at the terminal national centres, is not dealt with in this Recommendation.

It should be noted that the group link comprises any supplementary equipment for equalizing, filtering, etc., in the 60-108 khz band but not the terminal equipment (modems, etc.). Figure 1/H.14 illustrates these considerations.

B. CHARACTERISTICS OF CORRECTED GROUP LINKS

The characteristics mentioned in a) and b) below imply the use of a group pilot at 104.08 kHz. The use of a pilot in the middle of the group band requires different characteristics.

a) *Group-delay distortion*

The group-delay distortion over the band 68-100 kHz should not exceed $(5 + 10\ n)$ μs with respect to the value of the least group delay within that band. The parameter n is the number of through-group connection equipments encountered in the group link.

Note 1. – The following assumptions have been made in order to derive the above formula.

1) The group-delay distortion of through-group connection equipment can be corrected so as not to exceed 10 μs over the band 68-100 kHz. It should be noted that through-group connection equipment comprises the group demodulating equipment, the through-group filter and the group modulating equipment, (see Recommendation G.242). The equalization should be made in such a way that at least 6 group-delay maxima are obtained (6 being a provisional value).

2) An allowance of 5 μs has been made for the combined effects of the first group modulation equipment and the final group demodulation equipment encountered in the link (together with any group pilot stop filters) [8]. Correction may be necessary to achieve this.

3) The distortions in 1) and 2) above may be expected not to add up adversely.

4) In the case of "delayed transfer" or "multipoint links" an additional pilot suppression of 40 dB is required. Considering the group-delay characteristics of existing 104.08 kHz pilot stop filters, and assuming additional group-delay equalization, the requirement of 5 μs (see above) in the frequency band 68-100 kHz can be met by this 40 dB stop filter and the corresponding group translating equipment.

5) To respect these limits it may be necssary to avoid groups 1 and 5.

6) The use of a group containing the supergroup pilot should always be avoided.

Note 2. – In certain cases where disturbing signals outside the basic group band have to be expected, additional filtering has to be provided in the local lines. No group-delay distortion caused by such protection filters is included in the formula.

Note 3. – The effect of through-supergroup connection equipment can be significant, particularly for groups 1 and 5. Hence, when supergroup through-connection occurs, the avoidance of these groups [as mentioned in 5) of Note 1] becomes practically essential. If the group section which is made the subject of delay

---

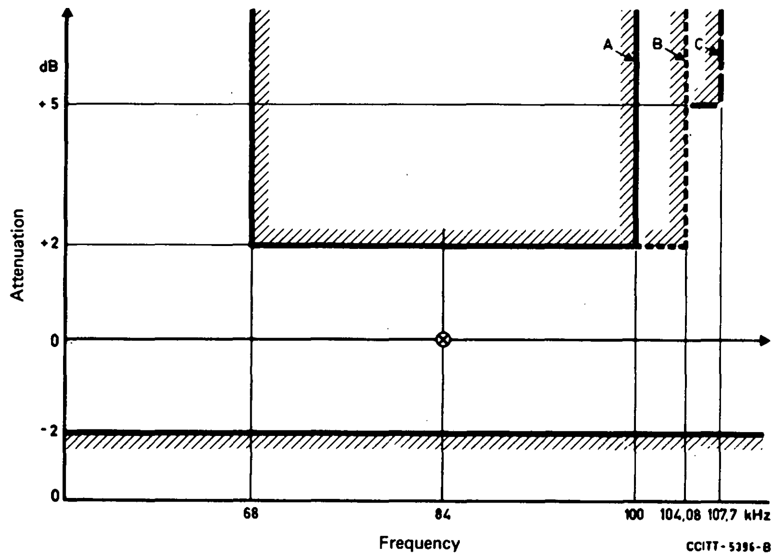
[8] Group pilot blocking filters for this purpose have to be chosen carefully with respect to the group delay characteristic. Filters which are similar with regard to the loss frequency characteristic may, depending on the design, be very different with regard to group delay distortion.

distortion equalization is considered to comprise a group modulating equipment, a line, a group demodulating equipment and a through-group filter, then the effects of the through-supergroup connection equipment can be readily taken into account.

Note 4. – If the group-delay distortion limits required for a particular service are stricter than those given by the formula, the additional equalization should preferably be introduced in the terminating equipments.

b) *Attenuation/frequency distortion*

The attenuation/frequency distortion of the whole link is given in Figure 2/H.14. It should be measured over the 60-108 kHz frequency range and equalized with a group link equalizer as necessary to meet the limits with respect to loss at 84 kHz.



A = these limits apply if the group reference pilot (104.08 kHz) is injected at an intermediate point on the link (e.g. the terminal national centre)

B = these limits apply if the group reference pilot is transmitted from end to end on the link (e.g. if it is injected by the terminal equipment)

C = these limits apply if the service channel is provided

FIGURE 2/H.14 – Limits for attenuation/frequency distortion

Note 1. – If the service channel is provided, additional equalization may be needed and there will be no possibility of using simplified through-group filters.

Note 2. – 84 kHz is the reference frequency for the purposes of specifying and measuring attenuation/frequency distortion. This is not in contradiction with the use of the group reference pilot at 104.08 kHz.

c) *Carrier leaks*

The leak from a carrier in the 60-108 kHz band shall not exceed −40 dBm0.

Note 1. – Although this value is an objective, it may in some cases prove impossible to achieve owing to the composition of the link, which will generally involve the use of both old and new types of equipment. In any case, no carrier leak in the band 60-108 kHz should exceed −35 dBm0.

Note 2. – For group links used in data transmission with modems in line with Recommendation V.36, problems may arise if the total carrier leak power exceeds −35 dBm0.

d) *Variations in level*

The following limits should not be exceeded:

– short-term variations
(for a few seconds) . . . . ± 3 dB

– long-term variations
(during long periods, including seasonal and daily variations) . . . . ± 4 dB

relative to the nominal level.

e) *Background noise*

This can be expected to be substantially uniformly distributed over the group band, and to have a value calculated in accordance with Recommendations G.222 and G.223, 4. For an actual link, a margin should be allowed as indicated in Recommendation G.226.

f) *Impulsive noise*

Under study [9].

g) *Frequency error*

Maximum frequency error shall not exceed 5 Hz.

Note. – According to Recommendation G.225, this condition should readily be met in practice.

h) *Phase changes with time*

Under study [9].

i) *Power handling capability*

Applied signals should be within the limits given in Recommendation H.52.

C. CHARACTERISTICS OF NON-CORRECTED GROUP LINKS

Under study [9].

---

[9] Under Questions 28/XV and 10/XVII.

Recommendation H.15

CHARACTERISTICS OF SUPERGROUP LINKS FOR THE TRANSMISSION OF WIDE-SPECTRUM SIGNALS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1976)

A. CONSTITUTION OF A LINK AND TERMINOLOGY

The constitution [10)] and terminology for supergroup links are analogous to those for group links described in Recommendation H.14.

B. CHARACTERISTICS OF CORRECTED SUPERGROUP LINKS

a) *Group-delay distortion*

Provisionally, the rule $(1 + 2n)$ µs over the band 325-512 kHz is recommended as the limit for a supergroup link with *n* through-supergroup connection equipments (i.e. modulating, demodulating and through-supergroup filter equipments). Supergroup links with corrected group-delay distortion should be restricted to supergroups 5, 6 and 7 in a mastergroup.

b) *Attenuation/frequency distortion*

Over the band 352-512 kHz the attenuation/frequency distortion should not exceed ± 2 dB with respect to the attenuation at 412 kHz.

Note. – The reference frequency for the purposes of defining *distortion* should be 412 kHz even if the supergroup reference pilot used for *regulating* purposes is 547.92 kHz.

c) *Carrier leaks*

The leak from a carrier in the 352-512 kHz band shall not exceed −40 dBm0.

Note. – Although this value is an objective, it may in some cases prove impossible to achieve owing to the composition of the link, which will generally involve the use of both old and new types of equipment. At all events, no carrier leak in the 352-512 kHz band should exceed −35 dBm0.

d) *Variations in level*

The following limits should not be exceeded:

– short-term variations
  (for a few seconds) . . . . ± 3 dB

– long-term variations
  (during long periods, including seasonal and daily variations) . . . . ± 4 dB

relative to the nominal level.

[10)] *Note by the Secretariat.* – It was not discussed, but it does not seem likely that the local telephone cable network could be used to extend a supergroup link, as is envisaged for a group link in Recommendation H.14, A.

e) *Background noise*

This can be expected to be substantially uniformly distributed over the supergroup band, and to have a value calculated in accordance with Recommendations G.222 and G.223, 4. For an actual link, a margin should be allowed as indicated in Recommendation G.226.

f) *Impulsive noise*

Under study [11].

g) *Frequency error*

Maximum frequency error shall not exceed 5 Hz.

Note. – According to Recommendation G.225, this condition should readily be met in practice.

h) *Phase changes with time*

Under study [11].

i) *Power handling capability*

Applied signals should be within the limits given in Recommendation H.53.

C. CHARACTERISTICS OF NON-CORRECTED SUPERGROUP LINKS

Under study [11].

Recommendation H.16

CHARACTERISTICS OF AN IMPULSIVE-NOISE MEASURING INSTRUMENT FOR WIDEBAND DATA TRANSMISSION

(Geneva, 1972)

The CCITT,

considering,

that impulsive noise is of interest in wideband data transmission and that there is a need for a simple pulse counter suitable for field use,

provisionally *recommends,*

that the instrument for impulsive-noise measurements should have the following characteristics:

a) *Types of measurements*

For the measurement of impulsive noise, the instrument should register a count whenever the instantaneous level applied to the input exceeds an adjustable threshold. This operation should be independent of the sense (or polarity) of the applied impulse.

For the measurement of circuit noise the instrument should provide means for indicating the average noise power.

---

[11] Under Questions 28/XV and 10/XVII.

b) *Input impedance*

The instrument should permit the measurements designated above on either balanced or unbalanced circuits at the nominal impedances which are used in wideband data transmission. On balanced circuits, the instrument should also be arranged to measure impulsive or circuit noise which is common to the two sides of the circuit with respect to earth.

Nominal input impedances should be provided as follows:

1) 75 ohms unbalanced;

2) 135 or 150 ohms balanced;

3) 135 or 150 ohms balanced with 20 000 ohms from each side of the circuit to a common 600 ohms which is returned to earth (the noise measurement is made across the 600 ohm resistor).

For the balanced input impedance (2. above) the balance of the input circuit in relation to earth should be such that when a 25-kHz sine wave, whose level is 70 dB higher than the instrument's threshold setting, is applied between the midpoint of the source impedance and the earth terminal of the instrument, the counter should not operate. Similarly, a 560-kHz sine wave, whose level is 42 dB higher than the threshold, when applied between the source impedance and the earth terminal of the instrument should not operate it. The above balance requirements shall hold for signal levels ranging up to 30 volts r.m.s.

Input arrangement 3) above is provided for use in measuring impulsive and circuit noise which is common to the two sides of a balanced circuit with respect to earth.

c) *Bandwidth*

A maximum bandwidth condition should be provided which has a nominal attenuation of 3 dB at 150 Hz and 560 Hz with respect to 25 kHz. In this condition the response should be within ± 1 dB in the frequency range from 275 Hz to 250 kHz and should provide attenuation of at least 10 dB at 50 Hz and 1500 Hz.

Provision should be made for measurements on other specific bandwidths such as group or supergroup bands. These bandwidths may be provided by means of plug-in or external filters.

Note. – The specific bandwidths which will be recommended are the subject of Question 5/CMBD.

d) *Sensitivity and accuracy*

For the measurement of impulsive noise, the threshold should be adjustable in steps of 1 dB for instantaneous levels from -60 to $+20$ dBm. For the measurement of circuit noise the sensitivity of the instrument should be -90 to $+10$ dBm. The accuracy of the instrument at the calibration frequency shall be ± 0.5 dB for any threshold setting or input polarity. The relative response to other signals should depend only on the attenuation characteristics for the maximum bandwidth or other selected bandwidths. The sensitivity of the instrument may be 30 dB less when used in the condition to measure circuit noise and impulsive noise common to the two sides of a balanced circuit with respect to earth.

e) *Counting rate*

Dead-time is defined as the time from the start of an impulse being registered until the counter is ready to register another impulse. A dead-time of 125 ± 25 ms should be provided within the instrument.

Thus, the maximum counting rate is nominally eight impulses per second. The capacity of the counter shall be at least 999.

f) *Calibration*

Calibration should be possible from an internal signal or from the peaks of an externally applied sine-wave signal. For measurement of impulsive noise the calibration should be such that with the threshold adjusted to +3 dBm, the peaks of a 0 dBm sine wave will just operate the counter.

g) A built-in timer, continuously adjustable from 5 to 60 minutes, should be provided. The accuracy should be within ± 10% of the setting.

h) *Temperature stability*

All of the preceding clauses shall be satisfied when the ambient temperature varies between +10 °C and +40 °C.

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 2

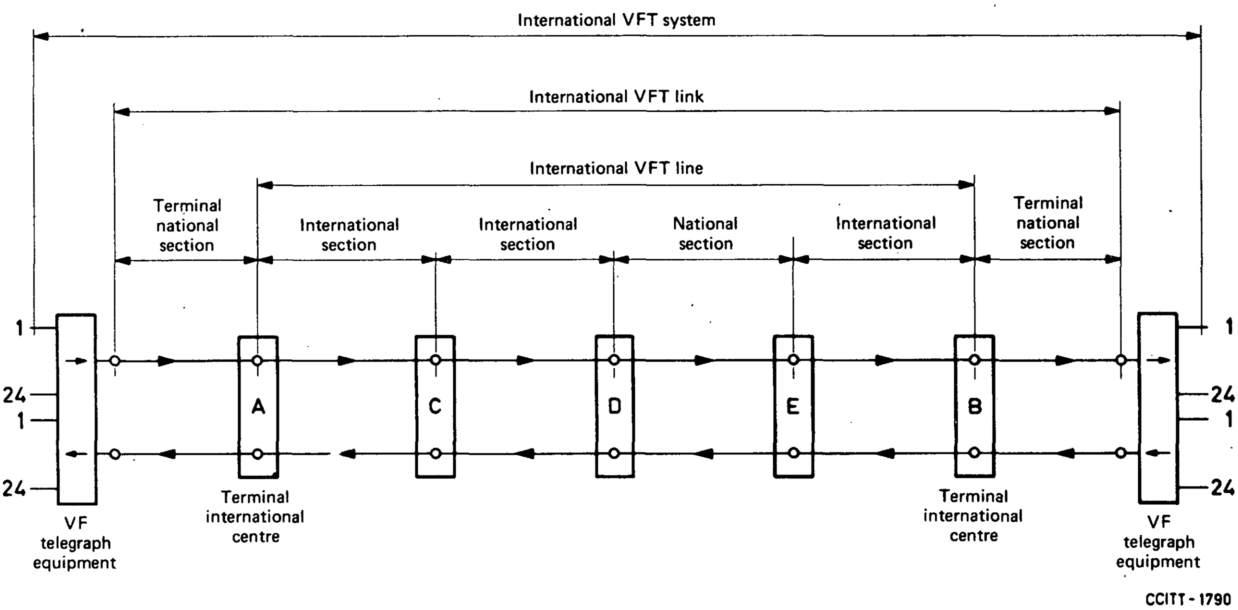
USE OF TELEPHONE-TYPE CIRCUITS FOR VOICE-FREQUENCY TELEGRAPHY

Recommendation H.21

COMPOSITION AND NOMENCLATURE OF INTERNATIONAL VOICE-FREQUENCY TELEGRAPH SYSTEMS

(Mar del Plata, 1968)

Figure 1/H.21 illustrates the composition of an international voice-frequency telegraph (VFT) system and the nomenclature used.



(At the intermediate centres C, D and E and at the terminal international centres A and B, the signals transmitted are at audio frequencies. At these points it is possible to make measurements.)

FIGURE 1/H.21 – **The components of an international VFT system**

a) *International voice-frequency telegraph system*

This is the whole of the assembly of apparatus and lines including the terminal VFT equipment. In Figure 1/H.21 the system illustrated provides 24 duplex international telegraph circuits, but other numbers of telegraph circuits can be provided.

b) *International VFT link*
(sometimes referred to as the bearer circuit)

1. Four-wire telephone-type circuits are used for VFT links. The link comprises two unidirectional transmission paths, one for each direction of transmission, between the terminal VFT equipments.

2. The VFT link consists of an international telegraph line together with any terminal national sections connecting the international telegraph line to the VFT telegraph terminal equipment and may be constituted entirely on carrier channels (on symmetric pair, coaxial pair or radio-relay systems) or on audio-frequency lines or combinations of such lines.

3. The normal links for VF telegraphy have no terminating units, signalling equipment or echo suppressors.

c) *International VFT line*

1. The international VFT line may be constituted by using a channel in a carrier group or channels in tandem on a number of groups. National and international sections can be interconnected to set up an international telegraph line. See Figure 1/H.21, but note that c) 2. below details the preferred method.

The international telegraph line could equally well be set up between, for example, only A and C or between C and D, in which case A and C, or C and D would be the terminal international centres.

2. Wherever possible an international telegraph line for a VFT link should be provided on channels of a single carrier group, thereby avoiding intermediate audio-frequency points. In some cases, such a group may not exist or, for special routing reasons, it may not be possible to set up the international telegraph line in the preferred way. In such cases, the international telegraph line will consist of channels in tandem on two or more groups with or without audio sections, depending on the line available and the routing requirements.

d) *Terminal national sections connected to the international telegraph line*

In many cases the VFT terminal equipment is remote from the terminal international centre of the international telegraph line (Figure 1/H.21), and such cases necessitate the provision of terminal national sections in order to establish international VFT links. These sections may be in short-distance local audio cables, amplified or unamplified, or may be routed in long-distance carrier groups or amplified audio plant as available.

Recommendation H.22

TRANSMISSION REQUIREMENTS OF INTERNATIONAL VOICE-FREQUENCY TELEGRAPH LINKS (AT 50, 100 AND 200 BAUDS)

(Mar del Plata, 1968; amended at Geneva, 1972)

A. LINKS ROUTED ON CARRIER SYSTEMS

Figure 1/H.21 shows the composition of an international circuit for voice-frequency (VF) telegraphy. The limits specified in the present Recommendation are based on the values between international terminal centres which are indicated in Recommendation G.151 for an international telephone circuit and which are applied approximately to the "international line" in Figure 1/H.21. A slight increase has been made to certain characteristics to allow for unloaded national sections connecting the centres to the VF telegraph equipments since most telegraph installations belonging to public services are fairly close to the international maintenance centres.

a) *Nominal insertion loss at 800 Hz*

The nominal insertion loss of the link at 800 Hz depends on the nominal relative power levels at the extremities of the telegraph link. These levels will be those normally used in the national network of the countries concerned so that it is not possible to recommend a particular nominal value for the insertion loss.

The nominal relative power level at the input to the link and the absolute power level of the telegraph signals at this point must be such that the limits concerning the power level per telegraph channel at a zero relative point on carrier systems are respected.

b) *Variation of insertion loss with time*

In accordance with Recommendation M.160:

1) the difference between the mean value and the nominal of the transmission loss value should not exceed 0.5 dB;

2) the standard deviation from the mean value should not exceed 1 dB.

However, in the case of circuits set up wholly or partly on older equipment, where the international line consists of two or more circuit sections, a standard deviation not exceeding 1.5 dB may be accepted.

c) *Sudden variations of insertion loss and short interruptions*

Such defects of the transmission path impair the quality of the telegraph transmission and should be reduced to the minimum possible.

d) *Overall loss/frequency distortion*

The variation with frequency of the 600-ohm insertion loss of the link with respect to the loss at 800 Hz must not exceed the following limits:

1. *Links with 4-kHz sections throughout* (Table 1/H.22)

TABLE 1/H.22

Frequency range (Hz)	Overall loss relative to that at 800 Hz
Below 300	Not less than −2.2 dB, otherwise unspecified
300- 400	−2.2 to +4.0 dB
400- 600	−2.2 to +3.0 dB
600-3000	−2.2 to +2.2 dB
3000-3200	−2.2 to +3.0 dB
3200-3400	−2.2 to +7.0 dB
Above 3400	Not less than −2.2 dB, otherwise unspecified

These limits are shown hatched in Figure 1/H.22.

Note. – The hatched limits in Figure 1/H.22 have been derived from the corresponding limits in Recommndation G.151 by adding a margin to allow for the presence of unloaded national sections and also for the fact that the composition of the international line may be more complicated. This will permit the establishment of most international circuits for VF telegraphy without additional equalization.

In favourable cases it will be possible to respect the limits in the graph in Recommendation G.151 which is shown as a broken line in Figure 1/H.22.

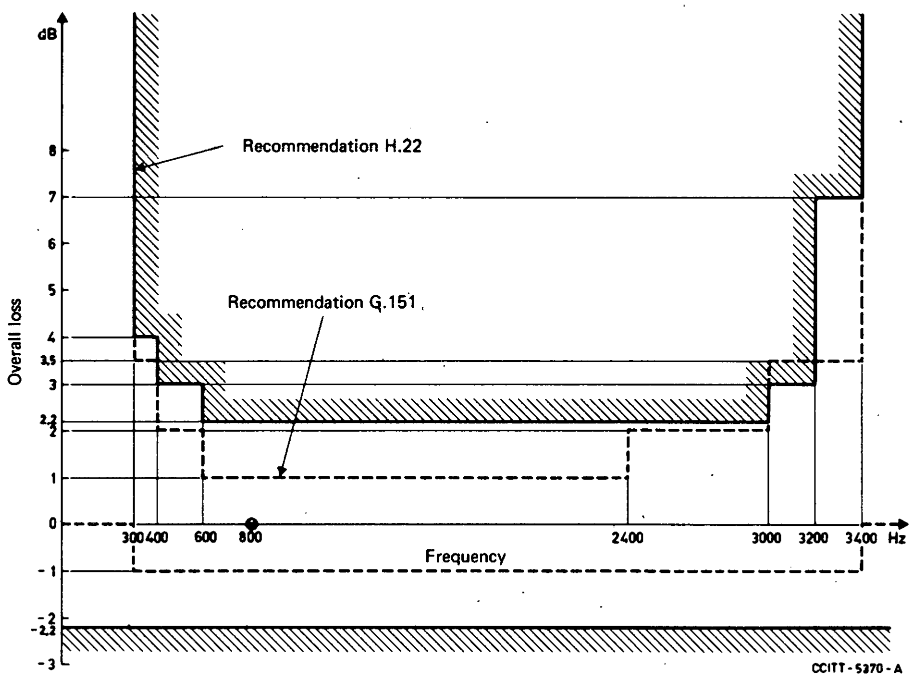


FIGURE 1/H.22 – Variation with frequency of the overall loss relative to the value measured at 800 Hz with 4 kHz end-to-end sections

2. *Links with one or more 3-kHz sections* (Table 2/H.22)

TABLE 2/H.22

Frequency range (Hz)	Overall loss relative to that at 800 Hz
Below 300	Not less than −2.2 dB, otherwise unspecified
300- 400	−2.2 to +4.0 dB
400- 600	−2.2 to +3.0 dB
600-2700	−2.2 to +2.2 dB
2700-2900	−2.2 to +3.0 dB
2900-3050	−2.2 to +6.5 dB
Above 3050	Not less than −2.2 dB, otherwise unspecified

e) *Noise*

1. *Uniform-spectrum random noise*

The mean psophometric noise power referred to a point of zero relative level should not exceed 80 000 pW0p (−41 dBm0p) [1)].

Note. – It was not possible to recommend a limit for the unweighted noise point level. The CCITT psophometer with the telephone weighting network should continue to be the instrument used for specifying and measuring random noise power levels on international telegraph links.

[1)] If recourse be had to synchronous operation, a higher noise level might be tolerated (such as −30 dBm0p for a particular telegraph system).

2. *Impulsive noise*

Impulsive noise should be measured with an instrument complying with Recommendation H.13 and used in the "flat" condition.

As a provisional limit for maintenance purposes, the number of impulsive noise peaks exceeding −18 dBm0 should not be more than 18 in 15 minutes.

Note. – Final values are still under study.

f) *Crosstalk*

1) The crosstalk ratio between the go and return channels of the link should be at least 43 dB.
2) The crosstalk ratio between the link and other carrier circuits is restricted by Recommendation G.151, D. to be not worse than 58 dB.

Crosstalk in any audio cables forming part of the terminal national sections should not normally significantly worsen the crosstalk ratio.

g) *Mean one-way propagation time*

The one-way propagation time referred to is the group delay as defined in the ITU *List of Definitions of Essential Telecommunication Terms* (Definition No. 04.17) calculated at a frequency of about 800 Hz.

It should be noted that VFT links routed over high-altitude satellite communication systems introduce mean one-way propagation times in excess of 260 ms.

h) *Group-delay distortion*

Practical experience obtained up to the present shows that it is not necessary to recommend limits for group-delay distortion for 50-baud VFT links even when they are composed of several sections each provided on telephone channels of carrier systems. There is little practical experience with higher speed telegraph systems.

It may happen that under adverse conditions some telephone channels of the link are of insufficient quality to provide 24 telegraph channels. In such a case, a better combination of telephone channels must be chosen for the telegraph service.

i) *Frequency drift*

The frequency drift introduced by the link must not be greater than 2 Hz. According to Recommendation G.225, this condition is fully met in practice even when the international line for VF telegraphy has the same composition as the 2500 km hypothetical reference circuit for the transmission system used.

j) *Interference caused by power supply sources*

When a sinusoidal measuring signal is transmitted over the link at a level of 0 dBm0 the level of the strongest unwanted side component should not exceed −45 dBm0 (see also Recommendation G.151 and Question 11/XV).

k) *Variation introduced by changeover to the reserve line or section*

1. *Change in insertion loss at 800 Hz*

Bearing in mind that the insertion loss of the normal line (or section) and the reserve line (or section) are both subject to variations with time, which in general will be uncorrelated, it is not possible to assign a limit to the change of insertion loss at 800 Hz introduced by the changeover procedure.

2. *Change in the insertion loss at other frequencies relative to that introduced at 800 Hz*

The insertion-loss distortion characteristic of the link when established over the normal route should be within 2 dB or less of that of the link when established over the reserve route. This limit applies over the frequency bands 300-3400 Hz or 200-3050 Hz as appropriate.

There should ordinarily be no difficulty in achieving the limit when only one portion of the link – for example, the international telegraph line or one section – has a reserve section. However, when two or more portions of the link are separately associated with reserve portions, it becomes difficult to ensure that all combinations of normal and reserve portions comply with the limit. In these circumstances, the best that can be done is to ensure that the insertion-loss characteristics of corresponding normal and reserve portions are as much alike as possible. Careful attention should be paid to the impedance of normal and reserve sections at the point where they are connected to the changeover apparatus so that errors due to changing mismatch losses are minimized. A suitable target would be for all impedances concerned to have a return loss against 600 ohms, non-reactive of not less than 20 dB over the appropriate band of frequencies.

3. The nominal relative power level at 800 Hz of the normal and reserve lines or sections at the changeover points for a particular direction of transmission should be the same. This level will be that normally used in the national network of the country concerned.

B. LINKS VIA AUDIO-FREQUENCY LINE PLANT

a) *Attenuation/frequency distortion*

Graph No. 6, Figure 2/H.22, shows the variations with frequency of the difference between the relative power levels at the origin and extremity of the link relative to the measured value at 800 Hz.

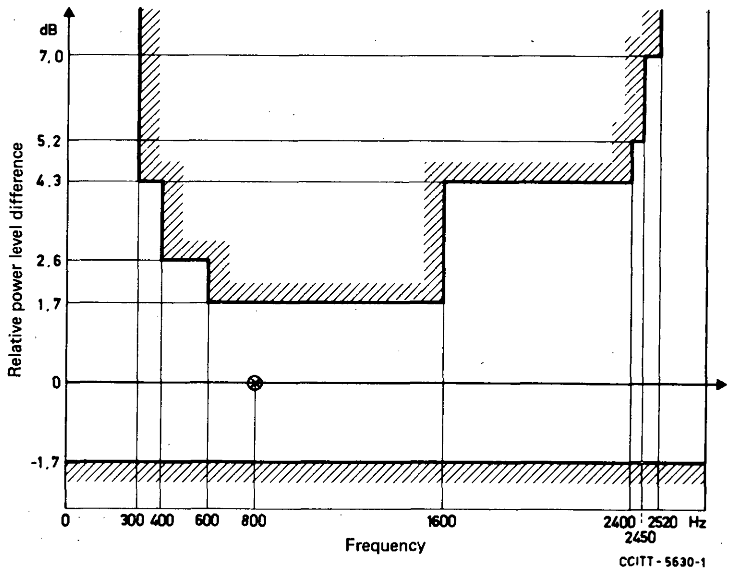


FIGURE 2/H.22. Graph No. 6 – Limits for the variation with frequency, relative to the value at 800 Hz, of the difference in relative power levels (in dB) between the input and output of a link used for VF telegraphy (set up on a telephone circuit using the band 300-2600 Hz)

The permissible tolerances for the relative power level at the output of the frontier repeater are the same as those for 4-wire repeaters, if maintenance measurements are made by sending a power giving 1 mW at a zero relative level point (as found from the telephone circuit level diagram) to the input of the link for VF telegraphy. These tolerances are shown in Graph No. 7, Figure 3/H.22.

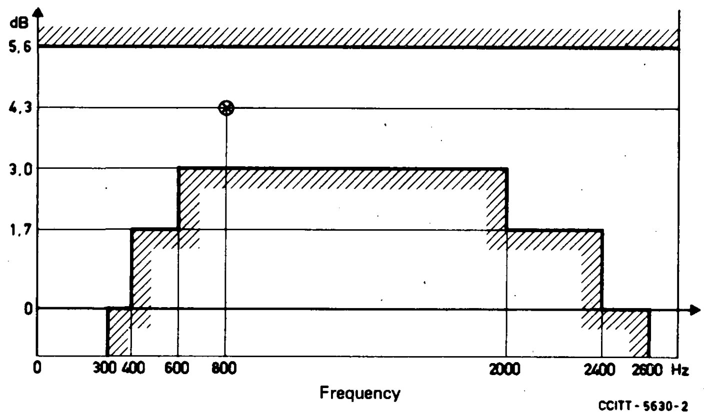


FIGURE 3/H.22. Graph No. 7 – **Maintenance limits for the absolute power level (in dB) at the output of a frontier repeater (frontier side) for an international circuit with a bandwidth of 300-2600 Hz and used for VF telegraphy (to be measured with a sent power at the origin of the VFT link such as to give 1 mW at a zero relative level point, deduced from the level diagram of the telephone circuit)**

It does not appear necessary to fix particular limits for the variations with frequency of the level measured at the output of the frontier repeater since these may be calculated easily from the limits allowed for the relative power level.

b) *Level variations with time*

The relative power level at the point at the receiving end where the changeover between the VFT telegraph circuit and its reserve circuit takes place must be as constant as possible with time. Furthermore, any interruption in the circuit, even for a very short duration, spoils the quality of the telegraph transmission. Great care must therefore be taken when measurements are made on circuits and repeaters, when changing-over batteries, etc. To draw the attention of the staff to this matter, it is desirable for circuits used for VFT to be specially marked at the terminal stations and in the intermediate repeater stations.

c) *Freedom from modulation*

It is desirable to make special arrangements to avoid any modulation on the circuits and in the repeaters. Such modulation may in particular be caused by the variation in battery voltages or by the connection of equipment for sub-audio telegraphy to the cable pairs.

Recommendation H.23

BASIC CHARACTERISTICS OF TELEGRAPH EQUIPMENTS USED IN INTERNATIONAL VOICE-FREQUENCY TELEGRAPH SYSTEMS [2)]

(Mar del Plata, 1968; amended at Geneva, 1976)

A. LIMITING POWER PER CHANNEL

a) *Amplitude-modulated voice-frequency telegraph (AMVFT) systems at 50 bauds*

Administrations will be able to provide the telegraph services with carrier telephone channels permitting the use of 24 VFT channels (each capable of 50 bauds) on condition that the power of the telegraph channel signal on each channel, when a continuous marking signal is transmitted, does not exceed 9 μW0.

For 18 telegraph channels only, the power so defined may be increased to 15 μW0 per telegraph channel so that even telephone channels with a relatively high noise level can then be used.

The power per telegraph channel should never exceed 35 μW0, however few channels there may be.

These limits are summarized in Table 1/H.23.

TABLE 1/H.23 – **Limiting power per telegraph channel when sending a continuous marking signal in AMVFT systems at 50 bauds**

System	Limiting power at zero relative level point per telegraph channel when sending a continuous marking signal	
	μW0	dBm0
12 telegraph channels or less	35	−14.5
18 telegraph channels	15	−18.3
24 telegraph channels	9	−20.5

b) *Frequency-modulated voice-frequency telegraphy (FMVFT) systems at 50 bauds*

The mean power transmitted to line by 50-baud FMVFT systems is limited to 135 μW0 when all channels of the system are sending. This gives the limits shown in Table 2/H.23 for the mean permissible power per telegraph channel at a zero relative level point.

Some Administrations have bilateral agreements to reduce the total mean power level of FMVFT systems to −13 dBm0 (50 μW0). The CCITT encourages such reduction where feasible. The above Administrations have made their own determination of the feasibility of operating at the reduced level. As a guide, other Administrations may wish to use the suggested link parameters provided by Study Group IX as given in the Annex to this Recommendation.

[2)] This Recommendation reproduces, for information, some characteristics given in Recommendations R.31 and R.35, Volume VII.

TABLE 2/H.23 – **Normal limiting power per telegraph channel in 50-baud FMVFT systems**

System	Permissible mean power at zero relative level point per telegraph channel	
	μW0	dBm0
12 telegraph channels or less	11.25	−19.5
18 telegraph channels	7.5	−21.3
24 telegraph channels	5.6	−22.5

B. TELEGRAPH CHANNEL FREQUENCIES

For international VF 24-channel, 50-baud, non-synchronous telegraph systems the frequency series consisting of odd multiples of 60 Hz has been adopted, the lowest frequency being 420 Hz as shown in Table 3/H.23. In the case of frequency-modulated systems, these frequencies are the centre frequencies of the telegraph channels, the frequency of the signal sent to line being 30 Hz (or 35 Hz) above or below the centre frequency according to whether A or Z polarity is being sent.

TABLE 3/H.23

Telegraph channel position	Frequency (Hz)	Telegraph channel position	Frequency (Hz)
1	420	13	1860
2	540	14	1980
3	660	15	2100
4	780	16	2220
5	900	17	2340
6	1020	18	2460
7	1140	19	2580
8	1260	20	2700
9	1380	21	2820
10	1500	22	2940
11	1620	23	3060
12	1740	24	3180

In addition a pilot channel using a frequency of 300 Hz or 3300 Hz can be used. For details of the normal frequencies used in other types of telegraph system, see Recommendations R.37, R.38 A and R.38 B.

ANNEX

(to Recommendation H.23)

Limits required by Study Group IX in respect of the bearer circuit for FMVFT if the total telegraph power is to be reduced to 50 microwatts (from 135 microwatts)

Loss/frequency distortion

The variation with frequency of the overall loss of the link with respect to the loss at 800 Hz should not exceed the following limits:

TABLE 1

Frequency range (Hz)	Overall loss relative to that at 800 Hz
Below 300	Not less than –2.0 dB, otherwise unspecified
300- 500	–2.0 to +4.0 dB
500-2800	–1.0 to +3.0 dB
2800-3000	–2.0 to +3.0 dB
3000-3250	–2.0 to +4.0 dB
3250-3350	–2.0 to +7.0 dB
Above 3350	Not less than –2.0 dB, otherwise unspecified

Random noise

The mean psophometric noise power referred to a point of zero relative level should not exceed 32 000 pW0p (–45 dBm0p), using a psophometer in accordance with Recommendation P.53.

Impulsive noise

The number of counts of impulsive noise which exceed –28 dBm0 should not exceed 18 in 15 minutes when measured with an impulsive noise counter in accordance with Recommendation H.13, h).

SECTION 3

TELEPHONE CIRCUITS OR CABLES USED FOR VARIOUS TYPES OF TELEGRAPH TRANSMISSION OR FOR SIMULTANEOUS TRANSMISSIONS

Recommendation H.32 [1)]

SIMULTANEOUS COMMUNICATION BY TELEPHONY AND TELEGRAPHY ON A TELEPHONY CIRCUIT

The CCITT,

considering

a) that the exceptional use of a leased telephone circuit for simultaneous communication by telephone and telegraph is envisaged in Recommendation D.1, 1.7.

b) that the CCITT has indicated conditions under which the simultaneous use of telephone circuits for telephony and telegraphy is technically tolerable;

c) that standardization of the characteristics of apparatus permitting simultaneous use of a telephone circuit for telephony and telegraphy is not justified, but that it is necessary to limit the power of the signals transmitted and to avoid the use of frequencies that will interfere with any telephone signalling equipment that may remain connected to the telephone circuit;

d) that new demands for the allocation of particular frequencies for special purposes frequently arise and the number of frequencies used for any one purpose should not be unnecessarily extended;

e) that the systems described below may be useful when the more modern systems advocated in Recommendation H.34 are not feasible,

unanimously recommends

1. that in the case of the simultaneous use of a telephone circuit for telephony and telegraphy, the telegraph signal, if continuously transmitted, should be maintained at or below a level of -13 dBm0;

2. that there should not be more than three telephone circuits of this type in a group of 12 telephone circuits and that the number of circuits of this type set up on a wideband carrier system should not exceed the number of supergroups in the system;

3. that the telegraph signals transmitted must not interfere with any signalling apparatus that may remain connected to the telephone circuit,

---

[1)] Recommendation H.31, Volume III, *Green Book* has been deleted.

and notes

that some Administrations have permitted the use, for simultaneous telephony and telegraphy, of the frequencies 1680 Hz and 1860 Hz both for amplitude and for frequency modulation.

Note. – If circuits equipped in accordance with the present Recommendation are used in a private network, it will be impossible to use push-button telephone sets or multifrequency signalling (e.g. Signalling System R2) in the network.

Recommendation H.34 [2)]

SUBDIVISION OF THE FREQUENCY BAND OF A TELEPHONE-TYPE CIRCUIT BETWEEN TELEGRAPHY AND OTHER SERVICES

(Geneva, 1972)

1. *General*

The specific case considered here is that of frequency subdivision at 2700 Hz of a 4-wire circuit into a main band (which can be used for telephony, data, phototelegraphy or facsimile transmission) and a secondary band, above the main band, reserved for frequency-modulation (FM) telegraphy.

The solution described in this Recommendation is recommended when the equipments are supplied by the Administration and also in cases where the circuit established on the main channel can be connected to the public telephone network. It should be pointed out that, in accordance with Recommendation D.1, 6.8, Administrations are not obliged to guarantee the quality of transmission of calls sent to, or received from, users on the public network over a leased circuit.

It is understood that any other system may be used on a leased circuit, provided the conditions concerning levels set forth in 5. below are observed; in this case, Administrations can give no guarantee concerning the quality of circuits, even between the users of a leased circuit.

2. *Main channel*

With the upper part limited in this way, the main channel can be used for:

a) telephone calls of a reduced quality plus an appropriate signalling system;

b) data transmission in accordance with Recommendation V.21, V.23 or V.30 with a return channel;

c) phototelegraph transmission in accordance with the normal conditions described in Recommendation T.1 (60 rpm with amplitude modulation; frequency modulation in any case is not advisable on circuits in land cables; amplitude modulation on submarine cables is still under study);

d) black-and-white facsimile transmission in accordance with Recommendation T.2 (120 rpm only, amplitude modulation or frequency modulation).

For services *b), c)* and *d)* above, the filter should be designed to keep the group delay distortion within tolerable limits for these services; the level condition stated in 5. below must also be complied with at all times.

With regard to service *a)*, where applicable, account should be taken of the telephony impairment (about 2 dB) due to the limitation of the frequency band (see Recommendation G.113).

