INTERNATIONAL TELECOMMUNICATION UNION

CCITT

THE INTERNATIONAL
TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE

BLUE BOOK

VOLUME III – FASCICLE III.6

LINE TRANSMISSION OF NON-TELEPHONE SIGNALS

TRANSMISSION OF SOUND-PROGRAMME AND TELEVISION SIGNALS

SERIES H AND J RECOMMENDATIONS

IXTH PLENARY ASSEMBLY
MELBOURNE, 14-25 NOVEMBER 1988

Geneva 1989

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PRELIMINARY NOTE

In this Fascicle, the expression "Administration" is used for shortness to indicate both a telecommunications Administration and a recognized private operating agency.

PART I

Series H Recommendations

LINE TRANSMISSION OF NON-TELEPHONE SIGNALS

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LINES USED FOR THE TRANSMISSION OF SIGNALS OTHER THAN TELEPHONE SIGNALS, SUCH AS TELEGRAPH, FACSIMILE, DATA, ETC., SIGNALS[1]

Part I 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 this Recommendation Series.

Section 6 of this Series is reserved for Recommendations concerning the characteristics of visual telephone systems.

Table 1 indicates the correspondence of Series H Recommendations to Recommendations of other Series.

TABLE 1

Series H Recommendations	*Recommendations of other Series*
H.12, § 1	M.1040 (Volume IV)
H.12, § 2	M.1025 (Volume IV)
H.12, § 3	M.1020 (Volume IV)
H.13	See Recommendation O.71 (Volume IV)
H.14, § 2	M.910 (Volume IV)
H.16	O.72 (Volume IV)
H.21	See also the Recommendations M.800 (Volume IV) and R.77 (Volume VII)
H.22	See also the Recommendation M.810 (Volume IV)
H.23	Extract of Recommendations R.31 and R.35 (Volume VII)
H.32	R.43 (Volume VII)
H.41	T.11 (Volume VII)
H.42	T.12 (Volume VII)
H.43	T.10 (Volume VII)
H.51	V.2 (Volume VIII)

[1] Excluding the transmission of sound-programme and television signals, which is the subject of the Series J Recommendations.

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SECTION 1

LINES USED FOR THE TRANSMISSION OF SIGNALS OTHER THAN TELEPHONE SIGNALS, SUCH AS TELEGRAPH, FACSIMILE, DATA, ETC., SIGNALS

1.1 Characteristics of transmission channels used for other than telephone purposes

Recommendation H.11

CHARACTERISTICS OF CIRCUITS IN THE SWITCHED TELEPHONE NETWORK

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.12

CHARACTERISTICS OF TELEPHONE-TYPE LEASED CIRCUITS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.13

CHARACTERISTICS OF AN IMPULSIVE NOISE MEASURING INSTRUMENT FOR TELEPHONE-TYPE CIRCUITS

(The text of this Recommendation can be found in Recommendation O.71 in Fascicle IV.4 of Volume IV of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.14

CHARACTERISTICS OF GROUP LINKS FOR THE TRANSMISSION OF WIDE-SPECTRUM SIGNALS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.15

CHARACTERISTICS OF SUPERGROUP LINKS FOR THE TRANSMISSION OF WIDE-SPECTRUM SIGNALS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.16

CHARACTERISTICS OF AN IMPULSIVE-NOISE MEASURING INSTRUMENT FOR WIDEBAND DATA TRANSMISSION

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

1.2 Use of telephone-type circuits for voice-frequency telegraphy

Recommendation H.21

COMPOSITION AND TERMINOLOGY OF INTERNATIONAL VOICE-FREQUENCY TELEGRAPH SYSTEMS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.22

TRANSMISSION REQUIREMENTS OF INTERNATIONAL VOICE-FREQUENCY TELEGRAPH LINKS (AT 50, 100 AND 200 BAUDS)

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.23

BASIC CHARACTERISTICS OF TELEGRAPH EQUIPMENTS
USED IN INTERNATIONAL VOICE-FREQUENCY TELEGRAPH SYSTEMS

(The text of this Recommendation can be found in Fascicle III.4
of the *Red Book*, ITU, Geneva, 1985)

1.3 Telephone circuits or cables used for various types of telegraph transmission or for simultaneous transmissions

Recommendation H.32

SIMULTANEOUS COMMUNICATION BY TELEPHONY AND TELEGRAPHY
ON A TELEPHONE-TYPE CIRCUIT

(The text of this Recommendation can be found in Fascicle III.4
of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.34

SUBDIVISION OF THE FREQUENCY BAND OF A TELEPHONE-TYPE CIRCUIT
BETWEEN TELEGRAPHY AND OTHER SERVICES

(The text of this Recommendation can be found in Fascicle III.4
of the *Red Book*, ITU, Geneva, 1985)

1.4 Telephone-type circuits used for facsimile telegraphy

Recommendation H.41

PHOTOTELEGRAPH TRANSMISSIONS ON TELEPHONE-TYPE CIRCUITS

(The text of this Recommendation can be found in Fascicle III.4
of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.42

RANGE OF PHOTOTELEGRAPH TRANSMISSIONS ON A TELEPHONE-TYPE CIRCUIT

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.43

DOCUMENT FACSIMILE TRANSMISSIONS ON LEASED TELEPHONE-TYPE CIRCUITS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

1.5 Characteristics of data signals

Recommendation H.51

POWER LEVELS FOR DATA TRANSMISSION OVER TELEPHONE LINES

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.52

TRANSMISSION OF WIDE-SPECTRUM SIGNALS (DATA, FACSIMILE, ETC.) ON WIDEBAND GROUP LINKS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation H.53

TRANSMISSION OF WIDE-SPECTRUM SIGNALS (DATA, ETC.) OVER WIDEBAND SUPERGROUP LINKS

(The text of this Recommendation can be found in Fascicle III.4 of Volume III of the *Red Book*, ITU, Geneva, 1985)

SECTION 2

CHARACTERISTICS OF VISUAL TELEPHONE SYSTEMS

Recommendation H.100

VISUAL TELEPHONE SYSTEMS

(former Recommendation H.61, Geneva, 1980;
amended at Malaga-Torremolinos, 1984
and at Melbourne, 1988)

1 Definition

The **visual telephone service** is generally a two-way telecommunication service which uses a switched network of broadband analogue and/or digital circuits to establish connections among subscriber terminals, primarily for the purpose of transmitting live or static pictures.

Special application one-way systems, e.g. surveillance and some information retrieval systems, or a non-switched videoconference service, can be regarded as degenerate cases of the visual telephone service.

The visual telephone service also includes the associated speech.

2 Facilities to be offered

The design of the visual telephone service shall be such as to offer at least the following basic facilities:

a) Transmission of live pictures such as head and shoulders of one person or a small group of persons, with moderate definition.

b) Transmission of the associated speech.

c) Transmission of graphics information such as drawings and documents with high definition (e.g. 625 lines or 525 lines).

d) Video conference service, with or without the use of split-screen techniques.

The above-mentioned services shall, in general, be bi-directional, although uni-directional operation should be possible. Also, some of the facilities can be omitted, if not required, in order to minimize costs.

Note – At the subscriber terminal, the use of ancillary equipments, e.g. for document reproduction, video tape recordings, etc., shall be possible.

3 System parameters

3.1 *Picture standards*

3.1.1 The video standards of the subscriber sets shall be compatible with, readily convertible to, or identical to, the local broadcast television standards.

3.1.2 Two classes of picture standards are recommended for the visual telephone system. They are given in Table 1/H.100.

TABLE 1/H.100

Picture standards

Class	Items	The region to which the figures should be applied	
		Regions where TV broadcasting uses 25 pictures per second	Regions where TV broadcasting uses 30 pictures per second
a	Number of horizontal scanning lines	625	525
	Pictures per second	25 (2:1 interleaved)	30 (2:1 interleaved)
	Aspect ratio	4:3	4:3
	Video bandwidth	5 MHz	4 MHz
b	Number of horizontal scanning lines	313	263
	Pictures per second	25 (2:1 interleaved)	30 (2:1 interleaved)
	Aspect ratio	4:3	4:3
	Video bandwidth	1 MHz	1 MHz

Class *a* standards are identical to the local broadcast video standards and will, in most cases, give sufficient definition for real-time picture transmission of a group of people (e.g. for conferencing) and of graphics material.

Class *b* standards give sufficient definition for real-time transmission of a head and shoulder picture of one person or a small group. For the transmission of graphics information or other still pictures with high definition, a slow-scan technique has to be applied. For instance, a system using 625 or 525 horizontal scanning lines and 5, or less, pictures per second which gives a Class *a* definition in the 1 MHz bandwidth.

Further study is required to define slow scanning parameters.

4 Characteristics relating to split-screen techniques for Class *a* television conference systems[1)]

In television conference systems which use split-screen techniques to make more effective use of the picture area, the following features for the terminals and transmitted signals are recommended. Preferred seating arrangement for such systems are given in Annex A.

4.1 *Picture format*

The transmitted picture should be 4 : 3 aspect ratio, split into upper and lower halves corresponding to the groups of seats. Viewed from the camera system, the left-hand group should be in the upper half and the right-hand group in the lower half.

The split should occur at the end of lines 166 and 479 for 625-line television systems and at the end of line 142 in Field 1 and line 141 in Field 2 for 525-line television systems, as shown in Figure 1/H.100.

1) Split-screen techniques for systems using Class *b* standards require future study.

Before display, the receive equipment may discard half lines and first and last lines which are liable to be averaged during standards conversion or vertical aperture correction of mixed signals.

4.2 *Identification signal for split-screen system*

4.2.1 *Analogue video signals*

The identification signal for split-screen system should be inserted in the vertical blanking period, because the control is required for each television frame or field.

The line where the identification signal is inserted and its signal format are under study.

4.2.2 *Digital video signals*

An identification signal for split-screen system should be provided. In the case of codecs in Recommendations H.120 and H.130 the format shall be that specified in Recommendation H.130.

4.3 *Compatibility with non-split-screen systems*

The simplest kind of a video telephone terminal is composed of a single camera and other equipments. These terminals may be interconnected with split-screen system terminals. In that case, mechanical masks (if used) for the two split-screen displays (aspect ratio = 4 : 1.5) need to be removed, or if a display with 4 : 3 aspect ratio needs to be installed additionally.

4.4 *Cameras and displays arrangement*

The entrance pupils of the TV camera optical system should be as near as possible to the centre of the TV display showing remote conferees, in order to minimize errors in eye contact angle.

Unless means are employed to place these pupils in line with the display, e.g. by use of half-silvered mirrors, the camera system should be sited above the display and central to it.

In order to keep the maximum horizontal errors as small as possible, the cameras used had better be in a cross-fire system, as for example in Figure A-1/H.100, and the camera/display assembly should be sited on the central axis of the terminal. However, in some cases, adoption of parallel-fire system as shown also in Figure A-1/H.100 is necessary due to a restriction in equipment arrangement.

Whether the two cameras are arranged in cross-fire or parallel-fire is left open to each Administration since the selection does not affect the interconnection of different systems.

4.5 *Picture processing methods at transmitting terminals*

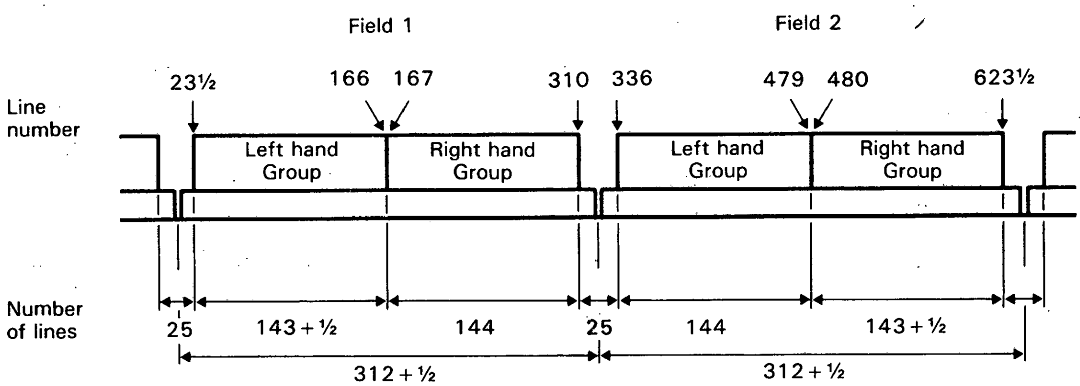
In order to obtain the correct relationship between the signals from the two cameras for split-screen working, the cameras should be synchronized but the vertical drive pulses should be rephased. The drive to one should be advanced by one quarter of the vertical period while the drive to the other should be retarded by the same amount. This causes a central strip of the target of each camera tube to be used and so minimizes the effects of distortions in the corners of the targets. Figure B-1*a)*/H.100 illustrates the preferred method.

Alternative methods which are not recommended although they do not give rise to problems of end-to-end compatibility are compared in Annex B.

4.6 *Receiving equipment*

The receiving equipment should be capable of working with discontinuities in the received signal that may be caused by switching between non-synchronous video sources.

Note – A split-screen device should be capable of working with a codec with the input and output frequency tolerances as specified in Recommendation H.120.

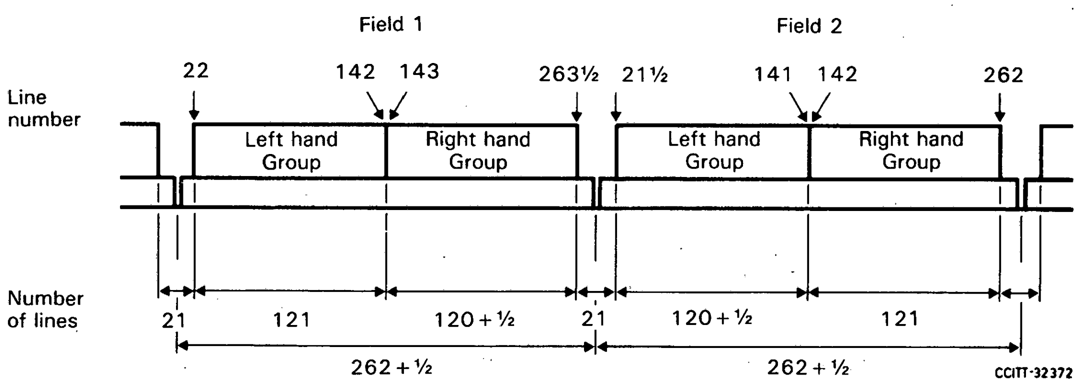


Left hand group: first complete lines 24 and 336
last complete lines 166 and 479

Right hand group: first complete lines 167 and 480
last complete lines 310 and 622

Lines 16-20 inclusive and 329-333 inclusive may contain identification, control or test signals.

a) 625-line television system



Left hand group: first complete lines 22 (Field 1, 2)
last complete lines 142 (Field 1), 141 (Field 2)

Right hand group: first complete lines 143 (Field 1), 142 (Field 2)
last complete lines 262 (Field 1, 2)

Lines 10-21 inclusive in Field 1 and 9½-21½ inclusive in Field 2 may contain identification, control or test signals.

b) 525-line television system

Note 1 – **The method of defining the line number is following Fig. 2-1 of CCIR Report 624 for 625 system, and Fig. 2-3 for 525 system.**

Note 2 – The notation which is used for line numbers is as follows. Line 23½ means that the picture starts (or finishes) half-way along line number 23. When totalling lines, a half-line is shown separately, e.g. 143 + ½.

FIGURE 1/H.100

Vertical format of split-screen video signal

ANNEX A

(to Recommendation H.100)

Seating arrangements when applying split-screen techniques for class *a* system

Preferred arrangements for video conferences using split-screen techniques are:

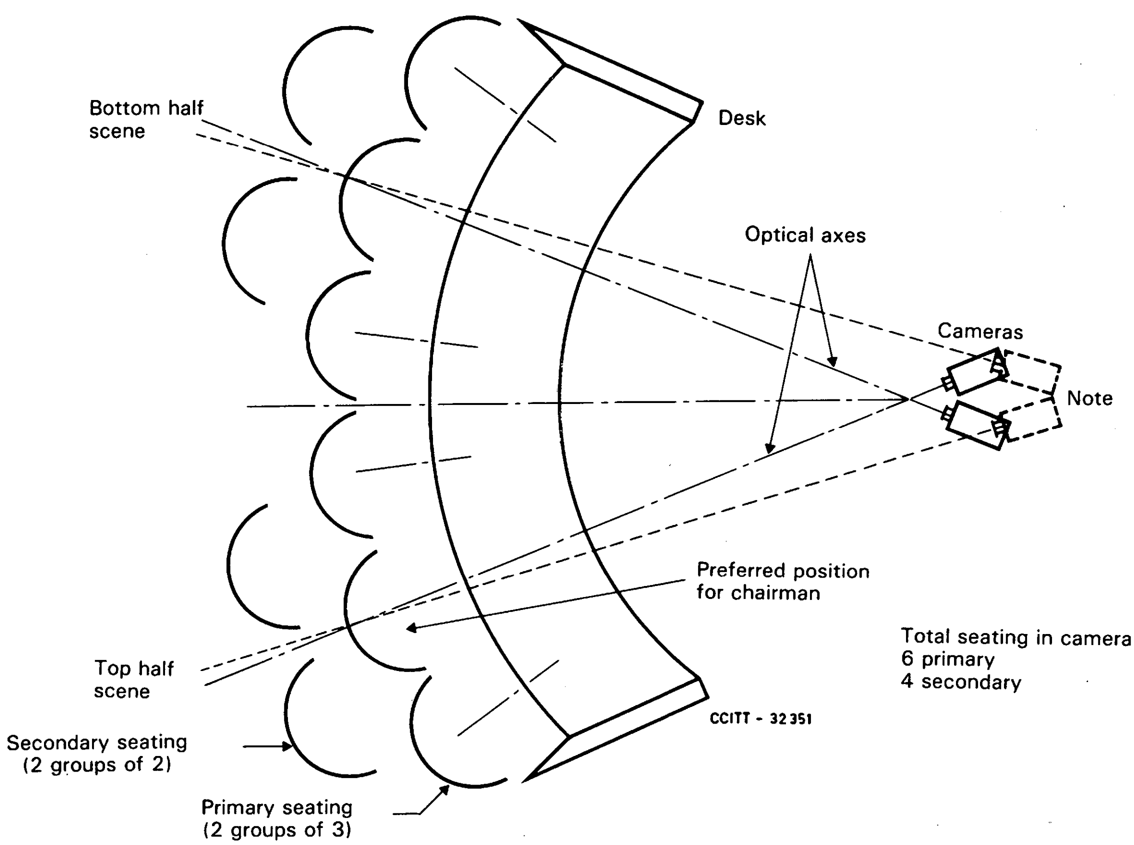
A.1 The conference terminal accommodation should be for 6 primary seats in two adjacent groups of 3 as shown in Figure A.1/H.100.

Provision for additional seating behind may be made, so long as allowance is made for the central gap between the two halves. For example, 4 additional persons may be seated on a second row as in the Figure.

A.2 The chairman's position should be in the centre of the left-hand group of seats (viewed from the camera) with user controls accessible from both this position and the one of the chairman's left.

Consequently, when split-screen pictures are displayed, stacked as received (i.e. shown as 3 over 3), the chairman's position is standardized as top centre.

The suite of 3 chairs containing the chairman's position should also be regarded as the primary position for occasions when only half of a studio is in use. Such standardization is necessary for connection of 3 studios in conference using time-division multiplex of pairs of TV signals to share a common trunk between two studios.



Note – Solid line cameras are for cross-fire. Dashed line cameras are for parallel-fire.