[2)] Recommendation H.33, Volume III, *Green Book* has been deleted.

3. *Telegraph channels*

The following are the preferred arrangements of telegraph channels in the secondary band in the case of a normal 300-3400 Hz telephone-type circuit:

1) four 120-Hz spaced channels (Nos. 121, 122, 123, and 124);
2) two 120-Hz channels and one 240 Hz channel (Nos. 123, 124 and 211);
3) two 240-Hz channels (Nos. 211 and 212);
4) one 480-Hz channel (No. 406).

The numbering, modulation and other characteristics of the telegraph channels should comply with Recommendations R.35, R.37, R.38 A and R.70 *bis*, as far as possible, considering the reduced transmission level which may result in sub-standard performance.

Where the upper limit is reduced to 3050 Hz (as in telephone channels complying with Recommendation G.235) it will only be possible to use two 120-Hz channels (Nos. 121 and 122) or one 240-Hz channel (No. 211) with the recommended frequency subdivision.

With the same subdivision, the main channel may be used for:

- telephone calls,
- facsimile (including phototelegraphy),

and the secondary channel for:

- data transmission by telegraph channel up to 200 bauds (100 bauds on submarine cables).

However, any private system may be used, depending on the characteristics of the portion of the band available.

4. *Filters*

For the protection of the telegraph channels from interference by speech components in the upper frequency range a filter must be used at the sending end. The recommended filter-attenuation characteristic is defined by the limits shown in Figure 1/H.34.

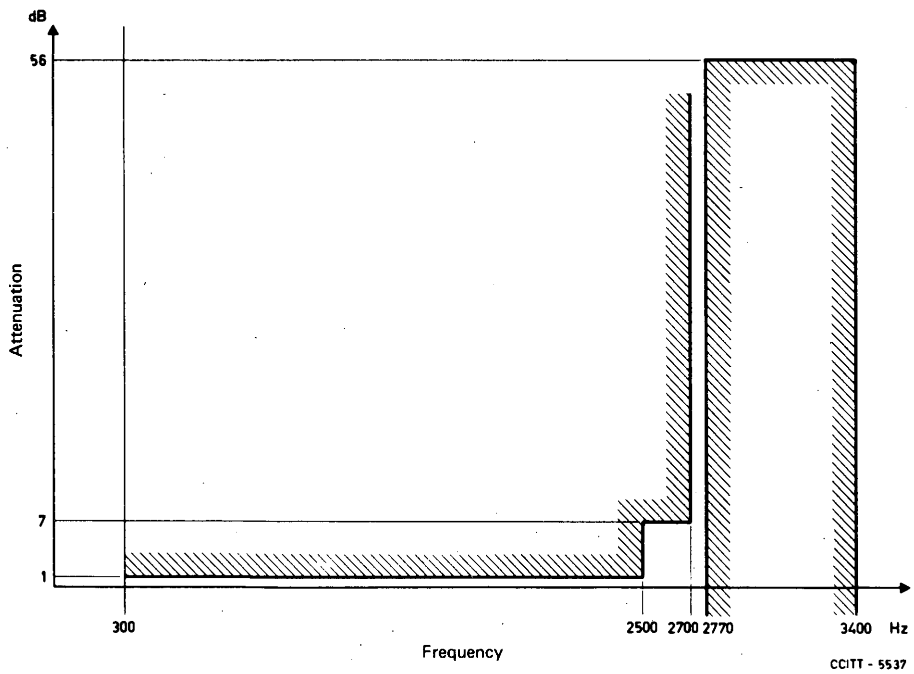


FIGURE 1/H.34 – **Limits for the attenuation characteristic of the separation filter at 2700 Hz**

Note. – This filter protects the telegraph channels from the signals transmitted on the main channel. The filters mentioned in Recommendation R.35 for protection in the opposite direction can be relied on; when the secondary channel is used for other purposes, special precautions should be taken to protect the main channel.

When non-speech transmission is required in the main band, the peak-to-peak delay distortion of this filter should not exceed 0.15 ms over the range 1100 to 2300 Hz. The maximum absolute delay in the range 300-2300 Hz should not exceed 3 ms.

Sufficient protection of the main band from interference by the telegraph signals in the secondary band is assured by a similar filter at the receiving end. It is assumed that the telegraph channels are provided with filters so as to meet the provisions of Recommendations R.35, R.37 or R.38 A.

5. *Levels*

The general principle concerning levels for each type of service is that the allowable mean signal power is proportional to the bandwidth assigned.

Of the maximum permissible 1-minute mean loading of 50 μW0 (– 13 dBm0), 10 μW0 are allocated to the secondary band and the remaining 40 μW0 to the main band. In the case of telephony this implies that normal levels for speech and signalling can be retained (given in Recommendation G.223 as 32 μW).

6. *Limitation of amplitude*

It may be desirable to impose a limit in the main band path so that the onset of nonlinearity in the common transmission path will not cause intermodulation and possible interference with the telegraph channels.

SECTION 4

TELEPHONE-TYPE CIRCUITS USED FOR FACSIMILE TELEGRAPHY

Recommendation H.41

PHOTOTELEGRAPH TRANSMISSIONS ON TELEPHONE-TYPE CIRCUITS

Note. – As far as carrier circuits are concerned, this Recommendation applies only to systems established on the basis of 12-channel groups; systems using 16-channel groups will form the subject of a later study.

Audio-frequency telephone circuits and carrier circuits can be used for phototelegraphy.

When normal audio-frequency circuits or carrier circuits are used, amplitude modulation offers some advantages over frequency modulation [1] and is therefore to be preferred for phototelegraph transmissions on circuits set up from end to end on cable or line-of-sight radio-relay links [2].

However, in the case of circuits subject to sudden level variations or to noise, frequency modulation may be preferable to amplitude modulation; Administrations could in this case come to an agreement to use frequency modulation for phototelegraph calls over such circuits; the provisions of Recommendation T.1 relative to the frequency-modulation characteristics should then be applied.

For these reasons, the CCITT

unanimously declares the view

that phototelegraph transmissions over telephone circuits require that the following conditions be observed, according to the way in which the circuits are used for phototelegraphy:

A. CIRCUITS PERMANENTLY USED FOR PHOTOTELEGRAPHY

It seems that these circuits are few. In any case, they should even more easily meet the characteristics given in B. below.

[1] In particular, with the same index of cooperation and speed, frequency modulation necessitates a wider frequency range than that of amplitude modulation to obtain a picture of the same quality.

[2] See Recommendation T.15 for phototelegraph transmissions over combined radio and wire circuits.

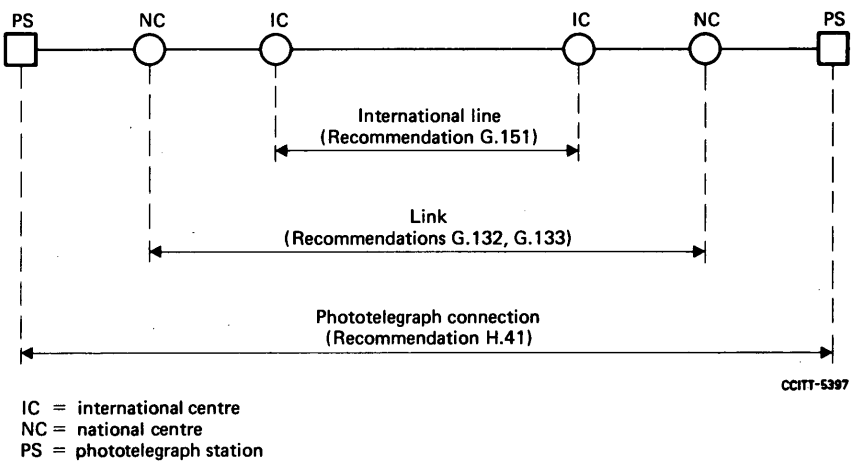
B. CIRCUITS USED NORMALLY (AND PREFERENTIALLY) FOR PHOTOTELEGRAPHY

a) *Types of circuit to be used*

Two-wire circuits have no practical value for phototelegraphy because of feedback phenomena.

For the same reason, 4-wire circuits should be extended to the phototelegraph stations on a 4-wire basis at the appropriate amplifier stations, the terminating units and echo suppressors always being disconnected.

The constitution of a phototelegraph connection is given in Figure 1/H.41.



Note. – The connection is not set up on a chain of switched circuits but on lines according to the terminology used by Study Group IV.

FIGURE 1/H.41 – Constitution of a phototelegraph connection

b) *Overall loss*

The same conditions apply to the overall transmission loss of 4-wire circuits used for phototelegraphy as apply, in general, for telephony.

c) *Sent signal power*

The emission voltage for the phototelegraph signal corresponding to maximum amplitude should be so adjusted that the power level of the signal is 0 dBm0 for amplitude-modulation phototelegraph transmission and –10 dBm0 for frequency-modulation phototelegraph transmissions. In the case of amplitude modulation, the level of the signal corresponding to black is usually about 30 dB lower than that of the signal corresponding to white.

d) *Relative levels*

If phototelegraph transmissions take place simultaneously from a transmitting station to several receiving stations, arrangements shall be made at the junction point so that, on the circuits following the junction point, the same power levels are maintained as those prescribed for individual transmissions.

e) *Attenuation/frequency distortion*

The limits for attenuation/frequency distortion on international circuits used for phototelegraphy are given in Recommendation G.151 concerning telephone circuits. The attenuation/frequency distortion between two terminal national centres shall therefore not exceed the limits indicated in Recommendation G.132 and it will not normally be necessary to compensate the distortion of the lines linking the phototelegraph stations to the terminal national centres in order to obtain, for amplitude-modulated phototelegraph transmission, an attenuation/frequency distortion between phototelegraph stations of less than 8.7 dB in the wanted band.

f) *Variation of circuit overall loss with time* [3), 4)]

1. The objective is that:

1.1 the difference between the mean value and the nominal of the transmission loss value should not exceed 0.5 dB;

1.2 the standard deviation from the mean value should not exceed 1 dB.

However, in the case of circuits set up wholly or partly on older equipment, where the international line consists of two or more circuit sections, a standard deviation not exceeding 1.5 dB may be accepted.

2. The method for achieving the above objective values is left to the discretion of Administrations (better maintenance, fitting of automatic regulators, etc.).

3. The assumption is made that these limits for the variation of loss with time of a single circuit may be compared to limits for loss measurements made on a set of circuits at a given time. Experience indicates that such a comparison has a practical validity although it has not been fully demonstrated at this time. Administrations are encouraged to use this Recommendation as giving currently practical limits for sets of circuits. This does not preclude the application of these limits to single circuits, should this prove practical at any time.

g) *phase distortion*

Phase distortion limits the range of satisfactory phototelegraph transmissions. Differences between the group delays of a telephone circuit, in the interval of the phototelegraph transmission, should not exceed

$$\Delta t \leqslant \frac{1}{2f_p}$$

f_p = maximum modulating frequency corresponding to the definition and scanning speed. (See Recommendation H.42.)

h) *Interference*

Interfering currents, whatever their nature, should not exceed the CCITT recommended limits for telephone circuits.

---

3) See Recommendation M.160 and Supplement No. 1.6 of Volume IV.2 of the *Green Book*.

4) The provisions specified under f) are provisional and need further study from the facsimile transmission point of view.

C. TELEPHONE CIRCUITS RARELY USED FOR PHOTOTELEGRAPHY

a) *Transmission characteristics*

It seems that the majority of the characteristics specified by the CCITT for modern telephone circuits are sufficient to permit phototelegraph transmissions on a circuit chosen at random in a group of circuits normally used for telephone working. However, it is not certain that such a circuit would have a sufficiently low phase distortion for such use, particularly channels 1 and 12 of a 12-circuit group, use of which is not advised. The influence of phase distortion is more noticeable in frequency modulation.

With amplitude modulation there is a further risk that phototelegraph transmissions will be subject to faulty modulation because the special precautions applied to circuits regularly used for phototelegraphy [see B.f) above] cannot be applied to circuits taken at random.

b) *Precautions concerning signalling*

As long as automatic switching for phototelegraph circuits is not envisaged, the signal receiver can be disconnected so that no signalling disturbances can occur even when frequency modulation is used. However, if frequency modulation is used for phototelegraph transmission and if it is impracticable to disconnect the signal receiver, then it would be desirable, in the case of the single-frequency system that a blocking signal be transmitted along with the picture signal to operate the guard circuit and render the receiver inoperative.

It is also apparent that the frequency of such a blocking signal should lie well outside the range of frequencies involved in the picture transmission and the frequency and the level of the blocking signal must depend on the characteristics of the VF receiver (or receivers in the case of a tandem international connection), as designed by different Administrations to meet the specification to be prescribed for international signalling.

In the case of the two-frequency international signalling system, the CCITT has indicated its view that no interference will take place.

Recommendation H.42

RANGE OF PHOTOTELEGRAPH TRANSMISSIONS ON A TELEPHONE-TYPE CIRCUIT

Note. – In case of carrier circuits, this Recommendation applies only to systems established on the basis of 12-channel group links. Systems using 16-channel group links will be the subject of subsequent study.

1. The differences between the group delays of the various frequencies and the width of the transmission band actually usable on a circuit for telephony give rise, when phototelegraph signals are started or stopped, to transient phenomena which limit the phototelegraph transmission speed.

2. The range of phototelegraph calls of satisfactory quality, for a given transmission speed, depends especially on the constitution of the circuit, i.e. on:

- the loading and length, in the case of audio-frequency circuits;
- the number of 12-channel group links used in the case of carrier circuits, and on the choice of the carrier frequency for amplitude-modulated photograph transmission, or on the mean frequency in the case of frequency modulation.

3. Phototelegraph transmission of satisfactory quality requires that the limits of difference between the group delays in the transmitted frequency band, as shown in Figure 1/H.42, are not to be exceeded.

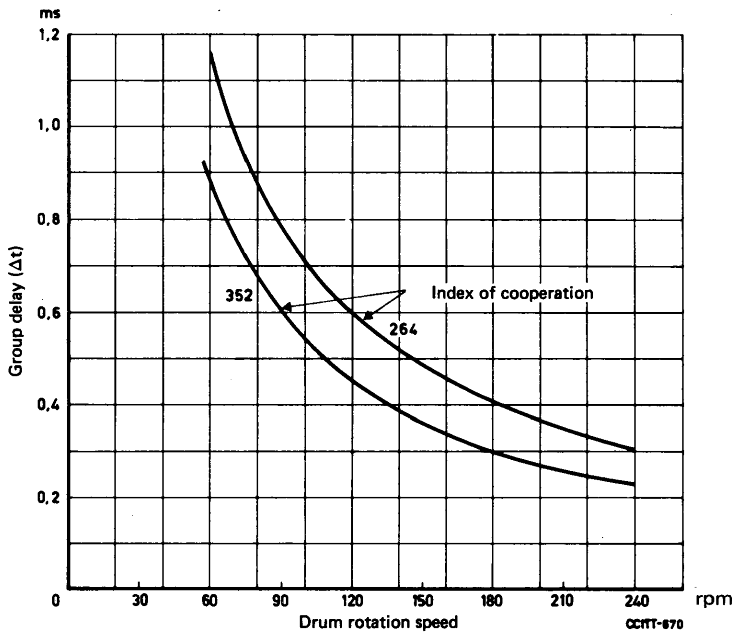


FIGURE 1/H.42 – Permissible group-delay distortion in the transmitted frequency band as a function of the phototelegraph transmission speed

Note. – The spot is assumed to have the same dimensions in both directions (square or circular).

4. The CCITT has recommended group-delay distortion limits for international telephone circuits (see Recommendation G.133):

For these reasons, the CCITT

unanimously recommends

that, as regards the effect of phase distortion on phototelegraph transmission quality, the carrier frequency (where amplitude modulation is used) or the mean frequency (when frequency modulation is used) must be chosen in such a way that it is as near as possible to the frequency which has the minimum group delay on the telephone circuit.

A. CIRCUITS PERMANENTLY USED FOR PHOTOTELEGRAPHY

1. It will generally be possible, by agreement between Administrations, to choose a circuit satisfying stricter limits than those specified above from the point of view of phase distortion.

2. Moreover, it will be possible to compensate phase distortions by inserting phase equalizers and to effect phototelegraph transmissions occupying the whole nominal band of the circuit.

B. CIRCUITS NORMALLY (OR PREFERENTIALLY) USED FOR PHOTOTELEGRAPHY

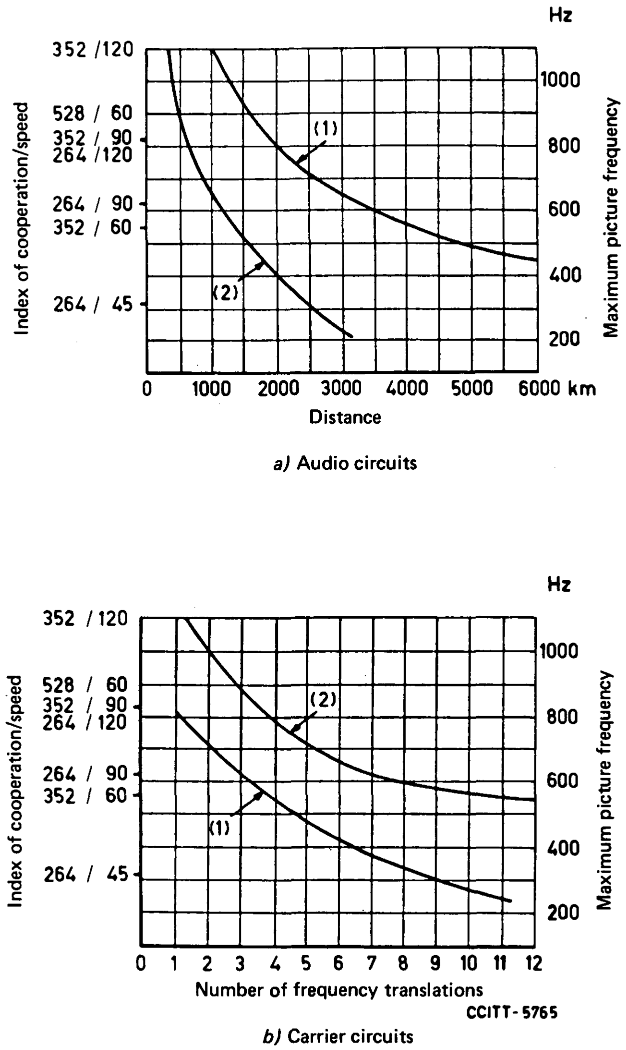
1. The greater the differences between the delays in the transmission intervals, the narrower should be the bandwidth chosen (leading to a lower phototelegraph definition or transmission speed).

2. Hence, audio-frequency circuits should in any case have only small loads.

3. Phase distortion is well within the limits indicated above, in the case of carrier circuits, if a single modern-type carrier system is considered (and considering especially the telephone channels in the middle of a 12-channel group of such a system).

4. Nevertheless, it would be unjustifiable from the financial point of view to make the aforementioned recommendation concerning phase distortion stricter, simply with a view to the occasional use of only a few circuits for high-speed phototelegraph transmissions.

5. The curves of Figure 2/H.42 give information on the relative performances of amplitude- and frequency-modulated phototelegraph transmissions over audio-frequency and carrier telephone circuits.



Curves (1) – AM carrier = 1300 Hz

Curves (2) – { FM = 1900 ± 400 Hz; AM carrier = 1900 Hz }

FIGURE 2/H.42 – Range of phototelegraph transmissions

C. TELEPHONE CIRCUITS RARELY USED FOR PHOTOTELEGRAPHY

If phototelegraph connections are set up on circuits selected at random from modern-type groups of telephone circuits (for example, by automatic switching), a circuit may be taken which has too high a degree of phase distortion, particularly if it has been set up on channel 1 or 12 of a 12-channel group, use of which is deprecated. It is impossible, in this case, to draw up general information on the range of phototelegraph transmissions; however, it will be possible to meet the conditions for a transmission of adequate quality if the phototelegraph connection comprises only one 12-channel group link and if transmission is effected in normal conditions as outlined in Recommendation T.1.

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 5

CHARACTERISTICS OF DATA SIGNALS

Recommendation H.51

DATA SIGNALS ON TELEPHONE-TYPE CIRCUITS

(Mar del Plata, 1968)

Use may be made either of circuits in the switched telephone network (see Recommendation H.11) or of leased telephone-type circuits of ordinary or special quality (see Recommendation H.12).

POWER LEVELS

The objectives in specifying data signal levels are as follows:

a) to ensure satisfactory transmission and to permit coordination 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). 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: −15 dBm0), probably has a standard deviation in the neighbourhood of 4 dB (see Supplement 5 to this Volume).

c) it is probable that Administrations will wish to fix specific values for the signal power level of data modulators either at the subscribers' 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)* above 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 CCITT

unanimously declares the following view:

A. DATA TRANSMISSION OVER LEASED TELEPHONE-TYPE CIRCUITS 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 −10 dBm0. When transmission of data is discontinued for any appreciable time, the power level should preferably be reduced to −20 dBm0 or lower.

3. For systems not transmitting tones continuously, for example amplitude-modulation systems, higher levels up to −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 μW (corresponding to a mean level of −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. See Recommendation G.224.

Note 1. — In suggesting these limits, the CCITT has in mind that the recommended maximum level of −5 dB referred to the zero relative level point for leased 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 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.41 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 transmission. 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 NETWORK

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 −10 dBm0 (simplex systems) or −13 dBm0 (duplex systems).

In systems not transmitting tones continuously, such as amplitude-modulation systems and multi-frequency 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 μW (representing a mean level of -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 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 a 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 4-wire subscribers' lines. If so, the levels to be used might be those proposed for leased circuits.

Recommendation H.52

TRANSMISSION OF WIDE-SPECTRUM SIGNALS (DATA, FACSIMILE, ETC.) ON WIDEBAND GROUP LINKS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1976)

Links meeting the provisions of Recommendation H.14 should be used.

a) *Power level*

1. The mean power level of the wideband signal over the range 60-108 kHz should not exceed $-15 + 10 \log_{10} 12 = -4$ dBm0.

2. In order to limit cross-modulation effects in wideband systems, the power level of any individual spectral component in the band 60-108 kHz should not exceed -10 dBm0 except for spectral components which are at multiples of 4 kHz, [see Recommendation G.221, b) 2].

With regard to its effect on non-telephone type signals, a discrete component is defined as a signal of sinusoidal form with a minimum duration of about 100 ms.

3. To protect the group or supergroup link pilots (used to establish wideband circuits) against other wide-spectrum signals (data, facsimile, etc.), it is recommended that the power spectrum emitted about the pilot frequency be limited in the equipment which transmits these signals (see Figure 1/H.52).

For continuous spectrum signals, the spectral density in the band $f_0 \pm 25$ Hz should not exceed -70 dBm0/Hz.

Other indications are given in Recommendation G.241, f).

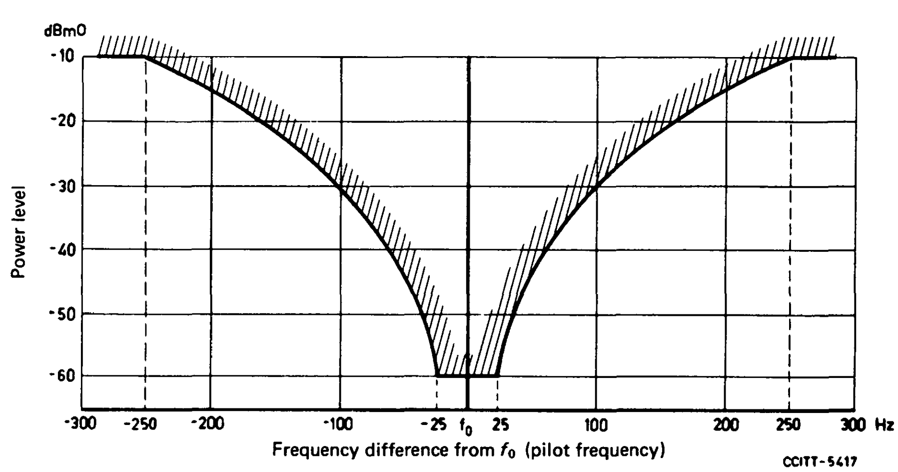


FIGURE 1/H.52 – Maximum permissible level of discrete frequency components of wide-spectrum signals (group and supergroup signals) in the vicinity of group and supergroup pilot frequencies

b) *Limitation of the power spectrum outside the band 60-108 kHz.*

The power level produced by the terminal equipment connected to the wideband group link shall not exceed −73 dBm0p in any 4-kHz band outside the range 60-108 kHz.

If the terminal equipment itself does not meet these conditions (e.g. a modem which just complies with the provisions of Recommendation V.35) an additional filter must be applied before the point of connection to the leased group circuit.

For some discrete frequencies (such as clock and carrier frequencies), with suitable frequency stability, a somewhat reduced requirement is permitted. This subject is under study.

Note. – An unweighted value of −50 dBm0 is proposed for the frequencies 48 and 56 kHz.

Recommendation H.53

TRANSMISSION OF WIDE-SPECTRUM SIGNALS (DATA, ETC.) OVER WIDEBAND SUPERGROUP LINKS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1976)

Links meeting the provisions of Recommendation H.15 should be used.

a) *Power level*

1. The mean power level of the wideband signal over the range 312-552 kHz should not exceed $-15 + 10 \log_{10} 60 = +3$ dBm0.

2. In order to limit cross-modulation effects in wideband systems, the power level of any individual spectral component in the band 312-552 kHz should not exceed −10 dBm0, except for spectral components which are at multiples of 4 kHz, [see Recommendation G.221, b) 2].

With regard to its effect on non-telephone type signals, a discrete component is defined as a signal of sinusoidal form with a minimum duration of about 100 ms.

3. In addition to 2. above, the energy spectrum transmitted in the neighbourhood of the pilot frequencies should be limited in accordance with Recommendation G.241, f).

b) *Limitation of the energy spectrum outside the band 312-552 kHz*

Under study.

PART III

Series J Recommendations

SOUND-PROGRAMME AND TELEVISION TRANSMISSIONS

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 1

GENERAL RECOMMENDATIONS CONCERNING SOUND-PROGRAMME TRANSMISSIONS

Recommendation J.11

HYPOTHETICAL REFERENCE CIRCUITS FOR SOUND-PROGRAMME TRANSMISSIONS

(Geneva, 1972; amended at Geneva, 1976)

Terrestrial systems and systems in the fixed-satellite service [1), 2), 3)]

The CCITT,

considering,

a) that there is a need to define a hypothetical reference circuit to enable design performance standards to be set;

b) that the hypothetical reference circuit should allow the different types of sound-programme circuits to be compared on a common basis;

recommends,

1. that the hypothetical reference circuit for sound-programme transmissions over a terrestrial system (shown in Figure 1/J.11) which may be provided by either radio or cable, should be characterized principally by:

- the overall length between audio points (B and C) of 2500 km;
- two intermediate audio points (M and M′) dividing the circuit into three sections of equal length;
- the fact that the three sections are lined up individually and then interconnected without any form of overall adjustment or correction;

2. that the hypothetical reference circuit for sound-programme transmissions over a system in the fixed-satellite service (shown in Figure 2/J.11) should be characterized principally by:

- one link, earth station – satellite – earth station,
- one pair of modulation and demodulation equipments for translation from baseband to radio frequency, and vice-versa.

[1)] Corresponding to CCIR Recommendation 502 (Rev. 76)

[2)] The hypothetical reference circuits defined in this Recommendation are based on analogue systems. The application to digital systems is for study under draft Study Programme 10A-1/CMTT (Rev. 76).

[3)] For maintenance purposes there may be a need to define other circuits of which an illustration is shown in the Annex.

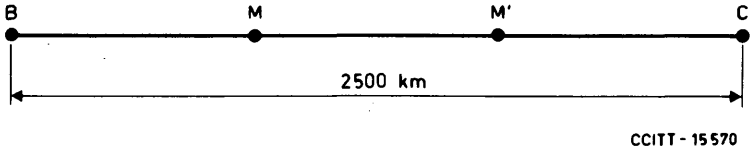


FIGURE 1/J.11 – **Hypothetical reference circuit for sound-programme transmissions over a terrestrial system**

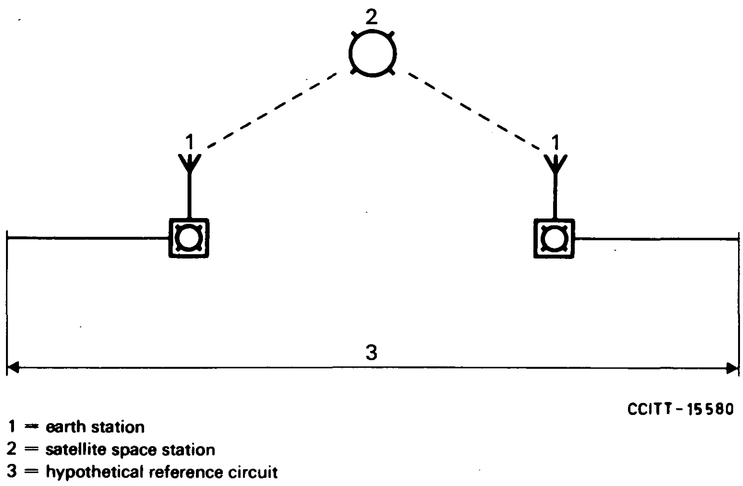


FIGURE 2/J.11 – **Hypothetical reference circuit for sound-programme transmissions over a system in the fixed-satellite service**

ANNEX

(to Recommendation J.11)

Illustration of an international sound-programme connection

Figure 1 illustrates a typical international sound-programme connection in which:

– point A, to be considered as the sending end of the international sound-programme connection, may be the point at which the programme originates (studio or outside location),

– point D, to be considered as the receiving end of the international sound-programme connection, may be a programme-mixing or recording centre or a broadcasting station,

– the local sound-programme circuit AB connects point A to the sending terminal station, point B, of the international sound-programme circuit BC,

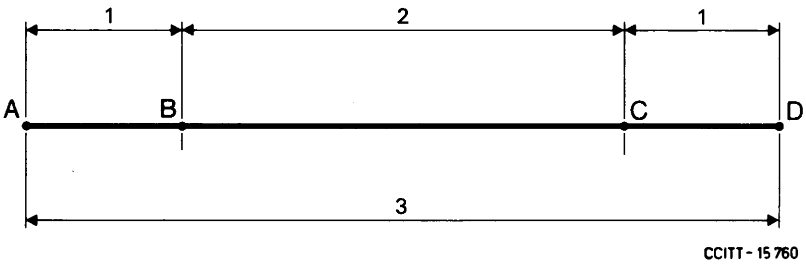
– the local sound-programme circuit CD connects point C, the receiving terminal station of the international sound-programme circuit BC, to the point D.

The hypothetical reference circuit must not be considered identical to any of the sound-programme circuits illustrated in Figures 1/J.11 and 2/J.11, or defined for maintenance purposes in Volume IV.1. However, some of these circuits may display the same structure as the hypothetical reference circuit. Such types of circuits are:

– an international sound-programme connection comprising three audio sections;

– a single sound-programme circuit made up of three audio sections.

In this case, the performance standards set for the hypothetical reference circuit may be applied to these circuits.



1 = local sound-programme circuit
2 = international sound-programme circuit
3 = international sound-programme connection

FIGURE 1 – **An international sound-programme connection**

Recommendation J.12

TYPES OF SOUND-PROGRAMME CIRCUITS ESTABLISHED OVER THE INTERNATIONAL TELEPHONE NETWORK

(former Recommendation J.11; amended at Geneva, 1972)

The CCITT recognizes the types of sound-programme circuits defined below.

Note. – For the purposes of this Recommendation and other Recommendations in the J Series, sound-programme circuits have been classified in terms of the nominal effectively transmitted bandwidth. For convenience, the corresponding type of circuit from the administrative point of view (see Recommendation E.330, Volume II.2) is given under each type of equipment in the following paragraphs.

1. *15-kHz sound-programme circuit*

This type of circuit is recommended for high-quality monophonic programme transmission and in certain arrangements is also recommended for stereophonic transmissions. This type of circuit corresponds to the "very wideband circuit" or "stereophonic pair", as appropriate, referred to in Recommendation E.330.

The performance characteristics of 15-kHz type sound-programme circuits suitable for both monophonic and stereophonic transmissions are defined in Recommendation J.21 and suitable methods of provision are given in Recommendation J.31.

2. *10-kHz type sound-programme circuit*

This type of circuit, previously known as the "normal programme circuit, type A", is recommended for monophonic transmission only. Originally regarded as suitable for high-quality transmissions, it may continue to be used for many years to come to provide a good quality of sound-programme transmission. This type of circuit corresponds to the "wideband circuit" referred to in Recommendation E.330. The performance characteristics of 10-kHz type sound-programme circuits are defined in Recommendation J.22 and suitable methods of provision are given in Recommendation J.32.

3. *6.4-kHz type sound-programme circuit*

This type of circuit was previously known as "normal programme circuit, type B" and is now recommended where the saving of bandwidth is of great importance, since this type of programme circuit when set up over carrier systems displaces only two telephone channels. A 6.4-kHz type sound-programme circuit provides a lower standard of programme transmission than those given in 1 and 2 above but is adequate for many purposes. This type of circuit falls within the category of "medium-band circuits" referred to in Recommendation E.330.

The performance characteristics of 6.4-kHz type sound-programme circuits are defined in Recommendation J.23, and methods or provision are given in Recommendation J.33.

Note 1. – A Question is under study (Question 3/XV) about the possibility of recommending a 5-kHz type sound-programme circuit, which would also fall within the category of "medium-band circuits" referred to in Recommendation E.330.

Note 2. – "Old-type programme circuits" are no longer the subject of CCITT Recommendations. Their general characteristics were described in former Recommendation J.41 and their methods of provision in former Recommendation J.42 (*White Book*, Volume III).

Note 3. – A technical recommendation on the use of telephone circuits for sound-programme transmission will be drawn up when there is an operating recommendation specifying the conditions under which such use is permissible.

Recommendation J.13

DEFINITIONS AND RESPONSIBILITIES FOR INTERNATIONAL SOUND-PROGRAMME CIRCUITS

(former Recommendation J.12; amended at Geneva, 1972)

a) *Definition of the constituent parts of an international sound-programme connection*

The following definitions apply to international sound-programme transmissions.

1. **international sound-programme transmission**

The transmission of sound over the international telecommunication network for the purpose of interchanging sound-programme material between broadcasting organizations in different countries. Such a transmission includes all types of programme material normally transmitted by a sound broadcasting service, for example, speech, music, sound accompanying a television programme, etc.

2. **broadcasting organization (send)**

The broadcasting organization at the sending end of the sound programme being transmitted over the international sound-programme connection.

3. **broadcasting organization (receive)**

The broadcasting organization at the receiving end of the sound programme being transmitted over the international sound-programme connection.

4. **international sound-programme centre (ISPC)**

A centre at which at least one international sound-programme circuit terminates and in which international sound-programme connections can be made by the interconnection of international and national sound-programme circuits.

The ISPC is responsible for setting up and maintaining international sound-programme links and for the supervision of the transmissions made on them.

5. **international sound-programme connection**

5.1 The unidirectional path between the broadcasting organization (send) and the broadcasting organization (receive) comprising the international sound-programme link extended at its two ends over national sound-programme circuits to the broadcasting organizations (see Figure 2/J.13).

5.2 The assembly of the "international sound-programme link" and the national circuits between the broadcasting organizations, constitutes the "international sound-programme connection". Figure 3/J.13 illustrates, by way of example, an international sound-programme connection as it might be encountered in practice.

6. **international sound-programme link** (Figure 2/J.13)

The unidirectional path for sound-programme transmissions between the ISPCs of the two terminal countries involved in an international sound-programme transmission. The international sound-programme link comprises one or more international sound-programme circuits interconnected at intermediate ISPCs. It can also include national sound-programme circuits in transit countries.

7. **international sound-programme circuit**(Figure 1/J.13)

The unidirectional transmission path between two ISPCs and comprising one or more sound-programme circuit sections (national or international), together with any necessary audio equipment (amplifiers, compandors, etc.).

8. **sound-programme circuit-section** (Figure 1/J.13)

Part of an international sound-programme circuit between two stations at which the programme is transmitted at audio frequencies.

The normal method of providing a sound-programme circuit-section in the international network will be by the use of carrier sound-programme equipment. Exceptionally sound-programme circuit-sections will be provided by other means, for example, by using amplified unloaded or lightly loaded screened-pair cables or by using the phantoms of symmetric-pair carrier cables.