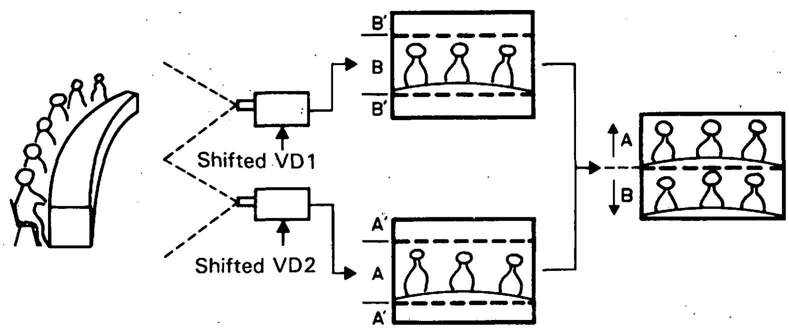
FIGURE A-1/H.100
Studio plan view

ANNEX B

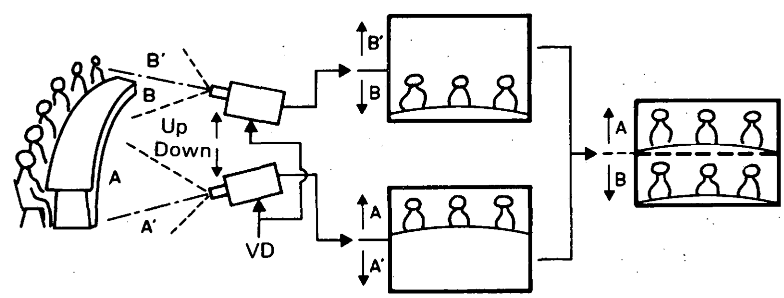
(to Recommendation H.100)

Picture processing methods in transmitting terminals

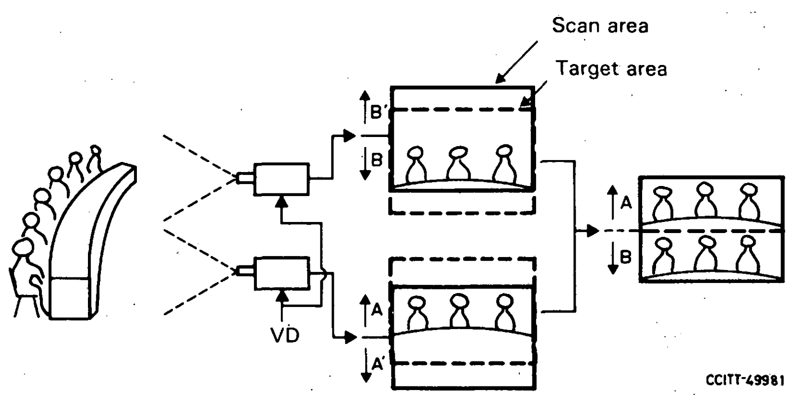
Alternative methods of obtaining the split-screen signal which are compatible with the recommended method and which might be useful for experiments and demonstrations are shown in *b)* and *c)* of Figure B-1/H.100. In method *b)*, the two cameras are directed upward and downward to pick up right and left halves of the conferencing room, respectively. Since circumferences of target and scanning areas are used, geometric and brightness distortions tend to occur. In method *c)*, vertical deflection currents are biased by the quantity corresponding to ± 1/4 of target height. Vertical deflection bias adjustment is needed every time cameras are exchanged. In method *a)*, the vertical driving pulses are phase-shifted by ± 1/4 V. The recommended method, *a)*, avoids the problems of methods *b)* and *c)*.



a) Vertical driving pulses are phase shifted



b) Cameras are directed upward and downward



c) Vertical deflection currents are biased

VD = vertical deflection

FIGURE B-1/H.100

Picture processing method at transmitting terminals

HYPOTHETICAL REFERENCE CONNECTIONS FOR VIDEOCONFERENCING USING PRIMARY DIGITAL GROUP TRANSMISSION

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The CCITT,

considering

(a) that there is growing evidence of a customer demand for a videoconference service;

(b) that circuits to meet this demand can, at present, be provided effectively by digital transmission using the primary digital group;

(c) that switched digital transmission networks known as the Integrated Digital Network (IDN) and Integrated Services Digital Network (ISDN) are under study, but the methods of exploiting these networks for the transmission of primary digital groups will not become clear until the studies have progressed further;

(d) that the existence of different digital hierarchies and different television standards in different countries complicates the problems of defining hypothetical reference connections;

(e) that a hypothetical reference connection may be used as a guide to simplify the problems of connections between countries with different television standards and digital hierarchies,

appreciating

that rapid advances are being made in research and development of video coding and bit-rate reduction techniques which may lead to further Recommendations being proposed for hypothetical reference connections for videoconferencing at bit rates which are multiples or sub-multiples of 384 kbit/s during subsequent study periods, so that this may be considered as the first of an evolving series of Recommendations,

noting

(a) that a hypothetical reference connection is a model in which studies relating to overall performance may be made, thereby allowing comparisons with standards and objectives; on this basis, limits for various impairments can be allocated to the elements of the connection;

(b) that such a model may be used:

- by an Administration to examine the effects on transmission quality of possible changes of impairment allocations in national networks,
- by the CCITT for studying the allocation of impairments to component parts of international networks,
- to test national rules for prima facie compliance with any impairment criteria which may be recommended by the CCITT for national systems;

(c) that hypothetical reference connections are not to be regarded as recommending particular values of impairments allocated to constituent parts of the connection, and they are not intended to be used for the design of transmission systems,

and recognizing

that the planning of the necessary transmission networks for a videoconference service will be facilitated if recommended hypothetical reference connections are available, even if only in a preliminary form without details of all transmission and switching arrangements, *recommends*

(1) that the hypothetical reference connection and means for digital transmission illustrated in Figures 1/H.110 and 2/H.110 shall be used as the model for studies of the overall performance of international videoconference connections, both intra-regional[1] and inter-regional[1], which are provided using minimum numbers of encoding and decoding equipments;

(2) that hypothetical reference connections of a more complex type, as, for example, those illustrated in Figure 3/H.110, being representative of many connections that may be employed in practice, should be studied further.

Note 1 – The hypothetical reference connection shown in Figure 1/H.110 contains the basic transmission elements, but is incomplete because switching has been excluded and the local ends and parts of the national network at each end of the connection have been left unspecified.

Note 2 – Because the arrangements of transmission systems interconnecting regions using different digital hierarchies have not yet been standardized, and because videoconferencing is likely to be a minority service in such transmission systems, it seems prudent to consider videoconference connections both where the primary hierarchical level on the inter-regional link is 1.5 Mbit/s and where it is 2 Mbit/s. In Figure 2b/H.110, the change between 2048 kbit/s and 1544 kbit/s transmission is placed at the 2048 kbit/s end of the long international network. The long distance part of the connection is thus operated at the lower bit rate. Where the international network is provided on a system which uses the 2048 kbit/s hierarchy, Figure 2c/H.110 maintains the efficiencies offered by the arrangement shown in Figure 2b/H.110, by making available the six vacated time slots for other use. Figure 2d/H.110 offers the possibility of improved picture quality compared with Figures 2b/H.110 and 2c/H.110 by making full use of the available 2048 kbit/s for the videoconferencing signal. This arrangement would require a 2048 kbit/s codec compatible with 525-line Video Standards, or the use of an external standards converter. This is for further study.

Note 3 – The lengths which have been assigned to the parts of the connections have been arbitrarily chosen, but have some consistency with existing CCITT and CCIR Recommendations. They are intended to be representative of long international connections, but not the longest possible. The lengths will likely require revision when studies on the error rates of digital paths have progressed to the stage when the error rates of the paths used in the connections can be predicted.

Note 4 – The propagation delay is one of the main factors to be studied based upon the structures and lengths of the connections in Figures 1/H.110, 2/H.110 and 3/H.110. However, in the absence of subjective test results, the specification of requirements for videoconferencing connections must await further study. This study and particularly operational experience are required to determine the extent to which Recommendation G.114, which applies to telephone connections, relates to videoconferencing connections.

Note 5 – In Figures 1/H.110 and 3/H.110, the codecs may be located anywhere within the international or national networks including the international gateway or the customer's premises.

Note 6 – The extensions beyond the codec shown as A or D in Figures 1/H.110 and 3/H.110 may include wideband analogue or high-speed digital transmission systems on terrestrial bearers. It is not expected that these transmission systems will have any significant influence on the quality of the picture or sound, or, on the propagation delay, other than that due to their length.

Note 7 – For inter-regional operation, television standards conversion between 525-line and 625-line video signals may be required. This conversion may be performed by the codecs themselves, or provided by external equipment.

Note 8 – The arrangements shown in Figure 2/H.110 provide for the simplest means of transmission. More complex means are possible and are not precluded.

Note 9 – The hypothetical reference connection shown in Figure 3/H.110 is of a more complex type than the connection shown in Figure 1/H.110, in that it includes codecs in cascade, and, possibly an external Television Standards Converter. The picture quality attainable with these more complex connections may be degraded with respect to that attainable using the connection illustrated in Figure 1/H.110. This and other aspects of the more complex connection must be studied further.

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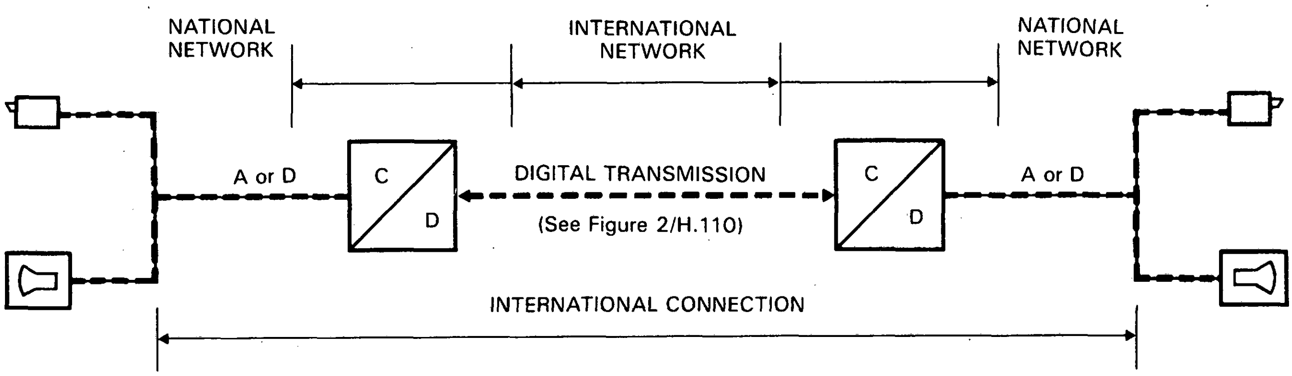
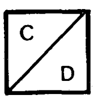
[1] The term "intra-regional" is used here to describe connections within a group of countries which share a common television scanning standard and a common digital hierarchy, and may or may not be in geographical proximity. The term "inter-regional" is used here to describe connections between groups of countries which have different television scanning standards and/or different digital hierarchies.



A or D – Wideband analogue or digital transmission or both all providing equivalent quality.

National option.

Digital transmission – Circuits for intra-regional or interregional digital transmission at the primary rate. This includes the international network and any national digital extensions thereof (see Figure 2/H.110).



– Types of codec which may be used in the hypothetical reference connections are indicated below. Each can work with others of the same type and can interwork with other types shown, using a remultiplexer, if required. Codecs which perform these functions are described in Recommendation H.120.

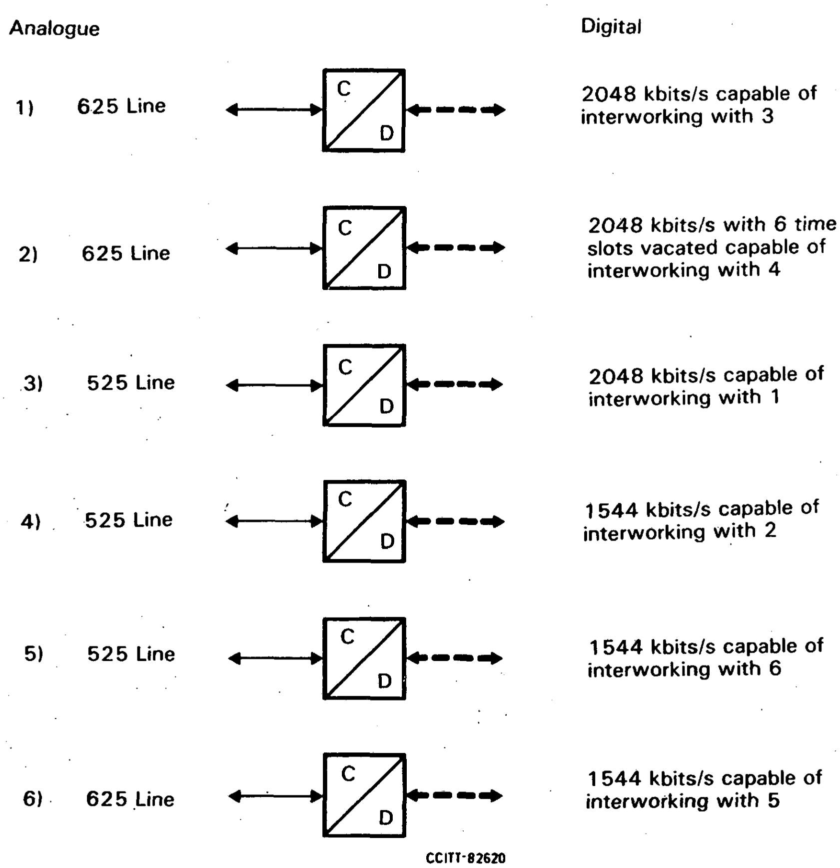
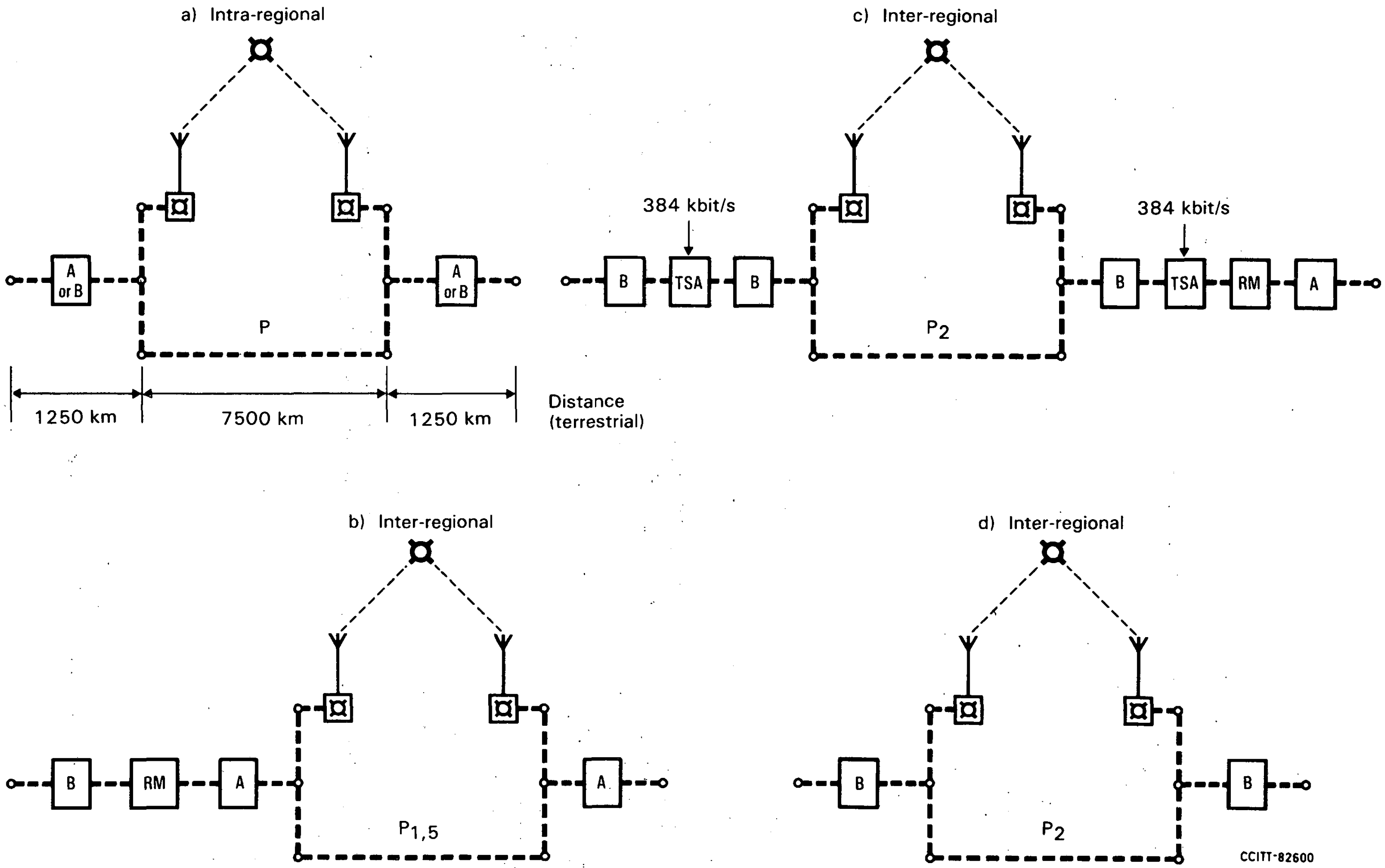


FIGURE 1/H.110

Hypothetical reference connection



Note – The distances shown in Figure 2a/H.110 are applicable to Figures 2b/H.110, 2c/H.110 and 2d/H.110. These distances refer to terrestrial transmission. Equivalent distances relating to satellite transmissions are for further study.

FIGURE 2/H.110

Means for digital transmission

Symbols for Figure 2/H.110

A — Termination of a 1544 kbit/s circuit with G.733 interface.

B — Termination of a 2048 kbit/s circuit with G.732 interface.

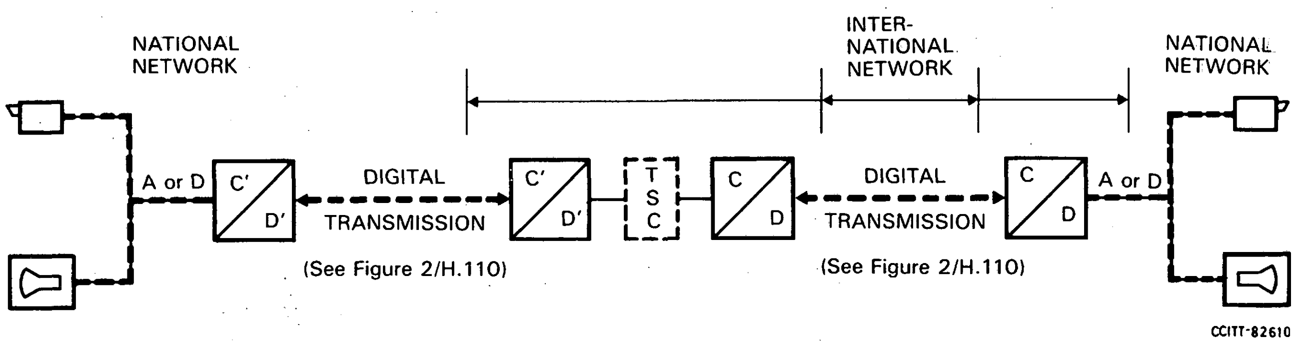
RM — Remultiplex Unit. This provides bit rate conversion between the 1544 kbit/s frame and 2048 kbit/s frame from which 6 timeslots have been vacated.

TSA — Optional Time Slot Access Unit. This provides means of inserting and extracting 384 kbit/s from the 2048 kbit/s frame which is not used for video conferencing.

P — Primary level of digital hierarchy ($y + n \times 384$ kbit/s, where $n = 5$ or 4 and $y = 128$ or 8 kbit/s, respectively).

$P_{1.5}$ — 1544 kbit/s.

P_2 — 2048 kbit/s.



Same symbols as Figure 1/H.110, and

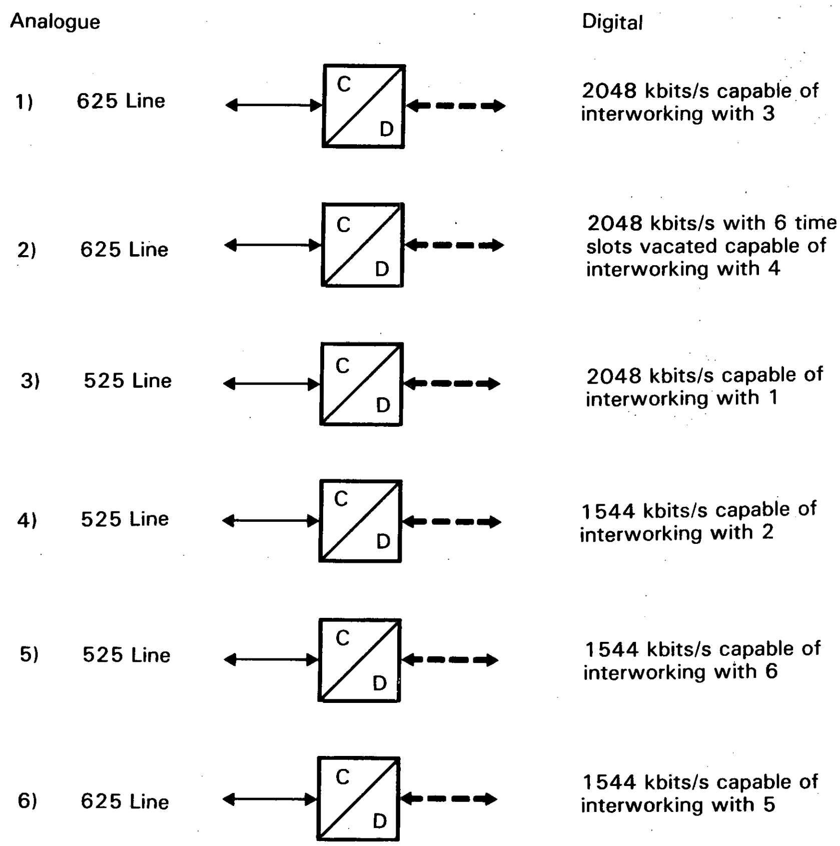
C'/D' — Codecs in the HRC of Figure 3/H.110 which may be any compatible (with one another) combination of those defined as C/D in Figure 1/H.110 but not capable of interworking with the specific C/D codecs in Figure 3/H.110.

TSC — External Television Standards Converter.
May or may not be required in the connection.

FIGURE 3/H.110

Complex hypothetical reference connection

Symbols for Figure 3/H.110:



Recommendation H.120

CODECS FOR VIDEOCONFERENCING USING PRIMARY DIGITAL GROUP TRANSMISSION

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The CCITT,

considering

(a) that there is growing evidence of a customer demand for a videoconference service;

(b) that circuits to meet this demand can, at present, be provided effectively by digital transmission using the primary digital group;

(c) that the existence of different digital hierarchies and different television standards in different parts of the world complicates the problems of specifying coding and transmission standards for international connections;

(d) that the eventual use of switched digital transmission networks should be taken into account,

appreciating

that rapid advances are being made in research and development of video coding and bit-rate reduction techniques which may lead to further Recommendations being proposed for videoconferencing at bit rates which are multiples or submultiples of 384 kbit/s during subsequent study periods so that this Recommendation may be considered as the first of an evolving series of Recommendations,

and noting

that it is a basic objective of the CCITT to recommend a unique solution for international connections as far as possible,

recommends

that the codecs having signal processing and interface characteristics described in §§ 1, 2 and 3 below, should be used for international videoconference connections.

Note – Codecs of types other than those described in this Recommendation are not precluded.

Introduction

Section 1 of this Recommendation specifies the codec, developed for operation with the 625-line, 50 field/s television standard and the 2048 kbit/s primary digital group. Its architecture has been chosen to permit variations in the detailed design of certain of the functional elements having the greatest influence on the picture quality. This enables future developments, aimed at improving the performance, to be incorporated without affecting the ability of different coders and decoders to interwork. For this reason, no details are given of such items as motion detectors or spatial and temporal filters. The Recommendation confines itself to the details necessary to enable a decoder correctly to interpret and decode the received signals.

The annexes to § 1 which can be found at the end of this Recommendation give details of some additional optional features which may be provided to supplement the basic design.

Under the general heading of codecs not requiring separate television standards conversion when used on interregional connections, § 2 describes a version of the codec for 525 line, 60 field/s and 1544 kbit/s operation which also provides automatic television standards conversion when connected to the version of the codec described in § 1 via a re-multiplexing unit (to convert between frame structures defined in §§ 2.1 and 2.3 of Recommendation G.704) at the junction of the 2048 and 1544 kbit/s digital paths. This codec is also suitable for use within regions using the 525-line, 60 field/s television standard and 1544 kbit/s transmission.

Other implementations of § 2 are to be studied, for example:

– a version of the codec for 625-line, 50 field/s and 2048 kbit/s operation capable of interworking with the codec described in § 3;

– a version of the codec for 525-line, 60 field/s and 2048 kbit/s operation capable of interworking with the codec described in § 1.

Section 3 of the Recommendation describes a codec for intra-regional use in 525-line, 60 field/s and 1544 kbit/s regions.

The frame structures associated with the codecs described in this Recommendation are to be found in Recommendation H.130.

As the codecs are complex items using combined intraframe and interframe picture-coding techniques which tend to be known only to specialists, Appendix I is provided giving a brief outline of the principles involved in the codecs of §§ 1 and § 2.

1 A codec for 625-lines, 50 fields/s and 2048 kbit/s transmission for intra-regional[1] use and capable of interworking with the codec of § 2

1.1 *Scope*

Section 1 defines the essential features of a codec for the digital transmission, at 2048 kbit/s, of signals for videoconference or visual telephone service in accordance with Recommendation H.100. The video input to the coder and output from the decoder in a 625-line, 50 field/s signal, according to the "Class *a*" standard of Recommendation H.100, or alternatively, the 313-line, 50 field/s signal of the "Class *b*" standard. Provision is also made for a sound channel and optional data channels. A brief description of the operation of the codec is given in Appendix I.

The Recommendation starts with a brief specification of the codec (§ 1.2) and a description of the video interface. This is followed by details of the source coder (§ 1.4) which provides analogue-to-digital conversion followed by recoding with substantial redundancy reduction in the face-to-face mode. Paragraph 1.5 deals with the video multiplex coder which inserts instructions and addresses into the digitized video signal to control the decoder so that it correctly interprets the signals received. Paragraph 1.6 describes the transmission coder which arranges the various digital signals (video, sound, data, signalling) into a form compatible with Recommendation G.732 for transmission over 2048 kbit/s digital paths. Paragraph 1.7 describes optional forward error correction facilities. Provision is made in the digital frame structure for the inclusion of other optional facilities such as a graphics mode, encryption and multipoint conferencing. Details of such facilities as are at present available are given in the annexes to this Recommendation.

1.2 *Brief specification*

1.2.1 *Video input/output*

The video input and output are standard 625-line, 50 field/s colour or monochrome television signals. The colour signals are in, or converted to, component form. Colour and monochrome operation are fully compatible.

1.2.2 *Digital output/input*

The digital output and input are at 2048 kbit/s, compatible with the frame structure of Recommendation G.704.

1.2.3 *Sampling frequency*

The video sampling frequency and the 2048 kHz network clock are asynchronous.

1.2.4 *Coding techniques*

Conditional replenishment coding supplemented by adaptive digital filtering, differential PCM and variable-length coding are used to achieve low bit-rate transmission.

1.2.5 *Audio channel*

An audio channel using 64 kbit/s is included. At present, coding is A-law according to Recommendation G.711, but provision is made for future use of more efficient coding.

1.2.6 *Mode of operation*

The normal mode of operation is full duplex.

[1] The term "intra-regional" is used here to describe connections within a group of countries which share a common television scanning standard and a common digital hierarchy, and may or may not be in geographical proximity. The term "inter-regional" is used here to describe connections between groups of countries which have different television scanning standards and/or different digital hierarchies.

1.2.7 *Codec-to-network signalling*

An optional channel for codec-to-network signalling is included. This conforms to emerging ideas in CCITT for switching 2-Mbit/s paths in the ISDN.

1.2.8 *Data channels*

Optional 2 × 64 kbit/s and 1 × 32 kbit/s data channels are available. These are used for video if not required for data.

1.2.9 *Forward error correction*

Optional forward error correction is available. This is required only if the long-term error rate of the channel is worse than 1 in 10^6.

1.2.10 *Additional facilities*

Provision is made in the digital frame structure for the future introduction of encryption, a graphic mode and multipoint facilities.

1.2.11 *Propagation delay*

When the coder buffer is empty and the decoder buffer full, the coder delay is less than 5 ms and the decoder delay is 130 ± 30 ms at 2 Mbit/s or 160 ± 36 ms when only 1.5 Mbit/s are in use[2].

1.3 *Video interface*

The normal video input is a 625-line, 50 field/s signal in accordance with CCIR Recommendation 472. When colour is being transmitted, the input (and output) video signals presented to the analogue/digital convertors (and from the digital/analogue convertors) are in colour-difference component form. The luminance and colour-difference components, E'_Y, $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ are as defined in CCIR Report 624. The analogue video input (and output) interface with the codec may be in the form of colour-difference components, colour components (R, G, B) or as a composite colour signal. The video interface is as recommended in CCIR Recommendation 656.

Optionally, any other video standard which can be converted to give 143 active lines per field may be used.

1.4 *Source coder*

1.4.1 *Luminance component or monochrome*

1.4.1.1 *Analogue-to-digital conversion*

The signal is sampled to produce 256 picture samples per active line (320 samples per complete line). The sampling pattern is orthogonal and line, field and picture repetitive. For the 625-line input, the sampling frequency is 5.0 MHz, locked to the video waveform.

Uniformly quantized PCM with 8 bits/sample is used.

Black level corresponds to level 16 (00010000).

White level corresponds to level 239 (11101111).

PCM code words outside this range are forbidden (the codes being used for other purposes). For the purposes of prediction and interpolation, the final picture element in each active line (i.e. picture element 255) is set to level 128 in both encoder and decoder.

In all arithmetic operations, 8-bit arithmetic is used and the bits below the binary point are truncated at each stage of division.

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[2] These are typical figures. The delays depend on the detailed implementation used.

1.4.1.2 *Pre- and post-filtering*

In addition to conventional anti-aliasing filtering prior to analogue-to-digital conversion, a digital transversal filtering operation is carried out on the 625-line signal to reduce the vertical definition of the picture prior to conditional replenishment coding. As a result of this process, 143 active lines per field are used instead of the 287½ active lines of the 625-line signal, although the effective vertical definition is greater than one-half of that of a normal 625-line display. An interpolation process in the decoder restores the 625-line signal waveform.

1.4.1.3 *Conditional replenishment coding*

A movement detector identifies clusters of picture elements which are deemed to be moving. The basic feature is a frame memory which stores 2 fields of 143 lines, each line containing 256 addressable points. The memory is updated at the picture rate and differences between the incoming signal and the corresponding stored values are used to determine the moving area in the coder. A similar frame memory must exist at the decoder and be similarly updated under the control of addressing information received from the coder. It is not necessary to specify the techniques used for movement detection because they do not affect interworking, although they do affect the resultant picture quality.

Detected moving areas are transmitted by differential PCM with a maximum of 16 quantization levels. The first picture element in each moving area is transmitted by PCM. Variable-length coding is used on the DPCM code words.

The first picture element of each cluster and the complete PCM lines, when they are transmitted to provide systematic or forced updating, are coded in accordance with § 1.4.1.1.

1.4.1.3.1 *DPCM prediction algorithm*

The algorithm used for DPCM prediction is:

$$X = \frac{A + D}{2}, \text{ where X is the sample being predicted. (See Figure 1/H.120.)}$$

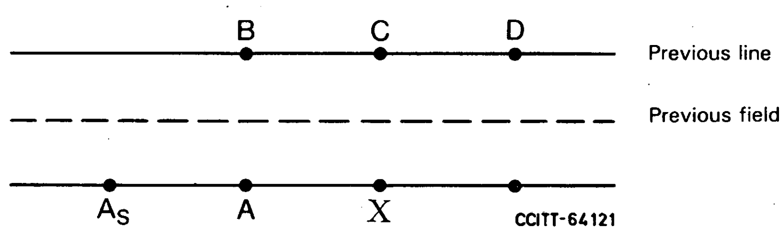


FIGURE 1/H.120
Identification of samples

For the purpose of prediction, line and field blanking are assumed to be at level 128 (out of 256).

1.4.1.3.2 *Quantization law and variable-length coding*

511 input levels are quantized to a maximum of 16 output levels. The quantizer does not assume the use of modulo 256 arithmetic.

The quantization law and associated variable-length codes which are used for both luminance and colour-difference picture elements in moving areas which are not horizontally subsampled are given in Table 1/H.120.

TABLE 1/H.120

Code table for non-horizontally-subsampled moving areas

Input levels	Output levels	Variable-length code	Code No.
−255 to −125	−141	1 0 0 0 0 0 0 0 0 1	17
−124 to − 95	−108	1 0 0 0 0 0 0 0 1	16
− 94 to − 70	− 81	1 0 0 0 0 0 0 1	15
− 69 to − 49	− 58	1 0 0 0 0 0 1	14
− 48 to − 32	− 39	1 0 0 0 0 1	13
− 31 to − 19	− 24	1 0 0 0 1	12
− 18 to − 9	− 13	1 0 1	10
− 8 to − 1	− 4	1 1	9
0 to 7	+ 3	0 1	1
8 to 17	+ 12	0 0 1	2
18 to 30	+ 23	0 0 0 1	3
31 to 47	+ 38	0 0 0 0 1	4
48 to 68	+ 57	0 0 0 0 0 1	5
69 to 93	+ 80	0 0 0 0 0 0 1	6
94 to 123	+107	0 0 0 0 0 0 0 1	7
124 to 255	+140	0 0 0 0 0 0 0 0 1	8

The end-of-cluster code is 1 0 0 1 and is designated as code number 11. The end-of-cluster code is omitted at the end of the last cluster in a line irrespective of whether it is a luminance cluster or a colour-difference cluster.

1.4.1.4 *Subsampling*

As the buffer fills, horizontal subsampling and field/field subsampling are introduced.

1.4.1.4.1 *Horizontal subsampling*

Horizontal subsampling is carried out only in moving areas. Normally, in this mode, only even elements are transmitted on even numbered lines and odd elements on odd numbered lines. This gives rise to a line quincunx pattern in moving areas.

Omitted elements are interpolated in the decoder by averaging the two horizontally adjacent elements.

Interpolated picture elements are placed in the frame stores. A moving area cluster will always start with a PCM value and finish with a transmitted DPCM picture element, even during subsampling. This means that in in some instances, the transmitted cluster needs to be extended by one element in comparison with the moving area declared by the movement detector. At the end of the active line, however, this cannot occur as clusters must not extend into blanking, so cluster shortening by one element can be necessary.

Adaptive element subsampling allows the transmission of normally omitted elements, either to remove interpolation errors or, to provide a softer switch to subsampling and thus improve the picture quality. The signalling of the extra elements is achieved by using, on horizontally subsampled lines only, 8 quantizing levels for normally transmitted elements and the remaining 8 levels for the extra elements. Also, a cluster can finish either on a normally transmitted element or an "extra" element.

During horizontally subsampled lines, the quantization law and variable-length code shown in Table 2/H.120 will be used for both luminance and colour-difference samples in moving areas.

TABLE 2/H.120

Quantization law and variable-length code table

Quantization		Variable-length codes			
Input range	Output levels	Normal elements	Code No.	Extra elements	Code No.
−255 to −41	−50	1 0 0 0 0 0 0 1	15	1 0 0 0 0 0 0 0 0 1	17
− 40 to −24	−31	1 0 0 0 0 1	13	1 0 0 0 0 0 0 0 1	16
− 23 to −11	−16	1 0 1	10	1 0 0 0 0 0 1	14
− 10 to − 1	− 5	1 1	9	1 0 0 0 1	12
0 to + 9	+ 4	0 1	1	0 0 0 1	3
10 to 22	+15	0 0 1	2	0 0 0 0 0 1	5
23 to 39	+30	0 0 0 0 1	4	0 0 0 0 0 0 0 1	7
40 to 255	+49	0 0 0 0 0 0 1	6	0 0 0 0 0 0 0 0 1	8

With regard to prediction, if element A is a non-transmitted element in a moving area, it is replaced by A_s (see Figure 1/H.120); if element D is part of a subsampled moving area, and not transmitted in the current frame, it is replaced by C.

1.4.1.4.2 *Field/field subsampling*

Either field can be omitted. In the omitted field, interpolation takes place only in those parts of the picture which are estimated to be moving. "Stationary" areas remain unchanged.

The estimated moving areas are formed from an OR function on the moving areas in the past and future fields, as shown in Figure 2/H.120. In the figure, x is a moving element if a OR b OR c OR d are moving.

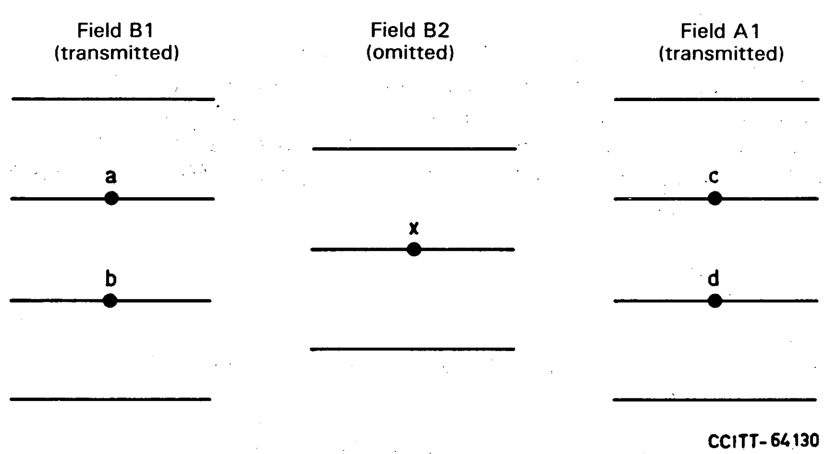


FIGURE 2/H.120

For the purpose of field interpolation, PCM lines are considered as non-moving and field blanking is assumed to be at a level of 128 out of 256.

In the interpolator for monochrome or luminance signals, the operations $\frac{a + b}{2}$ and $\frac{c + d}{2}$ are carried out before the combined average is taken. Thus

$$x = \frac{\left[\frac{a + b}{2}\right] + \left[\frac{c + d}{2}\right]}{2}$$

The interpolated values are placed in the frame store.

1.4.2 *Colour-difference components*

1.4.2.1 *Analogue-to-digital conversion*

The signal is sampled to produce 52 samples per active line (64 samples per complete line). The sampling pattern is orthogonal and line-, field- and picture-repetitive. For the 625-line input, the sampling frequency is 1.0 MHz, locked to the video waveform.

The $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ samples are sited so that the centre of the first colour-difference sample on any line is co-sited with the centre of the third luminance sample (addressed as number 2). The $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ signals are stored and transmitted on alternate lines of the coded picture. The first active line of Field No. 1 contains $(E'_B - E'_Y)$ and the first active line of Field No. 2 contains $(E'_R - E'_Y)$. The colour difference signal not being transmitted during any line is obtained at the decoder by interpolation.

The vertical filtering (see § 1.4.2.2) is arranged so that the effective vertical positions of the colour-difference samples in each of the 286 active lines coincide with those of the corresponding luminance samples.

Uniformly quantized PCM with 8 bits/sample is used.

The $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ signals are quantized using ± 111 steps with zero signal corresponding to level 128. The analogue video signals are amplitude-limited so that the digitized signals do not go outside that range (corresponding to levels 16 to 239). The video levels are set so that a 100/0/75/0 colour bar signal (see CCIR Recommendation 471 for explanation of nomenclature) will occupy levels 17 to 239.

As for the luminance signal, forbidden PCM code words are available for purposes other than transmitting video sample amplitudes.

1.4.2.2 *Pre- and post-filtering*

In addition to conventional anti-aliasing filtering prior to analogue-digital conversion, a digital transversal filtering operation is carried out on the 625-line signal to reduce the vertical definition of the picture prior to conditional-replenishment coding. As a result of this process, 72 active lines of $(E'_R - E'_Y)$ and 71 active lines of $(E'_B - E'_Y)$ are used in Field No. 2 instead of 287½ active lines per field of a 625-line signal. Similarly, Field No. 1 contains 72 active lines of $(E'_B - E'_Y)$ and 71 lines of $(E'_R - E'_Y)$. An interpolation process in the decoder restores the 625-line signal waveforms.

1.4.2.3 *Conditional replenishment coding*

Coloured moving areas are detected, coded and addressed separately from the luminance moving areas, but the same principles are employed.

Detected moving areas are transmitted by differential PCM with a maximum of 16 quantization levels. The first picture element in each moving area is transmitted by PCM. Variable-length coding is used on the DPCM code words.

Complete PCM lines are transmitted to provide systematic and forced updating coincident with luminance PCM lines.

1.4.2.3.1 *DPCM prediction algorithm*

The algorithm used for colour-difference signals is:

$$x = A \text{ (see Figure 1/H.120)}$$

1.4.2.3.2 *Quantization law and variable-length coding*

As for luminance component (see §§ 1.4.1.3.2 and 1.4.1.4.1).

1.4.2.4 *Subsampling*

Horizontal subsampling is carried out in exactly the same way as for the luminance signal, including adaptive element subsampling.

Field/field subsampling of the colour-difference signals is also similar to that of the luminance signal. Either field can be omitted and, in the omitted field, interpolation takes place only in those parts of the picture which are estimated to be moving. Stationary areas remain unchanged.