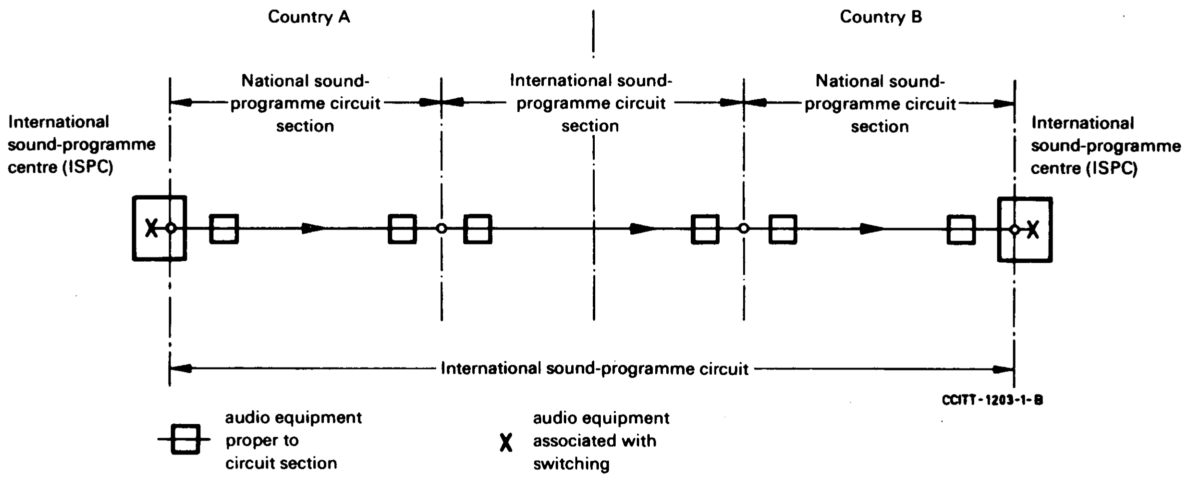


FIGURE 1/J.13 – **An international sound-programme circuit composed of two national and one international sound-programme circuit-section**

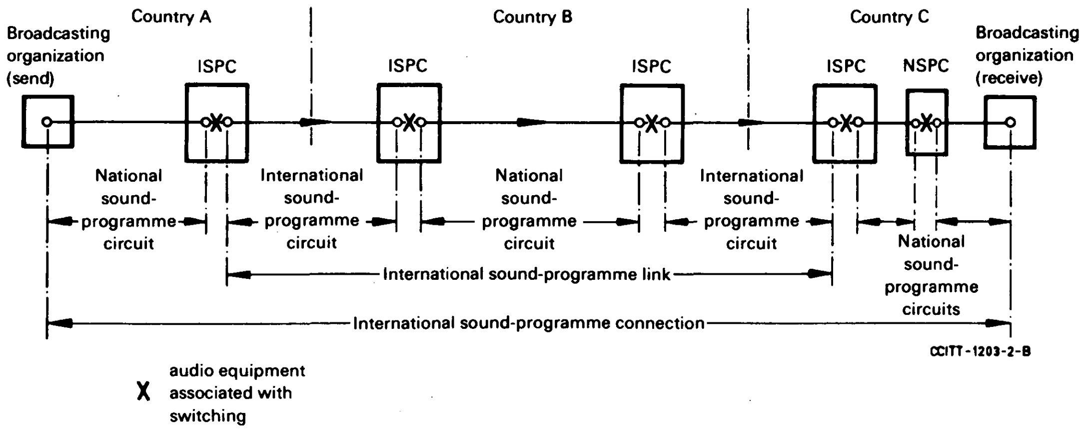
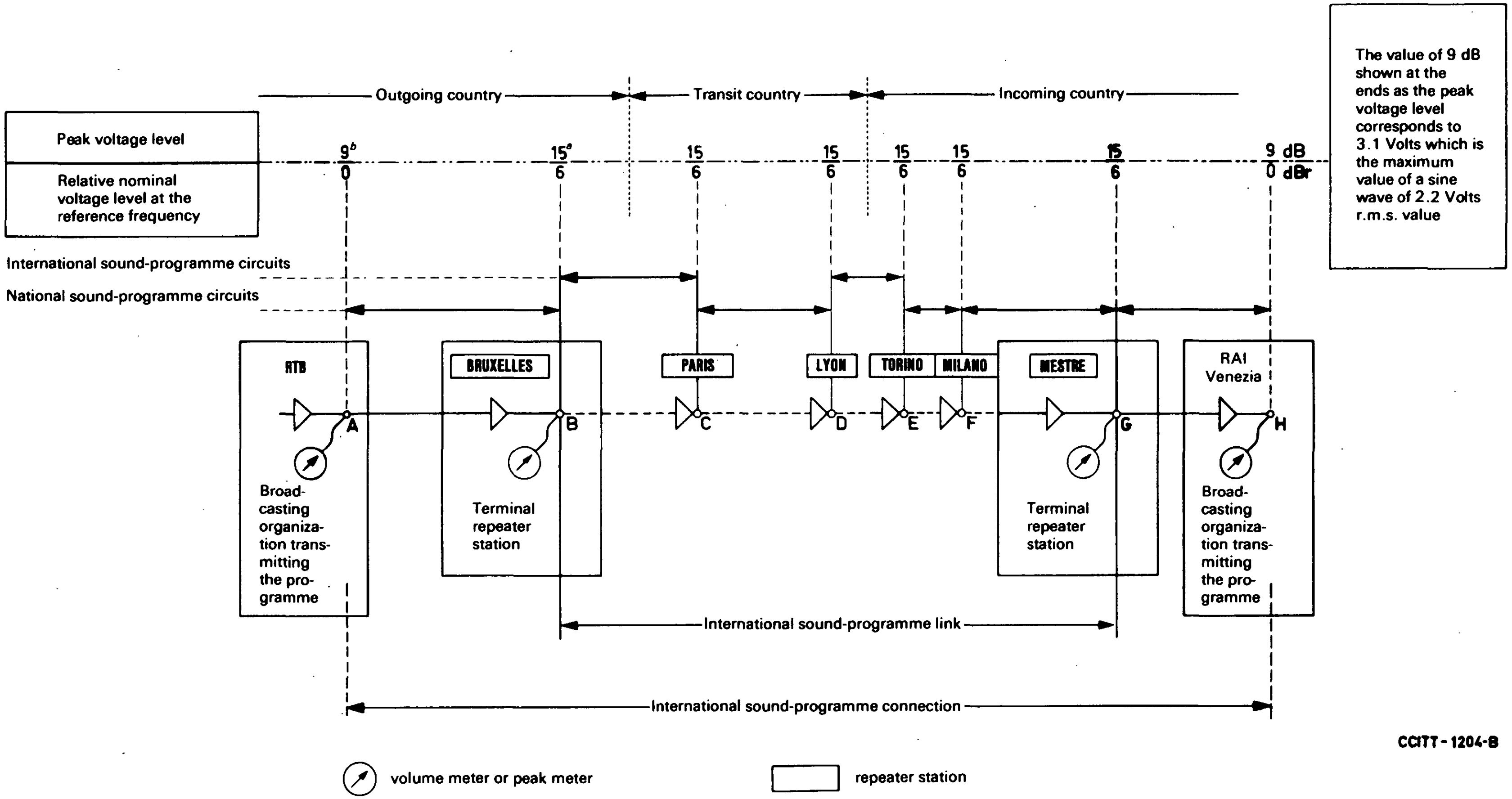


FIGURE 2/J.13 – **An international sound-programme link composed of international and national sound-programme circuits and extended on a national sound-programme circuit at each end to form an international sound-programme connection**



[a] These levels are appropriate to national sound-programme circuits in the countries concerned.

[b] The RTB in fact adopts levels of 15/6 at this point.

FIGURE 3/J.13 – **Diagram of an international sound-programme connection in which all the countries concerned have chosen +6 dBr as the nominal relative level at points at which circuits are interconnected**

9. **national circuit**

The national circuit connects the ISPC to the broadcasting authority; this applies both at the sending and at the receiving end. A national circuit may also interconnect two ISPCs within the same country.

10. **effectively transmitted signals in sound-programme transmission**

For sound-programme transmission, a signal at a particular frequency is said to be effectively transmitted if the nominal overall loss at that frequency does not exceed the nominal overall loss at 800 Hz by more than 4.3 dB. This should not be confused with the analogous definition concerning telephony circuits given in Recommendation G.151, A., Note 1.

For sound-programme *circuits*, the overall loss (relative to that at 800 Hz) defining effectively transmitted frequency is 1.4 dB, i.e. about one-third of the allowance.

11. **origin and extremity of an international sound-programme circuit**

The "origin" of an international sound-programme circuit is considered as the output of the first amplifier and the "extremity" as the output of the last amplifier of the circuit. In the case of Figure 3/J.13, the direct circuit Bruxelles-Paris, for example, is comprised between the points B and C.

In the case of a circuit on a carrier system for programme transmissions, the origin of the circuit is the input of the modulating equipment and the extremity is the output of the demodulating equipment.

b) *Technical responsibilities during an international broadcast programme transmission*

The "international sound-programme link" is, in all cases, the sole responsibility of the telephone Administrations.

The extreme national circuits may be the responsibility of either the Administration, the broadcast organization or the two together, depending on the local arrangements in each particular country.

Recommendation J.14

RELATIVE LEVELS AND IMPEDANCES ON AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

(former Recommendation J.13; amended at Geneva, 1972 and 1976)

a) *Level adjustment on an international sound-programme connection*

The CCITT recommends the use of the *constant voltage* method. If, to a zero relative level point of the international sound-programme connection, a zero absolute voltage level is applied (sine-wave signal of 0.775 volts r.m.s.) at one of the frequencies given in the maintenance instructions of Recommendation N.12, Volume IV.1 the absolute voltage level at the output of the last amplifier of each sound-programme circuit (Points B, C, D ... G of Figure 3/J.13) should be +6 dB (i.e. 1.55 volts r.m.s.) at 800 Hz, and should be within the given limits at other frequencies (see the appropriate Recommendations J.21, J.22, etc., for the limits applicable to the particular type of sound-programme circuit under consideration).

The zero relative level point is, in principle, the origin of the international sound-programme connection (Point A in Figure 3/J.13). Different conventions may be agreed between the telephone Administration and the broadcast organization within a country, provided that the levels on the international sound-programme link are unchanged.

A zero relative level point is, in principle, a point at which the sound-programme signals correspond exactly with those at the origin of the international sound-programme signals have been controlled in level by the broadcasting organization, such that the peak levels very rarely exceed +9 dB relative to the peak values reached by a sine-wave signal of 0.775 volts r.m.s. (for a 600-ohm resistor load, when levels are expressed in terms of dBm).

b) *Diagram of signal levels on an international sound-programme connection*

The signal levels given below are expressed in terms of r.m.s. values of sine-wave signals with reference to 0.775 volts.

The voltage level diagram for an international sound-programme connection, however made up, should be such that the voltage levels shown are not such as to exceed the maximum undistorted power which an amplifier can deliver to a sound-programme link when a peak voltage (i.e. +9 dB) is applied to a zero relative level point on the international sound-programme connection.

With these conditions, +6 dB is the nominal voltage level at the output of the terminal amplifiers of the sound-programme circuits making up the international sound-programme link (Points B, C, D . . . G of Figure 3/J.13).

The line amplifiers of the international sound-programme link should be capable of handling an upper voltage limit of at least +17 dB, when using the constant voltage method for lining-up.

From the statement above, that the voltage level at a zero relative level point can reach +9 dB, the relative level obtained at the output of an amplifier would be $+17 - 9 = +8$ dB. Assuming the maximum variation of this level with time to be ± 2 dB, a nominal relative level at the output of these amplifiers of $+8 - 2 = +6$ dB is obtained.

If a sound-programme circuit which is part of the international sound-programme link is set up on a group in a carrier system, the objective for a new design of equipment is that the relative level of the sound-programme circuit, with respect to the relative level of the telephone channel, should be chosen such that the mean value and the peak value of the load presented by the sound-programme channel should be no higher than that of the telephone channels which are replaced by the sound-programme channel. The effects of pre-emphasis and compandors should, where present, be taken into consideration.

It is recognized that this condition may not be observed in all cases, particularly in certain existing types of equipments. It is recommended that in those cases the zero relative level points of the sound-programme circuit and of the telephone channels should coincide.

It might be as well, however, if the equipment could, where possible, tolerate a maximum difference of ± 3 dB between the relative levels of the sound-programme and telephone transmissions, so that the best adjustment can be obtained, depending on any noise or intermodulation present, but at the same time observing the constraints imposed by the considerations on loading.

Note. – The relative level at which the modulated sound-programme signal is applied to the group link is given in Recommendation J.31 for 15-kHz type circuits, and in the Annex to Recommendation J.22 for 10-kHz and 6.4-kHz type circuits when pre-emphasis is employed.

c) *Definitions and abbreviations for new sound-programme signals*

Definitions and symbols are in current use to define relative levels for telephony. However, additional definitions and symbols are necessary for the absolute and relative levels in respect of sound-programme signals. The corresponding definitions and symbols for telephony and sound-programme signals are given below:

– the absolute (power) level, in decibels, referred to a point of zero relative level . . . . dBm0 [4)]

– the relative (power) level, in decibels . . . . dBr [4)]

[4)] These symbols traditionally relate to telephony relative levels.

– the absolute (power) level, in decibels, referred to a point of zero relative sound-programme level dBm0s
– the relative (power) level, in decibels, with respect to sound-programme signals. (This abbreviation is only applicable at points in a sound-programme circuit where the signals can nominally be related to the input by a simple scaling factor.) . . . . dBrs

Examples of the use of these symbols are given below.

Examples of the use of symbols for levels of sound-programme signals

As an example, Figure 1/J.14 illustrates three different test conditions applied to equipment in conformance with Recommendation J.31, which includes pre-emphasis and compandors. It also shows how random noise, originating on the line system, can be expressed in relation to the sound-programme circuit under quiescent conditions.

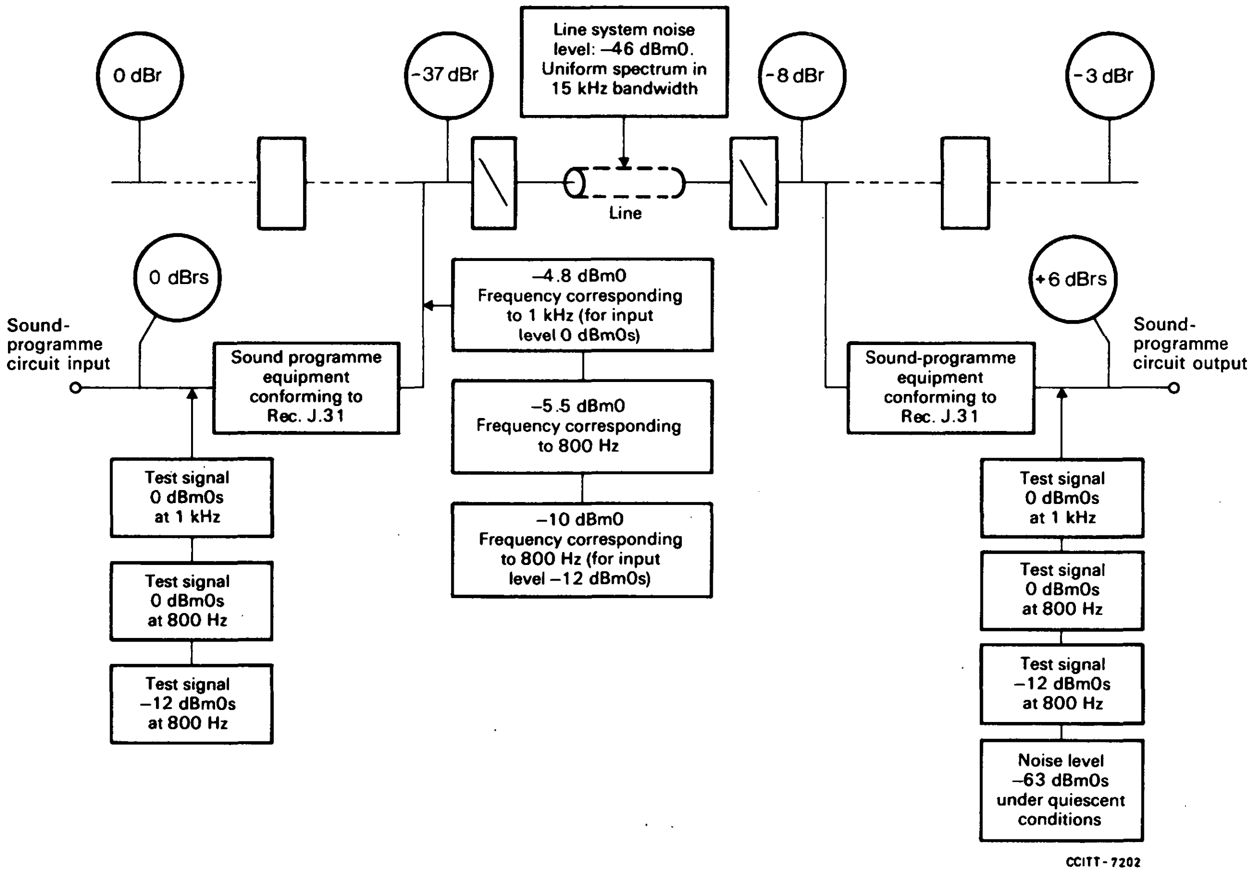


FIGURE 1/J.14 – Example illustrating the use of the symbols dBm0, dBr, dBrs, dBm0s for a sound-programme transmission circuit set up on a group link using equipment conforming to Recommendation J.31

Recommendation J.15

LINING-UP AND MONITORING AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

(former Recommendation J.14; amended at Geneva, 1972)

To comply with the provisions of Recommendation J.14, the lining-up and monitoring of an international sound-programme connection should ensure that, during the programme transmission, the peak voltage at a zero relative level point will not exceed 3.1 volts, which is that of a sinusoidal signal having an r.m.s. value of 2.2 volts. The methods for achieving this condition as well as the relevant performance requirements are given in Recommendations N.10 to N.18, Volume IV.1.

Some indication of the volume or of the peaks of the signals during programme transmission may be obtained by monitoring at the studio, in the repeater stations, or at the transmitter. One of the instruments, the characteristics of which are summarized in Table 1/J.15, may be used.

Since there is no simple relation between the readings given by two different instruments for all types of programme transmitted, it is desirable that the broadcast organization controlling the studio and the telephone Administration(s) controlling the sound-programme circuit should use the same type of instrument so that their observations are made on a similar basis.

In general the telephone Administration and the broadcast organization of a country agree to use the same type of instrument. It is desirable to reduce to a minimum the number of different types of instrument and to discourage the introduction of new types which only differ in detail from those already in service.

During programme transmission, the signal level at the output of the last amplifier controlled by the sending broadcast organization (Point A of Figure 3/J.13) should be monitored to see that the meter deflection of the measuring instrument is always lower than the peak voltage for the overall line-up, allowance being made for the peak factor of the programme involved.

It should be remembered that the amplitude range from a symphony orchestra is of the order of 60 to 70 dB, while the specification for sound-programme circuits is based on a range of about 40 dB. Before being passed to the sound-programme circuit, therefore, the dynamic ratio of the studio output needs to be compressed.

TABLE 1/J.15 – **Principal characteristics of the various instruments used for monitoring the volume or peaks during telephone conversations or sound-programme transmission**

Type of instrument	Rectifier characteristic (see Note 3)	Time to reach 99% of final reading (milliseconds)	Integration time (milliseconds) (see Note 4)	Time to return to zero (value and definition)
(1) "Speech voltmeter" UK Post Office type 3 (S.V.3) identical to the speech power meter of the ARAEN	2	230	100 (approx.)	equal to the integration time
(2) vu meter (United States of America) (see Note 1)	1.0 to 1.4	300	165 (approx.)	equal to the integration time
(3) Speech power meter of the "SFERT volume indicator"	2	around 400 to 650	200	equal to the integration time
(4) Peak indicator for sound-programme transmissions used by the British Broadcasting Corporation (BBC Peak Programme Meter) (see Note 2)	1		10 (see Note 5)	3 seconds for the pointer to fall 26 dB
(5) Maximum amplitude indicator used by the Federal Republic of Germany (type U 21)	1	around 80	5 (approx.)	1 or 2 seconds from 100% to 10% of the reading in the steady state
(6) OIRT – Programme level meter: type A sound meter; type B sound meter		for both types: less than 300 ms for meters with pointer indication and less than 150 ms for meters with light indication	type A: 10 ± 5; type B: 60 ± 10	for both types: 1.5 to 2 seconds from the 0 dB point which is at 30% of the length of the operational section of the scale

Note 1. – In France a meter similar to the one defined in line (2) of the table has been standardized.

Note 2. – In the Netherlands a meter (type NRU-ON301) similar to the one defined in line (4) of the table has been standardized.

Note 3. – The number given in the column is the index n in the formula $V_{(output)} = [V_{(input)}]^n$ applicable for each half-cycle.

Note 4. – The "integration time" was defined by the CCIF as the "minimum period during which a sinusoidal voltage should be applied to the instrument for the pointer to reach to within 0.2 neper or nearly 2 dB of the deflection which would be obtained if the voltage were applied indefinitely". A logarithmic ratio of 2 dB corresponds to 79.5% and a ratio of 0.2 neper to 82%.

Note 5. – The figure of 4 milliseconds that appeared in previous editions was actually the time taken to reach 80% of the final reading with a d.c. step applied to the rectifying integrating circuit. In a new and somewhat different design of this programme meter using transistors, the performance on programme remains substantially the same as that of earlier versions and so does the response to an arbitrary, quasi-d.c. test signal, but the integration time, as defined in Note 4, is about 20% greater at the higher meter readings.

Note 6. – In Italy a programme meter with the following characteristics is in use:

Rectifier characteristic: 1 (see Note 3)
Time to reach 99% of final reading: approx. 20 ms
Integration time: approx. 1.5 ms
Time to return to zero: approx. 1.5 s from 100% to 10% of the reading in the steady state.

Recommendation J.16

MEASUREMENT OF WEIGHTED NOISE IN SOUND-PROGRAMME CIRCUITS

(Geneva, 1972; amended at Geneva, 1976)

The noise objectives for sound-programme circuits are defined in terms of psophometrically weighted noise power levels at a zero relative level point. Psophometric weighting is used to ensure that the objectives and the results of measurements are directly related to the disturbing effect of the noise to the human ear. The psophometric weighting for sound-programme circuits consists of two operations:

– a frequency-dependent weighting of the noise signal, and

– a weighting of the time function of the noise signal to take account of the disturbing effect of noise peaks.

To achieve results which are comparable, it is recommended that for the measurement of noise in sound-programme circuits a measuring set should be used which conforms to the characteristics laid down in CCIR Recommendation 468-1, which is reproduced at the end of this Recommendation.

The following Annex gives symbols and definitions used in noise measurements.

ANNEX

(to CCITT Recommendation J.16)

Symbols and definitions used in noise measurements

A clear distinction should be made between measurements performed with equipment conforming to Recommendation P.53, B., *Green Book*, Volume V and those with equipment conforming to CCIR Recommendation 468-1.

It is recommended that the definitions and symbols in Table 1 be used.

TABLE 1 – **Definitions and symbols for the specification of noise measured on sound-programme circuits**

Definitions	Symbols
Unweighted noise power level, measured with an r.m.s. instrument and referred to a point of zero relative sound-programme level	dBm0s
Weighted noise power level, measured with an r.m.s. instrument and weighting characteristic complying with Recommendation P.53,B., *Green Book*, Volume V, and referred to a point of zero relative sound-programme level	dBm0ps
Unweighted noise level, measured with a quasi-peak measuring instrument complying with CCIR Recommendation 468-1 and referred to a point of zero relative sound-programme level	dBq0s
Weighted noise level, measured with a quasi-peak measuring instrument complying with CCIR Recommendation 468-1 and referred to a point of zero relative sound-programme level	dBq0ps

For noise measurements on sound-programme circuits, it is clear that the terms dBm0ps and dBm0s will ultimately fall into disuse, but the term dBm0s will continue to have a useful meaning with regard to signal power levels.

For specifications of sound-programme circuits, only the symbols shown in Table 1 are needed. However, for other purposes such as investigations of prototypes or bibliographic references, it might be more desirable to use r.m.s. values with the weighting curve of CCIR Recommendation 468-1 or quasi-peak values according to the weighting curve of Recommendation P.53, B, *Green Book*, Volume V. To this purpose the symbols dBm0ps (up to 15 kHz) and dBq0ps (up to 10 kHz) can be used.

CCIR RECOMMENDATION 468-1

MEASUREMENT OF AUDIO-FREQUENCY NOISE IN BROADCASTING, IN SOUND-RECORDING SYSTEMS AND ON SOUND-PROGRAMME CIRCUITS *

(1970 - 1974)

The CCIR,

unanimously recommends

that for the measurement of audio-frequency noise in broadcasting, in sound-recording systems and on sound-programme circuits, to obtain results in good agreement with subjective assessments **, the method of measurement defined below should be used:

1. *Weighting network*

The nominal response curve of the weighting network is defined together with the theoretical response of the passive network in Figure 1. Table I gives the values of this response at various frequencies.

The permissible differences between the response curve of measuring networks and this nominal curve are shown in the last column of Table I and in Figure 2.

Note 1. – It is considered unnecessary to use a different network for 10 kHz circuits and the recommended network would probably be suitable for narrower-band circuits.

Note 2. – If in measuring the signal-to-noise ratio of some equipments it is decided that the reference signal at 1000 Hz should pass through the weighting network, a stricter tolerance (± 0.2 dB) should be observed at that frequency.

* This Recommendation is also of interest to the CMTT.

** If, for technical reasons, it is desirable to measure unweighted noise, the method described in the Annex should be used.

TABLE I

Frequency (Hz)	Response (dB)	Proposed tolerance (dB)
31.5	−29.9	±2.0
63	−23.9	±1.4 [1]
100	−19.8	±1.0
200	−13.8	±0.85 [1]
400	− 7.8	±0.7 [1]
800	− 1.9	±0.55 [1]
1 000	0	±0.5
2 000	+ 5.6	±0.5 [1]
3 150	+ 9.0	±0.5 [1]
4 000	+10.5	±0.5 [1]
5 000	+11.7	±0.5
6 300	+12.2	0
7 100	+12.0	±0.2 [1]
8 000	+11.4	±0.4 [1]
9 000	+10.1	±0.6 [1]
10 000	+ 8.1	±0.8 [1]
12 500	0	±1.2 [1]
14 000	− 5.3	±1.4 [1]
16 000	−11.7	±1.65 [1]
20 000	−22.2	±2.0
31 500	−42.7	$\begin{cases} +2.8 \text{ [1]} \\ -\infty \end{cases}$

[1] This tolerance is obtained by a linear interpolation on a logarithmic graph on the basis of values specified for the frequencies used to define the mask, i.e., 31.5, 100, 1000, 5000, 6300 and 20 000 Hz.

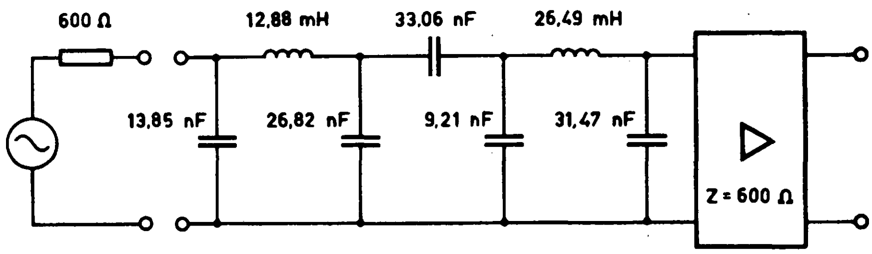


FIGURE 1*a* – **Weighting network**

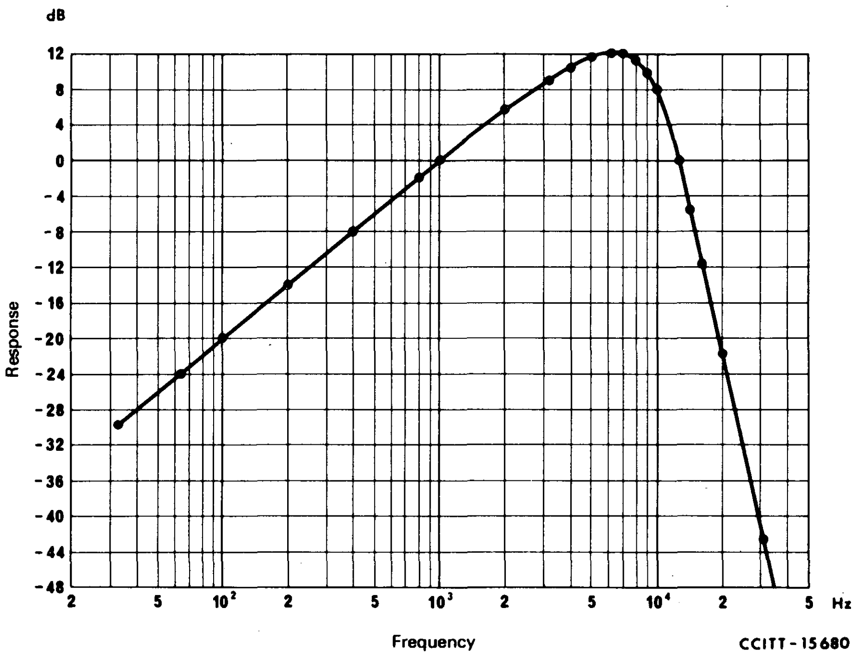


FIGURE 1*b* – **Frequency response of the weighting network shown in Fig. 1*a***

(A tolerance of at most 1% on the component values and a Q-factor of at least 200 at 10 000 Hz are sufficient to meet the tolerances given in Table I)

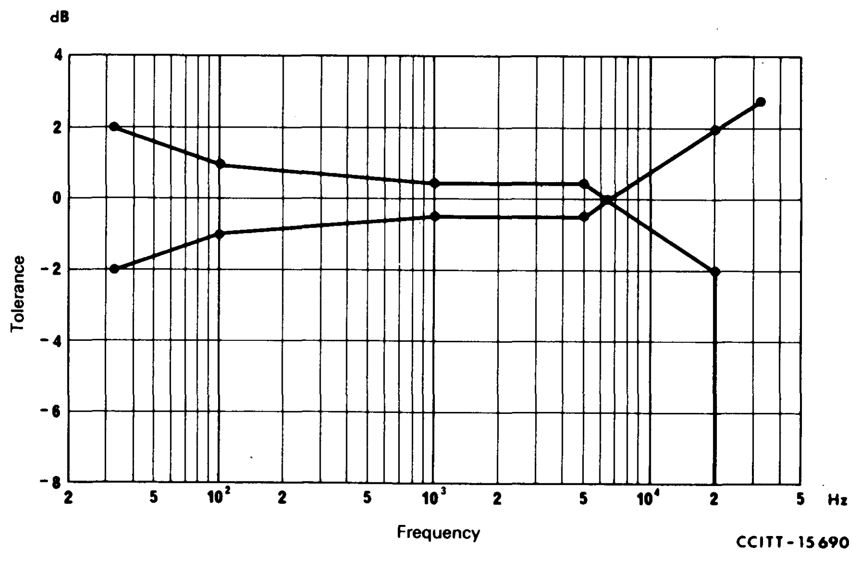


FIGURE 2 – **Maximum tolerances for the frequency response of the weighting network**

2. *Characteristics of the measuring device*

A quasi-peak value method of measurement should be used. It is defined by the time-response characteristic of the measuring set, as described in Table II.

TABLE II

Burst duration (ms) [1]	10	20	50	100	200
Percentage of steady signal reading (%)	48 (−6.4 dB)	52 (−5.7 dB)	59 (−4.6 dB)	68 (−3.3 dB)	80 (−1.9 dB)
Limiting values (%):					
– lower limit	41 (−7.7 dB)	44 (−7.1 dB)	50 (−6.0 dB)	58 (−4.7 dB)	68 (−3.3 dB)
– upper limit	55 (−5.2 dB)	60 (−4.4 dB)	68 (−3.3 dB)	78 (−2.2 dB)	92 (−0.7 dB)

[1] Method of measurement: a single burst of a 5000 Hz sine wave of a known duration is applied to the input of the instrument. The amplitude of the burst should correspond to about two-thirds of the reading scale. Table II shows, as a function of their durations, the limit values between which the reading shall fall; these values are expressed in percentages of the reading obtained for a steady signal of the same amplitude.

The dynamic properties of the device are further defined by the following procedure:

– a series of bursts of a 5000 Hz sine wave is applied to the input of the instrument, with a repetition frequency of 10 Hz and a burst duration of 5 ms. The amplitude of the burst should correspond to approximately two-thirds of the reading scale. The reading should reach 70% to 90% of the value corresponding to a steady signal of the same amplitude. This applies to all measurement ranges.

– the overload capacity of the whole chain preceding the reading device should be at least 20 dB in relation to the maximum indication of the scale. This condition applies to all ranges of sensitivity.

Note. – The Administration of the United Kingdom is proposing to use different dynamic characteristics for the measuring device [see Doc. 10/175 (United Kingdom), 1970-1974 and Report 398-2].

ANNEX

(to CCIR Recommendation 468-1)

Unweighted measurements

1. *Frequency response*

The frequency characteristic should lie between the following limits:

Below	31.5 Hz	: minimum slope: 12 dB/octave
Between	31.5 Hz and 50 Hz	: +0.5 dB to −3 dB
	50 Hz and 12.5 Hz	: +0.5 dB to −0.5 dB
	12.5 kHz and 16 kHz	: +0.5 dB to −3 dB
Above	16 kHz	: minimum slope: 18 dB/octave

2. *Detection*

Two kinds of measurement may be used:

– measurement of the r.m.s. value,
– measurement of the quasi-peak value.

The kind of measurement used should be stated when giving the result.

Recommendation J.17

PRE-EMPHASIS USED ON SOUND-PROGRAMME CIRCUITS IN GROUP LINKS

(Geneva, 1972)

The noise spectrum in group links is usually uniformly distributed, i.e. all parts of the frequency band are equally disturbed by the noise signal. Sound-programme signals, on the other hand, are not of uniform distribution. The mean power density of the signals tends to decrease towards higher frequencies. Furthermore, the sensitivity of the receiving part (consisting of the radio receiver, the loudspeaker and the human ear) in respect of noise is very dependent on the frequency. (This can be seen from the psophometric weighting curve which is a measure of the sensitivity of the complete receiving part.)

Taking these three facts together it appears to be advantageous to use pre-emphasis on sound-programme circuits set up on carrier systems.

The advantages which could be gained by using different pre-emphasis curves are rather small. It is recommended, therefore, that a single pre-emphasis curve should be used whenever pre-emphasis is applied to sound-programme circuits in group links.

It is futher recommended that the pre-emphasis attenuation curve should be that given by the following formula:

$$\text{Insertion loss between nominal impedances} = 10 \log_{10} \frac{75 + \left(\frac{\omega}{3000}\right)^2}{1 + \left(\frac{\omega}{3000}\right)^2} \text{ (dB)}$$

where ω is the angular frequency corresponding to frequency f.

The de-emphasis network should have a complementary curve.

The pre-emphasis curve calculated from this formula passes through the following points:

The measured pre-emphasis and de-emphasis curves should not depart by more than ± 0.25 dB from the theoretical curves when the measured levels at 800 Hz are made to coincide with the theoretical levels.

Note. – The formula given above defines only the "insertion-loss/frequency" characteristic. The level at which the modulated programme signal is given to the group link is different for the various types of sound-programme equipments and it depends on the modulation method and the type of compandors used. This information is given in the appropriate Recommendations (J.22, J.31).

f(kHz)	Insertion loss (dB)
0	18.75
0.05	18.70
0.2	18.06
0.4	16.48
0.8	13.10
2	6.98
4	3.10
6.4	1.49
8	1.01
10	0.68
∞	0

Recommendation J.18

CROSSTALK IN SOUND-PROGRAMME CIRCUITS SET UP ON CARRIER SYSTEMS

(Geneva, 1972)

This Recommendation outlines the principles followed by the CCITT in determining what limits are appropriately set for sources of crosstalk affecting sound-programme circuits and other principles which Administrations might apply to ensure that the objectives for intelligible crosstalk in sound-programme circuits are achieved in practice.

1. The causes of crosstalk arising in the transmission parts of telecommunications networks occur in:

a) frequency translating equipments at all levels, viz. audio, group, supergroup, and higher order translating equipments;

b) group, supergroup, etc., through-connection equipments (i.e. filter characteristics);

c) transmission systems, both the line (including repeater) and station equipments.

Different crosstalk mechanisms, e.g. inductive, capacitive and other couplings, intermodulation involving continuous fixed-frequency tones such as pilots, etc., operate in these equipments and systems. A particular channel may thus be disturbed by intelligible crosstalk from a number of potential disturbing sources.

However, because of the interconnections which occur at distribution points along the length of a sound-programme circuit, the same disturbing and disturbed signals are rarely involved in more than one exposure.

2. Only the more important crosstalk mechanisms are the subject of Recommendations (e.g. coaxial and balanced pair cable repeater section FEXT limits, Part I, Section 3); the limits are such that at least the objectives for intelligible crosstalk ratio between *telephone* circuits (58 dB, Recommendation G.151) may be met. In some cases it is practicable to take into account the more stringent objectives for *sound-programme* circuits (Recommendations J.21, J.22 and J.23). Certain crosstalk mechanisms, because they are not significant for telephony (e.g. near-end crosstalk limits for cable repeater sections), are not the subject of Recommendations; nevertheless, they may be significant in relation to sound-programme circuit objectives.

In principle, a probability of exposure can be attributed to each source of crosstalk, not all potential sources exerting their influence in every case. Given the respective probabilities and distributions, the risk of encountering low values of crosstalk attenuation could be calculated.

Without carrying out this analysis it is estimated that the risk of encountering adverse systematic addition for some sources is small and the allocation of the complete overall objective to a single source of crosstalk as the minimum value of crosstalk attenuation appears justifiable. For other sources, particularly where the equipments involved are specifically intended for sound-programme transmission, it is appropriate to require some higher minimum attenuation values so as to allow for some adverse addition (Recommendation G.242 specifying through-connection filter discrimination requirements against out-of-band components in the band occupied by sound-programme circuits is an example).

3. For these reasons meeting intelligible crosstalk objectives on sound-programme circuits in practice depends on:

a) reasonable care in the allocation of plant for sound-programme circuits, so that the principal crosstalk mechanisms, a single exposure to any of which may itself suffice to exceed the objective, are avoided.

Among these mechanisms are:

- far-end and near-end crosstalk at certain frequency bands in line-repeater sections (e.g. the lowest and highest frequency bands of coaxial systems);
- systematic addition of near-end crosstalk between go and return channels of a group link;

b) readiness to change allocated plant in the few cases where crosstalk is excessive because of systematic addition of two or more disturbing sources.

4. The CCITT limits agreed for crosstalk ratios between bands potentially occupied by sound-programme circuits are in terms of effects at single frequencies. The following factors need to be taken into account when assessing from such limits the probability of encountering intelligible crosstalk into real sound-programme circuits:

a) no method of assessing the subjective effects of intelligible crosstalk in the bands occupied by sound-programme circuits have as yet been standardized;

b) the intelligibility of crosstalk can be affected by:

- the use of emphasis in the disturbed circuit;
- noise masking effects;
- modulation arrangements (e.g. double sideband) in the disturbed circuit;
- frequency offsets and inversions;
- the use of compandors;

c) the mechanisms most liable to cause excessive intelligible crosstalk are, in general, highly frequency-dependent. These cases are those readily prevented by selective plant allocation advocated in 3. above;

d) crosstalk attenuation can, as a rule, be characterized by a mean value and a standard deviation; the mean value is usually several decibels higher than the worst value, which occurs with only a very small probability.

5. *Go-return crosstalk*

The assumptions made in the course of the CCITT study of go-return crosstalk in sound-programme circuits, and which served as the basis for the crosstalk limits prescribed in respect of group and higher-order translation equipments (Recommendation G.232, k) are given in the following:

a) the nominal maximum distance of the exposure to go-return crosstalk of two sound-programme circuits occupying opposite directions of the same group link is 560 km, i.e. 2/9 of the hypothetical reference circuit distance;

b) the equipments assumed to contribute to such go-return crosstalk are:

- 560 km of line;
- one pair of channel translations;
- one pair of group translations;
- three pairs of higher-order translations;
- two through connections.

The corresponding calculation is given in the Annex.

It was considered that the contribution of the line to go-return crosstalk can be limited to the range of values indicated in the Annex, given that precautions outlined in 3. above are exercised.

It is possible that, in the study of new transmission systems, the CCITT will be able to take such account of sound-programme circuit crosstalk objectives that these precautions may be relaxed somewhat. This study is in progress in the CCITT with respect to 60 MHz systems.

ANNEX

(to Recommendation J.18)

Calculations of overall go-return crosstalk between two sound-programme circuits occupying opposite directions of the same group link

Equipment	Crosstalk ratio limit (dB)	Crosstalk power per exposure in the disturbed circuit arising from a signal of 0 dBm0 on the disturbing circuit (pW)	Number of exposures	Total crosstalk power (pW)	Crosstalk ratio (dB)
Line	80 to 85 (single homogeneous section)	10 to 3	2 (2/9 h.r.c)	20 to 6	77 to 82
Channel translation	85	3	2	6	82
Group translation	80	10	2	20	77
Supergroup and higher translations	85	3	6	18	77.5
Through filters (cabling)	85	3	2	6	82
Totals (without compandors)				70 to 56	71.5 to 72.5
Totals (with programme-circuit compandors with a minimum companding advantage of 10 dB)				7 to 6	81.5 to 82.5

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 2

PERFORMANCE CHARACTERISTICS OF SOUND-PROGRAMME CIRCUITS

Recommendation J.21

PERFORMANCE CHARACTERISTICS OF 15-kHz TYPE SOUND-PROGRAMME CIRCUITS [1]

(Geneva, 1972; amended at Geneva, 1976)

Circuits for high-quality monophonic and stereophonic transmissions

The CCITT

unanimously recommends

that, taking account of the definition in 1. below, high-quality monophonic and stereophonic sound-programme transmissions should satisfy the requirements laid down in 2. and 3. below.

1. *Definition*

When the hypothetical reference circuit defined in Recommendation J.11 is composed of three "sound-programme carrier sections" the requirements indicated below should be met.

2. *Requirements at audio interconnection points*

2.1 *Measurement of characteristics*

When making measurements of the characteristics of a circuit, these should be made with the output terminated with a 600 ohm non-reactive load.

2.2 *Impedance and matching conditions*

The audio-frequency input impedance should be 600 ohms balanced; the tolerance on this value is a matter for further study.

It is provisionally recommended that the output impedance be balanced with respect to earth and be so low that the output level in the nominal transmission range does not decrease by more than 0.3 dB if the open-circuit output is loaded with 600 ohms. This output impedance is intended for connection to a nominal load impedance of 600 ohms.

[1] This Recommendation corresponds to CCIR Recommendation 505.