The estimated moving areas are formed by an OR function on moving areas in past and future fields in the same manner as for luminance (§ 1.4.1.4.2).

For colour-difference signals, the interpolated value of x is $\left(\frac{a+c}{2}\right)$ or $\left(\frac{b+d}{2}\right)$ when x is in Field 1 or Field 2, respectively.

Both field and horizontal subsampling take place simultaneously with subsampling of the luminance signal and they are signalled to the decoder in the same way.

1.5 *Video multiplex coding*

1.5.1 *Buffer store*

The size of the buffer store is defined at the transmitting end only and is 96 kbit/s. Its delay is approximately equal to the duration of one picture (40 ms).

At the receiving end, the buffer must be of at least this length, but in some implementations of the decoder it may be longer.

1.5.2 *Video synchronization*

The method used for video synchronization permits the retention of the picture structure. The required information is transmitted in the form of line start and field start codes (LST and FST).

1.5.2.1 *Line start code*

The line start code includes a synchronization word, a line number code and a digit to signal the presence of element subsampling.

It has the format:

0 0 0 0 0 0 0 0 | 0 0 0 0 1 0 0 0 | "S" | 3-bit line No. code |

"S" is a 1 if horizontal subsampling occurs on the TV line following the line start code. "S" is a "don't care" condition on empty or PCM lines.

The line number code comprises the least three significant digits of the line number, where Line 0 = first active line of Field 1 and Line 144 = first active line of Field 2.

Lines numbered 143 and 287 are non-coded lines, used for field synchronization and line number continuity.

1.5.2.2 *Field start code*

There are two field start codes, FST-1 and FST-2, where the first line of the field following FST-2 is interlaced between the first two lines of the field following FST-1. FST-1 indicates the start of the first field, starting with line number 0. FST-2 indicates the start of the second field, starting with line number 144, as shown in Figure 3/H.120.

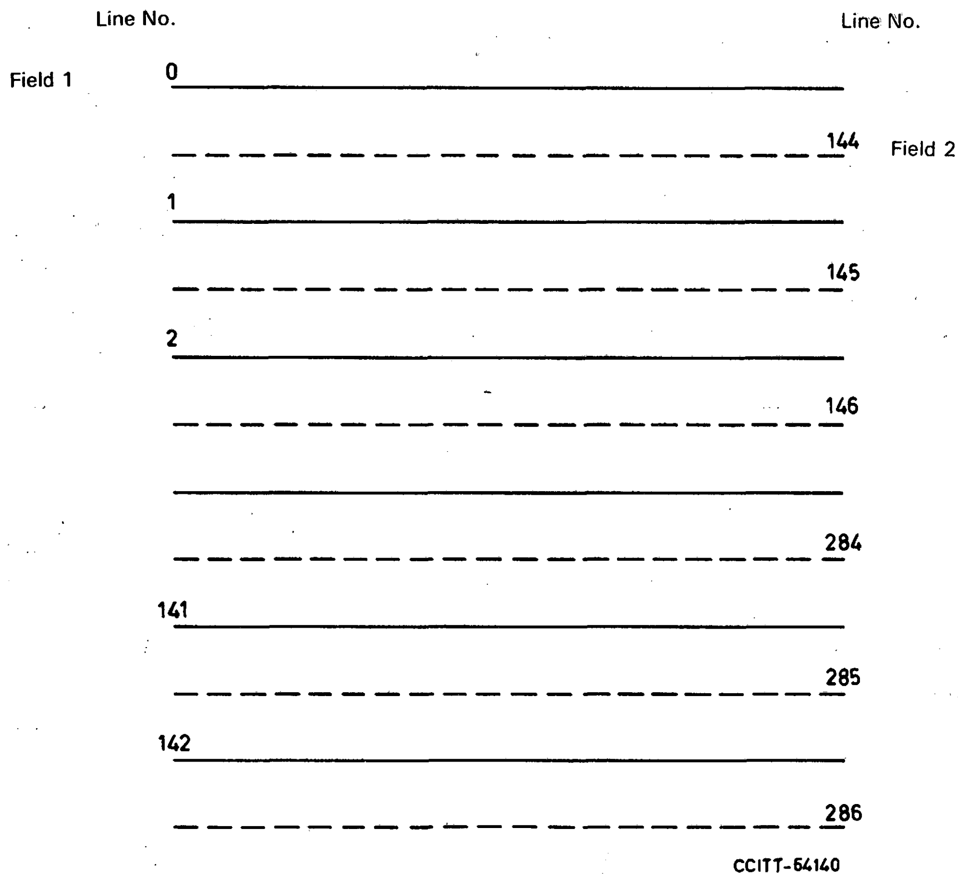


FIGURE 3/H.120

Each field start code comprises a line start code, followed by an 8-bit word, followed by the line start code of the first line of the next field.

The field start code is given in Figure 4/H.120.

LST					LST			
00000000	00001AAA	F	111	0000F11F	00000000	00001000	S	000

FIGURE 4/H.120

For FST-1, F = 1 and for FST-2, F = 0. A = 0 for normal operation. If required, A = 1 is used to signal that the buffer state is less than 6 kbits (used in switched multipoint applications). S is the subsampling digit as defined in § 1.5.2.1.

Field subsampling is signalled by two consecutive field start codes of the same number. For example:

FST – 1	field of data	FST – 1	field of data

signifies that field 2 has been omitted and that its moving areas must be interpolated as described in §§ 1.4.1.4.2 and 1.4.2.4.

1.5.3 *Addressing of moving areas*

The positions of the clusters of picture elements along each line which are deemed to form parts of moving areas are addressed by means of an address of the start of the cluster and an "end-of-cluster" code (EOC).

The form of coding is:

LST	PCM value	8-bit address of the PCM picture element	Variable-length DPCM coded moving area	EOC	PCM value	8-bit address	etc.

The PCM value is the amplitude of the first picture element of the cluster. When there is no colour-difference data, the EOC is omitted in the last luminance cluster of every line, i.e. both the LST and FST codes also signify end of cluster.

The EOC is 1001.

The address indicates the sample number along the line belonging to the first picture element of the cluster.

A cluster cannot start at the last element of the line, i.e. (11111111) is a forbidden cluster address, nor can it extend into line blanking even during subsampling.

The minimum gap between the end of one cluster and the start of the next is four picture elements, and the minimum length of a cluster is one picture element.

1.5.4 *Addressing of colour-difference data*

To permit the insertion of colour-difference data in a line containing moving picture elements, a colour escape code is inserted after the final luminance cluster in the line. This permits addresses to be re-used for colour clusters.

The escape code is 00001001 (an invalid PCM value) and follows the end of cluster code of the last luminance cluster (if any), otherwise it follows the line start code. This is followed by the addresses, variable-length codes and EOC codes of the subsequent coloured clusters, the sequence being terminated by the line start code of the next line.

The form of addressing of colour-difference moving areas is shown in Figure 5/H.120.

. . .	VLC luminance picture element	EOC	00001001	PCM value of first colour-difference picture element	Address of first colour-difference cluster	VLC codes	. . .

. . .	EOC	PCM	Address	. . . .	Line start code	. . . .

FIGURE 5/H.120

There are 52 colour-difference picture elements per line, the first of which is given an address with numerical value 4. The address range is therefore:

00000100 to 00110111.

A cluster cannot start at address (00110111) nor can it extend beyond this point, even during subsampling. The minimum gap between the end of one colour-difference cluster and the start of the next is 4 picture elements. The minimum cluster length is one picture element. Cluster bridging is not allowed between luminance and colour-difference clusters.

A monochrome decoder will discard the information between the colour escape code and the next line start code.

1.5.5 *PCM lines*

PCM lines are used for systematic or forced updating and are signalled as shown in Figure 6/H.120.

	Invalid PCM code	Invalid cluster address	PCM value of first picture element of line	254 × 8 bit PCM values	
LST	1 1 1 1 1 1 1 1	1 1 1 1 1 1 1 1	X X X X X X X X	X X . . .	1 0 0 0 0 0 0 0

FIGURE 6/H.120

With monochrome, all 256 elements in the line are transmitted using 8-bit PCM.

With PCM lines, the subsampling digit "S" is ignored at the receiver. PCM lines cannot be horizontally subsampled.

For the purpose of field interpolation, PCM lines are considered to be non-moving.

With colour signals, the colour-difference data will comprise 52 × 8-bit PCM values following the 256 × 8-bit luminance elements. The colour escape code is not transmitted. A monochrome decoder will discard the colour-difference picture elements.

1.6 *Transmission coding*

The transmission coder assembles the video, audio, signalling and optional data channels into a 2048-kbit/s frame structure which is in accordance with Recommendation G.704. It also provides justification facilities to enable the video sampling frequency to be independent of the network clock.

1.6.1 *Serial data*

With all serialized data (video, audio and addressing), the most significant digit is in the leading position. Positive logic is used throughout.

1.6.2 *Audio*

The audio is coded into 64 kbit/s using A-law PCM, as specified in Recommendation G.711.

In the coder, the delay difference between the coded audio and video when the buffer is empty should be within ± 5 ms. In the decoder, the delays must also be equalized, and the tolerance is under study.

The audio output should be muted in the event of loss of frame alignment.

1.6.3 *Transmission framing*

1.6.3.1 *General*

The frame structure is defined in Recommendation H.130 which specifies how the frame is structured and for what purposes the time slots are used. This information need not be repeated here.

Time slot 2 (odd) is allocated to codec-to-codec signalling and the functions of the various bits are specified in Recommendation H.130. In most cases, the action to be taken by the encoder and/or decoder according to whether each of these bits is 0 or 1 is evident from the specified purpose of the bit. In the few cases where this is not so, additional information is given below.

1.6.3.2 *Use of certain bits in each octet in the odd frames of time slot 2*

Studies to determine the most suitable methods for multipoint conferencing are still in progress, but from the preliminary results, a number of special features and facilities have been identified as being necessary and have therefore been included in the codec and frame structure. In "continutous presence" multipoint conferencing, a transmission channel may, at times, be shared by two codecs in different locations. This requires a reduction of the bit rate of each source so that the total bit rate is within the capacity of the channel. The "facilities bits", i.e. bits (see Recommendation H.130) 3.1.2 and 3.1.7, are used to signal the availability of this facility and the bits 4.9 and 4.15, signal the mode of operation and the active time slots in use at the output of the transmission coder. The details of how the foregoing bits are interpreted are given in Recommendation H.130.

Bits 3.7 to 3.15 also provide facilities whose principal usefulness is likely to be multipoint conferencing. Information on the use of these bits together with details of the use of bits 1 and 2, which are essential to the basic requirements of keeping the encoder and decoder in step, are given below:

Bit 1 – For clock justification

The frequency control arrangements are as follows:

The video sampling clock is locked to the line-scanning frequency of the incoming video signal, which has a permitted tolerance of ± 2 parts in 10^4.

The justification is controlled by a comparison frequency of (22 500/11) kHz, which is locked to the video clock.

The clock for the digital channel has a frequency of 2048 kHz ± 50 parts in 10^6.

The phase of the channel clock is compared with that of the comparison frequency and when the channel-clock phase exceeds that of the comparison frequency by 2 π radians, a 1 is transmitted. If the phase difference is less than 2 π radians, a 0 is transmitted.

Bit 2 – To signal buffer state

The degree to which the encoder buffer is filled, measured in increments of 1 K (1 K = 1024 bits), is signalled using an 8-bit binary code. The most significant bit (MSB) is in frame 1 of the multiframe, the second MSB in frame 2, etc. The buffer state is sampled at the start of the multiframe in which its state is transmitted.

Bit 3.7 – Fast update request

On receipt of this bit set to 1, the transmitter buffer is forced to decrease its fill and stabilize to a state of less than 6 K by preventing coded picture elements from entering the buffer. Bit A is set to 1 in the next FST. The two following fields are treated as complete moving areas and the encoder uses an arrangement for control of the subsampling modes to make the buffer overflow condition unlikely.

Bit 3.9 – Advance warning of interruption

This bit (set to 1) is used to warn a decoder that its received signal may be interrupted after the start of the next supermultiframe for a period of not more than two seconds. On receipt of bit 3.9 set to 1, a decoder will display a still picture for a period of not more than 2 s, or until an FST code is received with bit A set to 1.

Bit 3.11 – Sound power signal

This bit is used to signal the sound power in the audio channel. The power is integrated over a period of 16 ms (period of the supermultiframe), uniformly quantized to 8 bits and transmitted at supermultiframe rate. It is used during encrypted multipoint operations. In other cases, bit 3.11 is set to zero.

Bit 3.13 – Data distribution

This bit is set permanently to 0 in all encoders. When a 1 is received from the network (introduced, for example, by a multipoint control unit), the encoder will vacate the same time slots in its outgoing signal as signalled on the incoming stream by the settings of the relevant bits 4 (which identify the use of the time slots, see Recommendation H.130). It will confirm the action by transmitting the same bit 4 settings as received. This function should be carried out within 10 supermultiframe periods.

Bit 3.15 – Detection of looped ports

This bit is set to 1 in all codecs. It may be used by a multipoint control unit to detect whether one of its bidirectional 2 Mbit/s ports has been externally looped.

1.7 *Error correction*

Provision is made for the optional use of forward error correction. This is required if the channel error rate is worse than 1 in 10^6 for significant periods of time. The error corrector used is a (4095, 4035) five-error correcting BCH[3] code. The error-correcting decoder has the ability to correct up to 5 isolated errors and one burst of up to 16 errors in each block. At a channel error probability of 1×10^{-4}, the corrected error rate is 1.25×10^{-8}. The 60 parity bits which are required are obtained by removing the video from time slots 24 to 31 of frame number 15 of each multiframe.

Note – The question of whether error correction should be provided on the signal, on the link, or both, should be studied. Also under study is the question of whether the audio should be corrected by the same error corrector or whether a separate error-correcting codec should be used.

[3] BCH = Bose, Chaudhuri and Hocquengham.

2 Codecs not requiring separate television standards conversion when used on interregional connections

A codec for 525-line, 60 fields/s and 1544 kbit/s transmission for intra-regional use and capable of interworking with the codec of § 1

2.1 *Introduction*

Section 2 indicates the changes and additions which must be made to the text of § 1 in order to define the version of the codec for use with 525-line, 60 fields/s television standards and transmission at 1544 kbit/s. The two versions are capable of interworking via a re-multiplexing unit which can convert the Recommendation G.704, § 2.1 compatible frame structure on one side to the Recommendation G.704, § 2.3 compatible frame structure (with 6 time slots empty) on the other side.

The two versions of the codec are identical in most respects, the important differences (apart from the obvious ones arising from different input and output signals) being confined to the digital pre- and post-filters and the signals for the control of the buffers. Moreover, the detailed algorithms of the pre- and post-filters do not need to be specified to permit interworking. Only an outline of their mode of operation together with the few necessary specifications are therefore provided.

2.2 *Brief specification*

2.2.1 *Video input/output*

The video input and output are standard 525-line, 60 fields/s colour or monochrome television signals. The colour signals are in component form. Colour and monochrome operation are fully compatible.

2.2.2 *Digital output/input*

The digital output and input are at 1544 kbit/s, compatible with the frame structure of Recommendation G.704.

2.2.3 *Sampling frequency*

The video sampling frequency and 1544 kbit/s network clock are asynchronous.

2.2.4 *Coding techniques*

Conditional replenishment coding supplemented by adaptive digital filtering, differential PCM and variable-length coding are used to achieve low bit-rate transmission.

2.2.5 *Audio channel*

An audio channel using 64 kbit/s is included. At present, coding is A-law according to Recommendation G.711, but provision is made for future use of more efficient coding.

2.2.6 *Mode of operation*

The normal mode of operation is full duplex.

2.2.7 *Codec-to-network signalling*

An optional channel for codec-to-network signalling is included.

2.2.8 *Data channels*

Optional 2 × 64 kbit/s and 1 × 32 kbit/s data channels are available. These are used for video if not required for data.

2.2.9 *Forward error correction*

Optional forward error correction is available. This is required only if the long-term error rate of the channel is worse than 1 in 10^6.

2.2.10 *Additional facilities*

Provision is made in the digital frame structure for the future introduction of encryption, a graphic mode and multipoint facilities.

2.2.11 When the coder buffer is empty and the decoder buffer full, the coder delay is 31 ± 5 ms and the decoder delay is 176 ± 31 ms [1)].

2.3 *Video interface*

The normal video input is a 525-line, 60 fields/s signal in accordance with CCIR Report 624. When colour is being transmitted, the input (and output) video signals are in component form. The luminance and colour-difference components, E'_Y, $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ are as defined in CCIR Report 624. The video interface is as recommended in CCIR Recommendation 567.

2.4 *Source coder*

2.4.1 *Luminance component or monochrome*

2.4.1.1 *Analogue-to-digital conversion*

The signal is sampled to produce 256 picture samples per active line (320 samples per complete line). The sampling pattern is orthogonal and line, field and picture repetitive. For the 525-line input, the sampling frequency is 5.0 MHz, locked to the video waveform.

Uniformly quantized PCM with 8 bits/sample is used.

Black level corresponds to level 16 (00010000).

White level corresponds to level 239 (11101111).

PCM code words outside this range are forbidden (the codes being used for other purposes). For the purposes of prediction and interpolation, the final picture element in each active line (i.e. picture element 255) is set to level 128 in both encoder and decoder.

In all arithmetic operations, 8-bit arithmetic is used and the bits below the binary point are truncated at each stage of division.

2.4.1.2 *Pre- and post-filtering*

2.4.1.2.1 *Spatial filtering*

A digital filter reduces the 242½ active lines-per-field of the 525-line signal to 143 lines-per-field, the same number as in the 625-line version of the codec. In the decoder, the digital post-filter uses interpolation to restore the signal to 525-lines per picture.

2.4.1.2.2 *Temporal filtering*

A recursive temporal pre-filter with non-linear transfer characteristics is used in the coder to reduce noise in the signal and increase coding efficiency. The frame store used in this filter can also be used as the storage element of a frame interpolator with variable coefficients which is used to reduce the transmitted frame rate to a value less than that of the input video signal. In 525-line to 525-line transmission, the transmitted frame frequency is locked to the video clock and is approximately 29.67 Hz (29.97 Hz times 3057/3088) instead of the nominal video rate of 29.97 Hz. In 525-line to 625-line transmission, the transmitted frame frequency is nominally 25 Hz and is locked to the channel clock.

[1)] These are typical figures. The delays depend upon the detailed implementation used.

Because the (television) frames are leaving the coder more slowly than they are entering, the coding process is suspended for one frame every Nth input frame. N is approximately 100 for 525-line to 525-line operation and approximately 6 for 525-line to 625-line operation.

In the decoder, the digital post-filter incorporates a frame store in some versions of the 625-line codec where it is used in the line interpolation process. In the 525-line version, in addition to its use for line interpolation, it is used as a temporal interpolator with variable coefficients to provide an extra output frame during those periods when the decoding is temporarily suspended.

2.5 *Video multiplex coding*

2.5.1 *Buffer store*

The size of the buffer store is defined at the transmitting end only and is 160 kbits. Of this, 96 kbits is used for smoothing the video data in the face-to-face mode and the remainder is used to accomodate the action of the frame interpolator (see § 2.5.1.1 below) and the requirements of the graphics mode.

At the receiving end, the buffer must be at least this length but in some implementations of the decoder, it may be longer.

2.5.1.1 *Buffer control*

The amount to which the transmitting buffer is filled is used to control various coding algorithms (subsampling, etc.) and is signalled to the decoder to enable it correctly to interpret the received signals. In the 525-line codec, the transmission rate is less than the video input rate and hence the buffer tends to fill more rapidly than would be determined by the movement in the picture, only to empty again when the interpolator suspends the coding process.

To avoid incorrect changes in coding algorithms, the buffer-state signal is modified to take account of the progressively changing coefficients of the interpolator in the pre-filter. The buffer then operates as though the data is coming from a video source whose frame rate is uniform and the same as the transmitted frame rate.

2.6 *Transmission coding*

The transmission coder assembles the video, audio, signalling and optional data channels into a 1544 kbit/s frame structure which is compatible with Recommendation G.704.

2.6.1 *Serial data*

See § 1.6.1.

2.6.2 *Audio*

See § 1.6.2.

2.6.3 *Transmission framing*

The frame structure, compatible with Recommendation G.704 and also compatible with that of the 625-line version in § 1, is given in § 2 of Recommendation H.130.

2.6.3.1 *General*

See § 1.6.3.1.

2.6.3.2 *Use of certain bits in each octet in the odd frames of time slot 2*

The use of certain of the bits in time slot 2 (odd) differs slightly from that given for the codec in § 1. The differences are as follows:

Bit 1 – For clock justification

This bit is disregarded in 525-line decoders.

To permit interworking with the 626-line codecs of § 1, the 525-line coders must transmit a fixed bit-pattern which is used to control the frequency of the video clock in 625-line decoders. The exact form of the repetitive pattern need not be specified but it must contain seven "ones" and four "zeros" in 11 bits, e.g.:

1 0 1 1 0 1 0 1 1 0 1

Bit 2 – To signal buffer state

The degree to which the encoder buffer is filled, after correction for the interpolator (see § 2.5.1.1), is measured in increments of 1 K (1 K = 1024 bits), and signalled using an 8-bit binary code. When working to a 525-line decoder, the buffer state is sampled every 3057 channel-clock periods. When working to a 625-line decoder, the buffer state is sampled 10 times during every 525-line field period. When the buffer input is suspended for a frame period, the buffer sampling is stopped. The sampled values of the buffer state are stored prior to transmission. The store may hold between zero and 23 values which have been modified to take account of the interpolator coefficients at the times of sampling. The modified sample values are read out [as bit 2 of TS2 (odd)] at a uniform rate; the most significant bit (MSB) in frame 1 of the multiframe, the second MSB in frame 2, etc.

Bit 3.7 – Fast update request

On receipt of this bit set to 1, the transmitter buffer is forced to decrease its full and stabilise to a modified state of less than 6 K by preventing coded picture elements from entering the buffer. Bit A is set to 1 in the next FST. The two following fields are treated as complete moving areas and the encoder uses an arrangement for control of the sub-sampling modes to make the buffer overflow condition unlikely.

3 A codec for 525-lines, 60 fields/s and 1544 kbit/s transmission for intra-regional use

3.1 *Introduction*

A 1.5 Mbit/s interframe codec described under § 3, is capable of transmitting and receiving a single NTSC video signal and audio signal using an adaptive predictive coding technique with motion-compensated prediction, background prediction and intraframe prediction.

The aim of this codec is to effectively transmit video telephone and video conferencing signals which have relatively small movements. The video interface of the codec is a 525-line, 60 fields/s standard analogue television signal corresponding to the "Class *a*" standard of Recommendation H.100.

3.2 *Outline of codec*

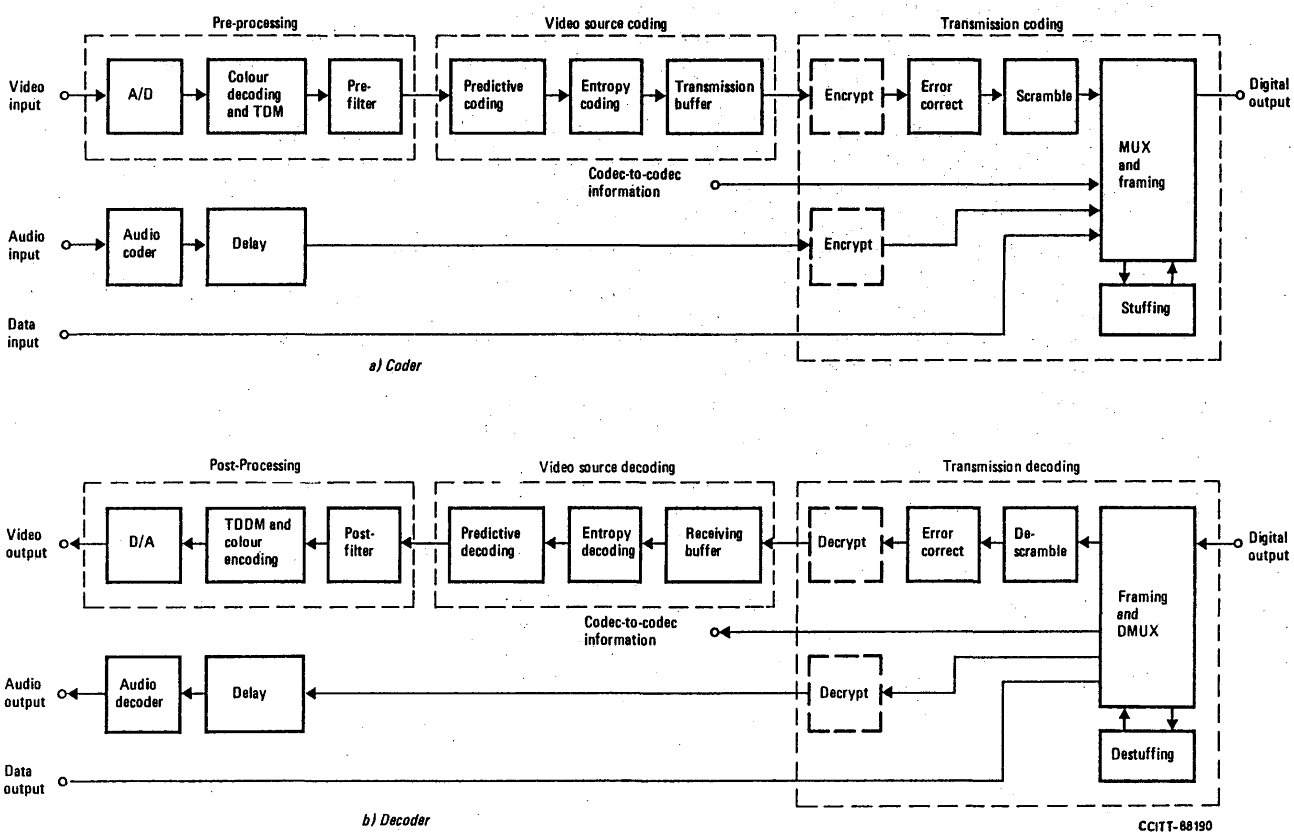
The essential parts of the codec block diagram are shown in Figure 7/H.120. The coder consists of three basic functional blocks, that is, pre-processing, video source coding and transmission coding.

In the pre-processor, the input analogue NTSC video signal is digitized and colour decoded into one luminance component and two chrominance components. These three components are time division multiplexed into a digital video form, whose noise and unnecessary signal components are removed by the pre-filter.

In the video source coder, the digital video signal is fed to the predictive coder where interframe and intraframe predictive coding techniques are fully utilized for minimizing prediction errors to be transmitted. The prediction error signal is next entropy-coded using its statistical properties to reduce redundancies. Since the coded error information is generated in irregularly spaced bursts, a buffer is used. If the buffer becomes full, the number of prediction error quantizing levels and/or picture elements to be coded is reduced to prevent any overflow.

In the transmission coder, coded video and audio signals are first encrypted on an optional basis. The coded video signal is then forward error correction coded and scrambled. The three signals, coded video, coded audio and optional data signals are multiplexed into a 1544 kbit/s digital format with a frame structure as defined in Recommendation H.130.

The decoder carries out a reverse operation.



TDM Time division multiplexing
TDDM Time division demultiplexing
MUX Multiplexing
DMUX Demultiplexing

FIGURE 7/H.120

Codec blockdiagram

3.3 *Brief specification*

3.3.1 *Video input/output*

NTSC signals are used for the video input/output signal, with monochrome signals being additionally applicable.

3.3.2 *Digital output/input*

The interface conditions for the digital output/input signal satisfy Recommendation G.703 specifications. The signal transmission rate is 1544 kbit/s.

3.3.3 *Sampling frequency*

The video sampling frequency is four times the colour sub-carrier frequency (f_{SC}) and asynchronous with the 1544 kHz network clock.

3.3.4 *Time division multiplexed (TDM) digital video format*

An NTSC signal is separated into a luminance component (Y) and two chrominance components (C_1 and C_2). A time division multiplexed signal composed of Y and time-compressed C_1 and C_2 is employed in the source coding as the standard digital video format.

3.3.5 *Coding algorithm*

Adaptive predictive coding supplemented by variable word-length coding is used to achieve low bit rate transmission. The following three predictions are carried out adaptively on a pel-by-pel basis:

a) motion-compensated interframe prediction for a still or slowly moving area,

b) background prediction for an uncovered background area, and

c) intraframe prediction for a rapidly moving area.

Prediction errors for video signals and motion vectors are both entropy-coded using the following two techniques:

i) variable word-length coding for non-zero errors, and

ii) run-length coding for zero errors.

3.3.6 *Audio channel*

An audio channel using 64 kbit/s is included. The audio coding algorithm complies with Recommendation G.722.

3.3.7 *Data channel*

An optional 64 kbit/s data channel is available, which is used for video if not required for data.

3.3.8 *Mode of operation*

The normal mode of operation is full duplex, with other modes, e.g. the one-way broadcasting operation mode, also taken into account.

3.3.9 *Transmission error protection*

A BCH error correcting code is used along with a demand refreshing method to prevent uncorrected errors from degrading the picture quality.

3.3.10 *Additional facilities*

Provision is made in the digital frame structure for the future introduction of such facilities as encryption, graphics transmission and multipoint communication.

3.3.11 *Processing delay*

The coder plus decoder delay is about 165 ms without that of a pre-filter and a post-filter.

3.4 *Video interface*

The video input/output signal of the codec is an analogue NTSC signal (System M) in accordance with CCIR Report 624.

3.5 *Pre- and post-processing*

3.5.1 *Analogue-to-digital and digital-to-analogue conversion*

An NTSC signal band-limited to 4.5 MHz is sampled at a rate of 14.3 MHz, four times the colour sub-carrier frequency (f_{SC}), and converted to an 8-bit linear PCM signal. The sampling clock is locked to the horizontal synchronization of the NTSC signal. Since the sampling frequency is asynchronous with the network clock, the justification information is coded and transmitted from the coder to the decoder.

The digital video data is expressed in two's complement form. The input level to the A/D converter is defined as follows:

– sinc tip level (−40 IRE) corresponds to −124 (10000100);

– white level (100 IRE) corresponds to 72 (01001000).

(IRE: Institute of Radio Engineers)

As a national option, a pad can be inserted before the A/D converter if a level fluctuation should be taken into account at analogue transmission lines connecting terminal equipment and codec.

At the decoder, the NTSC signal is reproduced by converting the 8-bit PCM signal to an analogue signal.

3.5.2 *Colour decoding and encoding*

The digitized NTSC signal is separated into the luminance component (Y) and the carrier band chrominance component (C) by digital filtering. The two baseband chrominance signals (C_1 and C_2) are obtained by digitally demodulating the separated carrier band chrominance component. The effective sampling frequency after colour decoding is converted to 7.2 MHz ($2\,f_{SC}$) and 1.2 MHz ($1/3\,f_{SC}$) for the luminance signal and chrominance signals respectively.

The replica of the NTSC signal is obtained by digitally modulating the C_1 and C_2 signals and adding to the Y signal at the decoder.

Filter characteristics for colour decoding and encoding are left to each hardware implementation since they do not affect interworking between different design codecs. Examples of recommended characteristics are described in Annex E.

3.5.3 *TDM signal*

A time division multiplexing (TDM) signal is constructed from the separated component signals.

First, the C_1 and C_2 signals are time-compressed to 1/6. Next, each of the time compressed C_1 and C_2 signals, with their horizontal blanking parts removed, is inserted into the Y signal horizontal blanking interval on alternate lines. C_1 is inserted on the first line of the first field and on every other line following throughout the frame, while C_2 is inserted on the second line of the first field and on every other line following throughtout the frame.

Active samples for the Y signal are 384 samples/line and 64 samples/line for the C_1 and C_2 signals. The TDM signal is constructed with these active samples and 7 colour burst samples (B), which are inserted into the top of the TDM signal.

As shown in Figure 8/H.120, the C_1 and C_2 signal sampling points coincide with that of the Y signal on every sixth sample. The C_1 and C_2 signals of only the odd lines are transmitted to the decoder.

At the decoder, each component signal is again demultiplexed from the TDM signal, and time-expansion processing of 6 times is carried out for the C_1 and C_2 signals.

Note – When a pad is inserted before the A/D converter as described in § 3.5.1, pre-emphasis (de-emphasis) with a compensating gain for the C_1, C_2 and colour burst signals is recommended at the source coder input (decoder output) to obtain better picture reproduction in coloured parts.

3.5.4 *Pre- and post-filtering*

In addition to conventional anti-aliasing filtering prior to analogue-to-digital conversion, the following two filtering processes should be used as pre-filtering for source coding:

a) temporal filtering to reduce random noise included in the input video signal;

b) spatial filtering to reduce aliasing distortion in subsampling.

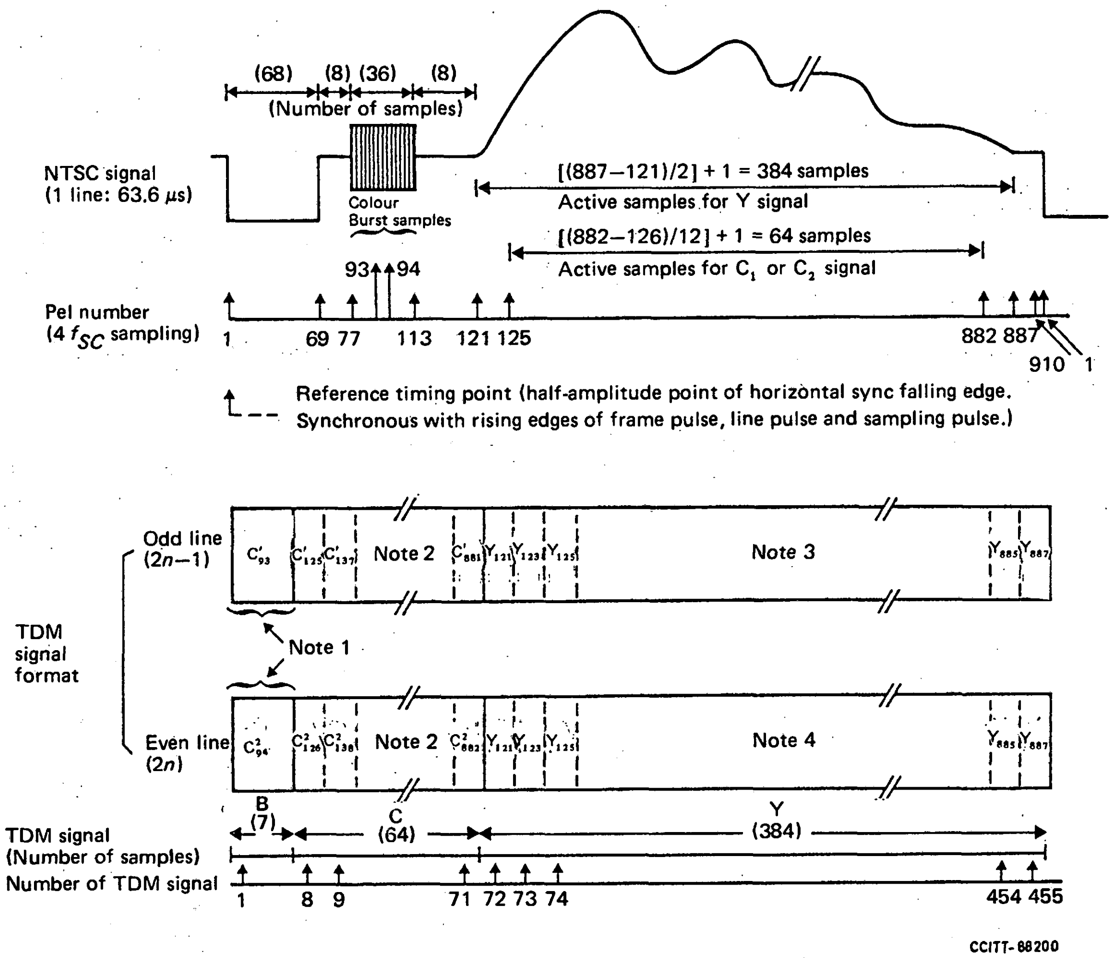
At the decoder, the following three filtering processes should be used as post-filtering in addition to conventional low pass filtering after digital-to-analogue conversion:

i) spatial filtering to interpolate the omitted picture elements in subsampling;

ii) spatio-temporal filtering to interpolate the omitted fields in field repetition;

iii) temporal filtering to reduce noise generated in the course of source coding.

Although these filtering processes are important for improving reproduced picture quality, their characteristics are independent of interworking between different design codecs. Hence, pre- and post-filtering is left to each hardware implementation.



Note 1 — Odd line samples. A colour burst sample is repeated seven times.

Note 2 — Odd line samples.

Note 3 — Odd line samples.

Note 4 — Even line samples.

FIGURE 8/H.120

TDM signal format

3.6 *Source coding*

3.6.1 *Configuration of source coder and decoder*

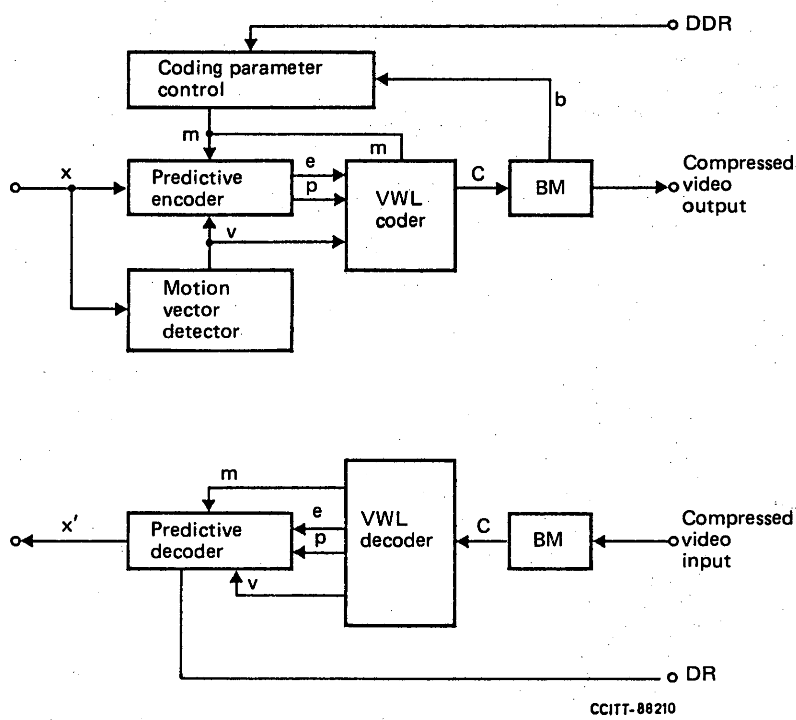
The video source coder and decoder configuration of this codec is outlined in Figure 9/H.120.

The predictive encoder converts the input video signal x into the prediction error signal e, using the motion vector v. This conversion is controlled by the coding mode m.

The variable word-length (VWL) coder codes e and v into the compressed data C using the variable length coding method. The transmission buffer memory (BM) smoothes out the irregularly spaced data C. The coding mode m is also coded.

The frame memory parity information p is used to check the identity of coder and decoder frame memory contents. If any parity error is detected, frame memories of both coder and decoder are reset by the demand refresh information (DR) and the demand refresh confirmation information (DDR).

At the decoder, the variable word-length (VWL) decoder decodes e, v, m and p, and the predictive decoder reproduces the video signal x'.



x Input video signal
x' Output video signal
m Coding mode
e Prediction error
v Motion vector
p Parity check information
C Compressed data
b Buffer memory occupancy information
VWL Variable word length
BM Buffer memory
DR Demand refresh request information
DDR Demand refresh confirmation information

FIGURE 9/H.120

Source coder and decoder configuration

3.6.2 *Predictive coding*

3.6.2.1 *Coding modes*

Five coding modes as summarized in Table 3/H.120 are provided. All of the samples are coded and transmitted in normal mode, while half of the samples are omitted in subsampling mode. In field repetition mode, one or more consecutive fields are omitted (called multi-field repetition, see Note 1). If field repetition mode and subsampling mode are used in combination, only a quarter or less of the original picture elements are coded and transmitted.

Subsampling is carried out in a quincunx way, namely by transmitting only odd-numbered pels on odd-numbered lines and even-numbered pels on even-numbered lines in each block-line (see Note 2).

In field repetition mode, either the odd or even fields are omitted. For the omitted fields, both the prediction error e and the motion vector v are set to 0.

Note 1 – If odd fields and even fields are mixed after field omission, a severe picture degradation takes place. Hence, 1 out of 2, 3 out of 4 or 5 out of 6 field omission is recommended.

Note 2 – Each block-line consists of 8 lines as defined in § 3.6.2.5.