This clause alone would, however, not rule out a large difference in the reactive parts of the output impedances of a stereophonic pair, and this in turn could lead to difficulties in meeting the limits of 3.2.2 below. This aspect needs further study.

For amplifiers which are intended for direct connection to audio frequency sound-programme lines, the reactive part of the output impedance should be restricted. A maximum value of 100 ohms for the series reactance part of the output impedance at frequencies in the transmitted range is provisionally recommended.

2.3 *Relative level*

The relative level on a sound-programme circuit at the audio-frequency amplifier output should be fixed at +6 dBrs [2)].

3. *Performance of the hypothetical reference circuit for 15-kHz type sound-programme circuits*

The values given below correspond to circuits operating with analogue techniques and are expected to be met on such transmission systems. Special additional parameters concerning digital techniques are under study (see 4. below).

3.1 *Parameters for monophonic sound-programme transmission*

3.1.1 Nominal bandwidth: 0.04 to 15 kHz.

3.1.2 Insertion gain at 0.8 or 1 kHz: this parameter should be measured at a sending level equivalent to − 12 dBm0 as specified by the CCITT for setting up sound-programme circuits.

3.1.2.1 Adjustment error: not to fall outside the range ± 0.5 dB.

3.1.2.2 Variation during 24 hours: not to exceed ± 0.5 dB.

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organizations to insert additional trimming attenuators.

3.1.3 The gain/frequency response referred to 0.8 or 1 kHz should comply with the following limits:

0.04 to 0.125	kHz:	+0.5 to −2.0 dB
0.125 to 10	kHz:	+0.5 to −0.5 dB
10 to 14	kHz:	+0.5 to −2.0 dB
14 to 15	kHz:	+0.5 to −3.0 dB

For the combined effect of three modulator and demodulator equipments, a tolerance of ± 0.5 dB from 0.125 to 10 kHz is considered the closest that can be met by equipments in practice. If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

The response should be measured using a test level of − 12 dBm0.

3.1.4 The difference between group delay at the given frequency and the minimum value of group delay should not exceed the following limits:

0.04 kHz:	55 ms
0.075 kHz:	24 ms
14 kHz:	8 ms
15 kHz:	12 ms

3.1.5 *Maximum noise power level*

The intention for the future is to standardize the use of the quasi-peak noise-measuring instrument specified in CCIR Recommendation 468-1 for weighted and unweighted measurements. This instrument differs from the programme psophometer specified by the CCITT in its Recommendation P.35, B., Volume V, *Green Book*, in respect of the weighting network and the detector part. For convenience, approximately comparable limits are given in Table 1/J.21. All values are referred to a point of zero relative level.

2) See the definition of zero relative level point in Recommendation J.14.

TABLE 1/J.21

Type of measurement	R.m.s value	Quasi-peak value [a]
Weighted noise		
– new network (CCIR Recommendation 468-1)	–47 dBm0ps (15 kHz)	–42 dBq0ps
– old network (CCITT Recommendation P.53)	–51 dBm0ps	–46 dBq0ps (10 kHz)
Unweighted noise	–41 dBm0s	–36 dBq0s

[a] The quasi-peak value includes the tolerance required for impulsive noise.

CCIR Report 493-1 indicates that if a compandor is used, an improved signal-to-noise ratio is necessary to avoid objectionable effects, for some programme material.

When using radio-relay systems the values given for both the weighted and unweighted noise should not be exceeded for more than 20% of any month. It seems acceptable for the limits to be exceeded by 4 dB and 12 dB for 1% and 0.1%, respectively, for any month.

3.1.6 The single-tone interference, measured selectively, should not exceed (–73 – Δps) dBm0s, in which Δps is the correction for the frequency being measured, given by the weighting characteristic in Recommendation 468-1.

In the case of sound-programme transmissions over carrier systems, occurrence of carrier leaks can be expected. For this reason, stop filters may be provided in the carrier-frequency path which can be switched in, if required, to suppress the tones otherwise audible in the upper frequency range from 8 to 15 kHz. For a hypothetical reference circuit, a 3-dB bandwidth of less than 3% for stop filters, referred to the midfrequency, is recommended. The use of stop filters influencing frequencies below 8 kHz should be avoided.

3.1.7 *Modulation interference from the power supply*

The highest-level unwanted side-component due to modulation of a sound-programme signal caused by interference from the a.c. line at the power supply output should not be greater than –45 dB, relative to the level of a sine-wave measuring signal applied to the sound-programme circuit (in accordance with CCIR Recommendation 474). The value for higher frequencies has to be determined, see CCIR Report 495-1 and Study Programme 5L/CMTT.

3.1.8 *Nonlinear distortion*

There are certain difficulties in giving a general recommendation on nonlinearity due to restrictions imposed by the CCITT on the levels and durations of test signals, see especially Recommendations N.21 and N.23, Volume IV.1. Pending progress with other test methods the following tests are recommended.

3.1.8.1 Harmonic distortion factors measured with single-tone test signals at +9 dBm0s should not exceed the following limits:

Frequency of test-tone (kHz)	Total harmonic distortion (%)	Second harmonic and third harmonic measured selectively (%)
0.04 to 0.125	1	0.7
0.125 to 7.5	0.5	0.35

The duration for which a single tone is to be transmitted at this level should be limited in accordance with the appropriate Recommendations of the N Series.

3.1.8.2 The difference tone factors [3] selectively measured by using two frequencies each at a level of +3 dBm0s should not exceed the following limits:

3.1.8.2.1 Frequencies 0.8 and 1.42 kHz corresponding to those prescribed in Recommendation O.31, Volume IV.2 for the 3rd-order difference tone, measured at 0.18 kHz: 0.5%

3.1.8.2.2 Frequencies 5.6 and 7.2 kHz for a 2nd-order difference tone, measured at 1.6 kHz: 0.5%

3.1.8.2.3 Frequencies 4.2 and 6.8 kHz for a 3rd-order difference tone, measured at 1.6 kHz: 0.5%

The measurements of 3.1.8.2.2 and 3.1.8.2.3 are intended for baseband transmission on physical circuits only and on modulation equipments in the local network.

3.1.8.3 Distortion products measured by shaped noise is under study. See CCIR Report 640.

3.1.9 Error in reconstituted frequency: not to be greater than 1 Hz.

3.1.10 *Intelligible crosstalk ratio*

The intelligible crosstalk ratio from other sound-programme circuits or from a telephone circuit into a sound-programme circuit should be measured selectively in the disturbed circuit at the same frequencies as that of the sinusoidal test-signal applied to the disturbing circuit, and should not be less than the following values:

0.04 kHz	: 50 dB
0.04 to 0.5 kHz	: straight-line segment with a linear scale in dB and a logarithmic scale in frequency.
0.5 to 5 kHz	: 74 dB
5 to 15 kHz	: straight-line segment with a linear scale in dB and a logarithmic scale in frequency.
15 kHz	: 60 dB.

3.1.11 *Error in amplitude/amplitude response*

When the level of a 0.8 or 1-kHz test signal is changed from +6 to −6 dBm0s or vice versa, the level difference at the receiving end should not lie outside the range 12 ± 0.5 dB. This level change of the test signal corresponds to that prescribed in Recommendation O.31, Volume IV.2.

3.2 *Additional parameters for stereophonic programme transmission*

3.2.1 The difference in gain between A and B channels should not exceed the following values:

0.04 to 0.125 kHz	1.5 dB
0.125 to 10 kHz	0.8 dB
10 to 14 kHz	1.5 dB
14 to 15 kHz	3 dB.

3.2.2 The phase difference between the A and B channels should not exceed the following values:

0.04 kHz	: 30°
0.04 to 0.2 kHz	: straight-line segment on a scale linear in degree and logarithmic frequency.
0.2 to 4 kHz	: 15°
4 to 14 kHz	: straight-line segment on a scale linear in degree and logarithmic in frequency.
14 kHz	: 30°
15 kHz	: 40°.

[3] Attention is drawn to the fact that in transmission systems using compandors, a 3rd-order difference tone may occur which exceeds the specified limit of 0.5%. This may occur when the difference between the two fundamental frequencies is less than 200 Hz. Thus, the components due to 3rd-order distortion will have frequencies which correspond to the difference between the two test frequencies. However, in these cases the subjective masking is such that a distortion up to 2% is acceptable.

3.2.3 The crosstalk ratio between the A and B channels should not be less than the following limits:

3.2.3.1 Intelligible crosstalk ratio, measured with sinusoidal test signal from 0.04 to 15 kHz: 50 dB.

3.2.3.2 Nonlinear crosstalk ratio measured with a sinusoidal test signal from 0.04 to 15 kHz: 60 dB.

4. *Transmission performance of the hypothetical reference circuit for 15-kHz type sound-programme circuits with particular reference to digital methods of transmission*

This section will deal with special additional parameters for digital systems. See Report 649 and Study Programme 10A-1/CMTT.

5. *Estimation of transmission performance of circuits shorter or longer than the hypothetical reference circuit*

See Study Programme 5K/CMTT.

Note. – For further work, CCIR Report 496-1 may be consulted. This Report also draws attention to certain differences between the above Recommendation and one drawn up by the OIRT.

Recommendation J.22

PERFORMANCE CHARACTERISTICS OF 10-kHz TYPE SOUND-PROGRAMME CIRCUITS [4]

(former Recommendation J.21; amended, Geneva, 1972 and 1976)

The CCITT

unanimously recommends

that, when the hypothetical reference circuit defined in Recommendation J.11 is assumed to be made of three 10-kHz type sound-programme circuit sections, the characteristics given below apply with the following reservations:

1) For an audio-frequency circuit, all the characteristics are valid, except for intelligible crosstalk.

2) For a circuit on a carrier system, all the characteristics are valid, except for intelligible crosstalk and noise. (See the Annex to this Recommendation.)

a) *Requirements at audio frequency interconnection points*

1. *Measurement of characteristics*

When making measurements of the characteristics of a circuit, this should be made with the output terminated with a 600 ohm non-reactive load.

2. *Impedance and matching conditions*

The audio input impedance should be 600 ohms balanced; the tolerance on this value is a matter for further study.

[4] This Recommendation corresponds to CCIR Recommendation 504.

It is provisionally recommended that the output impedance be balanced with respect to earth and be so low that the output level in the nominal transmission bandwidth does not decrease by more than 0.3 dB if the open-circuit output is loaded with 600 ohms. This output impedance is intended for connection to a nominal load impedance of 600 ohms.

For amplifiers which are intended for direct connection to sound-programme lines, the reactive part of the output impedance should be restricted. A maximum value of 100 ohms for the series reactance part of the output impedance at frequencies in the transmitted bandwidth is provisionally recommended.

3. *Relative level*

The relative level of a sound-programme circuit at the audio-frequency amplifier output should be fixed at +6 dBrs (see Recommendation J.14).

b) *Nominal bandwidth*

The nominal bandwidth is 0.05 to 10 kHz.

c) *Attenuation distortion*

Figure 1/J.22 shows the permissible limits for the variation of the received level with frequency (relative to the value measured at 800 Hz). The method of measuring this level variation with frequency is shown in Recommendation N.21, Volume IV.1.

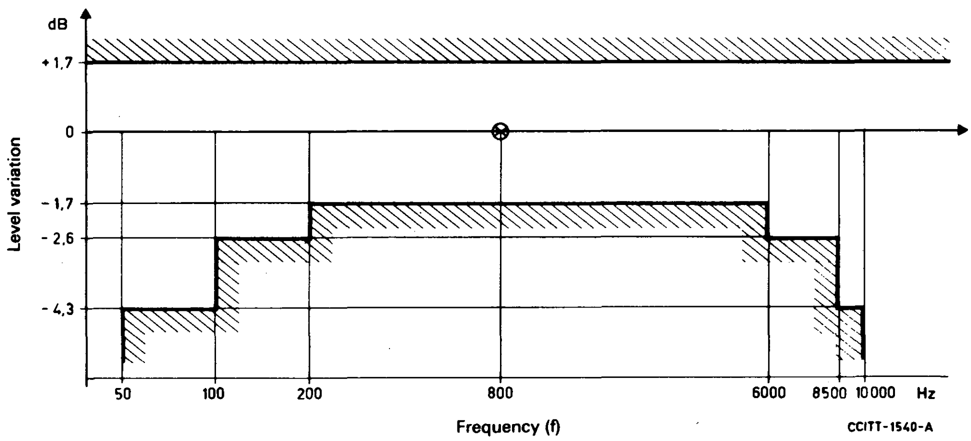


FIGURE 1/J.22 – **Variation of the received level with frequency relative to the value at 800 Hz**

When the circuit is set up on a carrier system, the curve applies to the combined three pairs of equipments for modulation from, and demodulation to, audio frequencies, as included in the hypothetical reference circuit for programme transmision.

d) *Group-delay distortion*

The difference between the group delay t_f, for the frequency f considered, and the minimum group delay t_{min}, should not exceed the following values:

8 ms for $t_{10\,000} - t_{min}$
20 ms for $t_{100} - t_{min}$
80 ms for $t_{50} - t_{min}$

e) *Maximum noise level*[5)]

Type of measurement	R.m.s. value	Quasi-peak value[a]
Weighted noise[b]		
– new network (CCIR Recommendation 468-1)	–44 dBm0ps (15 kHz)	–39 dBq0ps
– old network (CCITT Recommendation P.53)	–48 dBm0ps	–43 dBq0ps (10 kHz)
Unweighted noise[b]	–28 dBm0s	–23 dBq0s

[a] The quasi-peak value (for the measuring method see CCIR Recommendation 468-1) includes the impulsive noise allowance.
[b] For open-wire circuits, the given values are to be reduced by 8 dB.

f) *Intelligible crosstalk*

Provisional limits:

1. The near- or far-end crosstalk ratio (for speech) between two sound-programme circuits or between a telephone circuit (disturbing circuit) and a sound-programme circuit (disturbed circuit) should be at least 74 dB for cable circuits and at least 61 dB for open-wire lines.

2. The near- or far-end crosstalk ratio (for speech) between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) should be at least 58 dB for cable circuits[6)] and at least 47 dB for open-wire lines.

Note 1. – The CCITT draws the attention of Administrations to the fact that, for a 1000-km circuit, it is in some cases difficult to meet these limits, notably when using unscreened pairs or programme channels on carrier systems (see Recommendation J.18).

Note 2. – The CCITT draws the attention of Administrations to the fact that, because of crosstalk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above crosstalk limits between two sound-programme circuits, simultaneously occupying the go and return channels respectively of a carrier system (the most economical arrangement) because in those circumstances they occupy the same position in the line-frequency band (see Recommendation J.18).

g) *Change of relative level with time*

The 800-Hz relative level at the far end of the circuit should meet the defined conditions for attenuation/frequency distortion, and also during a given programme transmission should not change from its nominal value by more than ± 2 dB. In addition, for sound-programme circuits on special pairs or on the phantom circuits of unloaded symmetric pairs, the 800-Hz relative level at the output of a frontier amplifier should not change from its nominal value by more than ± 1 dB during a given programme transmission.

h) *Nonlinear distortion*

The total harmonic distortion coefficient for the 2500-km hypothetical reference circuit for programme transmissions should not exceed 4% (harmonic distortion attenuation of 28 dB) at any frequency within the

[5)] In the case of circuits on carrier systems, it is not always possible, in the absence of special precautions, to meet the limits recommended under e) (see the Annex).

[6)] See Question 9/XV.

band to be transmitted. The measurement being made with a sinusoidal signal (fundamental frequency) of +9 dBm0 connected to the origin of the circuit. The total harmonic distortion coefficient, k, being calculated from the formula

$$k = \sqrt{k_2^2 + k_3^2}$$

where k_2 is the 2nd order harmonic distortion coefficient and k_3 is the 3rd order harmonic distortion coefficient.

However, the following values should be considered as desirable design objectives for future developments:

k = 3% (30 dB), at fundamental frequencies below 100 Hz,
k = 2% (34 dB), at fundamental frequencies above 100 Hz.

Note. – Precautions should be taken in the measurement of harmonic distortion on circuits equipped with pre-emphasis networks. (See Recommendation N.21, Volume IV.1).

i) *Interference from the power supply*

The highest-level unwanted side component due to modulation of a sound-programme signal caused by interfering signals from power supply sources should not be of greater level than −45 dB relative to the level of a sine-wave test signal applied to the sound programme circuit.

j) *Error in reconstituted frequency*

The difference between the initial and reconstituted frequencies should not exceed 2 Hz.

k) *Single-tone interference*

Under study. (The subjective assessment of single-tone interference on high-quality circuits will be carried out by a method described in CCIR Report 623.)

ANNEX

(to Recommendation J.22)

Values of noise expected in practice on 2500 km circuits

Estimated noise power levels

Table 1 shows the noise values arising when sound-programme circuits (using pre-emphasis and de-emphasis in accordance with Recommendation J.17) are set up in place of three telephone channels, each of which conforms to the general noise objectives given in Recommendation G.222. The assumptions made for the purpose of the noise calculations are shown at the end of this Annex.

TABLE 1

	One-minute mean value	
	for not more than 20% of a month	for not more than 0.1% of a month
Noise power level weighted with the network of Recommendation P.53,B., *Green Book,* Volume V	−44.5 dBm0ps	−37.5 dBm0ps

Note. – The increased noise level shown as occurring for less than 0.1% of a month applies when the carrier circuit is established over a radio-relay system.

When 10-kHz and 6.4 kHz type sound-programme circuits which include pre-emphasis and de-emphasis networks according to Recommendation J.17 are set up on a carrier system it is recommended that, for reasons of overload, the level on such a circuit at a zero relative level (0 dBr) point, deduced from the level diagram of telephone circuits set up on the same 12-circuit group, should lie between a maximum of −1.5 dBm0 and a minimum of −4.5 dBm0, when the sound-programme signal is considered to be an 800-Hz tone at a level of 0 dBm0s.

The deduced level of −1.5 dBm0 could be considered as normal. A further 3-dB adjustment to permit a decrease down to −4.5 dB should, however, be included to cover the case of exceptional overloading if operational experience shows that in fact this is necessary.

Note. – Certain problems connected with the use of pre-emphasis on carrier systems have not yet been satisfactorily resolved. These are:

- the limitation of the level of testing tones, which is of concern to Study Group IV;
- the effect of pre-emphasis on the harmonic distortion requirements which the programme circuit should meet at high frequencies [7].

Use of compandors

Provided the compressor and the expander are of the same make, it is possible to obtain overall transmission characteristics as regards noise which conform to the CCITT Recommendations for the 2500-km hypothetical reference circuit, without introducing other factors that might impair transmission performance. The CCITT is now examining recommendations on the compressor and the expander, considered separately, so as to achieve the same results.

Assumptions and conventional terms

The expression dBm0ps is used to indicate noise power levels in a sound-programme circuit which have been psophometrically weighted according to Recommendation P.53, B. *Green Book*, Volume V, and measured in decibels relative to 1 mW at a point of zero relative sound-programme level (0 dBrs), in that circuit. The CCITT practice in the past has been to quote noise level for sound-programme circuits relative to "peak programme" or "maximum voltage" which is defined as a voltage of 2.2 volts r.m.s. (measured at the terminals of an impedance of 600 ohms) at a point of zero relative sound-programme level. The signal-to-noise ratio objective of 57 dB [previously given in Recommendation J.22, e), Volume III, *Green Book*] is thus equivalent to a noise power level of −48 dBm0ps.

The value for not more than 20% of a month was calculated for 10-kHz type circuits on the following assumptions:

– noise on one telephone channel (including the multiplex equipment) according to Recommendation G.222, weighted for telephony	−50 dBm0ps
– Bandwidth correction from 3.1 to 10 kHz	+ 5 dB
– Suppression of weighting for telephony (in the case of a uniform-spectrum noise)	+ 2.5 dB
– Improvement due to pre-emphasis [8] (see Recommendation J.17)	− 9 dB
– Effect of the relative level shifted by −1.5 dB at 800 Hz	+ 1.5 dB
– Weighting for sound-programme transmissions according to Recommendation P.53, B., *Green Book*, Volume V.	+ 5.5 dB
Total	−44.5 dBm0ps

The value for not more than 0.1% of a month was calculated on the basis of the noise variations to be expected on a radio-relay link used mainly for providing telephone circuits and conforming with Recommendation G.222.

[7] Measurements of harmonic distortion on programme circuits having pre-emphasis must be treated with reserve. This point is being studied by the CCITT.

[8] Set to have zero loss at 800 Hz.

Recommendation J.23

PERFORMANCE CHARACTERISTICS OF 6.4-kHz TYPE SOUND-PROGRAMME CIRCUITS [9)]

(former Recommendation J.31, B; amended at Geneva 1972 and 1976)

The CCITT,

unanimously recommends

that when the hypothetical reference circuit defined in Recommendation J.11 is assumed to be made up of three 6.4-kHz type sound-programme circuit-sections, the characteristics given below apply with the following reservations:

1) For audio transmission methods, all the characteristics are valid except for intelligible crosstalk.

2) For carrier transmission methods, all the characteristics are valid for the intelligible crosstalk and noise (see Recommendation J.18 and the Annex to Recommendation J.22).

a) *Band of frequencies effectively transmitted*

The band of frequencies effectively transmitted by the complete link should extend from 50 to 6400 Hz at least (see Recommendation J.13, Definition 10).

b) *Line levels*

(See Recommendation J.14.)

c) *Attenuation/frequency distortion*

Figure 1/J.23 shows the permissible limits for the variation, as a function of frequency (relative to the value measured at 800 Hz) of the relative voltage level (referred to 0.775 V) measured at the end of the circuit. The method of measuring this relative level is shown in Recommendation N.21, Volume IV.1.

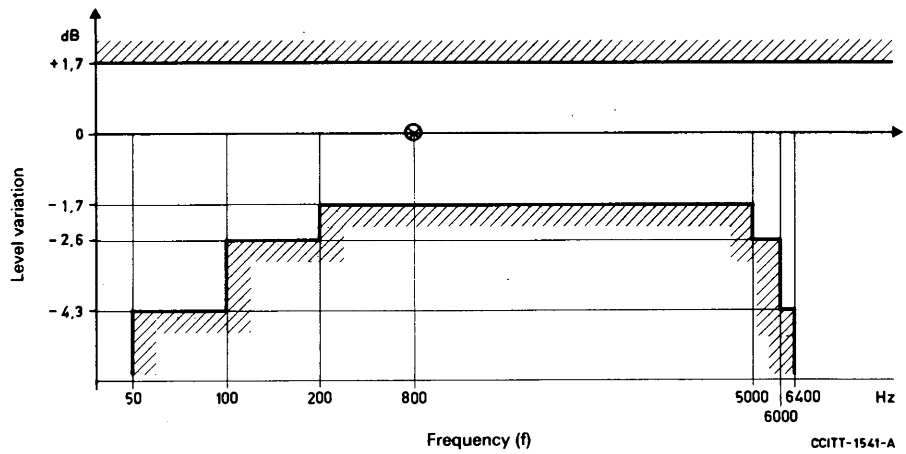


FIGURE 1/J.23 – **Variation, with frequency, of the relative voltage level at the end of a 6.4 kHz type sound-programme circuit. The values are shown relative to the value at 800 Hz.**

[9)] This Recommendation corresponds to CCIR Recommendation 503.

This curve applies to the overall combination of three pairs of equipments for modulation from, and demodulation to, audio frequencies, as included in the hypothetical reference circuit for sound-programme transmission.

d) *Group delay distortion*

The difference between the group delay t_f for the frequency f considered, and the minimum group delay t_{min}, should not exceed the values given below:

8 ms for $t_{6400} - t_{min}$
20 ms for $t_{100} - t_{min}$
80 ms for $t_{50} - t_{min}$

e) *Noise*

The recommended limits are the same as for a 10-kHz type sound-programme circuit set up on three carrier channels [see Recommendation J.22, e)]; the reservations in the Annex to that Recommendation also apply.

f) *Intelligible crosstalk*

The recommended limits are the same as for a 10-kHz type sound-programme circuit set up on three carrier channels [see Recommendations J.22, f) and J.18].

g) *Change of relative level with time*

The 800-Hz relative level measured at the far end of the circuit should meet the defined conditions for attenuation/frequency distortion and also, during a given programme transmission, should not change from its nominal value by more than ± 2 dB.

h) *Nonlinear distortion*

The recommended values are the same as for a 10-kHz type sound programme circuit set up on three carrier channels [Recommendation J.22, h).].

PAGE INTENTIONALLY LEFT BLANK

PAGE LAISSEE EN BLANC INTENTIONNELLEMENT

SECTION 3

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP SOUND-PROGRAMME CIRCUITS

Recommendation J.31

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 15-kHz TYPE SOUND-PROGRAMME CIRCUITS

(Geneva, 1972; amended at Geneva, 1976)

It is recognized that the overall objective given in Recommendation J.21 can be met by many different types of systems and that some solutions may be preferable to others for national networks, the choice depending on the particular requirements of an Administration.

It is, however, a basic objective of the CCITT to standardize a single solution to be adopted for international circuits. Furthermore, several Administrations have indicated that a single solution for international circuits will considerably ease the problem of providing these circuits.

The CCITT therefore recommends for international circuits the use of the solution described in A. below, in the absence of any other arrangement between the interested Administrations, including if necessary the Administrations of the transit countries. Other solutions which have been considered and are capable of meeting the recommended characteristics of Recommendation J.21 are described in Annexes 1, 2 and 3.

The characteristics of the group links, which have to be used in any case, are given in B. below.

A. CHARACTERISTICS OF AN EQUIPMENT ALLOWING TWO 15-kHz TYPE CARRIER-FREQUENCY SOUND-PROGRAMME CIRCUITS TO BE ESTABLISHED ON A GROUP

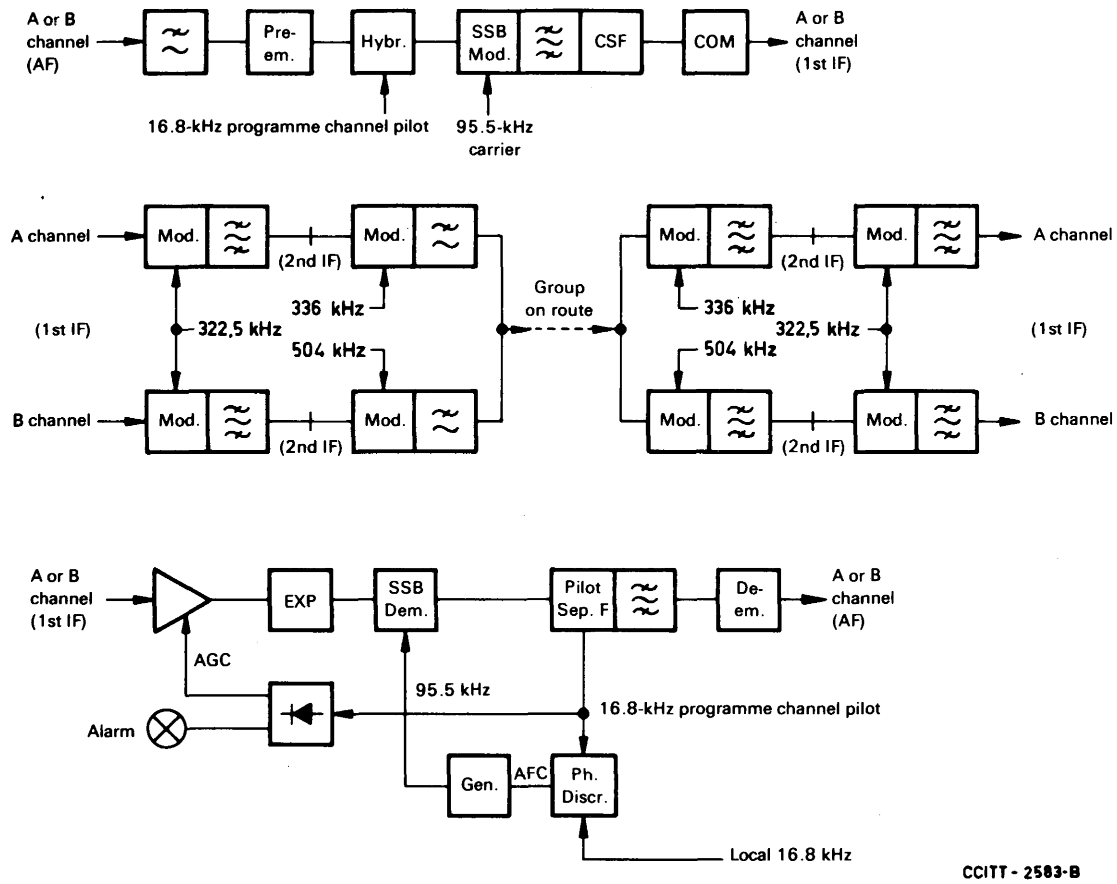
Introduction

An equipment allowing the establishment of 15-kHz type sound-programme circuits (in accordance with Recommendation J.21) on carrier telephone systems which conform to the noise objectives in Recommendation G.222 is defined here. The use of this equipment does not cause either a mean or a peak load higher than that of the telephone channels which it replaces [1]. The two sound-programme circuits set up on one group can be used either as two independent monophonic circuits or as a pair of circuits for stereophonic transmissions.

The following, covering frequency position, pre-emphasis, compandor and programme-channel pilot, are to be considered as integral parts of the Recommendation, forming the complete definition of the equipment covered by this Recommendation.

The block schematic of a suitable equipment is given in Figure 1/J.31.

[1] This is the objective given in Recommendation J.14 for new design of equipment.



Pre-em.	= pre-emphasis	Pilot Sep. F.	= pilot separating filter
Hybr.	= hybrid	De-em.	= de-emphasis
Mod.	= modulator	Ph. Discr.	= phase discriminator
SSB mod.	= single sideband modulator	AFC	= automatic frequency control
CSF	= carrier suppression filter	AGC	= automatic gain control
EXP	= expander	IF	= intermediate frequency
COM	= compressor	AF	= audio frequency
SSB dem.	= single sideband demodulator		

FIGURE 1/J.31 – **First modulation, auxiliary modulations and demodulation of the two-channel programme system**

a) *Frequency position in the basic group 60-108 kHz*

The frequency position in the basic group is shown in Figure 2/J.31. For both programme channels, the tolerance on the virtual carrier frequency is ± 3 Hz and the programme-channel pilot is fed in as 16 800 ± 0.1 Hz in the audio-frequency position.

Note. – Programme channel B can be replaced by telephone channels 1 to 6.

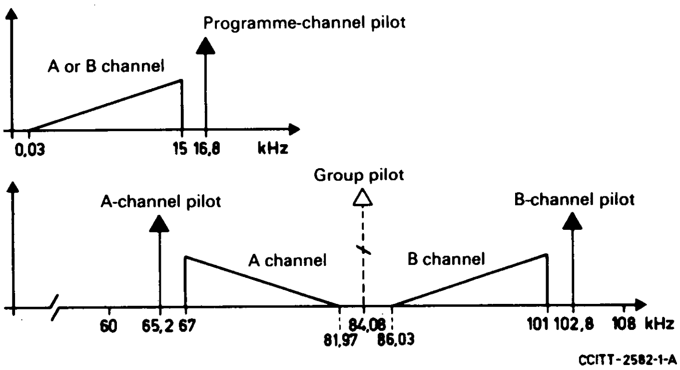


FIGURE 2/J.31 – **Line-frequency positions of the two-programme channels in the group**

b) *Intermediate frequency position* (see 1st IF in Figure 3/J.31)

Figure 3/J.31 gives an example of a modulation scheme which is suitable for deriving the line frequency positions shown in Figure 2/J.31, and in which two intermediate frequency stages are used. It is recommended that the first intermediate frequency (1st IF) be identical for each of the sound-programme channels A and B, and the inverted sideband be used based on suppressed carrier of 95.5 kHz.

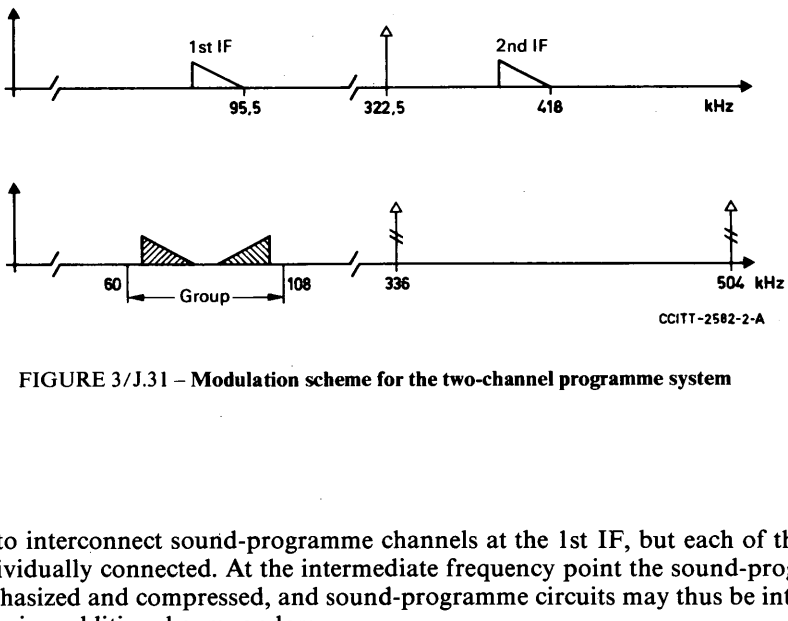


FIGURE 3/J.31 – **Modulation scheme for the two-channel programme system**

It is possible to interconnect sound-programme channels at the 1st IF, but each of the two programme channels must be individually connected. At the intermediate frequency point the sound-programme signal has already been pre-emphasized and compressed, and sound-programme circuits may thus be interconnected at the 1st IF without introducing additional compandors.

The relative level at the interconnection point is similar to the relative level in the carrier telephone system in the basic group at the receiving end (−30.5 dBr). The absolute level is determined by the pre-emphasis and compressor; the long-term mean power of the sound signal (A or B channel) is about 250 μW0.

The nominal impedance chosen in this example is 150 ohms balanced with a 26-dB return loss.

The programme channel pilot is through connected at 95.5 − 16.8 = 78.7 kHz, at a level of −12 dBm0 in the absence of a programme signal.

Special through-connection filters for the sound-programme channel are not required. The bandpass filters at the output of the second modulation stage (receiving end) have sufficient stopband rejection.

c) *Pre-emphasis and de-emphasis*

Pre-emphasis and de-emphasis should be applied before the compressor and after the expander respectively in accordance with Recommendation J.17, the 800-Hz attenuation of the pre-emphasis being set to 6.5 dB.

d) *16.8-kHz pilot signal*

At the sending end *the 16.8-kHz pilot signal* is fed in after the pre-emphasis and before the following modulator and compressor with a level of −29 dBm0 ± 0.1 dB. In the absence of a programme signal, this pilot level is increased by 17 dB by the compressor to −12 dBm0(*t*) [2)] on the carrier transmission path. After having passed through the expander, the pilot is branched off for control purposes after the demodulator and before the de-emphasis via a 16.8 kHz bandpass filter and is then suppressed in the transmission channel.

The control functions of the pilot are as follows: frequency and phase correction of the demodulator and compensation of the transmission loss deviations between compressor and expander. In view of the need to transmit stereophonic signals, the phase control should be sufficiently accurate so that the phase difference between the two channels does not exceed 1° even if the frequencies corresponding to the frequencies of the received pilots are in error by ± 2 Hz due to the carrier system.

e) *Compandor*

1. As shown in Figure 4/J.31 the compressor characteristic has a transition from the range of constant gain at low input levels to a range of constant loss at high input levels. Table 1/J.31 indicates the precise dependence of the compressor amplification as a function of the input level. The compressor and expander are controlled by the r.m.s. value of the sum of the voltages of programme and pilot signals.

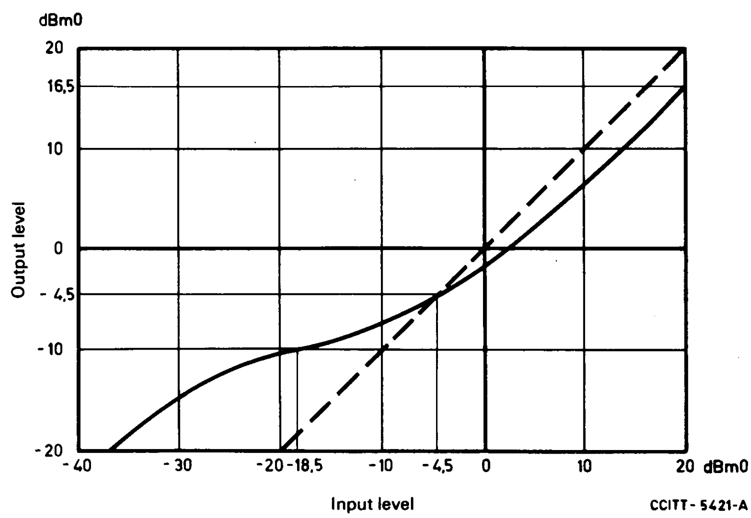


FIGURE 4/J.31 – **Characteristic of the compressor**

In Table 1/J.31, the compressor is pre-loaded by the pilot; in the absence of both programme and pilot, the gain of the compressor reaches the value of 22 dB.

The amplification of the expander is complementary to that of the compressor. The tolerance should also be ± 0.5 dB, or ± 0.1 dB as shown in Table 1/J.31.

[2)] dBm0(*t*) denotes that the level quoted is referred to a zero relative level point in a telephone channel.

TABLE 1/J.31 – **Compressor characteristic**

Programme signal level at the compressor input (dBm0)	Compressor gain (dB) (tolerance ±0.5 dB except at the point marked * where the tolerance is ±0.1 dB)
$-\infty$	+17.0 *
−40.0	+16.9
−35.0	+16.5
−30.0	+15.6
−25.0	+13.2
−20.0	+ 9.7
−15.0	+ 6.0 *
−10.0	+ 2.7
− 5.0	+ 0.2
− 4.5	0.0
0.0	− 1.3
+ 3.0	− 2.0 *
+ 5.0	− 2.3
+10.0	− 2.9
+15.0	− 3.2
+20.0	− 3.5

2. The attack and recovery times of the compressor are measured in 12-dB steps (see Recommendations G.162 and O.31) between the point of the unaffected level of −4.5 dBm0 and the level of −16.5 dBm0 and vice versa. In order to obtain as pronounced an envelope as possible in the oscillogram, the pilot is disconnected during this measurement and a test frequency is chosen which gives rise to an intermediate frequency that is approximately in the middle of the IF band. The attack and recovery times of the compressor are, as in Recommendation G.162, the times between the instant when the output voltage of the compressor is suddenly changed and the instant when, after the sudden change, the output voltage passes the arithmetic mean value between initial and final values.

The nominal values of the times so measured are:

attack time: 1 ms;
recovery time: 2.8 ms.

The subject of tolerances for these values is a matter for further study.

The transient behaviour of the expander is observed with the compressor and expander interconnected. If the same steps are then applied to the compressor input, the signal at the expander output should not deviate from the final steady-state value by more than ± 10%.

Note. – Since the initial and final values of the compressor output voltage in the case of this compandor are not in a 1:2 ratio because of the curved characteristic, the arithmetic means are here not 1.5 and 0.75, respectively, as in the case of the telephone compandor.

f) *Impedance at audio points*

The audio input-impedance should be 600 ohms balanced with a minimum return loss of 26 dB.

g) *Attenuation/frequency distortion due to the sending and receiving equipments*

The total attenuation distortion introduced by a sending and a receiving equipment should not exceed the following ranges:

40 to 125 Hz : +0.5 to −0.7 dB;
125 Hz to 10 kHz: +0.3 to −0.3 dB;
10 to 15 kHz : +0.5 to −0.7 dB;

relative to the gain at 800 or 1000 Hz.

h) *Suppression of carrier leaks at 10 kHz and 14 kHz*

Since, according to Recommendation H.14, carrier leaks may be of the order of −40 dBm0 and that Recommendation J.21, 3.1.6 requires a suppression to $(-73-\Delta ps)$ dBm0s for single tone interference, narrow-band crystal stop-filters should be available for insertion if required, and should have the following specifications:

1-dB bandwidth of the stopband:

at 10 kHz: $\leqslant \pm$ 150 Hz
at 14 kHz: $\leqslant \pm$ 210 Hz.

Attenuation for the midfrequencies:

at 10 kHz: $\geqslant$ 36 dB
at 14 kHz: $\geqslant$ 22 dB

Note. – The attenuation of these bandstop filters is sufficient without taking account of the compandor advantage.

The stopband attenuations should be maintained within ± 2 Hz referred to the above midfrequencies, in order to allow for the normal frequency variation of the carrier leaks.

In order to be able to use crystal bandstop filters of a simple design, it is recommended to assign them not to the AF position but to the corresponding IF position, additional allowance having to be made for the carrier frequencies used in the terminal equipment:

10 kHz corresponding to 85.5 kHz and
14 kHz corresponding to 81.5 kHz.

Note. – Contribution COM XV-No. 31 (study period 1973-1976) from the Federal Republic of Germany gives details of the calculation and numerical data for a possible filter characteristic.

i) *Interconnection*

When sound-programme circuits employing equipment in conformity with this Recommendation are interconnected, it is recommended that, where possible, the through connection should be performed either in the group-frequency position or in the position of the 1st IF. As described in b) above, interconnection in these positions will exclude unnecessary compandor stages from the through connection.

B. CHARACTERISTICS OF A GROUP LINK USED TO ESTABLISH TWO 15-kHz TYPE CARRIER-FREQUENCY SOUND-PROGRAMME CIRCUITS

The lining up of international group links is described in Recommendation M.460, Volume IV.1 in which information is given on the attenuation/frequency characteristics which should be obtained. To comply with the attenuation/frequency characteristics of sound-programme circuits in accordance with Recommendation J.21, it may be necessary to include a small amount of additional equalization.

Group links for programme transmission have to meet special requirements concerning carrier leaks and other interfering frequencies so that programme transmission conforms to the standard as defined in Recommendation J.21.

The basic requirement is that interfering frequencies appearing in the programme bands have to be suppressed to $(-73 - \Delta ps)$ dBm0s on the programme circuit [3]. For frequencies corresponding to audio frequencies above 8 kHz, additional suppression is possible by special spike filters in the terminal equipment of the programme circuit.

Group links to be used for programme transmission according to Recommendation J.21 and using programme terminal equipment according to Recommendation J.31 have to meet, therefore, the following requirements:

1. Carrier leaks [4] at 68, 72, 96 and 100 kHz and any single-tone interference signal falling outside the band of frequencies used for sound-programme transmission including the pilots (see Figure 2/J.31) should not be higher than −40 dBm0. This allows the necessary suppression to $(-73 - \Delta ps)$ dBm0s taking account of the amount of the narrow-band crystal stop-filter attenuation.

2. Carrier leaks at 76, 80, 88 and 92 kHz and any other single-tone interference signal falling within the band of frequencies used for sound-programme transmission including the pilots (see Figure 2/J.31), should not be higher than:

- for frequencies between 73 kHz and 95 kHz: −68 dBm0,
- for frequencies at 67 kHz and 101 kHz: −48 dBm0.

In the bands 67 to 73 kHz and 95 to 101 kHz the requirement is given by straight lines (linear frequency and dB scales) interconnecting the requirements given above [5].

It is necessary to consider whether additional requirements for the characteristics of group links for 15-kHz sound-programme transmission is needed beyond those covered in Recommendation M.460 (for example, group delay distortion in the case of stereophonic transmission bearing in mind the possibility of changeover to stand-by paths).

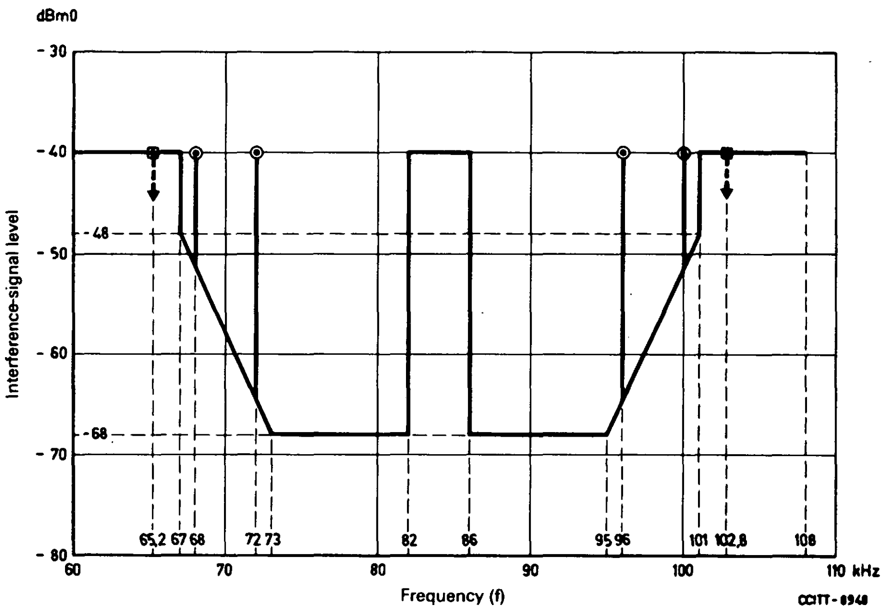
The above requirements are illustrated in Figure 5/J.31.

Note. – Figure 6/J.31 gives the permissible level of single-tone interference for the systems described in Annexes 1, 2 and 3 of this Recommendation, such that the basic requirement of $(-73 - \Delta ps)$ dBm0s mentioned above is met.

[3] This value has been specified in Recommendation J.21 by CMTT. CCIR Report 493-1 gives some additional information regarding the subjective impairments produced by interfering frequencies on a circuit using equipment conforming to Recommendation J.31.

[4] Having the frequency precision of carriers.

[5] These values are still under study. It has been assumed that the compandor gives a subjective improvement of at least 12 dB. CMTT is asked to confirm that this assumption is valid.

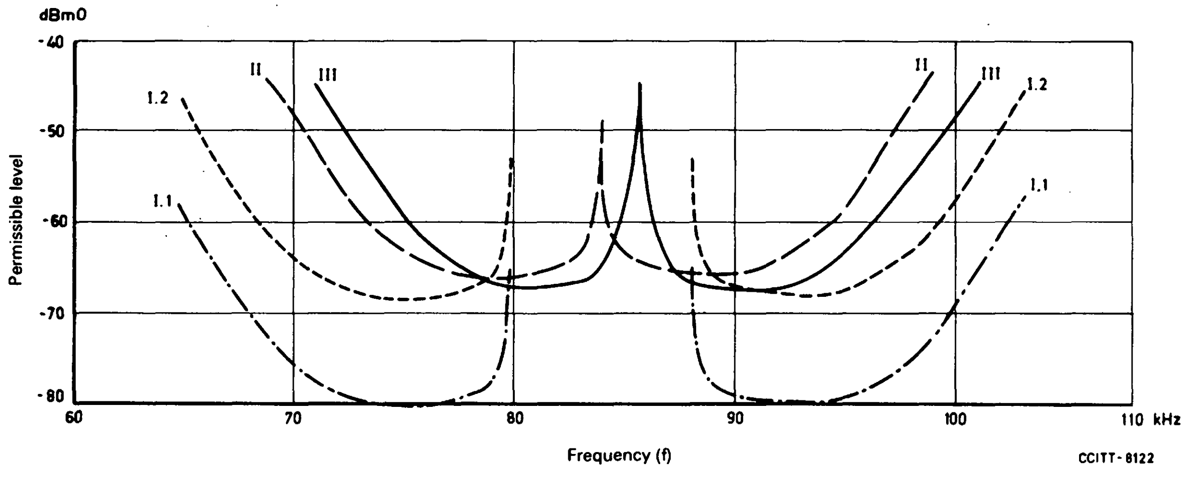


The continuous curve represents the general requirements for single-frequency interfering tones, with the following exceptions:

◉ carrier-leak frequencies at which the requirements are relaxed to –40 dBm0 are shown thus.

▣ values at these frequencies (A- and B-channel pilots) are still under study.

FIGURE 5/J.31 – **Mask for the carrier leaks and any other tone-interference signal falling within the group band**



Curve I.1: requirement for the system of Annex 1, *without* compandor gain
Curve I.2: requirement for the system of Annex 1, *with* compandor gain
Curve II: requirement for the DSB system of Annex 2
Curve III: requirement for the system of Annex 3

FIGURE 6/J.31 – **Permissible level of a single-frequency interference on the group link**

ANNEX 1

(to Recommendation J.31)

Single sideband system

(Contribution of the N.V. Philips Telecommunicatie Industrie)

This Annex concerns a single-sideband sound-programme transmission equipment incorporating pre- and de-emphasis combined with a compandor characterized by a separate FM control channel.

The equipment operates on group links of carrier telephone systems.

Both peak and average loads to the group are compatible with those of the replaced telephone channels.

a) *Frequency allocation in the group*

	Modulated programme frequencies	Compandor control channel	Synchronizing pilot
Channel A (inverted)	65 . . . . . 79.96 kHz	81.39 . . . 83.18 kHz	84 kHz
Channel B (erect)	88.04 . . . 103 kHz	84.82 . . . 86.61 kHz	

Channels A and B can be used for independent monophonic sound-programme circuits or combined into a stereophonic pair. Either channel A or B can be deleted and substituted by the corresponding telephone channels.

Group pilots at 84.08, 84.14 and 104.08 kHz and telephone channels 1 and 12 are compatible with this frequency allocation.

b) *Pre-emphasis*

Pre-emphasis takes place before compression by means of a network according to Recommendation J.17. The insertion loss at 800 Hz is 6.5 dB.

c) *Compandor*

1. *Steady-state characteristics*

The compandor has a separate frequency-modulated control channel containing the information on the degree of compression, as indicated in Table 1.

For the lowest programme-levels, the total improvement in signal-to-noise ratio will be 19.8 dB (when weighting by means of a psophometer according to Recommendation P.53, B., *Green Book*, Volume V).

TABLE 1

Compressor input level (dBm0)[a]	Compressor gain (dB)	Control channel frequency (kHz)	
		Channel A	Channel B
$-\infty$	17	81.39	86.61
−40	17	81.39	86.61
−35	16.9	81.40	86.60
−30	16.7	81.41	86.59
−25	15.9	81.43	86.57
−20	13.5	81.52	86.48
−15	9.5	81.70	86.30
−10	4.8	81.94	86.06
− 5	0	82.24	85.76
0	− 4.9	82.56	85.44
+ 5	− 9.6	82.90	85.10
+10	−11.8	83.18	84.82
+15	−11.8	83.18	84.82

[a] The relative level at the compressor input to be considered is 6.5 dB higher than that corresponding to an 800-Hz audio-frequency test-tone. With pre-emphasis and compressor, an audio input level of e.g. +6.5 dBm0s at 800 Hz will thus give rise to a compressor input level of 0 dBm0 and hence to a group level of −4.9 dBm0(*t*).

The level in the control channel is −17 dBm0(*t*).

The expander gain tracks that of the compressor with a tolerance of ±0.5 dB.

dBm0(*t*) denotes that the level quoted is referred to a 0 relative level point in a telephone channel.

dBm0s denotes that the level quoted is referred to the sound-programme circuit.

2. *Transient behaviour of the compressor*

Considering a 12-dB level step at the compressor input from −17 dBm0 to −5 dBm0 (point of unaffected level), the compressor attack time is defined as the time interval needed for the compressor output voltage to reach the arithmetical mean between initial and final values.

Taking the sudden level variation in the opposite direction yields the definition of the compressor recovery time.

The nominal values of attack and recovery time are respectively 2.4 and 4 ms.

3. *Transient behaviour of the expander*

With compressor and expander interconnected and when applying at the compressor input sudden level variations from −17 dBm0 to −5 dBm0 and vice versa, the expander output voltage should not deviate by more than 10% from the steady-state values.

d) *Synchronizing pilot*

A synchronizing pilot at 84 kHz with a level of −20 dBm0(*t*) is used in order to reduce frequency and phase errors due to the group link.

Frequency offset is reduced by a factor of 21.

At the transmitting and receiving terminals, the modulating and demodulating carriers should be phase-coherent with the synchronizing pilot in such a way that a frequency offset of 2 Hz does not give rise to a phase difference between the two channels of the stereophonic pair exceeding 1°.

ANNEX 2

(to Recommendation J.31)

Double-sideband system

(Contribution of L.M. Ericsson, ITT and Telettra)

a) *Frequency allocation*

Double-sideband modulation of a carrier frequency of 84.080 kHz. The sidebands are located in the band 69.080-99.080 kHz. The carrier is reduced in level, so that it can be used in the normal way for a group pilot.

b) *Pre-emphasis*

The pre-emphasis curve given in Recommendation J.17 should be used.

c) *Compandors*

Compandors are not an integral part of these systems.

d) *Levels of programme signal in carrier system*

The levels are such that a sine wave of 800 Hz applied at the audio input with a level of 0 dBm0s will appear at the group output, having been through a pre-emphasis network, as two sideband frequencies each with a level of +2 dB compared to the relative level of the telephone channels, that is +2 dBm0(*t*). This level should be adjustable over a range of about ± 3 dB.

e) *Group regulation*

Normal group regulation is available using 84.080 kHz. This frequency had the normal level and tolerances for a pilot as given in Recommendation G.241, b) and c).

f) *Carrier regeneration*

Different versions of this system rely respectively on the correct phase of the group pilot or on the use of an auxiliary pilot above the programme band (16.66 kHz or 16.8 kHz, for example, has been proposed for national systems); a frequency of 16.8 kHz should be reconsidered for international use; the sending terminal should, where necessary, be adapted to meet the needs of the receiving terminal in either respect. The level of any auxiliary pilot should not exceed −20 dBm0(*t*), i.e. referred to the telephone channel level in the group.

ANNEX 3

(to Recommendation J.31)

Transmitting of six sound-programme circuits on a supergroup link

(Contribution of Società Italiana Telecomunicazioni Siemens SpA)

A system for setting up on group links one monophonic programme circuit or two circuits combined into a stereophonic programme, is described in Annex 3 to Recommendation J.31, Volume III, *Green Book* and is widely used in Italy.

A new type of equipment for the transmission of six programme channels allocated in the band of a basic supergroup has been developed and successfully adopted experimentally.

The essential characteristic of this system is the utilization of a single sideband, modulated in amplitude, with a suppressed carrier of 86 kHz and a synchronous demodulation using a 16.8-kHz pilot in order to have no errors in the transmitted frequencies and no errors in the phase relation between the signals A and B for stereophonic programmes.

The carrier of 86 kHz is suitable for allocating the programme signal to that sideband which is unaffected by telephone carrier leaks and for avoiding intelligible crosstalk between telephone and programme channels.

The single-sideband modulation employs the phase-shift technique. By means of this the programme channel is allocated either to the lower sideband between 71 and 86 kHz or to the upper sideband between 86 and 101 kHz.

In a second modulation procedure the six sound-programmes are allocated to the band of the basic supergroup 312-552 kHz with the carriers 346 kHz, 382 kHz, 418 kHz, 454 kHz, 490 kHz and 526 kHz.

The measurements carried out show that the system complies with the values recommended in Recommendation J.21 for the high-quality circuits with equipments whose price renders the system economical, even for distances of some hundreds of kilometres.

Note. – The system is described in Contribution COM XV-No. 151 (study period 1973-1976).

Recommendation J.32

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 10-kHz TYPE SOUND-PROGRAMME CIRCUITS

(former Recommendation J.22; amended at Geneva, 1964, and Mar del Plata, 1968)

10-kHz type sound-programme circuits can be provided in wideband cables by the following methods:

a) *Special pairs for sound broadcasting*

If a broadcast programme is to be distributed to a number of intermediate points along the line (if this includes carrier telephone systems), it may be necessary to use a pair of conductors with a special screen for programme transmissions; or it may happen that it is preferable to transmit the broadcast programme over the carrier system itself or on the phantom of the unloaded pairs.

It should be remembered, however, that interstice pairs in a coaxial cable are principally intended for the maintenance and supervision of the telephone carrier system routed over the coaxial pairs.

b) *10-kHz sound-programme circuits routed over channels of a carrier telephone system in cable*

It is recommended to use the frequency band corresponding to three telephone channels of a carrier system to form a 10-kHz type sound-programme circuit. One such assembly of three channels may be used in this matter in a 12-circuit group.

The CCITT has already recommended the position defined below as Position I for this assembly of three channels to provide programme transmission in basic group B.

Position I: Frequency band used: 84-96 kHz
Virtual carrier frequency: 96 kHz

The CCITT no longer recommends the use of Position II, defined in the old Recommendation (*Red Book*, Volume III), in the international service.

The CCITT recommends also the following frequency arrangements in the basic Group B:

Position III: Frequency band used: 84-96 kHz
Virtual carrier frequency: 95.5 kHz

This position may be adopted whether a compandor is used or not.

Supplement No. 12 of this Volume indicates the improvement in crosstalk to be expected from offset of the carrier frequency and in particular from the use of Position III.

Note. – Some Administrations use a pilot inserted in the audio-frequency part of the sound-programme modulation equipment for the purpose of regulating the equivalent and supervising the link as a whole.

While, generally speaking, the provision of automatic group regulation should suffice to ensure satisfactory stability of the equivalent, a pilot like the one suggested by one of these Administrations might be useful when compandors (which increase variations of the equivalent) are used, or when the switching of sound-programme circuits to RF is envisaged or when frequency synchronization is required between the ends of the circuit.

With the limit that has been given in Recommendation J.14 for the "peak voltage" transmitted by one such assembly of three channels, these assemblies (used for sound-programme transmissions) may be placed in any basic group (or in all the basic groups) of a supergroup (or in all supergroups) of a carrier system on coaxial cable.

The CCITT has not limited the possible positions (in the basic supergroup) of the groups over which 10-kHz type sound-programme circuits can be routed, but it can be said that the basic groups (in a supergroup) which appear most appropriate for such circuits are groups 2, 3 and 4. These groups are subject to less attenuation distortion at the edges (produced by certain filters in the supergroup) than groups 1 and 5. The most appropriate supergroups in which to place the sound-programme circuits are those which are transmitted on the coaxial cable with the lowest carrier frequencies, because the frequency deviation (due to instability of the frequency generators) on the channels of these groups will be proportionately lower than the deviation on channels in supergroups transmitted at a high frequency. Supergroup 2 (the basic supergroup) has the additional advantage of having one stage of modulation less than the other supergroups.

In the case of a carrier system on symmetric pairs, it may be necessary to make a special choice of the group of the system and the pairs to be used in order that the conditions concerning crosstalk for the complete sound-programme circuit will be satisfied. (See Recommendations J.18 and J.22.)

c) *Use of phantom circuits on unloaded symmetric pairs equipped with carrier systems*

Experience has shown that the phantom circuits of symmetric pairs in cables equipped with carrier systems may allow transmission [as defined in Recommendation J.22, a)] from 50 Hz to 10 000 Hz. These circuits have the advantage that derivations at various repeater stations of the carrier system can easily be made, thus allowing the distribution of a radio programme or the picking up of a supplementary programme at various points along the line.

When such phantom circuits are used over long distances, it may be necessary to provide manual or automatic regulation to compensate for changes of attenuation with time.

d) *Use of the band of frequencies below 12 kHz*

The use of phantom circuits [see c) above] naturally depends on a multiple twin or a star quad cable being available. If only a pair cable is available, a possible solution would be to place the sound-programme transmission in the frequency band below 12 kHz, i.e. below the frequency band used for the carrier telephone channels; but this solution involves difficulties with filters or with crosstalk balancing frames, if any exist.

Recommendation J.33

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 6.4-kHz TYPE SOUND-PROGRAMME CIRCUITS

(former Recommendation J.31, A; amended at Geneva, 1972)

The CCITT recommends that, when an Administration wishes to provide a sound-programme circuit transmitted on a carrier system using a frequency band corresponding to two telephone channels, the circuit should occupy the frequency range 88 kHz to 96 kHz in the basic 12-channel group B frequency band and the virtual carrier frequency within this range should be 96 kHz, or as an alternative, 95.5 kHz [6].

SECTIONS 4 AND 5

Sections 4 and 5 have not yet been allocated.

---

[6] For the choice of groups and supergroups used, see Recommendation J.32.

SECTION 6

CHARACTERISTICS OF CIRCUITS FOR TELEVISION TRANSMISSIONS

Recommendation J.61

SPECIFICATIONS FOR A LONG-DISTANCE TELEVISION TRANSMISSION (System I excepted) [1), 2)]

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

The CCITT,

considering

the agreement reached by the Joint CCIR/CCITT Committee for television and sound transmission (CMTT) on a draft Recommendation concerning television transmission over long distances, common to the CCIR and the CCITT,

unanimously recommends

that, taking account of the definitions in 1. below, television transmissions over long distances should satisfy the requirements laid down in 2. and 3. below and their Annexes.

1. *Definitions*

1.1 *Definition of a long-distance international television connection* (see Figure 1/J.61)

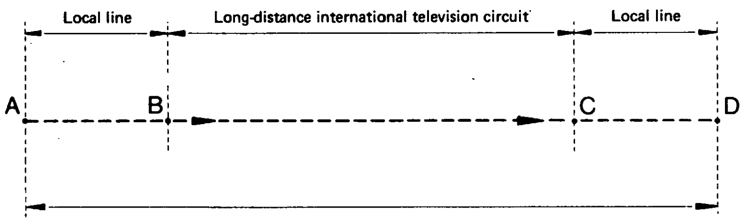


FIGURE 1/J.61 – **International television connection**

---

1) This Recommendation corresponds to CCIR Recommendation 421-3.
2) For System I see Recommendation J.62.

1.1.1 Point A, to be considered as the sending end of the international television connection, may be the point at which the programme originates (studio or outside location), a switching centre or the location of a standards converter.

1.1.2 Point D, to be considered as the receiving end of the international television connection, may be a programme mixing or recording centre, a broadcasting station, a switching centre or the location of a standards converter.

1.1.3 The local line AB connects point A to the sending terminal station, point B, of the international television circuit.

1.1.4 The long-distance international television circuit BC, comprises a chain of national and international television links. The precise locations (e.g. within buildings), to be regarded as the points B and C, will be nominated by the authorities concerned.

1.1.5 The local line CD connects point C, the receiving terminal station of the long-distance international television circuit, to the point D.

1.1.6 The combination AD, of the long-distance international television circuit BC, and the local lines AB and CD, constitutes the *international television connection.*

The requirements given in 2. and 3. below refer to the performance of long-distance international television circuits only; no requirements have been laid down for the local lines, AB and CD.

1.2 *Definition of the hypothetical reference circuit*

The main features of the television hypothetical reference circuit, which is an example of a long-distance international television circuit (BC in Figure 1/J.61) and which may be of either radio or coaxial-cable type, are:

- the overall length betwen video terminal points is 2500 km (about 1600 miles),
- two intermediate video points divide the circuit into three sections of equal length,
- the three sections are lined up individually and then interconnected without any form of overall adjustment or correction,
- the circuit does not contain a standards converter or a synchronizing-pulse regenerator.

Note 1. – The concept of the hypothetical reference circuit serves to provide a basis for the planning and design of transmission systems. Such a circuit has a length which is reasonably but not excessively long and, for a television circuit, a defined number of video-to-video sections. It is appreciated that, at the present time, international television circuits usually contain more than three video-to-video links in a length of 2500 km, but it is expected that the number will be reduced in the course of time. Annex 4 gives a provisional indication of the characteristics of circuits with more or fewer video sections than the hypothetical reference circuit.

Note 2. – In Canada and the United States of America, objectives are normally specified for circuits 6400-km long. The limits given in this Recommendation for 2500-km circuits for the 525-line system in Canada and the United States of America are therefore chosen to give an adequate performance in a 2500-km portion of a 6400-km circuit.

2. *Requirements at video interconnection points*

In this section the requirements apply at the video terminals of any long-distance television circuit, whatever its length.

2.1 *Impedance*

At video interconnection points, the input and output impedance of each circuit should be unbalanced to earth, with a nominal value of 75 ohms resistive and a return loss of at least 24 dB relative to 75 ohms. The return loss, relative to 75 ohms, of an impedance Z is

$$20 \log_{10} \left| \frac{75 + Z}{75 - Z} \right| \text{(dB)}.$$

Note 1. – In Canada and the United States of America, the impedance at video interconnection points should be either 124 ohms balanced to earth or 75 ohms unbalanced to earth, with a return loss of at least 30 dB.

Note 2. – In some countries, impedance is specified in terms of "waveform return loss" [see Doc. CMTT/9 (OIRT), 1963-1966 and CCIR Recommendation 451-2].

2.2 *Polarity and d.c. component*

At video interconnection points, the polarity of the signal should be *positive*, i.e. such that black-to-white transitions are positive-going.

The useful d.c. component, which is related to the average luminance of the picture, may or may not be contained in the video signal and need not be transmitted, or delivered at the output.

Any non-useful d.c. component unrelated to the video signal (e.g. the component due to d.c. valve supplies) should not cause more than 0.5 W to be dissipated in the 75 ohms load impedance. If the load impedance is disconnected, the voltage of this component should not exceed 60 volts.

2.3 *Signal amplitude*

At video interconnection points, the blanking level taken as the reference level, the nominal amplitude of the picture signal measured from the blanking level to the white level should be 0.7 volts (0.714 volts in Canada and the United States of America), while the nominal amplitude of the synchronizing signal measured from the blanking level to the tips of the synchronizing pulses should be 0.3 volts (0.286 volts in Canada and the United States of America), so that the nominal peak-to-peak amplitude of the video signal should be 1.0 volts (see Figure 2/J.61).

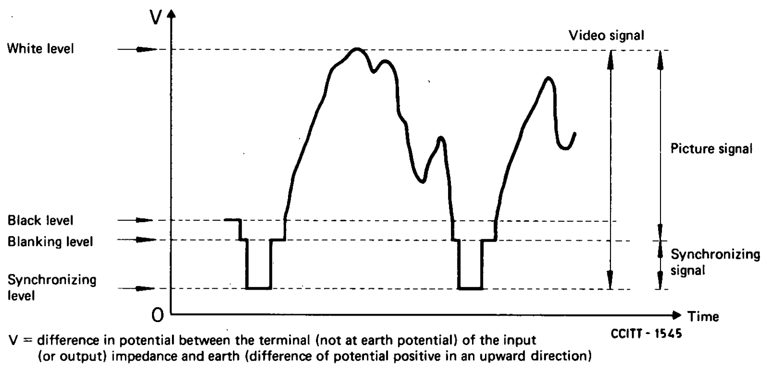


FIGURE 2/J.61

Theoretically, the amplitude should be measured with the useful d.c. component of the video signal restored, but in practice this is not necessary.

Note 1. – In the design of equipment, account should be taken of the losses in interconnecting cables when the video interconnection points are at some distance from the terminals of the modulating and demodulating equipment.

Note 2. – For colour in System M (Japan), the above specification applies to the luminance and synchronizing signals. For the chrominance signal, further study is required.

3. *Transmission performance of the hypothetical reference circuit*

In the following, the performance requirements are to be taken as design objectives applying to the hypothetical reference circuit as defined in 1.2 above.

It should be emphasized that the material contained in the following constitutes only a first step towards the solution of the general problem of determining methods of measuring and specifying the performance of television circuits of any length or degree of complexity.

3.1 *Insertion gain*

A long-distance international circuit, having the form of the hypothetical reference circuit should, at the time of setting up, have an insertion gain of 0 dB ± 1 dB (± 0.5 dB in Canada and the United States of America).

The insertion gain should be measured, using Test Signal No. 2 (described in Annex 1) and is defined as the ratio, in dB, of the amplitude of the bar (from black level to white level) at the output to the nominal amplitude of the bar at the input.

The measurement should be made under the following conditions:

a generator producing Test Signal No. 2, with an internal impedance of 75 ohms (resistive), is adjusted so that, if connected directly to a 75-ohm resistance, it would produce a line-synchronizing signal of 0.3 volts combined with a picture signal of 0.7 volts which may include 0.05 volts of pedestal. At the receiving end, the voltage between the black level and the white level (bar amplitude) is measured, using an oscilloscope connected across a resistance of 75 ohms. The ratio of this voltage to 0.65 volts, if pedestal is used, or 0.7 volts, if it is not (in dB), is the insertion gain of the television circuit.

Note. – In Canada and the United States of America somewhat different methods are used, but similar results are obtained.

3.2 *Variations of insertion gain*

The variations of insertion gain with time in the hypothetical reference circuit should not exceed the following limits:

- short-period (e.g. 1 s) variations: ± 0.3 dB (± 0.2 dB in Canada and the United States of America),
- medium-period (e.g. 1 hour) variations: ± 1.0 dB.

3.3 *Noise*

3.3.1 *Continuous random noise*

The signal-to-weighted noise ratio for continuous random noise is defined as the ratio, in dB, of the peak-to-peak amplitude of the picture signal (see Figure 2/J.61) to the r.m.s. [3] amplitude of the noise, within the range between 10 kHz and the nominal upper limit of the video frequency band of the system, f_c. The purpose of the lower frequency limit is to enable power supply hum and microphonic noise to be excluded from practical measurements.

For the hypothetical reference circuit, the signal-to-noise ratio should not be less than the values X given in Table 1/J.61 when measured with the appropriate lowpass filter described in Annex 2, the appropriate weighting network described in Annex 3, and an instrument having an "effective time constant" or "integrating time" in terms of power of 1 s (0.4 s in Canada and the United States of America).

[3] Administrations measuring the quasi-peak-to-peak amplitude of the noise are asked to establish the crest factor appropriate to their method of measurement and to express the results in terms of r.m.s. amplitude.

TABLE 1/J.61

System (see Report 624)	M (Canada and USA)	M (Japan) monochrome and colour	B, C, G, H	D, K, L	F	E
Number of lines	525	525	625	625	819	819
Nominal upper limit of video-frequency band f_c (MHz)	4	4	5	6	5	10
Signal-to-weighted noise ratio X (dB)	56	52	52	57	52	50

Note 1. – To obtain satisfactory transmission performance, television specialists believe that the signal-to-weighted noise ratio should fall neither below X (dB) for more than 1% of any month, nor below $X - 8$ dB for more than 0.1% of any month.

Note 2. – For the routine measurement of signal-to-noise ratio on real circuits, the noise can be measured with sufficient accuracy in the absence of the video signal. The error introduced by this method will not, in general, exceed 2 dB. More accurate devices and methods for measuring signal-to-weighted noise ratio when transmitting test signals, are described in Docs. XI/25, CCIR Moscow, 1958, CMTT/23, Monte Carlo, 1958, and CMTT/3, Paris, 1962, presented by the USSR.

3.3.2 *Periodic noise*

The signal-to-noise ratio for periodic noise is defined as the ratio, in dB, of the peak-to-peak amplitude of the picture signal (see Figure 2/J.61) to the peak-to-peak amplitude of the noise.

Note. – This definition has so far been used in specification clauses dealing with single-frequency noises and with power-supply hum (including the fundamental frequency and lower-order harmonics), but it may also prove to be useful for any case in which two or more sinusoidal components are in harmonic relationship.

The signal-to-noise ratio in the hypothetical reference circuit should not be less than the value given in Table 2/J.61.

TABLE 2/J.61

System	M (Canada and USA)	M (Japan)	B, C, G, H	D, K, L	F	E
Number of lines	525	525	625	625	819	819
Nominal upper limit of video-frequency band f_c (MHz)	4	4	5	6	5	10
Signal-to-noise ratio (dB) for power-supply hum (including the fundamental frequency and lower-order harmonics) [a]	35	30	30	30	30	30
Signal-to-noise ratio (dB) for single-frequency noise between 1 kHz and 1 MHz	59 [b]	50	50	50	50	50 [c]
Value (dB) to which the signal-to-noise ratio for single-frequency noise may decrease linearly between 1 MHz and f_c	43 [d]	30 [e]	30	30	30	30 [f]

[a] These figures apply only to hum added to the signal and not to hum which in transmission has modulated the amplitude of the signal and cannot be removed by clamping. The measurement should be made without clamping.

[b] This limit applies between 1 kHz and 2 MHz.

[c] For system E for frequencies below 1 kHz excluding power-supply hum (including both the fundamental frequency and lower-order harmonics), the signal-to-noise ratio may decrease linearly between the values 50 dB at 1 kHz and 45 dB at 100 Hz and between the value 45 dB at 100 Hz and 30 dB at 50 Hz.

[d] Value to which the signal-to-noise ratio may decrease, according to a linear function on a chart having a linear scale and a logarithmic frequency scale, for frequencies between 2 MHz and f_c (4 MHz).

[e] For Colour System M (Japan) the signal-to-noise ratio should not be less than 50 dB at 3.6 MHz.

[f] For system E this figure is reached at a frequency of 7 MHz and remains constant between 7 MHz and f_c (10 MHz).

3.3.3 *Impulsive noise*

The signal-to-noise ratio for impulsive noise is defined as the ratio, in dB, of the peak-to-peak amplitude of the picture signal (see Figure 2/J.61) to the peak-to-peak amplitude of the noise.

Provisionally, for the hypothetical reference circuit, a minimum signal-to-noise ratio of 25 dB for impulsive noise of a sporadic or infrequently occurring nature has been proposed for all systems, except System M (Canada and the United States of America), for which the requirement is 11 dB.

3.3.4 *Crosstalk*

This matter is still under study.

3.4 *Non-linear distortion*

Non-linear distortion affects both the picture and the synchronizing signals.

Non-linear distortions of the picture signal may be classified under three headings [4], namely:

- field-time non-linear distortion,
- line-time non-linear distortion,
- short-time non-linear distortion.

3.4.1 *Field-time non-linear distortion of the picture signal*

This matter is still under study.

3.4.2 *Line-time non-linear distortion of the picture signal*

Non-linear distortion of the picture signal is measured with Test Signal No. 3 (described in Annex 1), using a superimposed sine wave at a frequency 0.2 f_c.

The magnitude of the distortion is indicated by the ratio of the minimum peak-to-peak amplitude of the sine wave to the maximum amplitude along the saw tooth.

The sine wave may be displayed on an oscilloscope with the time base running at line frequency by using a bandpass filter to separate the sine wave from the rest of the signal. The display then has the form indicated in Figure 3/J.61 and the line-time non-linear distortion is indicated by changes in amplitude across the display.

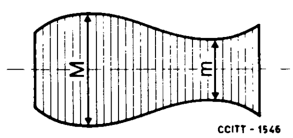


FIGURE 3/J.61

The non-linear distortion should be expressed as a percentage, in the form of 100 (1 – m/M) and should not be more than 20% for the hypothetical reference circuit. Alternatively, the result may, if desired, be expressed in dB in the form (20 $\log_{10}$ M/m) and for the hypothetical reference circuits should not exceed 2 dB.

For System M (Canada and the United States of America), the non-linear distortion is measured with a superimposed sine wave of 0.143 volts peak-to-peak at 3.6 MHz, and the results are expressed either as a percentage or in dB and should not be more than 13% or 1.2 dB respectively.

For colour System M (Japan), using the same test signal, the differential gain should not exceed 10%, and the differential phase should not exceed 5°.

3.4.3 *Short-time non-linear distortion of the picture signal*

This matter is still under study [5].

In Canada and the United States of America, the short-time non-linear distortion requirement is covered by the non-linearity distortion requirement given in 3.4.2 above.

[4] The corresponding terms in French are respectively : *distorsion de non-linéarité aux fréquences très basses, aux fréquences moyennes, aux fréquences élevées.*

[5] In several countries, such measurements are at present being made using Test Signal No. 3 with a higher value than $0.2 f_c$ for the frequency of the superimposed sine wave (see Doc. CMTT/41, Monte Carlo, 1958 – Chairman's Report).

3.4.4 *Non-linear distortion of the synchronizing signal*

For the hypothetical reference circuit, when the gain of the circuit is 0 dB, the amplitude, S, of the line-synchronizing signal, measured with Test Signal No. 3 (described in Annex 1), should lie between the limits of 0.21 volts and 0.33 volts (0.26 volts and 0.31 volts for Canada and the United States of America), irrespective of whether the intermediate lines are at black level, S_a, or at white level, S_b.

3.5 *Linear waveform distortion*

3.5.1 *Field-time waveform distortion*

3.5.1.1 Systems B, C, D, E, F, G, H, K, L

For the hypothetical reference circuit, using Test Signal No. 1 (described in Annex 1) the received waveform displayed on an oscilloscope should lie within the limits of the mask shown in Figure 4/J.61, provided that the oscilloscope is adjusted so that the half-amplitude points of the bar transitions coincide with M_1 and M_2, and the midpoints of the "black" and "white" portions coincide with A and B respectively.

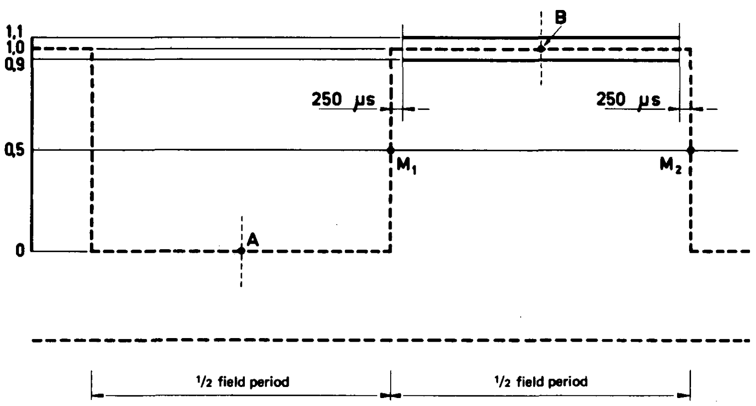


FIGURE 4/J.61 – Waveform response to Test Signal No. 1

3.5.1.2 System M

In Canada and the United States of America, with Test Signal No. 1, the variations about the level B should not exceed ± 5% when the signal is unclamped or ± 1% when the signal is clamped.

In Japan, with Test Signal No. 1, the tolerances are the same as for the 625- and 819-line systems.

3.5.2 *Line-time waveform distortion*

3.5.2.1 System M

In Canada and the United States of America, for the hypothetical reference circuit, using Test Signal No. 2 (described in Annex 1) with a rise-time of $2T$ (0.25 µs), the received waveforms displayed on an oscilloscope should lie within the limits of the corresponding mask, similar to that shown in Figure 5/J.61, but with a permitted variation about the level B of ± 1% provided that the oscilloscope is adjusted so that the half-amplitude points of the bar transitions coincide with M_1 and M_2, and the midpoints of the "black" and "white" portions coincide with A and B respectively.

In Japan, the conditions described below for the 625- and 819-line systems apply.

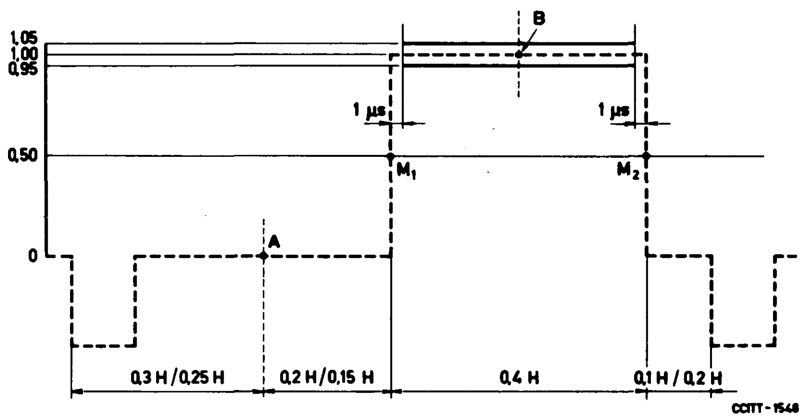


FIGURE 5/J.61 – Waveform response to Test Signal No. 2

3.5.2.2 Systems B, C, D, E, F, G, H, K, L

For the hypothetical reference circuit, using Test Signal No. 2 (described in Annex 1), with a rise-time of T (it may be necessary to use a rise-time of $2T$ for circuits which cut off sharply close to the nominal upper video-frequency limit), the received waveform displayed on an oscilloscope should lie within the limits of the mask shown in Figure 5/J.61, provided that the oscilloscope is adjusted so that the half-amplitude points of the bar transitions coincide with M_1 and M_2, and the midpoints of the "black" and "white" portions coincide with A and B respectively.