TABLE 3/H.120

Coding modes

Coding modes		Abbreviation	Operation
1	Normal	NRM	Full sampling
2	Field repetition	FRP	One or more fields omission
3	Subsampling	SBS	2: 1 per omission
4	Stop	STP	Suspension of coding
5	Refresh	RFS	Renewal of frame memory

3.6.2.2 *Adaptive prediction*

Prediction functions are adaptively selected on a pel-by-pel basis as shown in Figure 10/H.120. The selection is carried out so as to minimize probable prediction errors. This is accomplished using the two prediction status signals, which are determined by prediction reference signals, for the preceding pels located on the previous and the present lines.

When subsampling and/or field repetition are operated, omitted pels are interpolated in the prediction loop.

The notations defined for the i-numbered pel are as follows:

X_i : local decoder output,

Y_i : interpolator output,

M_i : motion compensated interframe prediction value,

B_i : background prediction value,

I_i : intraframe prediction value,

* : logical product, and

\+ : logical sum.

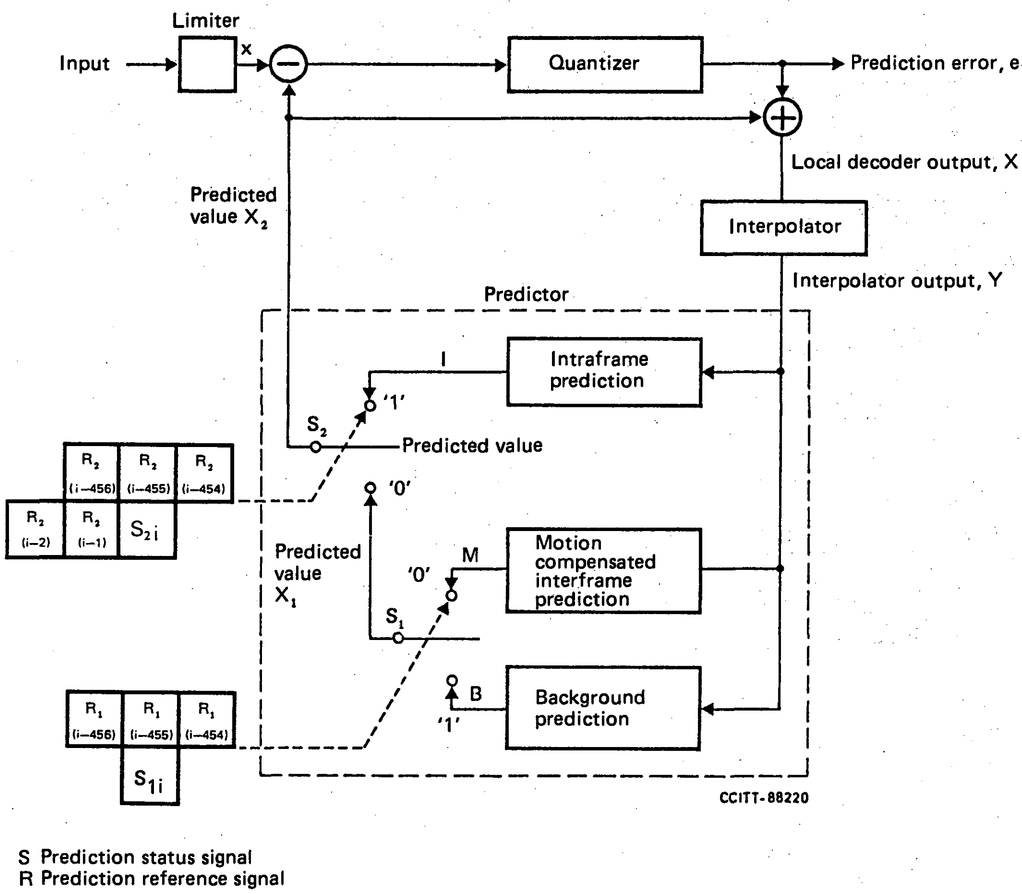


FIGURE 10/H.120

Adaptive prediction

3.6.2.2.1 *Motion-compensated interframe prediction/background prediction*

Prediction status signal S_{1i} for pel i is determined as

$$S_{1i} = R_1(i-455) * R_1(i-456) + R_1(i-456) * R_1(i-454) + R_1(i-454) * R_1(i-455) \quad (3\text{-}1)$$

where prediction reference signal $R_1(i)$ is

$$R_1(i) = \begin{cases} 0 \text{ if } |Y_i - B_i| \geqslant |Y_i - M_i|, \\ 1 \text{ if otherwise.} \end{cases} \quad (3\text{-}2)$$

Based on S_{1i}, prediction signal X_{1i} is given as

$$X_{1i} = \begin{cases} M_i, \text{ if } S_{1i} = 0, \\ B_i, \text{ if } S_{1i} = 1. \end{cases} \quad (3\text{-}3)$$

If pel i is either omitted due to subsampling and/or field repetition or forced intraframe coded or in burst B, its corresponding $R_i(i)$ is set to 0 regardless of equation (3-2).

3.6.2.2.2 *Interframe prediction/intraframe prediction*

Prediction status signal S_{2i} for pel i is determined as

$$S_{2i} = R_2 (i - 1) * R_2 (i - 455) \tag{3-4}$$

where prediction reference signal $R_2 (i)$ is

$$R_2 (i) = \begin{cases} 0 \text{ if } | Y_i - I_i | \geqslant | Y_i - X_{1i} |, \\ 1 \text{ if otherwise.} \end{cases} \tag{3-5}$$

Based on S_{2i}, prediction signal X_{2i} is given as

$$X_{2i} = \begin{cases} X_{1i} \text{ if } S_{2i} = 0, \\ I_i \text{ if } S_{2i} = 1. \end{cases} \tag{3-6}$$

If pel $(i - 1)$ is omitted due to subsampling, $R_2 (i - 2)$ is used instead of $R_2 (i - 1)$. On the other hand, if pel $(i - 455)$ is omitted, $R_2 (i - 454) * R_2 (i - 456)$ is used instead of $R_2 (i - 455)$. If pel i is forced intraframe-coded, its corresponding $R_2 (i)$ is set to 1 regardless of equation (3-5).

If pel i is omitted due to field repetition, its corresponding $R_2 (i)$ is set to 0 regardless of equation (3-5). When pel i is not forced-intraframe coded, $R_2 (i)$ in burst B is set to 0.

3.6.2.3 *Background generation*

The background prediction value is generated scene adaptively as

$$b_i = b_i^{-f} + v(k) \operatorname{sgn} (Y_i - b_i^{-f}) \, u(Y_i - Y_i^{-f}) \tag{3-7}$$

where

$$u(Y_i - Y_i^{-f}) = \begin{cases} 1 \text{ if } | Y_i - Y_i^{-f} | \leqslant L, \\ 0 \text{ if otherwise.} \end{cases} \tag{3-8}$$

$$v(k) = \begin{cases} 1, \text{ for one frame period in every block of } k \text{ frames} \\ 0, \text{ for consecutive } (k-1) \text{ frames following the frame of } v(k) = 1 \end{cases}$$

and

b_i is the background prediction value for the present frame,

b_i^{-f} is the background prediction value for the previous frame,

Y_i is the interpolator output for the present frame,

Y_i^{-f} is the interpolator output for the previous frame,

u is the still area detection function,

k is the background updata control parameter, and

L is the threshold value.

Parameters k and L are set as $K = 8$ and $L = 1$. It is noted that for hardware simplification, b_i^{-f}, instead of b_i, is used as background prediction value B_i (see Figure 11/H.120).

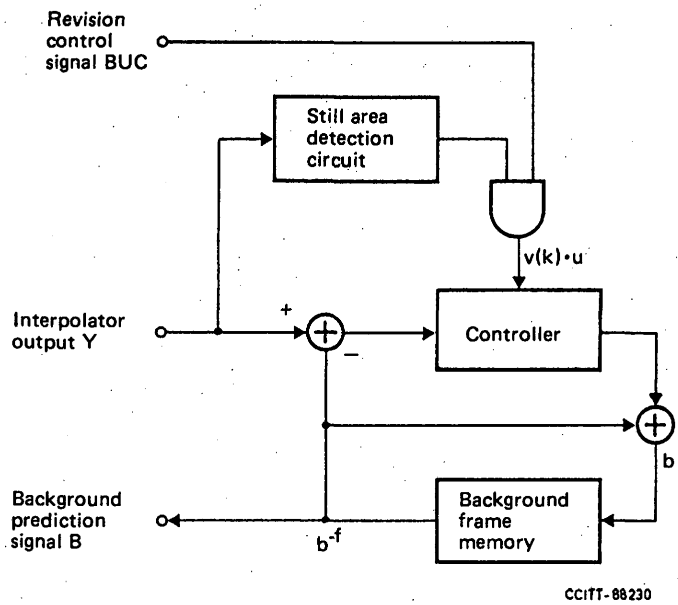


FIGURE 11/H.120

Background generation

3.6.2.4 *Forced intraframe prediction*

This codec usually used the demand refresh mode to prevent the defected picture due to transmission errors from being left in the decoder frame memory. The demand refresh mode is carried out if BWP (bit 3.15.4 in codec-to-codec information) = 0, which indicates that backward path from decoder to coder is available. However, the cyclic refresh mode is also provided, considering such applications as broadcasting communication where no backward path (from decoder to coder) is available. This mode is carried out when BWP = 1.

For either of the two refresh modes, the prediction function is forcedly set to intraframe prediction.

In the demand refresh mode, the motion frame memory and the background frame memory are updated block-line by block-line within a frame time by writing the interpolator output simultaneously. Once demand refresh starts by receiving DRR in the coder, the following DRR (demand refresh request command) received is disregarded for one second (see Note).

In the cyclic refresh mode, the two memories are simultaneously updated two lines at a time by writing the interpolator output. When a field is omitted due to field repetition, the background frame memory is updated by the signal which updates the motion frame memory. It should be noted that the command for the cyclic refresh mode is ignored in block-lines where updating based on the demand refresh mode is carried out.

Note – If a transmission error happens on the line from the codec A to the codec B, the decoder of the codec B detects the error occurrence and generates a demand refresh request information (DR). This DR is passed to the coder of the codec B and transmitted as a demand refresh request command (DRR) to the codec A. When the decoder of the codec A receives DRR, a demand refresh confirmation information DDR is passed to the coder of the codec A. Finally, demand refresh mode is operated along with transmission of a demand refresh mode command (DRM) from the codec A to the codec B.

3.6.2.5 *Definition of blanking and block-line and edge pels treatment*

3.6.2.5.1 The kinds of pels arranged on a horizontal scanning line (see Figure 8/H.120), for which prediction functions are defined, are as follows:

Burst B 7 pels,

Colour C 64 pels

Luminance Y 384 pels

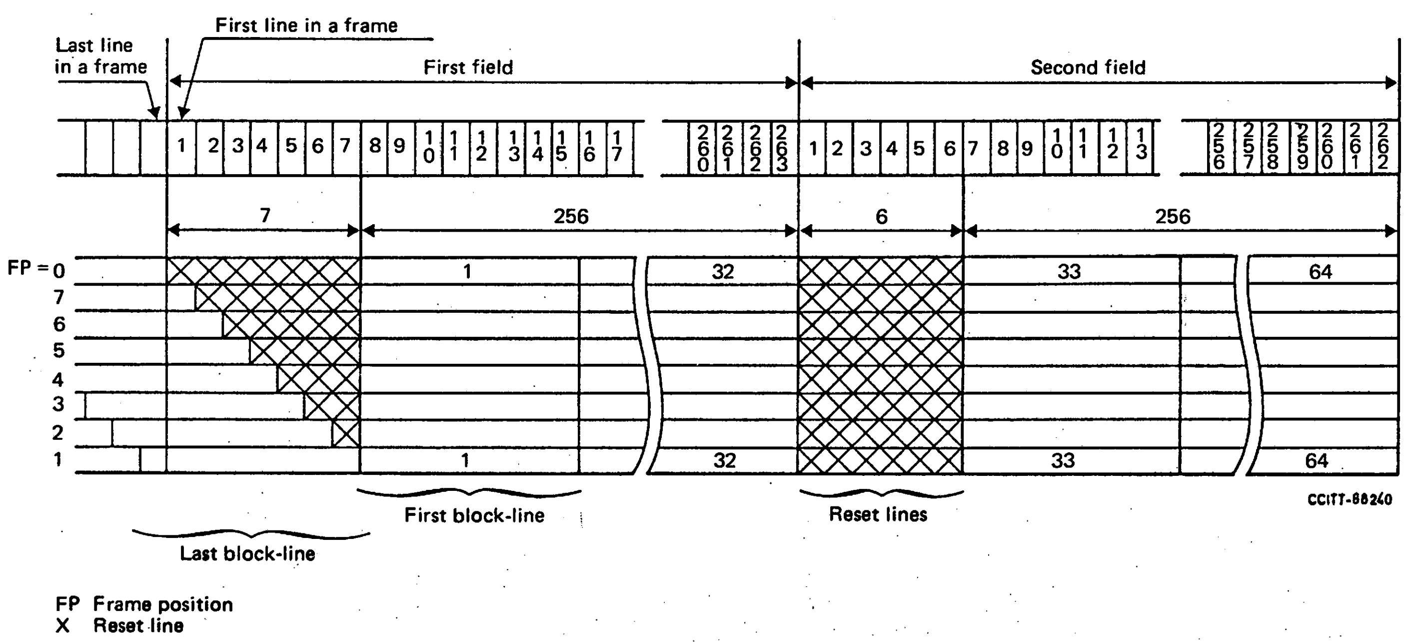
The vertical blanking periods are treated in the same way as active lines.

3.6.2.5.2 *Block-line*

See Figure 12/H.120.

In the first field, 8 lines consisting of the 8th to the 15th lines form the first block-line, with every 8 lines following this forming each a block-line. In the second field, 8 lines consisting of the 7th to the 14th lines form the 33rd block-line. Each field has 32 block-lines.

The last block-line in a frame is defined as the 8 lines which include the last line of the frame, or the line closest to the head line of the frame. The position of the video last line in the last block-line is coded as frame position.



Note – When the last line in a frame falls on the nth line of the last block-line, FP = mod(n,8).

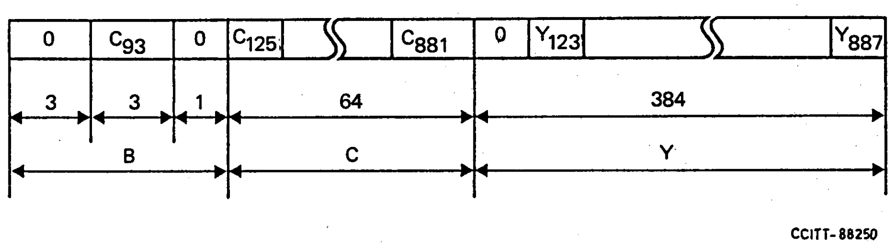
FIGURE 12/H.120

Definition of block-line and reset line

3.6.2.5.3 *Reset lines*

The lines which are excluded from block-lines are defined as reset lines. These reset lines are clampled to 0 in the predictive coding and decoding loops, or the corresponding prediction values X_2 and prediction errors e in Figure 10/H.120 are set to 0. The reset lines are prediction-coded in normal mode with adaptive prediction and by setting $v = 0$.

3.6.2.5.4 Edge pels suffer from crosstalk due to interpolation between B and C, C and Y, and Y and B. In order to prevent such crosstalk, the first 3 pels in B, the last pel in B and the first pel in Y are clampled to 0 at the source coder input IDM signal as in Figure 13/H.120.



Note – This figure shows an odd line in the first field. See Figure 8/H.120 as for pel numbers.

FIGURE 13/H.120

Zero insertion for preventing crosstalks due to interpolation

3.6.2.5.5 Edge pels are not treated specifically in the source coder and decoder. That is, video signals including reset lines and the 3 clamped pels in B are processed as if they were existing continuously (see Note). Consequently, even if a motion vector points to pels outside the active picture area, it functions as a delay control for the input time serial video signal.

Note – The right end of each line in the picture is assumed to be connected to the left end of the next line, and the lower end of each frame is assumed to be connected to the upper end of the next frame.

In the forced intraframe prediction mode, the prediction value for the first pel of each line is set to 0.

For the burst signal, no adaptive prediction nor subsampling is applied and no motion vectors are transmitted.

3.6.2.6 *Prediction and interpolation functions*

Prediction functions and interpolation functions are shown in Table 4/H.120 for all of the coding modes. It should be noted that motion vectors for the colour signal can be set to 0 without much loss of coding efficiency.

TABLE 4/H.120

Prediction and interpolation fuctions

Coding mode		Kind of pel	Prediction functions P(Z) (Note 1)			Interpolation functions I(Z) (Note 2)		
			P_Y (Z)	P_C (Z)	P_B (Z)	I_Y (Z)	I_C (Z)	I_B (Z)
Normal		Coded	Z^{-1}; $S_2 = 1$ Z^{-F+V}; $S_2 = 0$, $S_1 = 0$ $P_b(Z)$ (Note 3); $S_2 = 0$, $S_1 = 1$		Z^{-F}	1		
Subsampling		Coded	Z^{-2} (Note 4); $S_2 = 1$ Z^{-F+V}; $S_2 = 0$, $S_1 = 0$ $P_b(Z)$; $S_2 = 0$, $S_1 = 1$			1		
		Omitted	(not defined)			$\frac{1}{2}\left\{\frac{1}{2}(Z^{-1} + Z^{+1}) + \frac{1}{2}(Z^{-H} + Z^{+H})\right\}$	$\frac{1}{2}(Z^{-1} + Z^{+1})$	
Field repetition		Omitted	(not defined)			$\frac{1}{2}(Z^{-262H} + Z^{-263H})$	$Z^{-263\,H}$; first field $Z^{-262\,H}$; second field	
Refresh	NRM	Coded	Z^{-1}			1		
	SBS	Coded	Z^{-2} (Note 4)			1		
		Omitted	(not defined)			$\frac{1}{2}(Z^{-1} + Z^{+1})$		

Note 1 – S_1 and S_2 are prediction status signals defined in § 3.6.2.2.

Note 2 – To deal with fractions generated by operation of $(A+B)/2$, $(A+B+1)/2$ is executed and the 8 MSBS are used.

Note 3 – Background is generated as described in § 3.6.2.3.

Note 4 – Z^{-1}, if the previous pel is coded.

3.6.2.7 *Quantization*

Prediction errors for video signals are quantized using one of the four quantizing characteristics indicated in Table 5/H.120, that is, Q_0 (57 levels), Q_1 (57 levels), Q_2 (51 levels) and Q_3 (37 levels). The same set of quantizing characteristics are applied regardless of prediction functions.

TABLE 5/H.120

Quantizing characteristics

Q_0

Input range	Output level
0 to 1	0
2	1
3	2
4 to 5	3
6 to 7	5
8 to 9	7
10 to 11	10
12 to 14	13
15 to 17	16
18 to 20	19
21 to 23	22
24 to 26	25
27 to 29	28
30 to 32	31
33 to 37	35
38 to 42	40
43 to 48	45
49 to 54	51
55 to 60	57
61 to 67	64
68 to 74	71
75 to 81	78
82 to 88	85
89 to 95	92
96 to 102	99
103 to 109	106
110 to 116	113
117 to 123	120
124 to 255	127

Q_1

Input range	Output level
0 to 3	0
4 to 6	3
7 to 8	6
9 to 10	9
11 to 13	12
14 to 16	15
17 to 19	18
20 to 22	21
23 to 26	24
27 to 30	28
31 to 34	32
35 to 39	37
40 to 44	42
45 to 49	47
50 to 54	52
55 to 59	57
60 to 64	62
65 to 69	67
70 to 74	72
75 to 79	77
80 to 84	82
85 to 89	87
90 to 94	92
95 to 99	97
100 to 104	102
105 to 109	107
110 to 116	113
117 to 123	120
124 to 255	127

Q_2

Input range	Output level
0 to 4	0
5 to 8	5
9 to 12	10
13 to 17	15
18 to 22	20
23 to 27	25
28 to 32	30
33 to 37	35
38 to 42	40
43 to 47	45
48 to 52	50
53 to 57	55
58 to 62	60
63 to 67	65
68 to 72	70
73 to 77	75
78 to 82	80
83 to 87	85
88 to 92	90
93 to 97	95
98 to 102	100
103 to 107	105
108 to 112	110
113 to 118	115
119 to 124	121
125 to 255	127

Q_3

Input range	Output level
0 to 6	0
7 to 11	7
12 to 17	14
18 to 24	21
25 to 31	28
32 to 38	35
39 to 45	42
46 to 52	49
53 to 59	56
60 to 66	63
67 to 73	70
74 to 80	77
81 to 87	84
88 to 94	91
95 to 101	98
102 to 108	105
109 to 115	112
116 to 123	119
124 to 255	127

Note – Characteristics are symmetrical with respect to zero.