3.5.3 *Short-time waveform distortion*

3.5.3.1 System M

In Canada and the United States of America, where a test signal comprising a sine-squared pulse of half-amplitude duration $1/(2f_c)$s is used, the output signal should have a first-overshoot amplitude (negative), leading or trailing, not greater than 13% of the peak amplitude of the pulse.

In Japan, the test procedure is the same as that described for Systems B, C, D, E, F, G, H, K, L, the response being observed by means of the mask shown in Figure 6/J.61. For the chrominance channel, further study is required.

3.5.3.2 Systems B, C, D, E, F, G, H, K, L

Test Signal No. 2 is used, with a rise-time of $T = 1/(2f_c)$.

The response is observed by means of one of the masks shown in Figures 7/J.61 and 8/J.61, the oscilloscope being adjusted so that M coincides with the middle of the rise, and the black and white levels coincide with the segments α and β.

If ringing is present in the regions α and β, the peaks of the oscillations should be set symmetrically with respect to α and β.

For the hypothetical reference circuit, the response should be within the limits of the appropriate mask as follows:

– Figure 7/J.61 for Systems D, K.

– Figure 8/J.61 for Systems B, C, E, F, G, H (see Note).

µs	Upper limit	Lower limit
0.12	120	90
0.35	109	91
⩾ 1	105	95

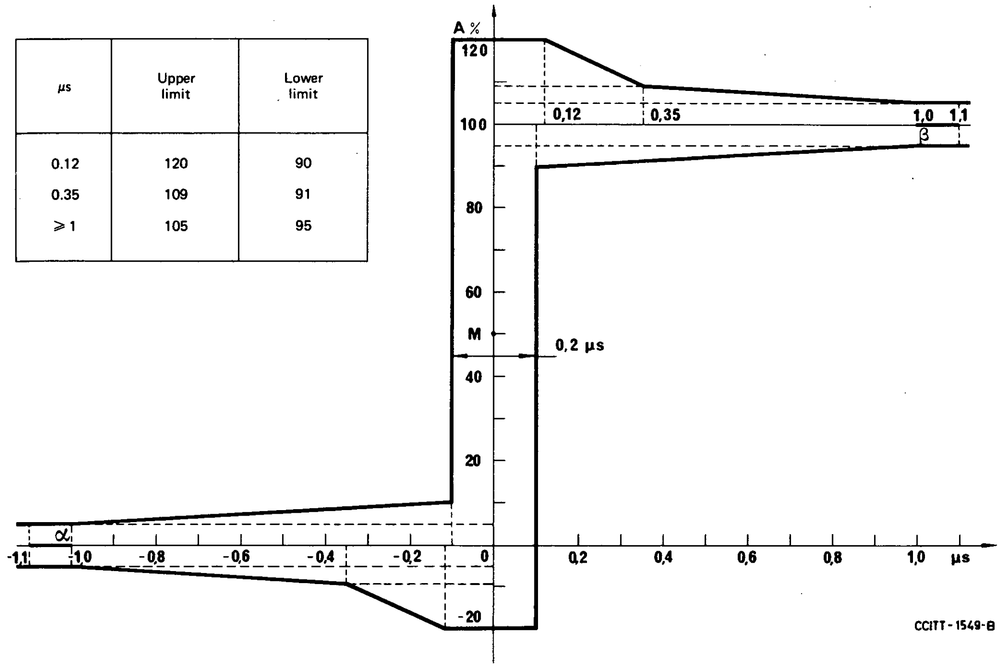


FIGURE 6/J.61 – **Mask for waveform response to Test Signal No. 2 for System M (Japan)**

µs	Upper limit	Lower limit
0.08	120	90
0.28	109	91
⩾ 1	105	95

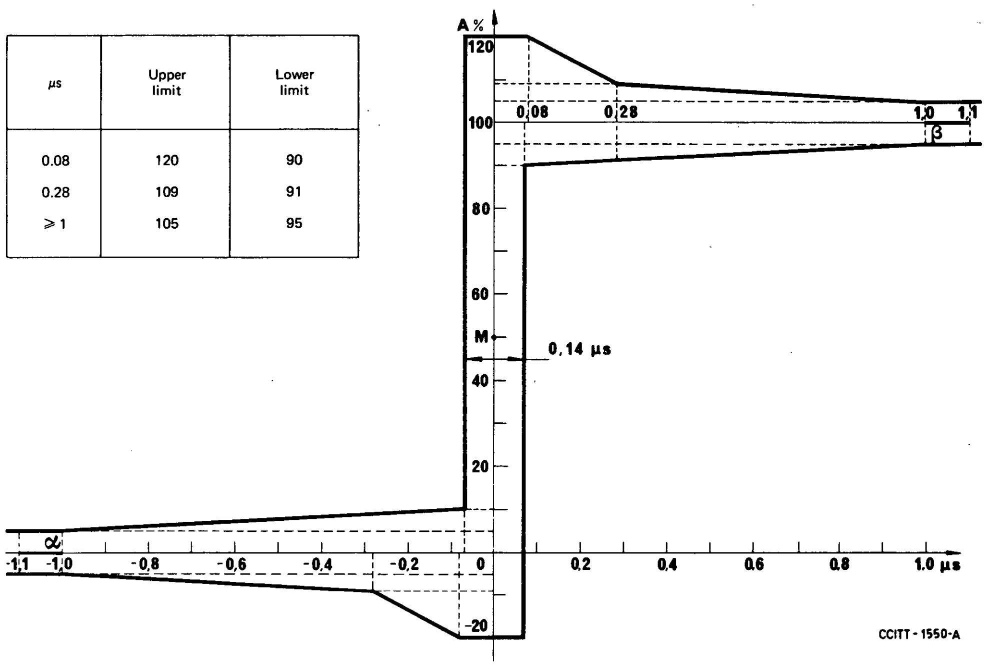
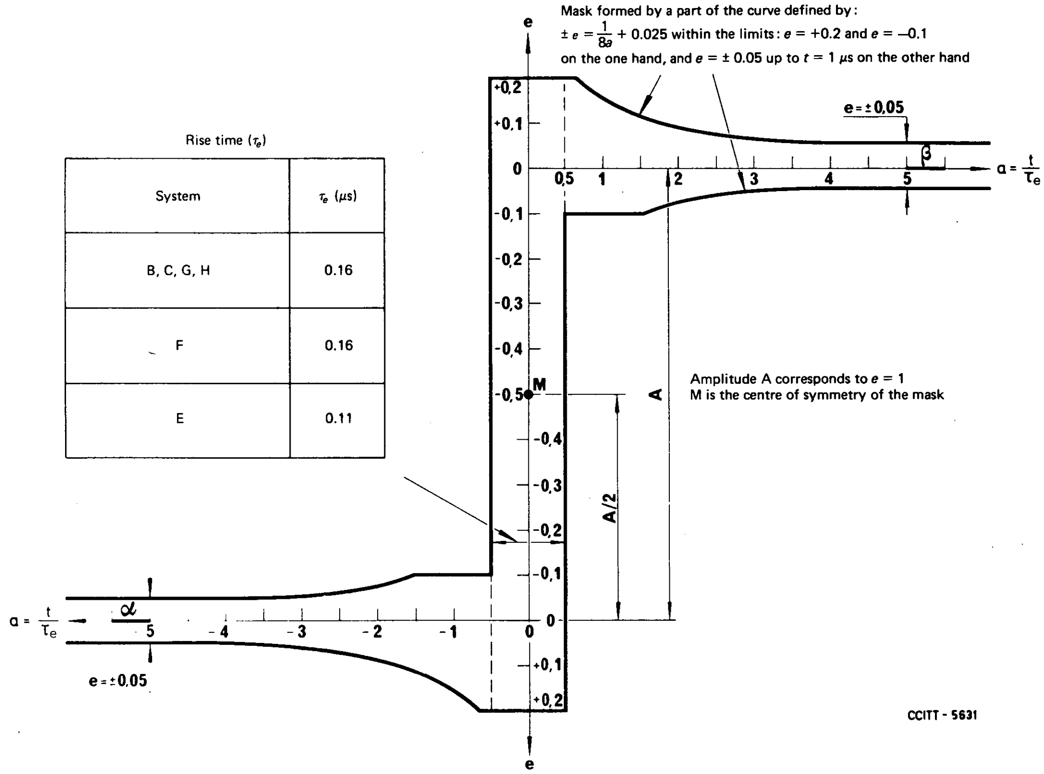


FIGURE 7/J.61 – **Provisional mask for waveform response to Test Signal No. 2 for Systems D, K**



System	τ_e (μs)
B, C, G, H	0.16
F	0.16
E	0.11

FIGURE 8/J.61 – Mask for waveform response to Test Signal No. 2 of Systems B, C, E, F, G, H

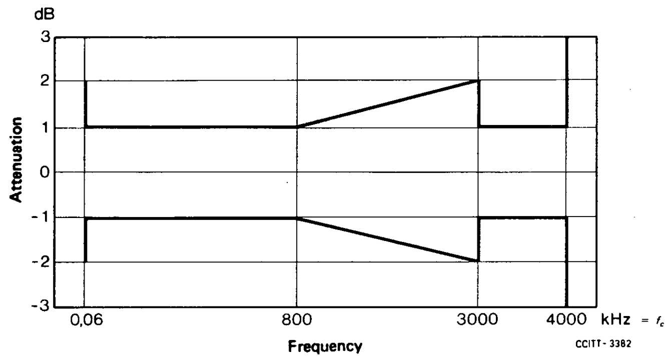
Note. – For the 625-line System L, the mask for the waveform response to Test Signal No. 2 is provisionally the mask of Figure 8/J.61 corresponding to the 819-line System E (f_c = 10 MHz).

3.6 *Steady-state characteristics*

3.6.1 *System M*

In Canada and the United States of America, the design-objective limits are shown by the lines B in Figures 10/J.61 and 11/J.61, the lowest frequency to which these limits apply being 0.0025 f_c.

In Japan the limits are as indicated below for the 625-line and 819-line systems, the appropriate value of f_c being 4 MHz. For colour, the attenuation/frequency limits are indicated in Figure 9/J.61; the envelope-delay/frequency limits require further study.

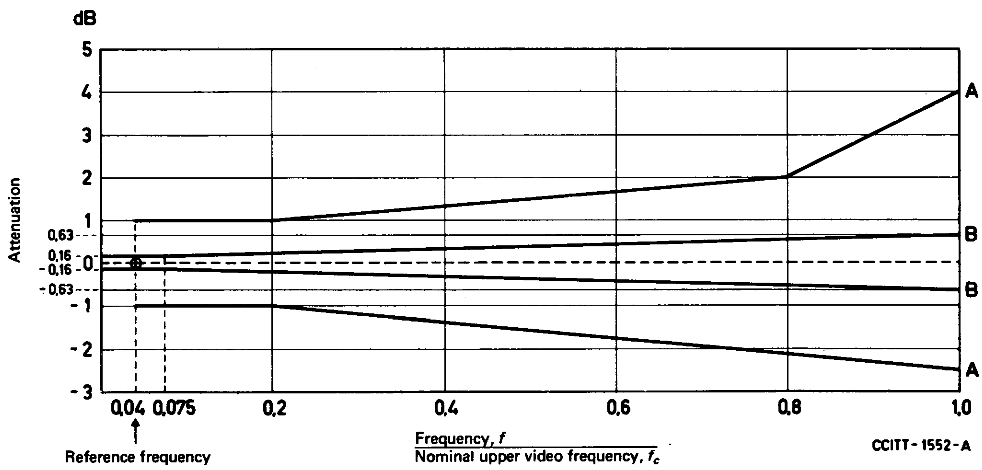


Note. – f_c : nominal upper limit of video-frequency band.

FIGURE 9/J.61 – **Limits for the attenuation/frequency characteristic of System M for colour television (Japan)**

3.6.2 *Systems B, C, D, E, F, G, H, K, L*

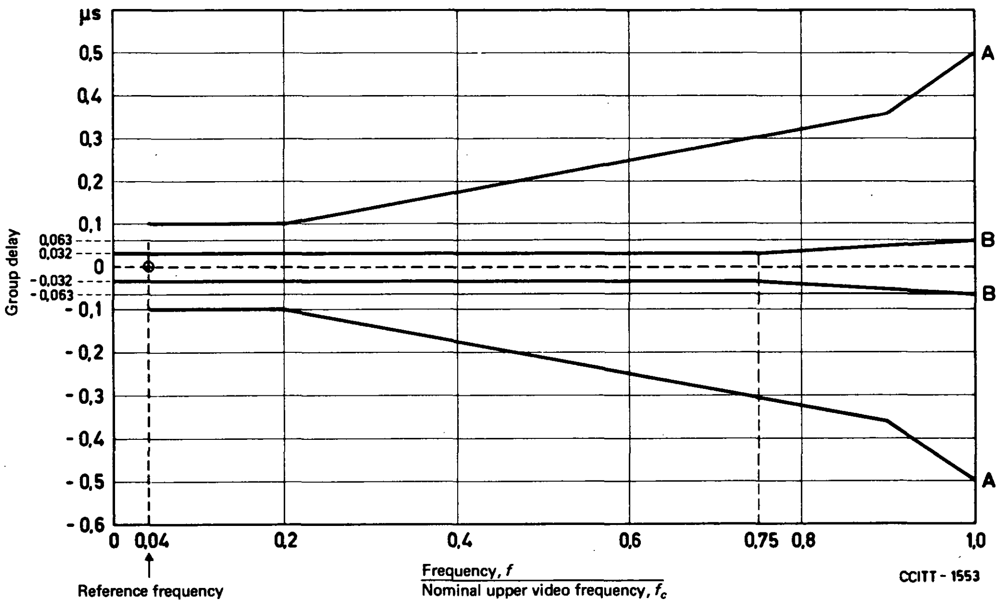
For the hypothetical reference circuit, the limits of the attenuation/frequency and group-delay frequency characteristics given in Figures 10/J.61 and 11/J.61 may be found useful by designers. In these figures, the abscissae show a single parameter which is the ratio between the frequency and the nominal upper video-frequency, f_c, of the system considered (normalized frequency).



Curves A: with nominal upper limits of the video-frequency band $f_c = 4$ MHz, System M (Japan); 5 MHz, Systems B, C, F, G, H; 6 MHz, Systems D, K, L; 10 MHz, System E.

Curves B: for System M (Canada and the United States of America), $f_c = 4$ MHz.

FIGURE 10/J.61 – **Limits for the attenuation/normalized-frequency characteristic for television systems**



Curves A: with nominal upper limits of the video-frequency band f_C = 4 MHz, System M (Japan); 5 MHz, Systems B, C, F, G, H; 6 MHz, Systems D, K, L; 10 MHz, System E.

Curves B: for System M (Canada and the United States of America), f_C = 4 MHz.

FIGURE 11/J.61 – **Limits for the envelope-delay/normalized-frequency characteristic for television systems**

ANNEX 1

(to Recommendation J.61)

Test signals

1. *Test Signal No. 1*

Test Signal No. 1 is used in the measurement of field-time waveform distortion. As shown in Figure 1, it comprises a square wave of field frequency superimposed upon line-synchronizing and blanking pulses. If desired, a field-synchronizing signal may be included and the pedestal may be omitted.

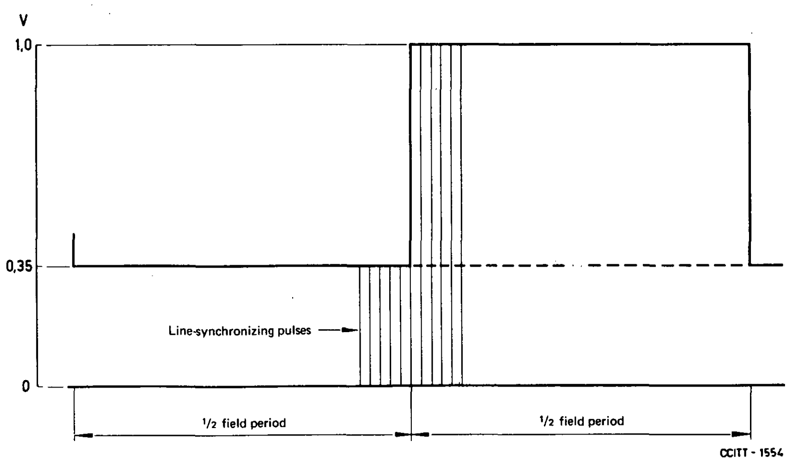


FIGURE 1 – **Test Signal No. 1**

2. *Test Signal No. 2* [6]

Test Signal No. 2 is used in the measurement of insertion gain, line-time waveform distortion and short-time waveform distortion. As shown in Figure 2, it comprises a half-line bar associated with line-synchronizing pulses. If desired, a field-synchronizing signal may be included. The interval between the half-line bar and the succeeding synchronizing pulse may be either 0.1 H or 0.2 H, where H is the line period. The pedestal may be omitted if desired.

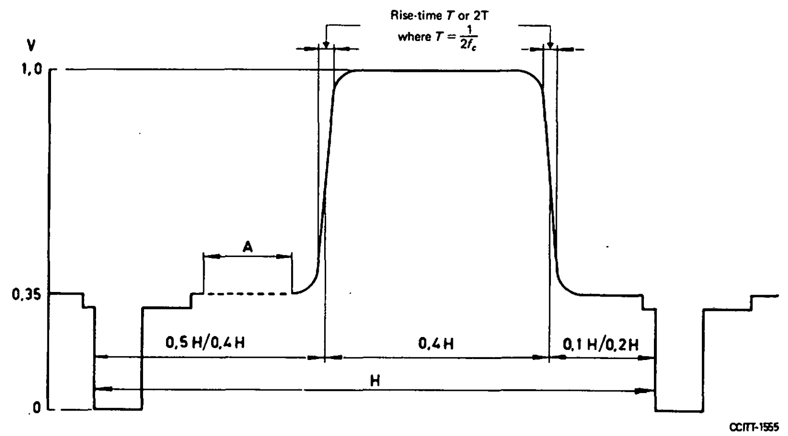


FIGURE 2 – Test Signal No. 2

The precise shape and rise time of each transition of the half-line bar may be determined by means of a shaping network, the design of which is based on "Solution 3" in a paper by W. E. Thomson [*Proc. I.E.E.*, Part III, 99, 373 (1952)]. Two alternative networks may be used giving rise times of T and $2T$, where $T = 1/(2f_c)$, and f_c is the nominal upper video-frequency limit of the system. (Annex IV of the paper by Thomson contains a description of the appropriate network.)

If desired, an additional feature such as a sine-squared pulse, of shape and half-amplitude duration determined by the above-mentioned shaping networks, or a high-frequency burst, can be added in the space marked A. For Systems D and K, a sine-squared pulse of half-amplitude duration T or $2T$ is used.

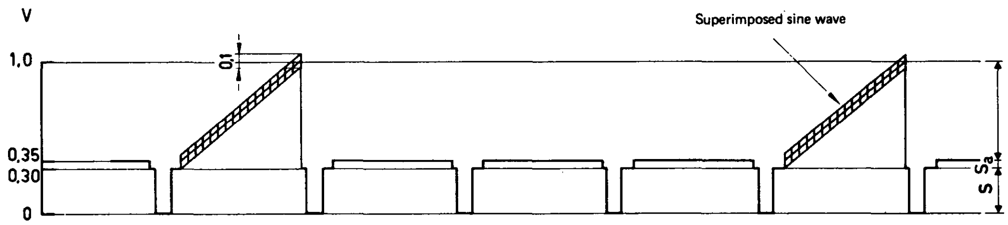
3. *Test Signal No. 3* [6]

Test Signal No. 3 is used in the measurement of non-linear distortion. As shown in Figure 3, it is a signal in which the "picture" portion of every fourth line consists of a sine wave of 0.1 volts peak-to-peak amplitude superimposed on a saw-tooth, the three intermediate lines being set either to black level or to white level by means of a switch at the sending end. If desired, a field-synchronizing signal may be included and the pedestal may be omitted.

For measuring line-time non-linear distortion, the frequency of the superimposed sine wave is 0.2 f_c.

At the receiving end of a circuit, any variation of the sine-wave amplitude over the duration of the saw-tooth is taken as indicative of non-linear distortion.

[6] Considerable errors in measurement occur when using Test Signals Nos. 2 and 3, if the signal-to-noise ratio is less than 30 dB (see Doc. CMTT/2, Paris, 1962).



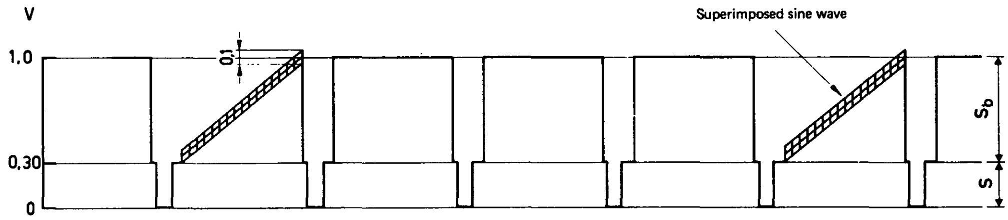


FIGURE 3 – **Test Signal No. 3**

ANNEX 2

(to Recommendation J.61)

Lowpass filter for use in measurements of continuous random noise

ANNEX 2

(to Recommendation J.61)

Lowpass filter for use in measurements of continuous random noise

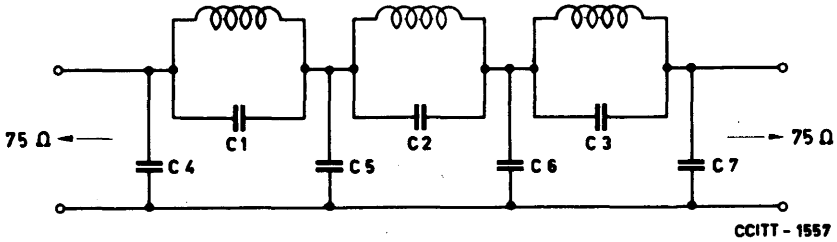


FIGURE 1 – **Lowpass filter**

TABLE 1

	Nominal upper video-frequency limit: f_C (MHz)[a]		
	L (μH)	C (pF)	f (MHz)
1	$14.38/f_C$	$497.6/f_C$	$1.8816\, f_C$
2	$7.673/f_C$	$2723/f_C$	$1.1011\, f_C$
3	$8.600/f_C$	$1950/f_C$	$1.2290\, f_C$
4		$2139/f_C$	
5		$2815/f_C$	
6		$2315/f_C$	
7		$1297/f_C$	

[a] For System M (Canada and the United States of America), a value of f_C = 4.2 MHz is adopted for the design of the lowpass filter used for noise measurement.

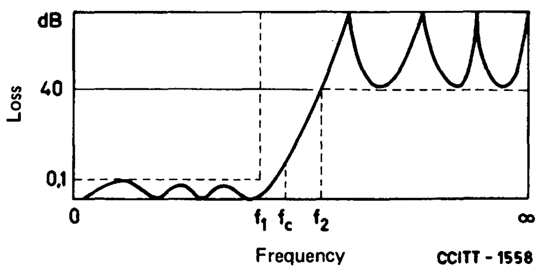
Note 1. – Each capacitance quoted is the total value, including all relevant stray capacitances, and should be correct to ± 2%.

Note 2. – Each inductor should be adjusted to make the insertion loss a maximum at the appropriate indicated frequency, f (MHz).

Note 3. – The theoretical insertion loss curve of Figure 2 corresponds to an infinite Q-factor. In practice, Q should be at least of the order of 100 at frequency f_C.

Note 4. – Limits for the insertion-loss/frequency characteristics are specified indirectly by the indicated tolerances on the component values.

f/f_c	dB	f/f_c	dB	f/f_c	dB
0.98	0.1	1.02	7.3	1.06	23.0
0.99	0.5	1.03	10.9	1.07	27.7
1.00	1.8	1.04	14.8	1.08	33.3
1.01	4.2	1.05	18.8	1.09	41.0



Theoretical insertion loss f_1 = $0.9\, f_C$ by design.

Ringing frequency = f_C by design.

$f_1 = 0.9807\, f_C$

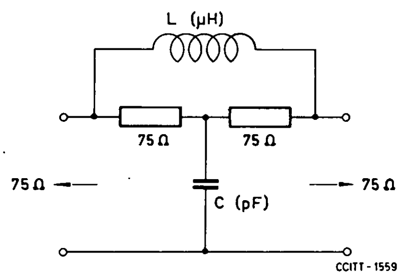
$f_2 = 1.0897\, f_C$

FIGURE 2 – **Lowpass filter characteristic**

ANNEX 3

(to Recommendation J.61)

Continuous random-noise weighting networks



$$L\ (\mu H) = 75\ \tau(\mu s)\ ;\quad C\ (pF) = \frac{\tau(\mu s)}{75} \cdot 10^6$$

$$\text{Insertion loss (dB)} = 10\ \log_{10}\ [1 + (2\pi\ \tau f)^2]$$

FIGURE 1 – **Weighting network**

TABLE 1

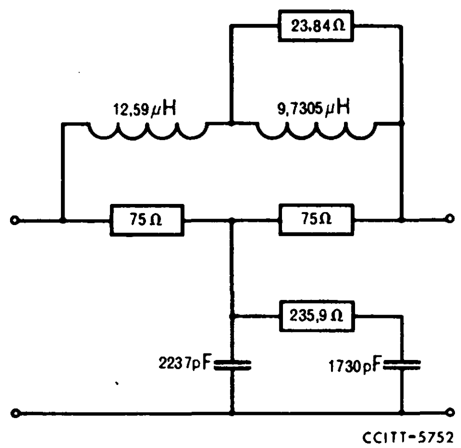
System	f_c (MHz)[a]	τ (μs)	τf_c	Theoretical weighting (dB), for	
				"White" noise	"Triangular" noise
M (Canada and U.S.A.)	See Note 1			6.1	10.2
M (Japan)	4	0.415	1.66	8.5	16.3
B, C, G, H	5	0.33	1.66	8.5	16.3
D, K, L	6	0.33	2.0	9.3	17.8
F	5	0.33	1.66	8.5	16.3
E	10	0.166	1.66	8.5	16.3

[a] f_c is the nominal upper video-frequency limit of the system (MHz).

Note 1. – For System M (Canada and the United States of America), the following weighting characteristic is used:

Frequency (MHz)	0.01	0.05	0.10	0.50	1.00	2.00	3.00	4.00
Weighting (insertion loss) (dB)	0	0	0.3	2.8	4.7	8.1	10.8	13.0

A weighting network, such as that shown in Figure 2, may be used:



$$\text{Insertion loss (dB)} = 10 \log_{10} \frac{[1 + (f/f_1)^2]\,[1 + (f/f_2)^2]}{[1 + (f/f_3)^2]}$$

where: $f_1 = 0.270$ MHz, $f_2 = 1.37$ MHz and $f_3 = 0.390$ MHz

FIGURE 2 – **Example of a weighting network**

Note. – For Colour System M (Japan), the weighting curve of Figure 3 is used for colour [see WATANABE, K. Effects of continuous random noise on colour television pictures. *Electrical Telecomm. Laboratory.* Report No. 1528, N.T.T., Japan (1964)].

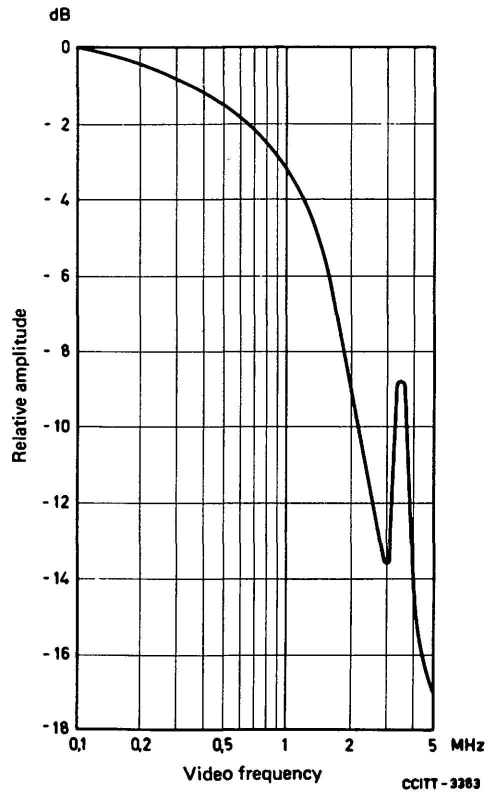


FIGURE 3 – **Weighting curve for continuous random noise of a 525-line television system**

ANNEX 4

(to Recommendation J.61)

Circuits having more or fewer video sections than the hypothetical reference circuit

1. *Introduction*

The purpose of this Annex is to give some indication of the design objectives of hypothetical circuits that have more or fewer video-to-video sections than the three sections of the hypothetical reference circuit defined in 1.2 of the text. The values calculable from Tables 1 and 2 provide only indications of the probable design objectives. These values should be used with caution when considering specifications of actual circuits, because the law of addition is not precisely known for every type of impairment.

TABLE 1

n	$\left(\frac{n}{3}\right)^{1/h}$		
	$h = 1$	$h = 3/2$	$h = 2$
1	0.33	0.48	0.58
2	0.67	0.76	0.82
3	1.00	1.00	1.00
4	1.33	1.21	1.15
5	1.67	1.41	1.29
6	2.00	1.59	1.41
7	2.33	1.76	1.53
8	2.67	1.92	1.63
9	3.00	2.08	1.73
10	3.33	2.23	1.83
11	3.67	2.38	1.91
12	4.00	2.52	2.00
13	4.33	2.66	2.08
14	4.67	2.79	2.16
15	5.00	2.92	2.24

TABLE 2

Reference to text of this Recommendation	Characteristic	D_3 expressed in	h	Notes
3.1	*Insertion gain* (tolerance)	dB	2	
3.2	*Insertion gain variation* short period medium period	 dB dB	 2 2	
3.3.1	*Continuous random noise*			1
3.3.2	*Periodic noise* Power-supply hum 1 kHz to 1 MHz 1 MHz to f_c	 amplitude of noise	 2 2 2	 2 ; 3 4 4
3.3.3	*Impulsive noise*	amplitude of noise		5
3.4 3.4.2 3.4.4	*Nonlinear distortion* Picture signal Synchronizing signal	 $1 - \left(\frac{m}{M}\right) \times 100\,\%$ %	 3/2 3/2	 3
3.5 3.5.1 3.5.2 3.5.3	*Linear waveform distortion* Field-time Line-time Short-time overshoot and ringing Rise-time	 % % mask µs	 1 2 2 no law	 3 6 ; 3 6 ; 3 3
3.6	*Steady-state characteristics* Attenuation/frequency Envelope-delay/frequency	 dB µs	 3/2 3/2	 7 7

Note 1. – For circuits on coaxial cables, quadratic addition ($h = 2$) applies to random noise expressed in terms of r.m.s. voltage. For circuits on radio-relay links, see CCIR Recommendation 289-1.

Note 2. – Considering the probability of arithmetic addition of power-supply hum in circuits of few sections, it may be advisable to put $h = 1$ when $n \leqslant 3$.

Note 3. – Further information is given in Doc. CMTT/49 (OIRT), 1966-1969.

Note 4. – Considering the probability of arithmetic addition when periodic noise consists of a few components that are very close in frequency, it may be advisable to put $h = 1$, when the number of such components is small.

Note 5. – When each of a number of sources of impulsive noise is operative for a small percentage of the time (e.g. < 0.1 %), arithmetic addition of the percentage will apply.

Note 6. – For Systems D and K, the method outlined in Doc. CMTT/60, 1963-1966, could be used.

Note 7. – In Canada and the United States of America, the practice is to use $h = 2$.

2. *Laws of addition*

If D_3: design objective as expressed in this Recommendation, or the parameter derived therefrom and indicated in Table 2, permitted in the hypothetical reference circuit,

and D_n: design objective, or the parameter mentioned above, permitted in n sections,

then $$D_n = D_3 \left(\frac{n}{3}\right)^{1/h},$$

where h has the value 1, 3/2 or 2 in accordance with Table 2; $h = 1$ gives linear or arithmetic law of addition, $h = 3/2$ gives the "three-halves power" law of addition, and $h = 2$ gives quadratic addition or the root sum of squares.

Calculated values of $\left(\frac{n}{3}\right)^{1/h}$, are given in Table 1.

3. *Examples of the use of Tables 1 and 2*

3.1 In the hypothetical reference circuit, if the tolerance on gain is $\pm$ 1 dB, the tolerance on gain for a video section will (with $h = 2$) be:

$$D_1 = D_3 \left(\frac{1}{3}\right)^{1/2} = D_3 \times 0.58 = \pm 0.58 \text{ dB}.$$

3.2 In the hypothetical reference circuit, if the tolerance on the signal-to-noise ratio is 50 dB, the tolerance on the signal-to-noise ratio for a 9-section circuit will be calculated as follows (with $h = 2$):

noise amplitude for the hypothetical reference circuit: D_3

noise amplitude for the 9-section circuit:

$$D_9 = D_3 \left(\frac{9}{3}\right)^{1/2} = D_3 \times 1.73$$

signal-to-noise for the 9-section circuit:

$$\frac{S}{D_9} = \frac{S}{D_3} \times \frac{1}{1.73}$$

or, in dB: $\left(\frac{S}{D_9}\right) \text{dB} = 50 - 4.8$, i.e. about 45 dB.

3.3 In the hypothetical reference circuit, if the tolerance on non-linearity is 20%, the tolerance on non-linearity for a video section will be (with $h = 3/2$):

$$D_1 = D_3 \left(\frac{1}{3}\right)^{2/3} = D_3 \times 0.48$$

$$D_1 = 20 \times 0.48 = 9.6\ \%.$$

Recommendation J.62

SPECIFICATION FOR A LONG-DISTANCE TELEVISION TRANSMISSION
(System I only) [7), 8)]

(Mar del Plata, 1968)

The CCITT,

considering

the agreement reached by the Joint CCIR/CCITT Committee for television and sound transmission (CMTT) on a draft Recommendation concerning television transmission over long distances, common to the CCIR and the CCITT;

unanimously recommends

that, taking account of the definitions in A. below, television transmissions over long distances for System I should satisfy the requirements laid down in B. and its Annex.

The requirements for the transmission of other systems are contained in Recommendation J.61. The existence of this Recommendation does not necessarily imply that the requirements for other systems will later be included in this Recommendation or that their requirements will be changed in form.

A. DEFINITIONS

1. *Definition of a long-distance international television connection* (see Figure 1/J.61)

1.1 Point A, to be considered as the sending end of the international television connection, may be the point at which the programme originates (studio or outside location), a switching centre or the location of a standards converter.

1.2 Point D, to be considered as the receiving end of the international television connection, may be a programme mixing or recording centre, a broadcasting station, a switching centre or the location of a standards converter.

1.3 The local line AB connects point A to the sending terminal station, point B, of the international television circuit.

1.4 The long-distance international television circuit BC, comprises a chain of national and international television links. The precise locations (e.g. within buildings), to be regarded as the points B and C, will be nominated by the Administrations concerned.

1.5 The local line CD connects point C, the receiving terminal station of the long-distance international television circuit, to the point D.

1.6 The combination AD, of the long-distance international television circuit BC, and the local lines AB and CD, constitutes the *international television connection.*

2. *Definition of the hypothetical reference circuit*

The main feature of the television hypothetical reference circuit, which is an example of a long-distance international television circuit (BC in Figure 1/J.61) and which may be of either radio or coaxial-cable type, are:

- the overall length between video terminal points is 2500 km (about 1600 miles);
- two intermediate video points divide the circuit into three sections of equal length;

7) This Recommendation corresponds to CCIR Recommendation 451-2.

8) For the other systems see Recommendation J.61.

- the three sections are lined up individually and then interconnected without any form of overall adjustment or correction;
- the circuit does not contain a standards converter or a synchronizing-pulse regenerator.

Note. – The Annex to A. gives a provisional indication of the characteristics of circuits having more or fewer sections than the hypothetical reference circuit.

ANNEX

(to A. of Recommendation J.62)

Circuits having more or fewer sections than the hypothetical reference circuit

1. *Introduction*

The purpose of this Annex is to give some indication of the design objectives of hypothetical circuits that have more or fewer video-to-video sections than the three of the hypothetical reference circuit defined in A.2 of the text. The values calculable from Tables 1 and 2 provide only indications of the probable design objectives, which should not be used directly when studying the design of equipment because the law of addition is not precisely known for every type of impairment.

TABLE 1

n	$\left(\frac{n}{3}\right)^{1/h}$		
	$h = 1$	$h = 3/2$	$h = 2$
1	0.33	0.48	0.58
2	0.67	0.76	0.82
3	1.00	1.00	1.00
4	1.33	1.21	1.15
5	1.67	1.41	1.29
6	2.00	1.59	1.41
7	2.33	1.76	1.53
8	2.67	1.92	1.63
9	3.00	2.08	1.73
10	3.33	2.23	1.83
11	3.67	2.38	1.91
12	4.00	2.52	2.00
13	4.33	2.66	2.08
14	4.67	2.79	2.16
15	5.00	2.92	2.24

TABLE 2

Reference to B.4 of the text	Characteristic	D_3 expressed in	h	Notes
4.1	*Insertion gain* (error)	dB	2	
4.2	*Insertion gain variations*	dB	2	
	Continuous random noise			
4.3.1	Luminance channel			1
4.3.2	Chrominance channel			1
	Periodic noise			
4.4	Power-supply hum	noise voltage	2	2
4.4	Single-frequency	noise voltage	2	3
4.5	*Impulsive noise*	noise voltage		4
4.6	*Crosstalk*	crosstalk voltage	3/2	
	Nonlinear distortion of the picture signal			
4.7.1	Luminance channel	%	3/2	
	Chrominance channel			
4.7.2	Differential gain	%	3/2	
4.7.2	Differential phase	degrees	3/2	
4.8	*Nonlinear distortion of the synchronizing signal*	%	3/2	
	Linear waveform distortion			
4.9.1	Luminance channel	%	3/2	
4.9.2	Chrominance channel	%	3/2	
	Luminance-chrominance inequalities			
4.10.1	Gain inequality	%	2	5
4.10.2	Delay inequality	ns	2	5
	Steady-state characteristics			
–	Attenuation/frequency			
–	Envelope-delay/frequency			

2. *Laws of addition*

If D_3: design objective as expressed in this Recommendation, or the parameter derived therefrom and indicated in Table 2, permitted in the hypothetical reference circuit,

and D_n: design objective, or the parameter mentioned above, permitted in n sections,

then
$$D_n = D_3 \left(\frac{n}{3}\right)^{1/h}$$

where h has the value 1, 3/2 or 2 in accordance with Table 2; $h = 1$ gives linear or arithmetic law of addition, $h = 3/2$ gives the "three-halves power" law of addition, and $h = 2$ gives quadratic addition or the root sum of squares.

Calculated values of $\left(\frac{n}{3}\right)^{1/h}$, are given in Table 1.

Note 1. – For circuits on coaxial cables, quadratic addition ($h = 2$) applies to random noise expressed in terms of r.m.s. voltage. For circuits on radio-relay links, see CCIR Recommendation 289-1.

Note 2. – Considering the probability of arithmetic addition of power-supply hum in circuits of few sections, it may be advisable to put $h = 1$ when $n \leqslant 3$.

Note 3. – Considering the probability of arithmetic addition when periodic noise consists of a few components that are very close in frequency, it may be advisable to put $h = 1$ when the number of such components is small.

Note 4. – When each of a number of sources of impulsive noise is operative for a small percentage of the time (e.g. $< 0.1\%$), arithmetic addition of the percentage will apply.

Note 5. – Quadratic addition ($h = 2$) for gain and delay inequalities is based on the assumption that positive and negative values are made equally likely by the use of correcting networks or equivalent means.

B. REQUIREMENTS FOR SYSTEM I

1. *Introduction*

In the following are given the methods of testing and the limits and tolerances applicable to the hypothetical reference circuit for System I, i.e. the 625-line system having a nominal video bandwidth of 5.5 MHz.

2. *Basic concepts*

The requirements are based upon two concepts. The first follows from the fact that the composite colour signal may be regarded as the sum of a luminance signal (similar to a monochrome signal, including the line- and field-synchronizing pulses) and a chrominance signal (the modulated subcarrier conveying the hue and saturation information, and the colour burst). A colour-television link may therefore be regarded as the combination of a "luminance channel" and a "chrominance channel" in overlapping frequency bands. For both specifying and testing purposes, it is convenient to deal with:

– the permissible distortion and noise impairments in these two channels taken separately;
– the permissible inequalities of gain and delay of the two channels taken together.

The requirements for the luminance channel are assumed to be identical with those for monochrome transmission.

The second concept is based upon the consideration that it is sufficient in practice to specify and test the performance of the chrominance channel as though it were intended to carry a simple double-sideband amplitude modulated signal. The test signals thus include subcarrier elements modulated by waveforms chosen to suit the nominal bandwidth of the chrominance channel.

Applying these concepts to System I for the purpose of this Recommendation, the luminance-channel band is deemed to extend up to 5 MHz, and the chrominance-channel band from approximately 3.5 to 5.5 MHz, i.e. the baseband of the chrominance signal is deemed to extend up to 1 MHz. These assumptions do not imply restrictions upon the transmission of any luminance-signal components in the range 5.0 to 5.5 MHz, or chrominance-signal components below 3.5 MHz.

3. *General requirements*

3.1 *Impedance*

At points of video interconnection, the input and output impedance of each link should be unbalanced to earth with a nominal value of 75 ohms resistive and a return loss of at least 30 dB relative to 75 ohms.

The conventional frequency-domain interpretation of this requirement is that the return loss should be at least 30 dB at any frequency within the video band. A time-domain interpretation is, however, more convenient and useful because the technique of measurement is simpler and the results are more directly related to the picture impairments caused by mismatched impedances. The "waveform return loss", as it may be termed, is measured with a television-type test signal and the result is taken as the ratio, expressed in dB, of the peak-to-peak voltages of the "picture" portions of the incident and reflected waveforms. This result is numerically the same as the conventional one if the return loss is independent of frequency. Provisionally, it is required that the waveform return loss, relative to 75 ohms, shall be at least 30 dB when measured with each of the test signals shown in Figures 1/J.62, 2/J.62 and 4/J.62.

3.2 *Polarity and d.c. component*

At points of video interconnection, the polarity of the signal should be "positive", i.e. such that black-to-white transitions are positive-going.

The useful d.c. component, which is related to the average luminance of the picture, may or may not be contained in the signal and need not be transmitted or delivered at the output.

Any non-useful d.c. component unrelated to the signal (e.g. the component due to d.c. supplies) should not cause more than 100 mW to be dissipated in a 75-ohm load impedance. If the load impedance is disconnected, the voltage of this component should not exceed 5 volts.

3.3 *Signal amplitude*

At points of video interconnection, the nominal peak-to-peak amplitude of the picture luminance signal, between blanking level and white level, should be 0.7 volts, and the nominal amplitude of the synchronizing pulses should be 0.3 volts. The nominal peak-to-peak amplitude of the video signal is thus 1.0 volt although it is recognized that this value may occasionally be exceeded during transmission of colour signals.

4. *Transmission performance requirements*

4.1 *Insertion gain*

Insertion gain should be measured under the following conditions. At the sending end, a 75-ohm generator of the $2T$ pulse-and-bar test signal shown in Figure 1/J.62 should be adjusted so that, if connected directly to a 75-ohm load, the bar amplitude would be 0.7 volts and the synchronizing pulse amplitude 0.3 volts. The sine-squared pulse is ignored in this application. At the receiving end, the bar amplitude (between the points A and B shown in Figure 10/J.62) should be measured with a 75-ohm oscilloscope. The ratio, expressed in dB, of this amplitude to 0.7 volts is taken as the insertion gain.

After initial or routine adjustment, the insertion gain should be within the limits 0 ± 0.5 dB.

4.2 *Variations in the insertion gain*

Any variations of insertion gain with time should not exceed the following limits:

- short-period variations (e.g. 1 s): ± 0.2 dB;
- medium-period variations (e.g. 1 h): ± 0.5 dB.

Long-period variations are not specified because they would generally be corrected by the normal routine adjustments.

The foregoing refers only to insertion gain as defined in 4.1 above. When considering variations of gain with time, the permissible limits of the luminance-chrominance gain inequality given in 4.10.1 below should not be overlooked.

4.3 *Continuous random noise*

The signal-to-weighted noise ratio for continuous random noise is defined as the ratio, expressed in dB, of the nominal peak-to-peak amplitude of the picture luminance signal to the r.m.s. amplitude of the noise measured under the following conditions:

- the noise is passed through a specified bandpass filter to delimit the effective frequency range, and also through a specified weighting network, or equivalent;
- the measurement is made with an instrument having, in terms of power, an effective time constant or integrating time of 1 s.

4.3.1 *Luminance channel*

The nominal frequency range is 10 kHz to 5 MHz. The lower limit is determined by the high-pass member of the junction filter shown in Figure 6/J.62. Its purpose is to exclude power-supply hum and microphony noise. The upper limit is determined by the lowpass filter shown in Figure 7/J.62. The weighting network is shown in Figure 8/J.62; it has a time constant of 200 ns giving a weighting effect of 6.5 dB for flat random noise and 12.3 dB for triangular random noise.

The signal-to-weighted noise ratio should not fall below 52 dB for more than 1% of any month nor below 44 dB for more than 0.1% of any month.

4.3.2 *Chrominance channel*

The nominal frequency range is 3.5 to 5.5 MHz, determined by the combined bandpass filter and weighting network shown in Figure 9/J.62. For For each subcarrier sideband, the filter provides a weighting effect which is approximately equal to that of the luminance weighting network in the 0- to 1-MHz band.

The signal-to-weighted noise ratio should not fall below 46 dB for more than 1% of any month nor below 38 dB for more than 0.1% of any month.

4.4 *Periodic noise*

The signal-to-noise ratio for periodic noise is defined as the ratio, expressed in dB, of the nominal peak-to-peak amplitude of the picture luminance signal to the peak-to-peak amplitude of the noise.

For power-supply hum including lower-order harmonics, the signal-to-noise ratio should not be less than 35 dB. The measurement is made through the lowpass member of the junction filter shown in Figure 6/J.62.

For single-frequency noise between 1 kHz and 5.5 MHz, the signal-to-noise ratio should not be less than 55 dB.

4.5 *Impulsive noise*

The signal-to-noise ratio for impulsive noise is defined as the ratio, expressed in dB, of the nominal peak-to-peak amplitude of the picture luminance signal to the peak-to-peak amplitude of the noise.

For impulsive noise of a sporadic or infrequently-occurring nature, the signal-to-noise ratio should not be less than 25 dB.

4.6 *Crosstalk*

Crosstalk between two circuits is measured with a specified video test signal applied to the input of the disturbing circuit and an oscilloscope at the output of the disturbed circuit, which is otherwise quiescent. The signal-to-crosstalk ratio is defined as the ratio, expressed in decibels, of the nominal peak-to-peak amplitude of the picture luminance signal to the peak-to-peak amplitude of the "picture" portion of the crosstalk waveform.

At present, definitive limits can be specified only for two particular cases; for other forms of crosstalk further study is required. The specifications given in the two following paragraphs are strictly applicable only when the disturbing circuit, as well as the disturbed circuit, is designed to transmit System I signals, but they may serve as a guide under comparable conditions of service with other systems.

If the crosstalk is substantially undistorted, the signal-to-crosstalk ratio should not be less than 58 dB when measured with the test signal shown in Figure 1/J.62 applied to the disturbing circuit.

If the crosstalk is substantially "differentiated" (i.e. crosstalk voltage proportional to frequency), the signal-to-crosstalk ratio should not be less than 50 dB when measured with the test signal shown in Figure 4/J.62 applied to the disturbing circuit.

4.7 *Nonlinear distortion of the picture signal*

Line-time nonlinearity distortions in the luminance and chrominance channels are measured with the test signal shown in Figure 3/J.62, consisting of a 5-riser staircase, with superimposed subcarrier, in every fourth line. Separate measurements are made with the three intermediate lines at black level and white level, and the higher value of distortion is taken as the result.

4.7.1 *Luminance channel*

At the receiving end, the test signal is passed through a differentiating and shaping network (see Doc. CMTT/3, Monte Carlo, 1958), whose effect is to eliminate the subcarrier and transform the staircase into a train of 5 pulses of approximately sine-squared shape with 2-μs half-amplitude duration. Comparing the amplitudes of the pulses, the numerical value of the distortion is found by expressing the difference between the largest and smallest amplitude as a percentage of the largest.

The distortion should not exceed 12%. In addition, when the test signal is sent at 3 dB above normal amplitude (i.e. 1.4 volts peak-to-peak), the distortion should not exceed 24%.

4.7.2 *Chrominance channel*

At the receiving end, the subcarrier is filtered from the rest of the test signal and its six sections are compared in amplitude and phase. Taking the blanking-level section of the subcarrier as reference, the differential gain is defined as the largest departure from the reference amplitude, expressed as a percentage, and the differential phase is defined as the largest departure from the reference phase-angle, expressed in degrees. (It seems desirable to seek a method of deriving numerical values which are more closely related to picture impairment.)

Provisionally, the differential gain should not exceed ± 8%, and the differential phase should not exceed ± 4°. In addition, when the test signal is sent at 3 dB above normal amplitude, the distortions should not exceed ± 16% and ± 8° respectively.

4.8 *Nonlinear distortion of the synchronizing signal*

The distortion is expressed in terms of percentage departure of the midpoint amplitude of the line-synchronizing pulse from its nominal amplitude, i.e. 0.3 volts for a circuit having zero insertion gain as defined in 4.1 above. Using the staircase test signal shown in Figure 3/J.62, separate measurements are made with the three intermediate lines at black level and white level, and the higher value of distortion is taken as the result.

The distortion should not exceed ± 10%. In addition, when the test signal is sent at 3 dB above normal amplitude, the distortion should not exceed ± 20%.

4.9 *Linear waveform distortion*

4.9.1 *Luminance channel*

The short-time, line-time and field-time linear distortions in the luminance channel are found from the waveform responses to the pulse-and-bar and 50-Hz square-wave test signals shown in Figures 1/J.62 and 2/J.62. The result is expressed as a rating factor *K* by the method described below in the Annex to B.

The rating factor should not exceed 3%.

4.9.2 *Chrominance channel*

The short-time and line-time linear distortions in the chrominance channel are found from the waveform responses to the pulse-and-bar modulated subcarrier test signal shown in Figure 4/J.62. The result may be expressed by a rating factor analogous to that of the luminance channel but a limit cannot be proposed until more experience has been gained.

4.10 *Luminance-chrominance inequalities*

Gain and delay inequalities between the luminance and chrominance channels are measured with the composite test signal shown in Figure 5/J.62. It consists essentially of added luminance and chrominance signal elements which are equal in single-peak amplitude and coincident in time. At the receiving end, two calibrated variable networks are adjusted to annul any inequality of amplitude or delay.

4.10 *Gain inequality*

The gain inequality, expressed as the percentage departure of the amplitude of the chrominance element from the amplitude of the luminance element, both measured at the midpoint of the bar, should not exceed ± 10%.

4.10.2 *Delay inequality*

The delay inequality should not exceed ± 100 ns.

ANNEX

(to B. of Recommendation J.62)

Linear waveform distortion, luminance channel

1. *Introduction*

This Annex describes two complementary methods of specifying the linear transmission performance of a luminance channel. The first, or "routine-test method", is rapid but less precise because it relies on direct oscilloscopic observation of the responses to prescribed test signals, and because the spectrum of one of these signals unavoidably extends beyond the nominal 5-MHz limit of interest. The second, or "acceptance-test method", is slow but more precise because a process of computation applied to a series of waveform ordinates enables irrelevant information to be eliminated and certain measuring equipment errors to be corrected.

The performance limits are given in terms of a rating factor K, for which numerical values are assigned in the individual specifications of links and equipment. Rating factors may range from 0.5% ($K = 0.005$) for a short-distance link up to several per cent for a chain of long-distance links.

2. *Routine-test method*

To meet a specified rating factor K, the responses to the pulse-and-bar and 50-Hz square-wave test signals shown in Figures 1/J.62 and 2/J.62 should fall within the following limits.

2.1 *2*T *bar response*

The limits are indicated by the oscilloscope mask shown in Figure 10/J.62. In effect, the oscilloscope is to be adjusted so that the half-amplitude points of the bar transition coincide with M_1 and M_2, and the mid-points of the $H/2.5$ "black" and "white" portions coincide with A and B respectively. The response should then fall within the $\pm K$ limits indicated by the full lines, which extended to $H/100$ from the half-amplitude point of each transition.

2.2 *2*T *pulse response*

The limits are indicated by the oscilloscope mask shown in Figure 11/J.62. In effect, the oscilloscope is to be adjusted so that:

- the sweep velocity corresponds with the time scale indicated;
- the "black" level of the response coincides with the horizontal axis;
- the peak of the response falls on the unit-amplitude line;
- the half-amplitude points of the response are symmetrically disposed about the vertical axis.

2.3 *2*T *pulse/bar ratio*

The ratio of the amplitude of the $2T$ pulse response to the amplitude of the $2T$ bar response should fall within the limits $1/(1 \pm 4K)$, where the pulse amplitude is the difference between the "black" level sand the peak of the response, and the bar amplitude is the difference between the points A and B already defined. The limits are included in the mask shown in Figure 10/J.62.

2.4 T *pulse response*

To meet the luminance-channel requirements, the T pulse response should not show appreciable ringing at a frequency below 5.0 MHz, irrespective of the assigned rating factor. This is only of acedemic interest for System I because the chrominance-channel requirements are such that the ring frequency should not be less than 5.5 MHz.

Other limits cannot be specified rigidly because the spectrum of the T pulse extends far beyond 5 MHz, and the response must therefore contain irrelevant information. A partial solution is found in the insertion of a "5.3-MHz link filter" between the link and the oscilloscope. This is a member of a series of delay-equalized lowpass filters designed to have good waveform responses; its insertion loss is almost constant up to 5.0 MHz, then increases by about 3 dB at 5.3 MHz (the ring frequency) and 20 dB at 5.7 MHz. Being dominant in determining the overall upper cut-off characteristic, it substantially attenuates the irrelevant components of the response. The T pulse/bar ratio of the overall response is then a useful feature for measurement; it is closely related to the ratio which forms the basis of restriction (3) in the acceptance-test method (see 3.2 below).

It has been found empirically that, to meet a specified rating factor K, the T pulse/bar ratio of the link + filter should fall within the limits $0.84/(1 \pm 6K)$. Thus, for a rating factor of 1%, the ratio should be between 79% and 89%. As the formula indicates, a ratio of 84% is given by the filter alone.

Other features of interest in the T pulse response of the link + filter are the lobes of ringing immediately before and after the main lobe of the response. The following is a rough guide to the maximum amplitudes to be expected under normal conditions:

Lobe	Upper limit of lobe amplitude expressed as percentage of bar amplitude	
	$K = 1\%$	$K = 5\%$
First lobe (negative), leading or trailing	12	20
Second lobe (positive), leading or trailing	8	12

Although the amplitudes of other lobes may be of importance in some cases, it is not possible to offer further general guidance at present.

2.5 *50-Hz square-wave response*

The limits are indicated by the oscilloscope mask shown in Figure 12/J.62. As for the $2T$ bar response, the oscilloscope is to be adjusted so that the waveform passes through the four marked points, the line-synchronizing pulses being ignored.

3. *Acceptance-test method*

3.1 *2T bar response*

The limits are identical with those given in 2.1 above for the routine-test method.

3.2 T *pulse response*

From the measured T pulse response and the measured or assumed response of the measuring equipment itself, the "filtered impulse response" is derived and expressed in the form of a normalized time series [see N. W. LEWIS, *Proc. I.E.E.*, Vol. 101, Part III (1954)]. The "main" term of this series represents the ideal or non-distorting part, and the "echo" terms represent the distorting part. To meet a specified rating factor K, the amplitudes of the echo terms should be such that each of the following four restrictions is met.

Let the time series representing the filtered impulse response be

$$B(rT) = \ldots B_{-r}, \ldots B_{-1}, B_0, B_{+1}, \ldots B_{+r}, \ldots$$

and assume that this has already been normalized so that $B_0 = 1$.

Let the serial product of $B(rT)$ and the series [½, 1, ½] be

$$C(rT) = \ldots C_{-r}, \ldots C_{-1}, C_0, C_{+1}, \ldots C_{+r}, \ldots$$

where $C_r = \frac{1}{2} B_{r-1} + B_r + \frac{1}{2} B_{r+1}$

Restriction (1) is then given by

$$\frac{1}{8}\left|\frac{C_r}{C_0} - \frac{1}{2}\right| \leq K \quad r = \pm 1$$

$$\text{and} \quad \frac{1}{8}\left|r\,\frac{C_r}{C_0}\right| \leq K \quad \begin{cases} -8 \leq r \leq -2 \\ +2 \leq r \leq +8 \end{cases}$$

$$\text{and} \quad \left|\frac{C_r}{C_0}\right| \leq K \quad \begin{cases} r \leq -8 \\ +8 \leq r \end{cases}$$

Restriction (2) is given by $\frac{1}{4}\left|\left(\frac{1}{C_0}\sum_{-8}^{+8} B_r\right) - 1\right| \leq K$

Restriction (3) is given by $\frac{1}{6}\left|\left(\sum_{-8}^{+8} B_r\right) - 1\right| \leq K$

Restriction (4) is given by $\frac{1}{20}\left\{\left(\sum_{-8}^{+8} |B_r|\right) - 1\right\} \leq K$

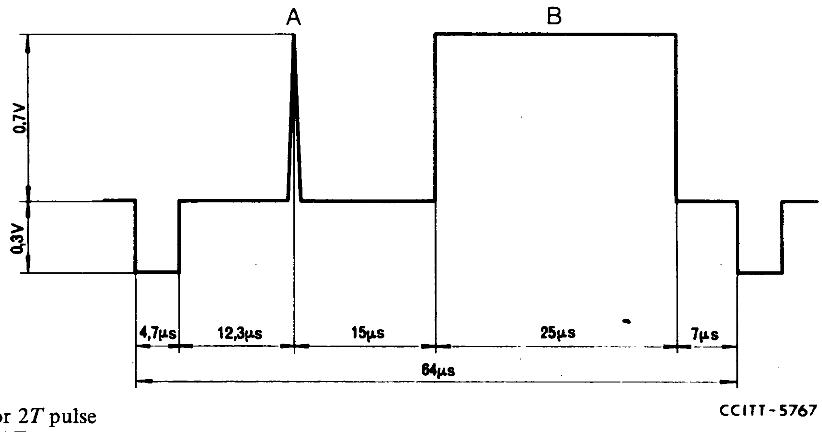
The series $C(rT)$ represents fairly closely the response to a $2T$ pulse. Restriction (1) is thus approximately equivalent to the limits indicated in Figure 11/J.62 for the $2T$ pulse response in the routine-test method. Restriction (2) is similar to the limits placed on the $2T$ pulse/bar-ratio in the routine-test method. Restriction (3) is equivalent to limits placed on the pulse/bar ratio of the response to a hypothetical pulse-and-bar test signal in which the pulse is an ideal filtered impulse. Restriction (4) is an upper limit placed on the average amplitude, ignoring signs, of the 16 central echo terms.

3.3 *50-Hz square-wave response*

The limits are identical with those given in 2.5 above for the routine-test method.

4. *Gain/frequency characteristic*

As a precaution against possible overloading effects, the insertion gain at any frequency between 50 Hz and 5 MHz should not exceed the gain at the line-repetition frequency by more than an amount in dB numerically equal to the percentage rating factor, e.g. 1 dB for a rating factor of 1% ($K = 0.01$).



A: T pulse or $2T$ pulse
B: T bar or $2T$ bar
T = 100 ns

Note. – For the design of the shaping-network, see MacDiarmid and Philips, *Proc. I.E.E.*, Vol. 105B, 440 (1958).

FIGURE 1/J.62 – T and $2T$ pulse-and-bar test signals
(for insertion gain, and short-time and line-time linear distortions in the luminance channel)

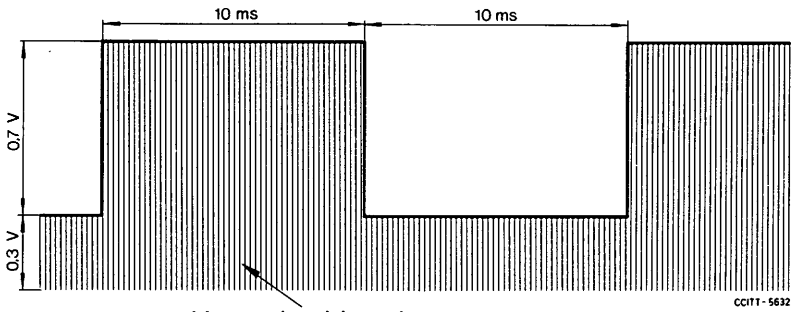
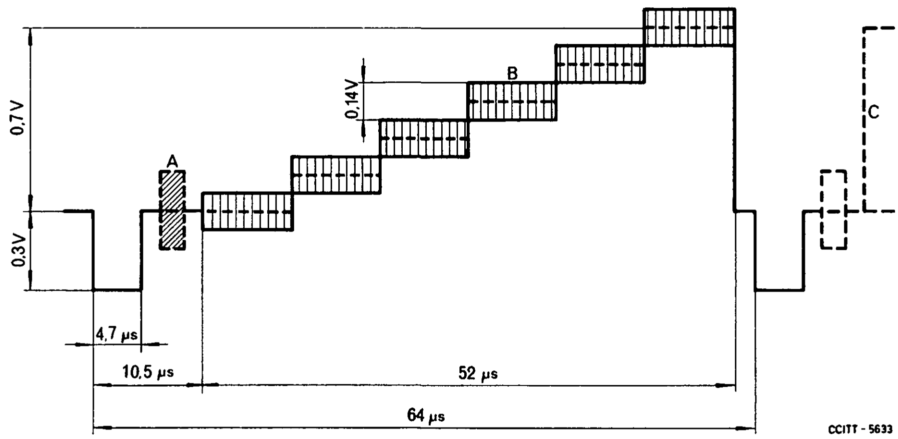
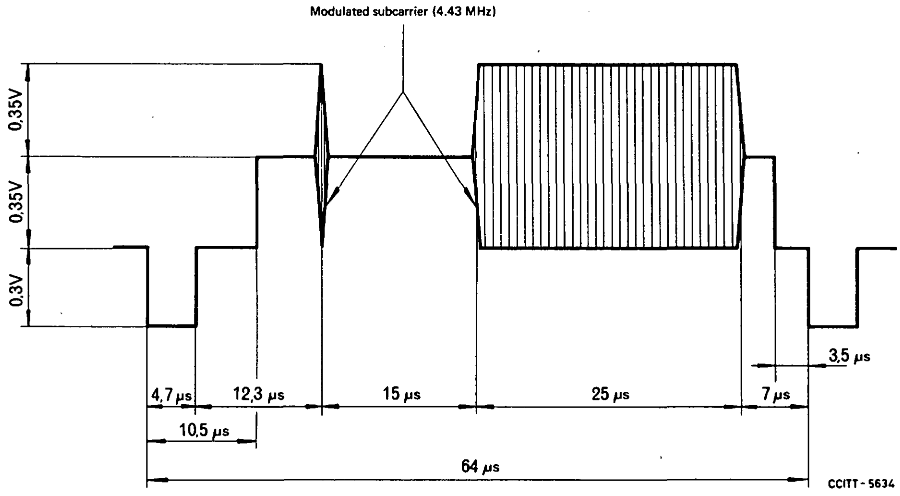


FIGURE 2/J.62 – **50-Hz square-wave test signal (for field-time linear distortion in the luminance channel)**



A: optional colour burst
B: superimposed subcarrier (4.43 MHz)
C: 3 lines at black level or 3 lines at white level

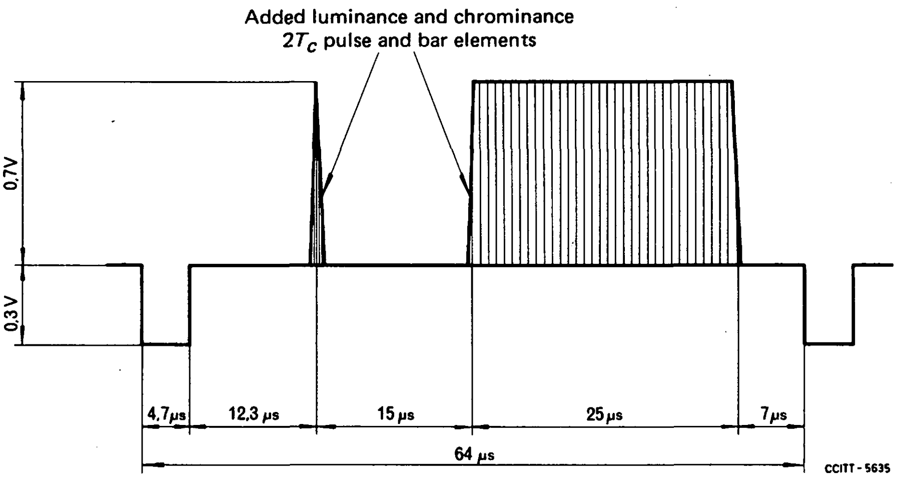
FIGURE 3/J.62 – **Staircase test signals (for all nonlinear distortions)**



$T_C = 500$ ns

Note. – For the design of the shaping-network, see MacDiarmid and Phillips, *Proc. I.E.E.*, Vol. 105B, 440 (1958).

FIGURE 4/J.62 – T_C and $2T_C$ **pulse-and-bar test signals**
(for short-time and line-time linear distortions in the chrominance channel)



$T_C = 500$ ns

FIGURE 5/J.62 – **Composite test signal**
(for luminance-chrominance gain and delay inequalities)

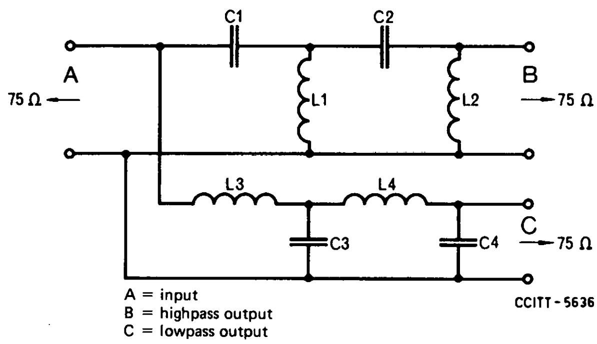


Table of values

Component	Value	Tolerance
C1	139 000	± 5%
C2	196 000	
C3	335 000	
C4	81 200	
L1	0.757	± 2%
L2	3.12	
L3	1.83	
L4	1.29	

Note 1. – Inductances are given in mH, capacitances in pF.
Note 2. – The *Q*-factor of each inductor should be equal to, or greater than, 100 at 10 kHz.

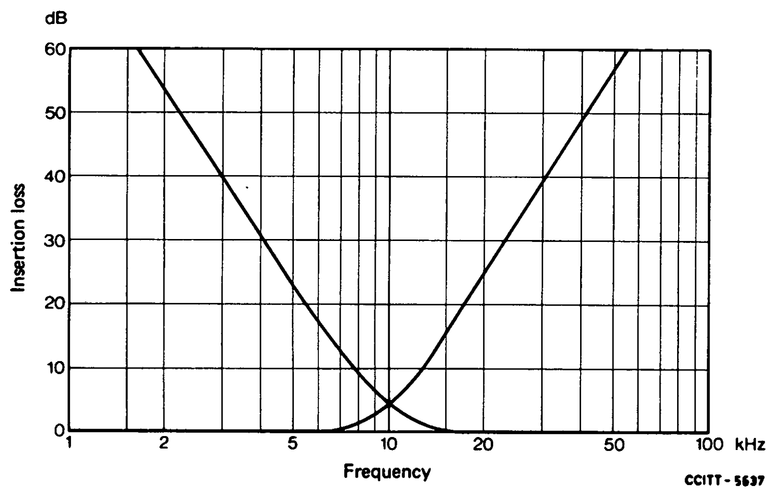


FIGURE 6/J.62 – Junction filter
(for noise measurement)

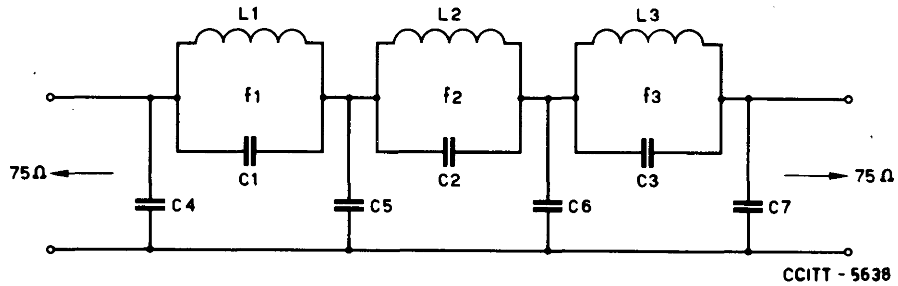


Table of values

Component	Value	Tolerance
C1	100	See Note 2
C2	545	
C3	390	
C4	428	
C5	563	
C6	463	
C7	259	
L1	2.88	See Note 3
L2	1.54	
L3	1.72	
f_1	9.408	
f_2	5.506	
f_3	6.145	

Note 1. – Inductances are given in μH, capacitances in pF, frequencies in MHz.
Note 2. – Each capacitance quoted is the total value, including all relevant stray capacitances, and should be correct to ± 2 %.
Note 3. – Each inductor should be adjusted to make the insertion loss a maximum at the appropriate indicated frequency.
Note 4. – The Q-factor of each inductor measured at 5 MHz should be between 80 and 125.

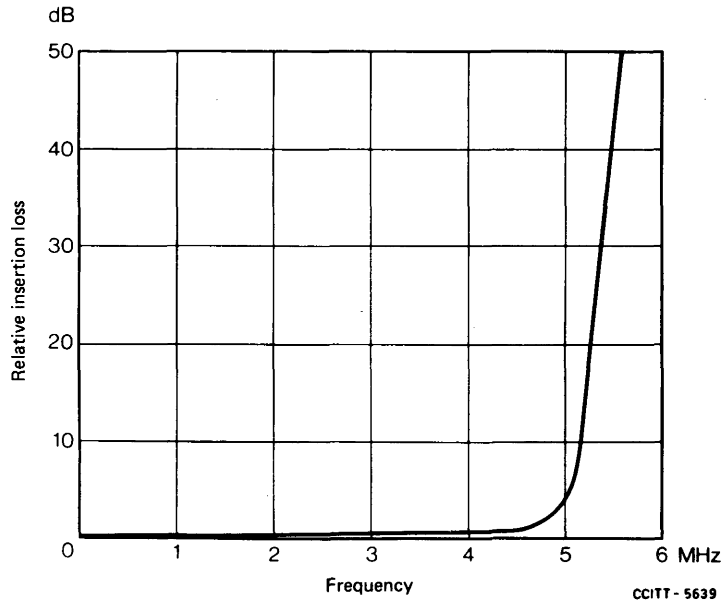


FIGURE 7/J.62 – **Lowpass filter**
(for random noise in the luminance channel)

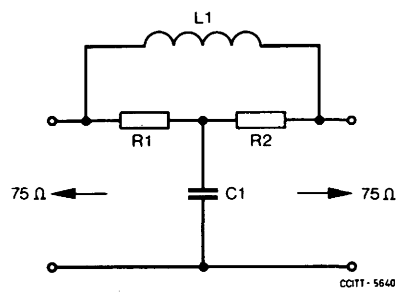


Table of values

Component	Value	Tolerance
C1	2660	± 1%
L1	15	
R1	75	
R2	75	

Note 1. – Inductance is given in μH, capacitance in pF, resistance in ohms.
Note 2. – The *Q*-factor of inductor L1 should be equal to, or greater than, 25 at 8 MHz.
Note 3. – Insertion loss = $10 \log_{10} [1 + (2\pi\tau f)^2]$ dB, where τ = 200 ns.

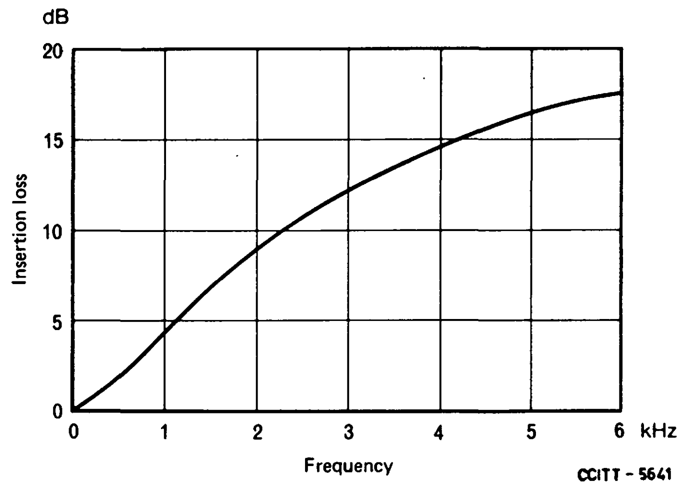


FIGURE 8/J.62 – Weighting network
(for random noise in the luminance channel)

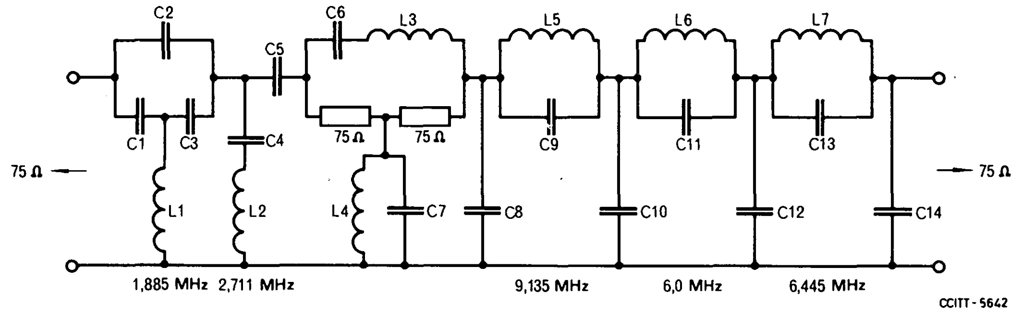


Table of values

Component	Value	Tolerance	Component	Value	Tolerance
C1	496.0		C12	311.4	
C2	89.47		C13	619.2	± 1%
C3	292.1		C14	187.5	
C4	715.8		L1	2.960	
C5	1239.0		L2	4.814	
C6	194.3	± 1%	L3	6.650	
C7	1182		L4	1.093	See Note 2
C8	385.7		L5	2.149	
C9	141.3		L6	0.7476	
C10	418.6		L7	0.9846	
C11	941.2				

Note 1. – Inductances are given in μH, capacitances in pF.

Note 2. – L3 is adjusted to resonate with C6, and L4 with C7 at 4.428 MHz. L1, L2, L5, L6 and L7 are adjusted to give maximum insertion loss at the appropriate indicated frequencies.

Note 3. – The *Q*-factor of each inductor should be equal to, or greater than, 100 between 3 MHz and 6 MHz.

Note 4. – The insertion loss is equal to, or greater than, 35 dB at frequencies above 6 Mhz.

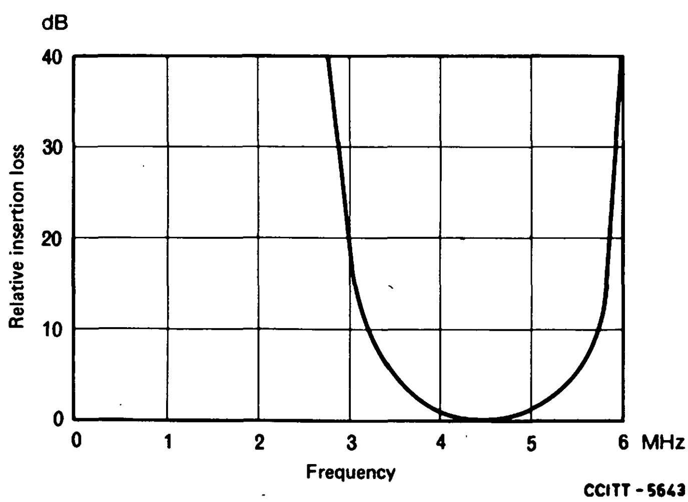


FIGURE 9/J.62 – **Bandpass filter and weighting network**
(for random noise in the chrominance channel)

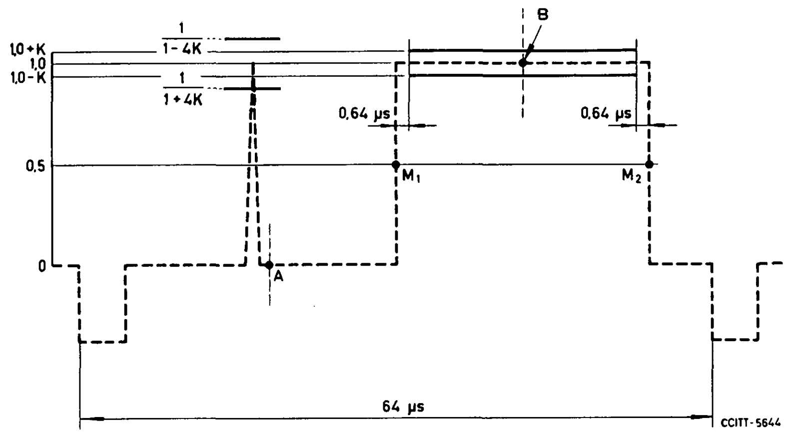
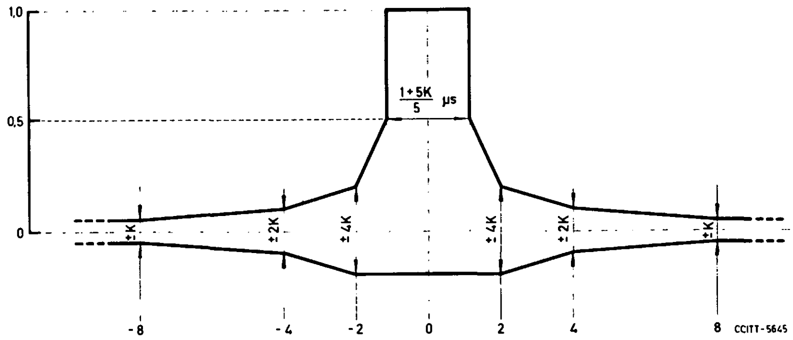


FIGURE 10/J.62 – 2T bar response and 2T pulse/bar ratio



Unit interval: 100 ns

FIGURE 11/J.62 – 2T pulse response

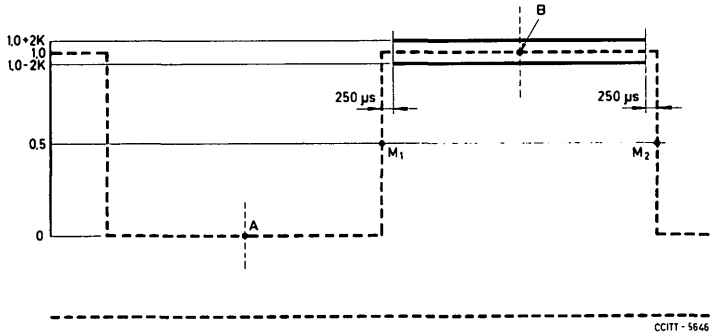


FIGURE 12/J.62 – 50-Hz square-wave response

SECTION 7

GENERAL CHARACTERISTICS OF SYSTEMS FOR TELEVISION TRANSMISSION OVER METALLIC LINES AND INTERCONNECTION WITH RADIO-RELAY LINKS

Recommendation G.331 gives details of the 2.6/9.5-mm type of coaxial cable standardized by the CCITT.