3.6.2.8 *Limiter in prediction loop*

No limiter is allocated in the prediction loop. Accordingly, the input signal x for the prediction loop is limited to $-124 \leqslant x \leqslant 123$ so that the local decoder output X is maintained in the $-128 \leqslant X \leqslant 127$ range.

3.6.2.9 *Frame memory parity check*

Parity is counted for each bit plane of the interpolator output during a video frame period from the 1st to the 64th block-line as defined in Figure 12/H.120. If block-lines are omitted in field repetition mode, parity is not counted during these omitted block-lines.

Eight odd parity bits are sent to the decoder, where they are compared with the parity bits of the decoder interpolator output to detect uncorrected errors. If any difference between received and counted parity bits is found, a demand refresh is requested from the decoder to the coder.

3.6.2.10 *Suspension of coding operation*

When the information is generated to the degree that the transmission buffer memory overflows, coding operation is suspended by setting $e = 0$ and $v = 0$. This stop mode is defined only in the coder. Interpolation and prediction functions for this mode are defined as either of NRM, SBS, FRP or RFS modes according to the control of the coding parameter controller.

3.6.3 *Motion vector transmission*

3.6.3.1 *Block size*

A block for motion compensation consists of 8 lines (vertical) by 16 pels (horizontal).

3.6.3.2 *Maximum tracking range*

Motion vectors are tracked in the range of +7 to −7 lines (vertical) and +15 pels (horizontal) at its maximum. The decoder should be able to reproduce any vector in this maximum range.

3.6.3.3 *Definition of vector direction*

The motion vector $v\,(v_x,\ v_y)$ is defined as

$$v_x = x_a - x_b$$

and (3-9)

$$v_y = y_a - y_b$$

where the block positions in the present frame and in the corresponding previous frame are $(x_a,\ y_a)$ and $(x_b,\ y_b)$ respectively. The x and y directions are identical to those of the horizontal and vertical scanning. This definition means that delay in the interframe prediction loop increases for v_x, v_y greater than 0.

3.6.3.4 *Motion detection method*

A motion vector is detected for each block by the interframe block matching method. Detailed detection methods are left to each hardware implementation (see Note).

Note – When multi-field repetition is employed, the detected vector for the previous transmitted frame can be utilized as the initial value for vector detection in the present frame to be omitted, and the detected vector for the present frame can be utilized as the initial value for vector detection in the next frame, and so on.

3.6.4 *Coding parameter control*

3.6.4.1 *Control method*

Coding control is carried out by selecting quantizing characteristics described in § 3.6.2.7 and coding modes described in § 3.6.2.1.

3.6.4.2 *Control timing*

Coding parameters are controlled according to the timing and commands as shown in Table 6/H.120.

TABLE 6/H.120

Coding parameter, control unit and commands

Coding parameter	Control unit	Commands
Normal	Frame Block-line (8 lines) Block (8 × 16 pels)	SBC = 1, IFM = 1, FRP = 1 and TRANS (SBS: off)
Quantization	Block-line	QC1 and QC2
Field repetition	Block-line (Note)	FRP = 0
Subsampling	Frame	SBC = 0 and FRP = 1
	Block	TRANS (SBS: on) and FRP = 1
Stop	arbitrary	Prediction error e = 0, motion vector v = 0
Demand refresh	Block-lines	DRM = 0 and IFM = 0
Cyclic refresh	Two lines	DRM = 1, IFM = 0 and CRM 1, 2

Note – Consecutive 32 block-lines from the first through the 32nd block-lines or from the 33rd through the 64th block-lines are omitted for ordinary field repetition. Other methods are also possible using the FRP command controlled in the unit of block-line.

3.6.4.3 *Control sequence*

The control sequence is determined based on buffer memory occupancy and other control information. Since this sequence does not affect interworking between different design codecs, it is left to each hardware implementation. However, the codec operating principle is that the coder determines all operating modes, which are transmitted with the coded video data to the decoder as a combination of commands. The decoder reproduces the video signal according to the received commands and data. A control sequence example is shown in Annex F.

3.6.5 *Entropy coding*

3.6.5.1 *Configuration of entropy coding*

The configuration of entropy coding is shown in Figure 14/H.120. The entropy coder compresses the data of the prediction error *e* and motion vector *v*, which are provided by the source coder, using variable length coding. These compressed data are multiplexed with coding mode data *m* and fed into the transmission buffer memory. The multiplexed data format is outlined in Figure 15/H.120.

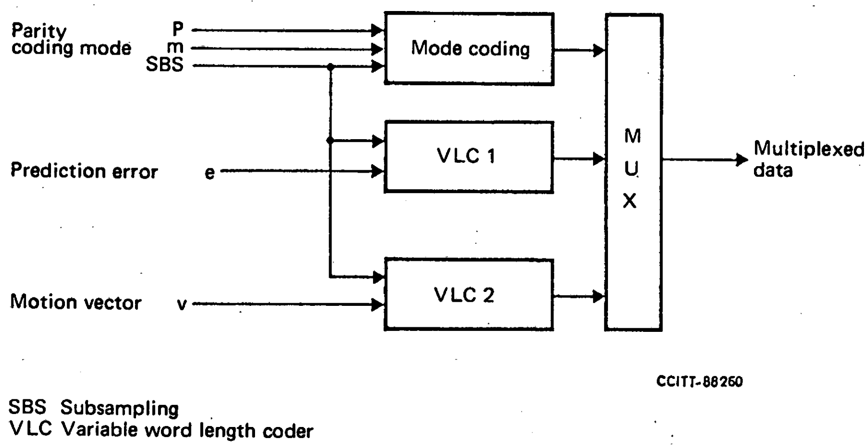


FIGURE 14/H.120

Configuration of entropy coding

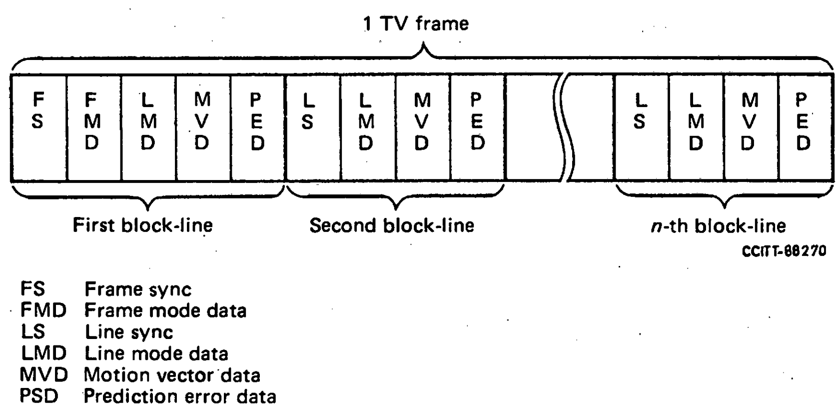


FIGURE 15/H.120

Multiplexed data format

3.6.5.2 *Commands for coding modes and data structure*

The commands for coding modes and data structure are defined as follows:

3.6.5.2.1 *Frame sync (FS)*

The frame sync is a unique word to designate the start of a video frame and its value is: 0000000000000010.

3.6.5.2.2 *Frame mode data (FMD)*

The format of the frame mode data is given in Figure 16/H.120.

Head	PT	FM1	BC	FM2	FM3	FM4
	1∅	1∅	1∅	1∅	1∅	3∅

byte (∅)

FIGURE 16/H.120

a) *Parity data (PT)*

Odd parity for each of the 8-bit planes of the interpolator output during the previous frame period (MSB first).

b) *Frame mode 1 (FM1)*

The format of the frame mode 1 is given in Figure 17/H.120.

SBC	BRC	BUC	DRM	1	FP2	FP1	FP0

FIGURE 17/H.120

i) *Subsample control (SBC)*

When SBC = 0, subsampling is carried out throughout the frame excluding burst signals, reset lines and block-lines with FRP = 0. See § 3.6.2.1.

ii) *Background revision control (BRC)*

When BRC = 0, the contents of the motion frame memory are transferred to the background frame memory during this frame period. See § 3.6.2.4.

iii) *Background update control (BUC)*

When BUC = 0, the background frame memory is updated. If BRC is operating, it has priority. See § 3.6.2.3.

iv) *Demand refresh mode (DRM)*

When DRM = 0, coding is carried out with demand refresh mode. See § 3.6.2.4.

v) *Frame position, FP2-FP0* (see Note)

This 3-bit word designates the position of the head line of the video frame or the first line in the first field (MSB first). See Figure 12/H.120.

Note – FP bits are employed for preventing degradation in the case where the input signals are asynchronously switched to another signal where those signals have different sync phase or different sync frequency. For this purpose, the horizontal sync pulse interval in the codec, namely the picture element numbers per line should be kept as 455 samples, even in the transition period. Furthermore, input signal switching which takes place during the reset line periods should be ignored.

c) *Buffer control (BC)*

The staying time of FS in the transmission buffer memory is coded into an 8 bit word (MSB first). See § 3.6.6.1.

d) *Frame mode 2 (FM2)*

The format of frame mode 2 is given in Figure 18/H.120.

1	DRR	CMS	CRM1	CRM2	SF1	MAF	1

FIGURE 18/H.120

i) *Demand refresh request (DRR)*

When DRR = 0, the decoder requests a demand refresh to the coder. See § 3.6.2.9.

ii) *Colour/monochrone state (CMS)*

Colour (bit = 1) / monochrome (bit = 0) where monochrome is optional and the default mode is colour.

iii) *CRM1, CRM2: cyclic refresh mode*

This 2-bit word designates the position of the two lines in a block-line which is cyclic refreshed. See Figure 19/H.120. See also § 3.6.2.4.

iv) *Spare frame mode (SF1)*

v) *Mode addition flag (MAF)*

When MAF = 0, FM4 is added.

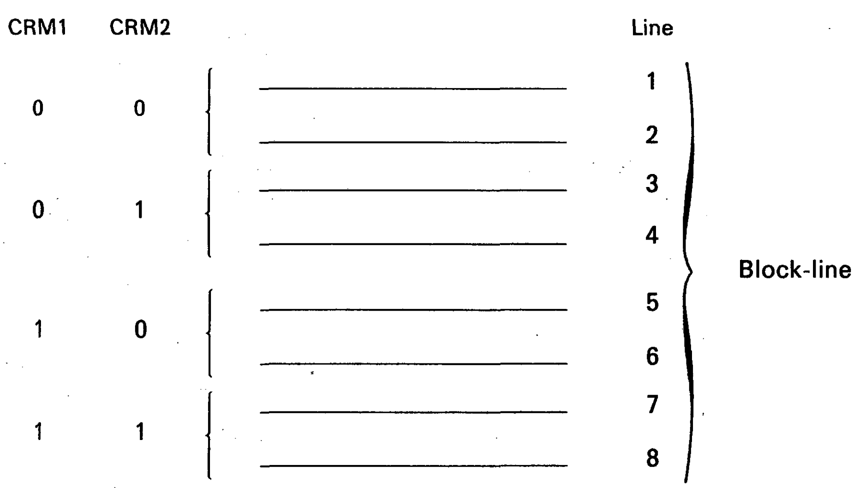


FIGURE 19/H.120

e) *Frame mode 3 (FM3)*

National option 8-bit data. If not used, a code consisting of all ones (11111111) is inserted.

f) *Frame mode 4 (FM4)*

Byte 1	SF2	SF3	SF4	1	SF5	SF6	SF7	SF8
Byte 2	SF9	SF10	SF11	1	SF12	SF13	SF14	SF15
Byte 3	SF16	SF17	SF18	1	SF19	SF20	SF21	SF22

SF2-SF22 Spare frame mode.

3.6.5.2.3 *Line sync (LS)*

The line sync is a unique word to designate the start of a block-line and its value is: 0000000000000011.

3.6.5.2.4 *Line mode data (LMD)*

The format of the line mode data is given in Figure 20/H.120.

Head	QC1	QC2	IFM	1	LSK	FRP	SL1	SL2	LDN
	1 byte								1 byte

FIGURE 20/H.120

a) *QC1, QC2: Quantizing characteristics*

QC1	QC2	Characteristics (Table 5/H.120)
0	0	Q_0
0	1	Q_1
1	0	Q_2
1	1	Q_3

b) *Forced intraframe prediction mode (IFM)*

When IFM = 0, the prediction function is fixed to intraframe prediction, throughout this block-line if DRM = 0, and at the two lines designated by CRM1 and CRM2 if DRM = 1. See § 3.6.2.4.

c) *Line skip (LSK)*

When LSK = 0, the following byte (LDN, line data number) designates the number of block-lines which are skipped. See § 3.6.5.5. LDN is coded similarly as the number of vector data, VDN. When LDN = n, consecutive $(n + 1)$ block-lines are the same. Therefore $0 \leqslant n \leqslant 63$.

d) *Field repetition (FRP)*

When FRP = 0, this block-line is omitted because of field repetition. This is valid even if IFM = 0. See § 3.6.2.1.

e) *SL1, SL2: Spare line mode*

3.6.5.2.5 *Motion vector data (MVD)*

The format of the motion vector data is given in Figure 21/H.120.

n7	n6	n5	1	n4	n3	n2	n1	VD	Note
VDN								*n* byte	

Note – Dummy, see § 3.6.5.4.6.

FIGURE 21/H.120

a) *Vector data number (VDN)*

Designates the byte number of the following vector data (VD) (in natural binary code, MSB first).

b) *Vector data (VD)*

Variable length coded motion vector data.

3.6.5.2.6 *Prediction error data (PED)* (variable word length coded)

The format of the prediction error data is given in Figure 22/H.120.

	Dummy
PED	0
(Integer) byte	

FIGURE 22/H.120

3.6.5.3 *Prediction error coding (VLC 1)*

See Figure 14/H.120.

3.6.5.3.1 *Coding method*

The quantizing level number corresponding to prediction error e is coded based on its statistical characteristics. For $e = 0$, variable word length coding is carried out using code V or F stating the quantizing level number (see Table 7/H.120). For $e = 0$, run length code R is used to state a run (RL) of ineffective pels. Note that if RL = 1, a variable word length code V_0 or F_0 to state $e = 0$ is used (see Table 8/H.120).

TABLE 7/H.120

Variable length code for non-zero amplitude prediction errors

Level number	Code length	V Code	
V_0	4	0 1 1 1	
1	2	1 S	
2	5	0 1 1 0	S
3	7	0 1 0 1	1 1 S
4	7	0 1 0 1	1 0 S
5	8	0 1 0 1	0 1 1 S
6	8	0 1 0 1	0 1 0 S
7	8	0 1 0 1	0 0 1 S
8	8	0 1 0 1	0 0 0 S
9	9	0 1 0 0	1 1 1 1 S
10	9	0 1 0 0	1 1 1 0 S
11	9	0 1 0 0	1 1 0 1 S
12	9	0 1 0 0	1 1 0 0 S
13	9	0 1 0 0	1 0 1 1 S
14	9	0 1 0 0	1 0 1 0 S
15	9	0 1 0 0	1 0 0 1 S
16	9	0 1 0 0	1 0 0 0 S
17	10	0 1 0 0	0 1 1 1 1 S
18	10	0 1 0 0	0 1 1 1 0 S
19	10	0 1 0 0	0 1 1 0 1 S
20	10	0 1 0 0	0 1 1 0 0 S
21	10	0 1 0 0	0 1 0 1 1 S
22	10	0 1 0 0	0 1 1 1 0 S
23	10	0 1 0 0	0 1 0 0 1 S
24	10	0 1 0 0	0 1 0 0 0 S
25	10	0 1 0 0	0 0 1 1 1 S
26	10	0 1 0 0	0 0 1 1 0 S
27	10	0 1 0 0	0 0 1 0 1 S
28	10	0 1 0 0	0 0 1 0 0 S

Level number	Code length	F code
F_0	4	0 0 0 1
1	6	1 1 1 1 1 S
2	6	1 1 1 1 0 S
3	6	1 1 1 0 1 S
4	6	1 1 1 0 0 S
5	6	1 1 0 1 1 S
6	6	1 1 0 1 0 S
7	6	1 1 0 0 1 S
8	6	1 1 0 0 0 S
9	6	1 0 1 1 1 S
10	6	1 0 1 1 0 S
11	6	1 0 1 0 1 S
12	6	1 0 1 0 0 S
13	6	1 0 0 1 1 S
14	6	1 0 0 1 0 S
15	6	1 0 0 0 1 S
16	6	1 0 0 0 0 S
17	6	0 1 1 1 1 S
18	6	0 1 1 1 0 S
19	6	0 1 1 0 1 S
20	6	0 1 1 0 0 S
21	6	0 1 0 1 1 S
22	6	0 1 0 1 0 S
23	6	0 1 0 0 1 S
24	6	0 1 0 0 0 S
25	6	0 0 1 1 1 S
26	6	0 0 1 1 0 S
27	6	0 0 1 0 1 S
28	6	0 0 1 0 0 S

Note – S denotes sign. S = 0 for positive, S = 1 for negative.

TABLE 8/H.120

Run length code R for zero amplitude prediction errors

RL (Note 1)	Code length	Code word R				Remark
2	5	0 0	0 0 1			
3	5	0 0	0 0 0			
4	6	0 0	1 0 1 0			
5	6	0 0	1 0 0 1			
6	6	0 0	1 0 0 0			
7	7	0 0	1 0 1 1 1			
8 to 11	7	0 0	1 1 0 X X			X = 11 − RL
12	8	0 0	1 1 1 1 0 1			
13	8	0 0	1 1 1 1 0 0			
14 to 17	8	0 0	1 1 1 0 X X			X = 17 − RL
18 to 25	9	0 0	0 1 1 1 X X	X		X = 25 − RL
26 to 33	10	0 0	0 1 1 0 0 X	X X		X = 33 − RL
34 to 37	10	0 0	0 1 0 1 0 0	X X		X = 37 − RL
38 to 64	12	0 0	0 1 0 0 1 X	X X X X		X = 69 − RL
MK1	13	0 0	1 0 1 1 0 Y	Y Y Y Y Y		Y = 0 to 63
MK2	14	0 0	1 1 1 1 1 1	Y Y Y Y Y Y		
MK3	14	0 0	1 1 1 1 1 0	Y Y Y Y Y Y		
MK4 to 7	15	0 0	0 1 1 0 1 X	X Y Y Y Y Y Y		X = 7 − MK
MK8 to 15	16	0 0	0 1 0 1 1 X	X X Y Y Y Y Y Y		X = 15 − MK
MK16 to 19	16	0 0	0 1 0 1 0 1	X X Y Y Y Y Y Y		X = 19 − MK
MK20 to 34	18	0 0	0 1 0 0 0 1	X X X X Y Y Y Y	Y Y	X = 35 − MK
MK35 to 49	19	0 0	0 1 0 0 0 0	1 X X X X Y Y Y	Y Y Y	X = 50 − MK
MK50 to 56	19	0 0	0 1 0 0 0 0	0 1 X X X Y Y Y (Note 2)	Y Y Y	X = 57 − MK

Note 1 – RL = 64 × (MK number) + 1 + Y, 0 ⩽ Y ⩽ 63.

Note 2 – The maximum run length is (455 − 3) × 8 = 3616. Corresponding MK and Y turn out to be 56 and 31, respectively. Hence, 0 ⩽ Y ⩽ 31 for MK = 56.

3.6.5.3.2 *Scanning sequence*

Entropy coding for a video frame is carried out from the first to the last block lines, excluding reset lines. The frame sync (FS) and frame mode data (FMD) are codes in the first block-line. When the last line falls on the *n*th line of the last block-line, the frame position is set to FP = md (*n*, 8). FP is transmitted to the decoder as a part of the frame mode data (see Note 1).

Since the first three pels of each line are clamped to 0 in the predictive source coder, and the reset lines are so defined as described in § 3.6.2.5.3, the pels to be entropy coded can be indicated as in Figure 23/H.120 (see Note 2).

The scanning sequence is a block scan as shown in Figure 24/H.120. The first block after scan conversion consists of 4 pels × 8 lines = 32 pels.

Note 1 – Without asynchronous switching of input video signals, the last block-line coincides with the 64th block-line and FP = 0.

Note 2 – Entropy coding is not carried out for the reset lines defined in Figure 12/H.120. The number of reset lines in the first field varies according to the FP value.