Recommendations have also been made by the CCITT concerning the following systems for transmitting television signals over such pairs:

- 4-MHz system, which can be used to transmit 405-line television (Recommendation J.71);
- 6-MHz system, for transmitting 625-line television (5-MHz video-frequency band) or the Belgian 819-line system (Recommendation J.72);
- 12-MHz system, which can be used to transmit telephony and television simultaneously (Recommendation J.73).

Recommendation J.71

4-MHZ SYSTEM FOR TELEVISION TRANSMISSION

The 4-MHz coaxial cable system, defined in Recommendation G.337, can be used for transmitting 405-line television signals. For this it is recommended that the following conditions be met:

a) *Carrier frequency and sidebands*

It is unanimously recognized that the use of a method of transmission with a vestigial sideband is necessary for the type of television transmission considered. It is assumed that the video signal to be transmitted corresponds to an image consisting of 405 lines and that at the studio output the spectrum of the video signal has a relatively sharp cut-off at the two extremities, these being respectively 30 Hz and 3 MHz. It is also assumed that the originating broadcast authority has corrected as far as possible for aperture distortion and other distortions in the camera chain.

With a coaxial cable of the type standardized by the CCITT (see Recommendation G.331) and with a repeater spacing of the order of 9 kilometres (used for 4-MHz telephone carrier systems [1]) on coaxial cable of this type), it is possible to transmit an upper sideband of width around 3 MHz and a vestigial lower sideband of width 500 kHz (value considered satisfactory provisionally even though there is not yet sufficient experience on this subject). If the construction of the cable has been of high quality as regards the regularity of impedance and if the equalization and phase compensation have been good, it may be assumed that the signal applied at the input will be faithfully reproduced at the output.

---

[1]) See Recommendation G.338

For television transmissions of the type considered it is recommended to employ in Europe a carrier frequency of nominal value 1056 kHz, it being understood that, during a transmission, the carrier frequency should not change by more than a few hertz.

In the present state of the technique, it is not yet possible to design transmitting and receiving terminal equipments independently of each other.

b) *Polarity of modulation*

The advantages or disadvantages of positive polarity (where the signal increases with the luminance) or negative polarity, have not yet been established for television transmissions on metallic lines, but, on the other hand, it is desirable not to have to use inversion apparatus at the interconnection of two circuits. Hence it is recommended that the polarity of modulation adopted at the origin of a chain for international television transmission should be conserved throughout the length of this chain.

c) *Ratio of amplitude between video and synchronizing signal*

It is recommended that the ratio:

$$\frac{\text{amplitude of video signal}}{\text{amplitude of synchronizing signal}}$$

in the modulated wave should equal 7/3.

d) *Depth of modulation*

The limit allowed for the "reference modulation coefficient" defined below is provisionally fixed at 50%.

Note. – The modulation coefficient, for a given signal s, is defined as follows: V_s is the voltage (peak-to-peak) of the video signal considered. This signal amplitude modulates a carrier, the amplitude of which varies between the two limits V_M and V_m with $V_M - V_m = V_s$ when the two sidebands are retained.

By definition, the modulation coefficient τ is

$$\tau = \frac{V_s}{V_M + V_m} = \frac{V_M - V_m}{V_M + V_m}$$

It will be seen that this definition coincides with the usual definition when the signal s is sinusoidal.

After partial suppression of the lower sideband, the amplitude ratios considered above are approximately retained and the modulation coefficient

$$\tau = \frac{V_M - V_m}{V_M + V_m}$$

remains for all intents and purposes the same.

The modulation coefficient defined above is essentially a function of the type of signal transmitted and differs according to whether the d.c. component of a video signal is retained or not.

However, there should be a limit to the highest modulation coefficient that is possible, considering all the possible types of waveform in order to limit the detection distortion which appears due to the partial suppression of the lower sideband.

The choice of the highest modulation coefficient determines the modulation coefficient for all other types of signal.

Also, the maximum amplitude of the modulated carrier that is possible should have an upper limit in order to limit the non-linear distortion. The ratio between the video signal and the basic noise is then low since the modulation coefficient is itself low. It would appear from this that there should be a lower limit to the modulation coefficient. The choice is therefore a compromise between the two requirements.

When the complete video signal is as defined above (with, for example, a negative polarity of modulation) and the d.c. component has been suppressed, it is easy to determine the type of video signal corresponding to the higher coefficient of modulation. It is this coefficient which corresponds to the transmission of white spots on a dark background (Figure 1/J.71) (it might be considered that the average value of the synchronizing signal is negligible compared with V_s).

The corresponding coefficient of modulation τ_R is called the "reference coefficient of modulation".

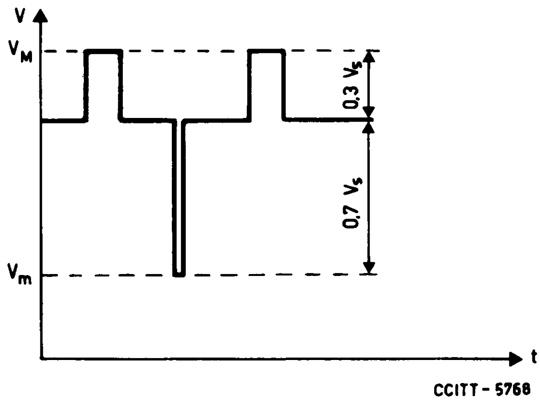


FIGURE 1/J.71

e) *D.c. component*

It is recommended that the d.c. component of the complete video signal should be suppressed for transmission to line.

f) *Repeater input and output impedance*

The return loss between repeater input and output impedances and a pure resistance of 75 ohms should be at least 20 dB at the carrier frequency used for television; the limit permitted for such return loss may decrease progressively to 15 dB at the upper and lower edges of the band of frequencies transmitted for monochrome television.

Note 1. – Under these conditions, at the 1056-kHz carrier frequency and at adjacent frequencies, the overall resultant value of echo in a single repeater section of normal length (sum of the three terms as defined in the Annex to Recommendation J.73) that is obtained is considerably better than the value of 70 dB recommended. The value of 70 dB is, in fact, easily throughout the transmitted band in the case of the 4-MHz system, except at the lower limit of the vestigial sideband, say from 0.5 to 0.7 MHz. A lower figure is in any case acceptable here, as the energy of the signal is small at these frequencies.

Note 2. – The CCITT considered it unnecessary to specify other repeater characteristics.For the time being, the arrangements in the case of a cable crossing a frontier should be the subject of bilateral agreement between the countries concerned. (See Recommendation G.352 for the comparable case of telephony on a 2.6/9.5-mm type coaxial cable crossing a frontier.)

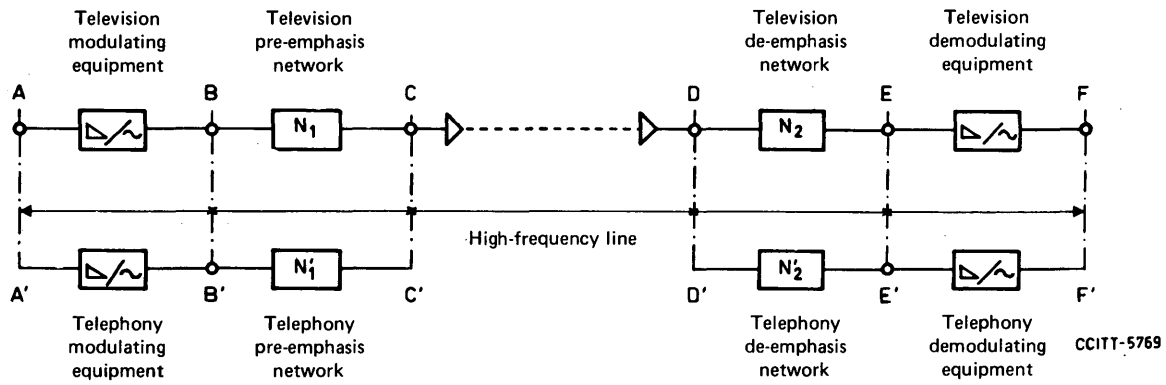
Recommendation J.72

6-MHz SYSTEM FOR TELEVISION TRANSMISSION

The 6-MHz coaxial cable system, defined in Recommendation G.337, is normally used for transmitting 625-line television signals (with a 5-MHz video-frequency band) or the 819-line Belgian system. It is then recommended that the following conditions be met:

a) *Characteristics at intermediate distribution points*

Interconnection between different high-frequency lines or between lines and television modulating or demodulating equipments should only be made at points corresponding to B and E in Figure 1/J.72, which might be called "television carrier-frequency interconnection points". (In the same way, points B′ and E′ are "telephony carrier-frequency interconnection points" when telephone channels have to be transmitted to line.) At such interconnection points, the line pilots are suppressed, and the gain/frequency characteristic between points B and E (or B′ and E′) is a horizontal straight line throughout the transmitted frequency band, since networks N_1 and N_2 (or N'_1 and N'_2) have compensating inverse characteristics. It is not necessary to specify the characteristics of these networks precisely, since they depend on the particular line system used for the high-frequency line, CD.



Note 1. – Points A and F are video interconnection points. Points B and E are television interconnection points at carrier frequency. Points B′ and E′ are telephony interconnection points at carrier frequency.

Note 2. – The modulator characteristics should be defined between the points A and B. The demodulator characteristics should be defined between the points E and F.

Note 3. – Between BE and B′E′ the gain/frequency characteristic at the high frequency is uniform.

Note 4. – The networks N_1, N_2, etc., should be chosen to facilitate the matching to the line and to present to the modulation and demodulation equipment standard level conditions, etc.

Note 5. – If a number of high-frequency lines of different types are interconnected or if derivations are made at an intermediate point, pre-emphasis and de-emphasis networks, etc., will be necessary at junction points to enable the interconnection to be made at a point of defined level and independent of frequency.

Note 6. – For alternative telephony and television transmissions, switching should be carried out at points C and D.

FIGURE 1/J.72 – **Systems of vestigial-sideband television transmission on coaxial cable**

The impedance of the input and output circuits corresponding to points B and E in Figure 1/J.72 must be specified. The recommended value is 75 ohms unbalanced to earth and the return loss against a pure 75-ohm resistance should not be less than 24 dB.

When it is necessary, as an alternative, to insert pre-emphasis and de-emphasis networks, N'_1 and N'_2, the 6-MHz line, CD, can be used alternatively for television or telephony, under the conditions given in Recommendation G.337, B.

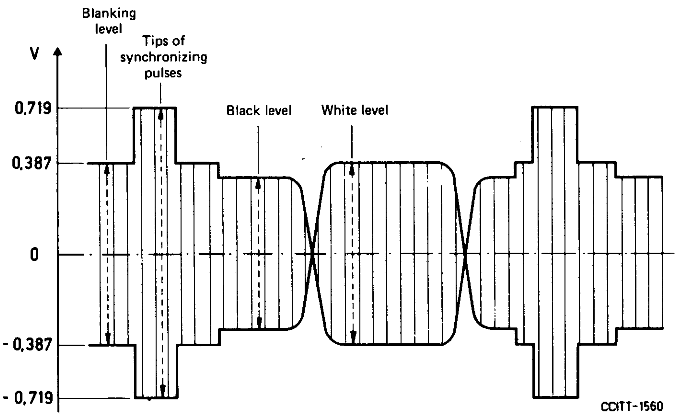
Note. – So that two line-systems made by different manufacturers can be interconnected at an intermediate point, such as a frontier repeater station, it is first necessary to correct for pre-emphasis (produced, for example, by network N_1 of Figure 1/J.72) by means of an inverse network (such as N_2 of Figure 1/J.72) and thus to create a point similar to E. Between B and E the line has a gain/frequency characteristic which is a horizontal straight line.

b) *Carrier frequency*

The nominal frequency of the video-signal carrier should be 1056 kHz with a tolerance of ± 5 Hz.

c) *Modulation ratio*

Amplitude modulation should be used. The modulation ratio should be greater than 100% (as shown in Figure 2/J.72) so that when the carrier is modulated by a signal at the blanking level, the amplitude of this signal should equal that of the carrier modulated by the white level (assuming that the d.c. component is transmitted).



Note. – The voltages shown are the values measured:

– at the output of the modulation equipment (point B, Figure 1/J.72) in 6-MHz systems (Recommendation J.72);
– at a zero relative level point for television transmission in the 12-MHz system (Recommendation J.73).

FIGURE 2/J.72 – Envelope of carrier modulated by Test Signal No. 2

When Test Signal No. 2 (see Annex 1 to Recommendation J.61) is sent to the modulated input (point A in Figure 1/J.72), the nominal value of the peak voltage of the modulated carrier at the output of the modulating equipment (point B in Figure 1/J.72) and at the input to the demodulating equipment (point E in Figure 1/J.72) should be as follows (see Figure 2/J.72):

- white level or blanking level, 0.387 volt – i.e. the peak value of a sinusoidal signal dissipating a power of 1 mW in a 75-ohm resistance;
- synchronizing signals, 0.719 volt – i.e. the peak voltage of a sinusoidal signal dissipating a power of 3.45 mW in a 75-ohm resistance.

d) *Shaping of the vestigial sideband*

It has not appeared possible to recommend a signal system, and the existing shaping filters, whose characteristics are given in the following Annex, should be used. In these systems, the modulation and demodulation equipments play equal parts in shaping the vestigial-sideband signal.

Interconnection between two different systems should be dealt with by bilateral agreement between the Administrations concerned.

e) *Pilots*

Pilots should be injected at the input to the first line-amplifier (after point C in Figure 1/J.72) and should be blocked after the last line-amplifier (before point D in Figure 1/J.72).

To facilitate the interconnection of line systems, it is recommended that the following frequencies and levels should be standardized for the pilots in each of the two systems defined in the Annex to this Recommendation.

1st System. – At point B in Figure 1/J.72, the levels of the various pilots should have the following values:

308 kHz: −29.7 dBm
4142 kHz: −22.2 dBm
6142 kHz: −20.3 dBm

2nd System. – At point B in Figure 1/J.72, the level of the pilots shall have the following values.

308 kHz: −39.0 dBm
4092.45 kHz: −39.8 dBm
6200 kHz: −41.7 dBm

Administrations which use different systems should reach bilateral agreement in connection with all the necessary arrangements for interconnecting their line systems at a frontier repeater station.

Whatever system is used, the CCITT recommends that the frequency of the line pilots should have a relative tolerance of $\pm 1.0 \times 10^{-5}$.

f) *Interference*

Recommendation J.61, 3.3 shows the overall target design value referred to the hypothetical reference circuit for television transmission.

It is provisionally recommended that the overall random noise should be allocated on the basis of 50% to the line and 50% to the three pairs of modulators and demodulators.

g) *Repeater input and output impedance*

The return loss between repeater input and output impedances and a non-reactive resistance of 75 ohms should be at least 20 dB at the carrier frequency used for television.

The limit permitted for such return loss may decrease progressively to 15 dB at the upper and lower edges of the band of frequencies transmitted for monochrome television.

Note 1. – Under these conditions, at the 1056-kHz carrier frequency and at adjacent frequencies, the overall resultant value of echo in a single repeater section of normal length (sum of the three terms as defined in the Annex to Recommendation J.73) that is obtained is considerably better than the value of 70 dB recommended. The value of 70 dB is, in fact, easily achieved throughout the transmitted band in the case of 6-MHz systems, except at the lower limit of the vestigial sideband, say from 0.5 to 0.7 MHz. A lower figure is in any case acceptable here, as the energy of the signal is small at these frequencies.

Note 2. – The CCITT considered it unnecessary to specify other repeater characteristics. For the time being, the arrangements in the case of a cable crossing a frontier should be the subject of bilateral agreement between the countries concerned. (See Recommendation G.352 for the comparable case of telephony on a 2.6/9.5-mm type coaxial cable crossing a frontier.)

ANNEX

(to Recommendation J.72)

Methods used in 6-MHz systems for shaping the television signal transmitted to line

1st System

1. The vestigial-sideband region should cover a band from approximately 500 kHz above the carrier frequency to about 500 kHz below the carrier frequency. This represents a reasonable compromise between the difficulty of designing a narrowband filter and the difficulty of extending the frequency range of the repeaters in the low-frequency direction.

The same filter is to be used at the transmitting and receiving terminals. The loss of each filter at the carrier frequency should therefore be 3 dB relative to the filter loss at high frequencies where the transmission is single sideband. Thus, after passing through two filters, the sidebands associated with very low video frequencies will be attenuated by 6 dB, i.e. to one-half of the voltage of the single sideband by which high-frequency information is transmitted. Thus in-phase addition of the two very-low-frequency sidebands produces video information of the same amplitude as the high-frequency information.

With higher video frequencies the vestigial-sideband filters will attenuate the two sidebands unequally. If the filters are phase-equalized the two sidebands will, in the process of demodulation, add in-phase to give the video output.

The requirement, then, to obtain a video output which is independent of frequency is that the sum of the two corresponding sideband amplitudes shall be constant for all video frequencies.

There are many possible filter characteristics which will fulfil this requirement. The mathematically simplest is that producing a linear voltage/frequency characteristic as shown in Figure 3/J.72.

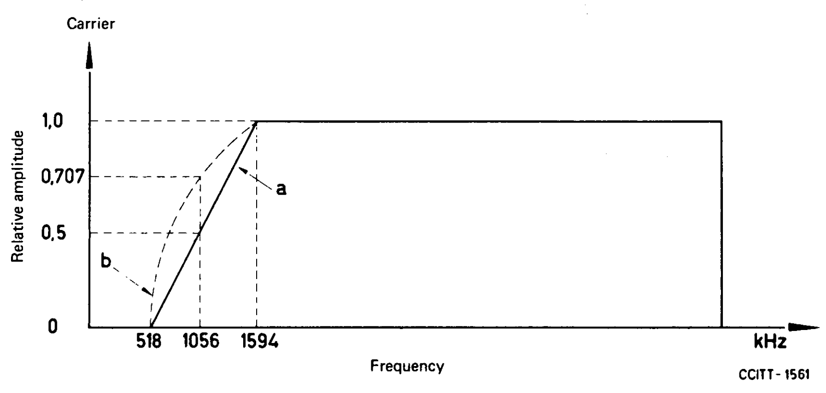


FIGURE 3/J.72 – **Partial suppression of the lower sideband**

It is, however, not possible to realize this characteristic with practical filters. The difficulty occurs at both extremes of the vestigial-sideband region where the linear characteristic has a discontinuity. Practical filters would round off these regions and it is better to take such limitations into account in specifying the required characteristic.

A characteristic which produces the required sideband amplitudes and takes into account the limitation of practical filters is:

For one send or one receive filter

$$D(f) = 10 \log_{10} ERF(y) \text{ (dB)}$$

where $D(f)$ is the filter loss at frequency f relative to the loss in the single-sideband region;

$$y = \frac{f - f_0}{K}$$

where f_0 is the carrier frequency; K is a constant defining the rate of cut-off of the unwanted sideband.

The function $ERF(y)$ is the error function of y defined as:

$$ERF\ (y) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{y} \exp\left(-\frac{y^2}{2}\right) dy$$

This function will be found tabulated in mathematical tables.

As regards the constant, K, for the definition of the rate of cut-off of the unwanted sideband, a value of $K = 215$ kHz has been proposed by the Cuban Telephone Company.

It is believed that sending and receiving vestigial-sideband terminals using rather different K values may interwork satisfactorily, provided that each terminal equipment is phase-equalized.

2. In specifying the accuracy with which such a characteristic should be met, the limits set should take into account the fact that, for a given error in the video output, a much wider tolerance can be allowed at frequencies at which the filter attenuation is high.

This is satisfied by the following expression:

$$E = \left| 10^{-\frac{D_a}{10}} - 10^{-\frac{D_f}{10}} \right|$$

where

E is a constant to be specified and which determines tolerances,
D_a is the actual filter discrimination (dB),
D_f is the specified filter discrimination (dB).

2nd System

The lower sideband is attenuated by the combination of transmitting modulator and receiving demodulator, in such a way that between 518 and 1594 kHz the change of amplitude with frequency is linear (see Figure 3/J.72, curve a). At the points where the voltage passes through the values 0 and 100%, there is only negligible rounding of the characteristic. The total filter attenuation required is divided equally between the sending and receiving equipments. Each component filter has an attenuation of 3 dB at the carrier frequency and of 9 dB at 400 kHz below the carrier frequency. Bearing in mind the preceding recommendation, the shape of the amplitude/frequency characteristic of a single filter is therefore that of curve b of Figure 3/J.72. The increase is proportional to the square root of the difference between the frequency considered and 518 kHz, the value being 0 at 518 kHz, 0.707 at 1056 kHz, and 1.0 at 1594 kHz.

No proposal can be made for the allowable deviation from this nominal curve but, for information, it is pointed out that the following characteristic is achieved by the Nyquist filters used by the Federal Republic of Germany. The combined attenuation of the two filters gives a linear voltage/frequency characteristic. When this characteristic passes through 0 and 100% of the voltage, the rounding off is insignificant. The overall attenuation of the two filters is equally divided between the sending and the receiving equipment. Each partial filter has a 3-dB attenuation at the carrier and a 9-dB attenuation at 400 kHz below the carrier. Thus the slope of the Nyquist filter characteristic is linear between the frequencies.

f_1 = 518 kHz and
f_2 = 1594 kHz.

Recommendation J.73

USE OF A 12-MHz SYSTEM FOR THE SIMULTANEOUS TRANSMISSION OF TELEPHONY AND TELEVISION

(amended at Geneva, 1964)

The 12-MHz coaxial pair system is defined in Recommendation G.337 and its use for telephony transmission in Recommendations G.332 and G.337.

Any 12-MHz system equipped for television transmission should be capable of transmitting the signals used in all those television systems defined by the CCIR having a video bandwidth not exceeding 5 MHz [if necessary, by means of the switching (in terminal equipments only) of certain components].

This Recommendation has been drafted only for the transmission of monochrome television systems defined by the CCIR up to 1964.

a) *Carrier frequency*

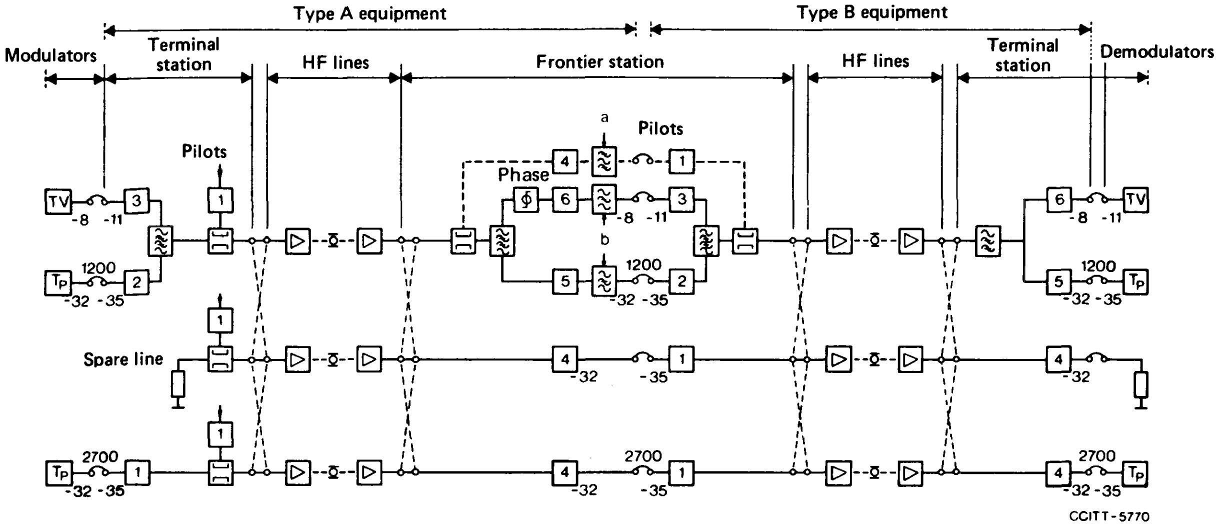
The CCITT recommends the use of a carrier frequency of 6799 kHz with a tolerance of ± 100 Hz for the transmission of all the television signals indicated above. The video band transmitted over the cable should be 5-MHz wide, whatever television system is to be used. The level provisionally recommended for this carrier has been defined for the interconnection points and is shown in Figures 1/J.73 and 2/J.73 (see Note 3).

b) *Modulation ratio*

Amplitude modulation has to be used. The modulation ratio has to be higher than 100% (as indicated in Figure 2/J.72), so that, when the carrier is modulated by a signal corresponding to blanking level its amplitude is equal to that of the carrier when it is modulated by a signal corresponding to the white level, assuming that the d.c. component is transmitted.

When Test Signal No. 2 (see Annex 1 to Recommendation J.61) is applied at a video junction point, the nominal peak voltage of the modulated carrier, at a point where the relative level for the television transmission is zero, should be as follows:

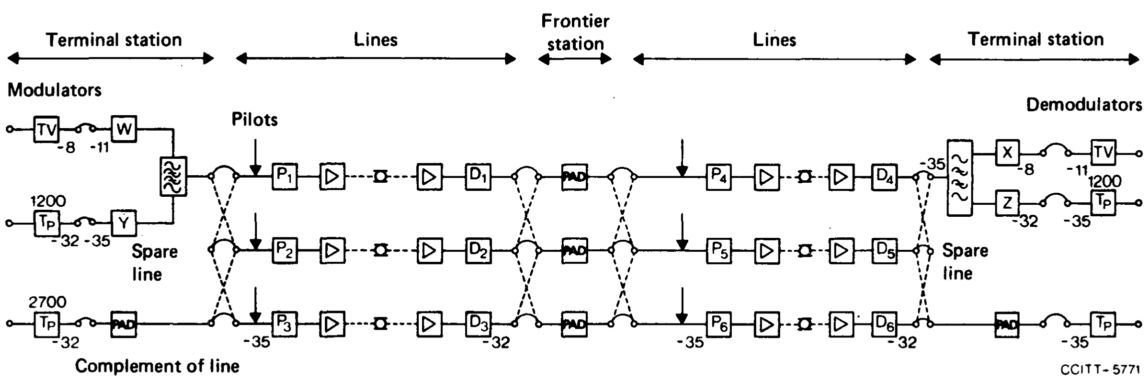
– for white or blanking level, 0.387 volt (i.e. the peak voltage of a sine-wave signal dissipating a power of 1 mW in a resistance of 75 ohms);
– for the synchronizing signals, 0.719 volt (i.e. the peak voltage of a sine-wave signal dissipating a power of 3.45 mW in a 75-ohm resistance).



1 4 pre-emphasis (respectively de-emphasis) for telephony 2700 channels
2 5 pre-emphasis (respectively de-emphasis) for telephony 1200 channels } simultaneous transmission
3 6 pre-emphasis (respectively de-emphasis) for television

a pilot pass-filter
b pilot stop-filter

FIGURE 1/J.73 – General case of interconnection of 12-MHz lines



Nominal relative levels on the line for each telephone channel in the "all-telephony" case

W X differential pre-emphasis and de-emphasis for television } common for all types of 12-MHz line
Y Z differential pre-emphasis and de-emphasis for telephony 1200 channels

P_1 D_1 pre-emphasis and de-emphasis for telephony, 2700 channels, particular to each design of 12-MHz line

Notes to Figures 1/J.73 and 2/J.73

1. Interconnection of pilots, e.g. blocking and re-injecting or by-passing, should be agreed between Administrations.
2. The level of the line pilots is fixed at −10 dBm0 for the all-telephony case. When the line is used to transmit telephony and television simultaneously, different values of pre-emphasis may be required; although the absolute levels of the pilots will remain the same, they may no longer be at −10 dBm0.
3. The television levels shown are those of the modulated carrier, relative to that of the idealized reference signal described in b) of this Recommendation. (See also Figure 2/J.72.)
4. The characteristics of the filters in Figure 1/J.73 (used for separating and combining the telephony and television bands so that the necessary arrangements for pre-emphasis and de-emphasis can be made) must be agreed between Administrations.

FIGURE 2/J.73 – **Showing use of differential emphasis networks to simplify interconnection of 12-MHz lines of different designs**

c) *Vestigial-sideband shaping*

The shaping of the vestigial-sideband signal has to be carried out entirely at the transmit point. Provisionally, the vestigial sideband should not exceed a width of 500 kHz. Figure 3/J.73 shows the frequency arrangement recommended for television transmission over the 12-MHz system.

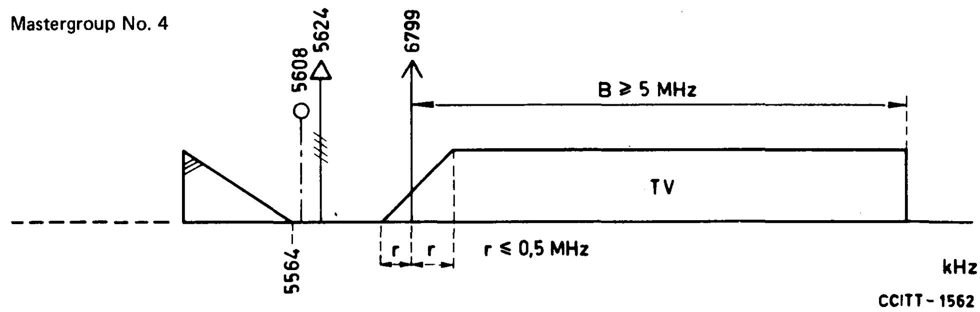


FIGURE 3/J.73 – **Frequency allocation for television transmission on a 12-MHz system**

d) *Relative power levels and interconnection at a frontier section*

It is not possible to recommend relative power levels at the output of intermediate repeaters since they are very closely linked to the inherent design of each Administration's system.

When interconnection between two telephone systems is effected via a cable section that crosses a frontier, in accordance with Recommendation G.352, each Administration should accept, on the receiving side, the level conditions which normally apply to the incoming system used in the other country. It may be possible to comply with this condition simply by insertion of a correcting network at the receiving end. The repeater section crossing the frontier should then be less than 4.5 km long. The details being agreed directly between the Administrations concerned before the repeater stations are sited.

Where a line is to be used alternatively for "all-telephony" or for "telephony-plus-television", such a solution is not generally applicable. In this case, one of the frontier stations may act as a main station having the necessary types of pre-emphasis and de-emphasis networks to permit interconnection at flat points at the recommended levels. Figure 1/J.73 shows how this may be done in the general case and also shows how, at terminal stations, the same interconnections levels are used when connecting the line to telephony and television translating equipment.

However, if a common differential characteristic can be agreed for all types of 12-MHz line, then free interconnection of the full line-bandwidth becomes possible, both nationally (e.g. between working and spare lines) and internationally (between national systems of different designs). This method leads to the simpler interconnection arrangement of Figure 2/J.73.

In this arrangement, the circuit is always lined up for "all-telephony". For telephony-plus-television, the emphasis characteristic used for the "all-telephony" case is modified by the insertion, at the terminal equipment stations only, of differential pre-emphasis and de-emphasis networks additional to those used for "all-telephony" transmission.

e) *Repeater input and output impedance*

The return loss between repeater input and output impedances and a non-reactive resistance of 75 ohms should be at least 20 dB at the carrier frequency used for television.

The limit permitted for such return loss may decrease progressively to 15 dB at the upper and lower edges of the band of frequencies transmitted for monochrome television.

Note. – Under these conditions, at the 6799-kHz carrier frequency and at adjacent frequencies, the overall resultant value of echo in a single repeater section of normal length (sum of the three terms as defined in the Annex to this Recommendation) that is obtained is considerably better than the value of 70 dB recommended. The value of 70 dB is, in fact, easily achieved throughout the transmitted band.

f) *Interference*

Recommendation J.61, 3.3 indicates the overall values relative to the hypothetical reference circuit for television transmissions which are taken as objectives for design projects.

In the experience of certain Administrations, the weighted psophometric power can be distributed between the terminal equipment and the line in the ratio of 1 to 4.

In particular, the Administration of the Federal Republic of Germany uses, for the 12-MHz system, the following signal-to-weighted noise ratio:

for terminal modulation equipment: 70 dB
for terminal demodulation equipment: 64 dB
for a line 840 km in length: 58 dB

These values result in a signal-to-noise ratio of 52 dB at the end of the reference circuit.

ANNEX

(to Recommendation J.73)

Impedance matching between repeaters and coaxial pair in television transmission

Such impedance matching, for television systems having repeater sections of about 9 km, was formerly specified by stating an overall limit, as follows (taken from pages 269 and 270 of Volume III *bis* of the CCIF *Green Book*, Geneva, 1956).

"Let:

Z_L be the measured line impedance at frequency f see from a repeater station (see Figure 1);

Z_E the measured output impedance at frequency f of the repeater station equipment seen from the line;

Z_R be the measured input impedance at frequency f of the repeater station equipment seen from the line;

A be the total line attenuation al, at frequency f, between two adjacent repeater stations, a being the measured attenuation coefficient of the coaxial pair and l the distance between the two adjacent repeater stations concerned.

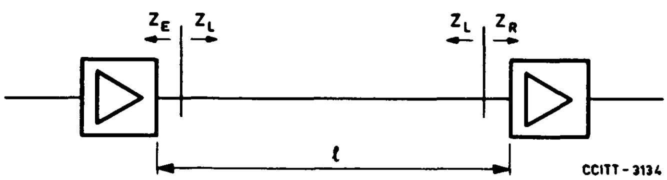


FIGURE 1 – Repeater section of a coaxial pair

Then the value N is defined by the formula

$$N = 2A + 20 \log_{10} \left| \frac{Z_E + Z_L}{Z_E - Z_L} \right| + 20 \log_{10} \left| \frac{Z_L + Z_R}{Z_L - Z_R} \right| \text{ (dB)}$$

Provisionally the condition indicated below should be met.

In the case of a television transmission system, N should be of the order of 70 dB at frequencies adjacent to the virtual carrier frequency used for the line transmission. At frequencies remote from the carrier frequency, lower values of N might be acceptable."

Since then, the CCITT has recommended limits for return loss at the input and output of repeaters, as given in the following Recommendations:

- Recommendation J.71, f), for the 4-MHz system and Recommendation J.72, g), for the 6-MHz system both of which have approximately 9-km repeater sections and a carrier frequency of 1056 kHz;
- Recommendation J.73, e), for the 12-MHz system, with approximately 4.5-km repeater sections and a carrier frequency of 6799 kHz.

These give more stringent limits than the overall limit shown above, which becomes redundant for these systems. However, if in the future the CCITT should define other television transmission systems having a large number of closely spaced repeaters, this overall limit may again become important; in that case it will be necessary to revise and correct the old recommendation quoted above, using more precise terms to specify the impedances concerned.

Recommendation J.74

METHODS FOR MEASURING THE TRANSMISSION CHARACTERISTICS OF TRANSLATING EQUIPMENTS

a) No special measuring method is necessary for the carrier.

b) An oscilloscope can be used, for example, to measure the modulation ratio.

c) No special method is recommended for measuring pre-emphasis.

d) An oscilloscope can be used, for example, to measure the voltages at the input to the modulating equipment and the output from the demodulating equipment.

e) The following is an example of a method which can be used to measure the random noise at the modulator output:

The input and output video terminals of the modulator are closed with 75-ohm resistances and the modulator is set to give an output carrier power of 1 mW. The random noise power can then be measured with a selective measuring instrument, and the result is given relative to the video-frequency bandwidth for the television system concerned.

To measure noise produced by the demodulator, 1 mW of carrier power is sent to its input, and the random noise at the output is measured at the output terminals with a selective measuring instrument.

This method can also be used to measure parasitic noise having a recurrent waveform.

Note. – Methods for measuring parasitic noise in television are being studied.

Recommendation J.75

INTERCONNECTION OF SYSTEMS FOR TELEVISION TRANSMISSION ON COAXIAL PAIRS AND ON RADIO-RELAY LINKS

A. TELEVISION TRANSMISSION ONLY

Direct video transmission over long, e.g. more than about 15 km, coaxial cables is unsatisfactory, because of the likelihood of picking up interference and the difficulties of low-frequency equalization: it is therefore necessary to transmit the television signal as a modulated carrier transmission, usually with a vestigial sideband.

On the other hand, the television signal can be transmitted directly in the baseband of a radio-relay system as a video signal; in general it is advantageous to do so, since this minimizes distortion and enables a better signal-to-noise ratio to be obtained as compared with a modulated signal with vestigial sideband, transmitted in the baseband. This procedure is recommended by the CCIR.

Interconnection between television channels on radio-relay and cable systems will therefore normally take place at video frequencies.

Levels and impedances at interconnection points should then conform to Recommendation J.61.

Exceptionally, in special cases, the video signal can be transmitted over short cables or a vestigial-sideband television signal can be transmitted on short radio-relay links, to allow direct interconnection at line frequencies (radio-relay link baseband). Special arrangements may be necessary in such cases in respect of signal level, pre-emphasis and pilots, to maintain the recommended standard of transmission performance.

B. TELEPHONY AND TELEVISION TRANSMISSION, ALTERNATIVELY OR SIMULTANEOUSLY, ON COAXIAL PAIRS OR RADIO-RELAY LINKS

a) *Interconnection between a coaxial cable system having alternative transmission of telephony and television and a radio-relay link with the same alternative transmission*

It is recommended that the following conditions should be met at the interconnection point:

- For telephony transmission, the frequency arrangements, the relative power levels of the telephone channels and the frequency of the pilots should be as indicated in Recommendation G.423.
- For television transmission, interconnection should generally be made at video frequencies. Levels and impedances at interconnection points should then conform to Recommendation J.61.

b) *Interconnection between a coaxial system having simultaneous telephony and television transmission and a radio-relay link with the same simultaneous transmission*

On all radio-relay links designed for such simultaneous transmission it is intended to transmit video-frequency television signals in the lower part of the baseband and telephony signals in the upper part. Since these arrangements are incompatible with those which are recommended by the CCITT for simultaneous telephony and television transmission on coaxial cables (Recommendation J.73), it will normally be possible to consider interconnection at video frequencies only for the television channel, and interconnection at group, supergroup or supermastergroup points for telephony.

However, by agreement between the Administrations concerned, direct interconnection may be achieved in special cases on a short system (on cable or radio) by using a frequency allocation recommended for the other type of system.

Recommendation J.76

LOCAL LINES FOR TELEVISION TRANSMISSIONS

(Geneva, 1964)

The CCITT has not issued any Recommendations concerning the characteristics of "local lines" for television transmissions as defined in Recommendation J.61, 1.1.3.

By way of information Annexes 57 to 60 (Part IV of Volume III, *Blue Book*) describe the arrangements made in various countries:

a) to connect the sending end of an international television connection to the sending terminal station of a long-distance international television circuit, and from the receiving terminal of such a circuit to the receiving end of international television connection;

b) to ensure satisfactory transmission over those local circuits and equipments which are the responsibility of the telecommunications authority.

Similar information may be found in the following articles:

MYHRMAN (A.): Video amplifying equipment for television program transmission, *Ericsson Review*, No. 2, 1963.

MAEDA (K.): Coaxial cable video transmission system, *Japan Telecommunication Review*, Volume 1, No. 2.

HORIGUCHI (T.): Transistorized coaxial cable video transmission system, *Japan Telecommunication Review*, Volume 9, No. 1.

ISBN 92-61-00351-6

PRINTED IN SWITZERLAND