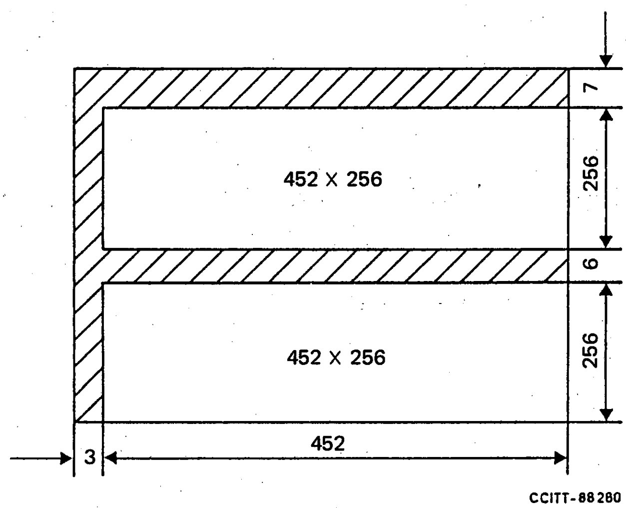


FIGURE 23/H.120

Entropy-coded pels

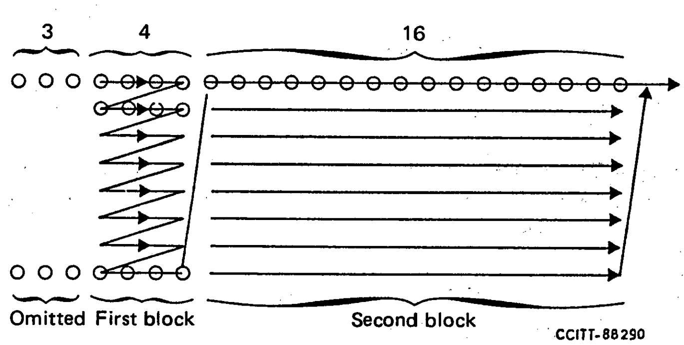


FIGURE 24/H.120

Scanning sequence

3.6.5.3.3 *Group of codes*

See Table 9/H.120.

TABLE 9/H.120

	Symbol	Number of codes	Length of codes
Amplitude code No. 1	F	57	4, 6
Amplitude code No. 2	V	57	2-10
Run length code	R	3615	5-19

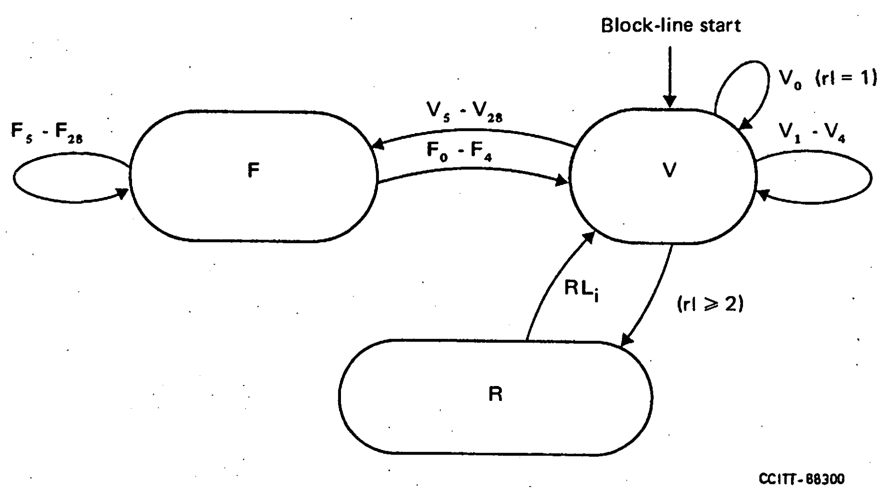
F Pseudo fixed length code for stating quantizing level number. This code is introduced to shorten the maximum code length.

V Variable length code for stating quantizing number.

R Variable length code for stating run length of ineffective pels for RL $\geqslant$ 2.

3.6.5.3.4 *Code transition rule*

The rule is shown in Figure 25/H.120 and a prediction error coding example is given in Annex F.



Note 1 – RL is a length of the run to be coded, while rl is a number of continued pels whose e = 0.

Note 2 – Prediction error data start with an R or V code. R code is used if rl $\geqslant$ 2. Otherwise V code is used.

Note 3 – The code can shift to V even if RL $\geqslant$ 2 to prevent a buffer memory underflow.

FIGURE 25/H.120

Code transition rule for prediction error data

The following points should be noted:

a) the starting code is either the V or R code;

b) the last run in a block-line may not be transmited since the LS or FS command can be utilized as a termination of the last run;

c) coding is carried out assuming that omitted pels due to subsampling do not exist;

d) some 0s are filled at the tail tail-end of the PED as dummies to make the total number of bits for the block-line data a multiple of 8.

3.6.5.3.5 *Code assignments for F and V*

See Table 7/H.120.

Code assignments are common to the four quantizing characteristics Q_0, Q_1, Q_2 and Q_3.

3.6.5.3.6 *Code assignments for R*

See Table 8/H.120.

3.6.5.4 *Motion vector coding (VLC 2)*

3.6.5.4.1 *Coding method*

A motion vector v, is first coded with predictive coding whose output Δv, is variable length coded throughout a block-line.

3.6.5.4.2 *Predictive coding*

The prediction algorithm is the previous block prediction which is

$$\Delta v = v - v_1 \qquad (3\text{-}10)$$

where v and v_1 represent the present and the previous block vectors. The opeation is carried out for each x and y component in two's complement form. The operated results are expressed with 5 bits for the x component and 4 bits for the y component neglecting carries (MSB first). Note that the decoder carries out the inverse operation $v = v_1 + \Delta v$ in two's complement from neglecting carries.

The motion vector for the first block (horizontal blanking) is set to (0,0).

3.6.5.4.3 *Variable length coding*

For $\Delta v = (0,0)$, a run length of zero is coded. For $\Delta v \neq (0,0)$, variable length coding is applied with their code lengths shown in Figure 26/H.120.

The coding of Δv is carried out for the 28 vectors of the 2nd to the 29th blocks.

The last run of $\Delta v = (0,0)$ may not be transmitted since VDN states the total bits of VD.

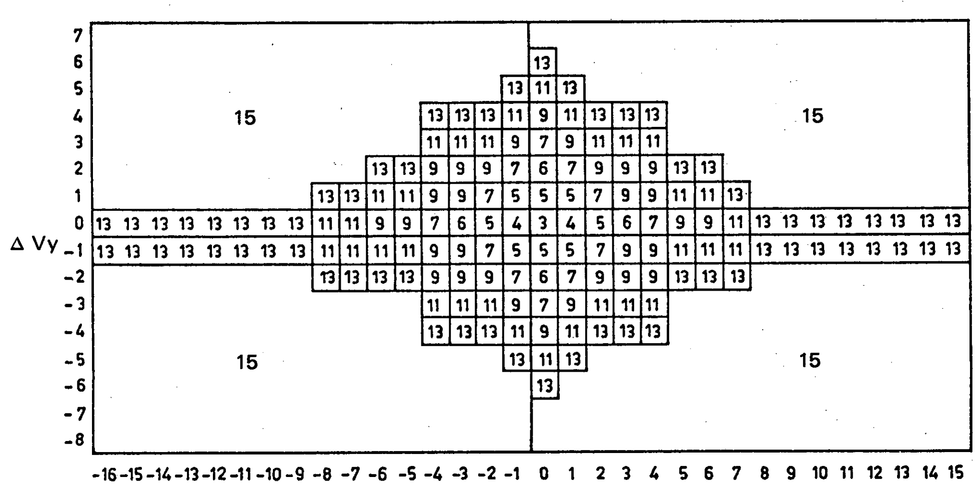


FIGURE 26/H.120

Word length for motion vector prediction errors

3.6.5.4.4 *Code assignments*

The codes are assigned as shown in Table 10/H.120, where the maximum code length is 15. The variable length codes consist of 541 codes, or 512 codes for Δv, 28 codes for run length and one TRANS code for transition of subsampling ON/OFF.

TABLE 10/H.120

Variable length code and run length code for motion vector data

ΔVx	ΔVy	Code length	Code word			Number of codes
± 1	0	4	0 0 1	Sx		2
± 1 0 ± 2	± 1 ± 1 0	5 5 5	1 1 1 1 1 1	1 SxSy 0 1 Sy 0 0 Sx		8
0 ± 3	± 2 0	6 6	1 0 1 1 1 0 1 1	1 Sy 0 Sx		4
± 1 ± 2 0 ± 4	± 2 ± 1 ± 3 0	7 7 7 7	1 0 0 1 0 0 1 0 0 1 0 0	1 1 SxSy 1 0 SxSy 0 1 1 Sy 0 1 0 Sx		12
± 3 ± 1 ± 2 ± 3 ± 4 ± 4 ± 5 ± 6 0	± 1 ± 3 ± 2 ± 2 ± 1 ± 2 0 0 ± 4	9 9 9 9 9 9 9 9 9	1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0	1 1 1 SxSy 1 1 0 SxSy 1 0 1 SxSy 1 0 0 SxSy 0 1 1 SxSy 0 1 0 SxSy 0 0 1 1 Sx 0 0 1 0 Sx 0 0 0 1 Sy		30
−8 to 7	−5 to +5 (see Figure 26/H.120)	11	1 0 0 0 0 1	X X X X Sy	[X] = ΔVx	32
−16 to 15	−6 to +6 (see Figure 26/H.120)	13	0 1 0 0 0 0 1	X X X X X Sy	[X] = ΔVx	64
−16 to 15	−8 to +7 (see Figure 26/H.120)	15	1 0 0 0 0 0	X X X X X Y Y Y Y	[X] = ΔVx [Y] = ΔVy	359

RL	Coder length	Code word		Number of codes
1	3	0 0 0		1
2	4	0 1 1 1		1
3 to 6	6	0 1 1 0 X X	X X = 6 − RL	4
7 to 12	7	0 1 0 1 X X X	X X X = 12 − RL	6
13 to 20	8	0 1 0 0 1 X X X	X X X = 20 − RL	8
21 to 28	9	0 1 0 0 0 1 X X X	X X X = 28 − RL	8
TRANS	6	0 1 0 1 1 1		1

Note 1 – Sx and Sy denote signs: $S_i = 0$ for positive, $S_i = 1$ for negative.

Note 2 – XX..X and YY..Y are expressed in two's complement form (MSB first).

3.6.5.4.5 *Transition code for subsampling (TRANS)*

The code TRANS indicates transition between ON and OFF for subsampling (SBS). For the first block in a block-line, SBS is set to OFF. Subsampling is then set to ON at the block just after the first TRANS code is inserted, and returned to OFF at the block just after the second TRANS code is inserted. The same sequence follows on. The TRANS code is expressed as a 6-bit word. When SBS = 0, transition code is disregarded in the decoder.

3.6.5.4.6 *Dummy code insertion*

When a vector data (VD) for a block-line does not have exactly 8 multiple bits, a dummy code consisting of 1 to 7 bits is inserted at the tail of the vector data.

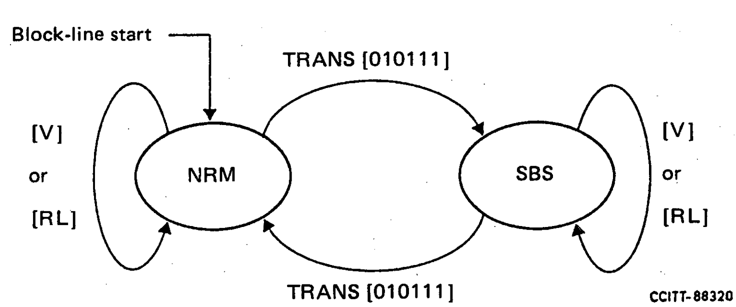
The dummy code has 1 as the head, 0s as the body and 1 as the tail (see Table 11/H.120).

TABLE 11/H.120

Number of dummy bits	Dummy code
1	1
2	11
3	101
4	1001
5	10001
6	100001
7	1000001

3.6.5.4.7 *Code transition rule*

The rule is indicated in Figure 27/H.120 with a motion vector coding example given in Annex F.



Note – The same motion vector coding method is applied for both NRM and SBS modes.

FIGURE 27/H.120

Code transition rule for motion vector data and normal/subsampling mode change

3.6.5.5 *Block-line skipping*

If block-lines continue, in which all the prediction error *e*, data and motion vector *v* data are 0, and whose line mode data (QC1, QC2, IFM, FRP, SL1, SL2) are identical, their number is run-length coded with natural binary code as skipped block-lines. A run ends when it encounters FS or a block-line with new line mode data or some $e \neq 0$ or some $v \neq 0$. A run also ends when variable length code V_0 appears due to underflow prevention.

3.6.6 *Buffer memory*

3.6.6.1 *Receiving buffer control*

The staying period of FS in the transmission buffer is counted with a 1/16 input video line frequency clock and transmitted to the decoder as BC command. The staying period is represented in eight bit binary code. Similarly, the staying period in the receiving buffer is counted and the operation of the receiving buffer is so controlled as to make the total delay time caused by the two buffer memories constant.

Note – This control method is applicable even when the read out speed for the transmission buffer varies.

3.6.6.2 *Memory size*

Transmission buffer memory size B_S is defined as 180 kbits, while receiving buffer memory size B_R should be more than 220 kbits considering the variation of the transmission buffer read out speed.

Note – The delay time due to the transmission and receiving buffer memories becomes about 165 ms for B_S = 180 kbits and B_R = 220 kbits.

3.6.6.3 *Underflow prevention*

If the occupancy of the transmission buffer decreases to a threshold, the run length coding for the prediction error is prohibited and variable length code V_0 is used.

3.6.6.4 *Overflow prevention*

If the occupancy of the transmission buffer increases to another threshold, Stop Mode for setting forcedly all the prediction error and motion vector data to 0 is applied.

3.7 *Audio coding*

An audio channel using 64 kbit/s is included. The audio coding algorithm complies with Recommendation G.722.

Since video coding and decoding introduces a significant delay as described in § 3.3.11, the encoded audio signal should be delayed by the corresponding time in the coder and decoder to obtain the proper synchronization between video and audio at the decoder. The delay inserted in the audio coder should be the sum of a half of the buffer memory delay and other video coding process delay, while the delay inserted in the audio decoder should be the sum of a half of the buffer delay and other video decoding process delay.

3.8 *Transmission coding*

3.8.1 *General*

The transmission coder assembles the video, audio, optional data and codec-to-codec information channels into a 1544 kbit/s digital stream. With all serialized data, the most significant digit leads.

3.8.2 *Encryption*

Video and audio signals can be independently encrypted on an optional basis. Their algorithms are under study. Keys and other control information can be transmitted through the message channel provided in the codec-to-codec information channel.

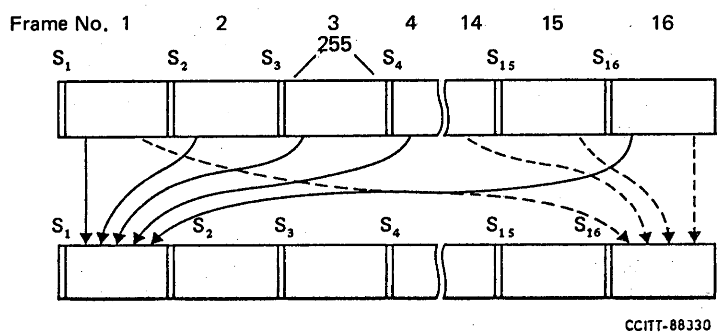
3.8.3 *Error correction*

An encoded (and encrypted) video signal is forward error corrected by a (255, 239) two-error correcting BCH code with the following polynomial generator:

$$g(x) = (1 + x^2 + x^3 + x^4 + x^8)(1 + x + x^2 + x^4 + x^5 + x^6 + x^8)$$

One framing bit is added to each 255 bit error correction frame, and 16 such frames are assembled into one large frame as shown in Figure 28/H.120. The frame alignment pattern is 0001101*y* (*y*: for future multiframe alignment signal use). The other 8 bits are used for controlling purposes, whose protocol is under study.

To correct a burst error of up to 32 bits, 16 phase interleaving is employed. The bit allocation rule is also indicated in Figure 28/H.120. Note that framing bits are excluded from interleaving.



S_1 S_3 S_5 S_7 S_9 S_{11} S_{13} = 0001101

S_{15} Multiframe alignment signal

S_2 S_4 S_6 ... S_{16} Control information

FIGURE 28/H.120

Error correction frames and interleaving

3.8.4 *Scrambling*

An error corrected video signal is scrambled with an 8-stage pseudo-random pulse generator to reduce stuffing required by the network restrictions. At each error correction frame bit, the scrambler is reset. The polynomial generator and the scrambled output pattern following the reset pulse for the input of all zeros are as follows:

$$1 + x^4 + x^5 + x^6 + x^8,$$

1111010011 . . . 1001111011.

3.8.5 *Frame structure and stuffing*

Recommendation H.130, § 3 is applied.

ANNEX A

(referring to § 1 of Recommendation H.120)

Graphics option – 625 line

A.1 *Introduction*

In order to comply with the requirements of Recommendation H.100, an optional graphics mode may be provided giving improved definition at the expense of ability to convey movement. Two suitable arrangements are as follows:

A.2 *Graphics codec for videoconferencing graphics*

A.2.1 *Facilities*

The graphics mode provides, for still pictures, a capability of full 625-line luminance definition and colour definition which is better than that of the PAL and SECAM systems. It provides a limited capability for conveying movement, sufficient to permit pointing at items under discussion on the display. When the codec is in or adjacent to, the conference room, an alternative frozen-frame mode enables the face-to-face picture frozen for about 1.5 s while the graphics picture is being transmitted, to restart while the frozen graphics picture was displayed on another monitor.

The definition in the graphics mode is sufficient to permit good reproduction of one-half an A4 page of typescript.

A.2.2 *Coding*

The luminance and colour-difference signals are sampled at 12.5 MHz and 12.5/3 MHz respectively, the sampling frequency being locked to the television line-scanning frequency.

The samples are converted to PCM with 6 bits/sample. The luminance signal has a two-level half-sampling-frequency dither signal added to it which reduces the quantization distortion to approximately that of 7-bit coding.

Only the active picture area is sampled. Thus, there are 639 luminance samples along the line and two fields of 288 lines.

One colour-difference sample occurs every three luminance samples. Two of the 6 bits from a colour-difference sample are added to each of the three associated luminance samples, giving three 8-bit words for three luminance samples plus one colour-difference sample.

The $(E'_R - E'_Y)$ component is associated with the first, third, fifth, etc., active lines of Field No. 1, with the $(E'_B - E'_Y)$ component on the intervening lines, the pattern being reversed in Field No. 2.

The colour-difference samples are delayed with respect to the luminance samples to which they are attached, so that when decoded, they are coincident with the luminance output. The centre of the first colour-difference element on a line is co-sited with the second luminance element. Similarly, the centre of the 213th colour-difference element is co-sited with the 638th luminance element.

The luminance signal is amplitude-limited so that its PCM values are confined to the range:

Black level	000000
White level (700 mV): transmission between	100111 and 111000
Maximum level (750 mV)	111011

The colour-difference signals are limited to the range: 000000 to 111111 (0 to 63) with black level at 100000 (32). The 100/0/75/0 colour bar signal (see CCIR Recommendation 471 for explanation of nomenclature) fills the range: 000100 to 111100 (4 to 60). Prior to transmission, the colour difference codes are converted to the two's complement form by inversion of the most significant bit. This gives the range: 100000 to 011111 (−32 to 31) with the black level at 000000. The colour bar signal then occupies the range: 100100 to 011100 (−28 to 28).

A.2.3 *Transmission and synchronization*

A.2.3.1 *General*

The PCM words, formed as above, are transmitted to give a continuous update of the picture store in the receiver. The update pattern, which has been chosen to give smooth transitions on a changing picture, transmits every 19th luminance sample (with associated colour-difference data). The sequence of every 19th sample continues from one line to the next, as if the 639 elements in one active line are followed immediately (that is, without a gap for line blanking) by the 639 active elements in the next line. The use of this continuous sequence of samples makes line addressing unnecessary. A field synchronization code, followed by the address (in the range of 0 to 18) of the first luminance sample of the first active line, provides all the necessary synchronizing information.

The field synchronization code comprises 8 bytes of the form 1111001 or 11111100, which are invalid PCM values. The order of the last two pairs of bits, 0011 or 1100, in each of the first seven bytes, representing 0 and 1 respectively, signals the address of the first element of the field. In the 8th byte, 1100 signals a first field (starting line 23) and 0011 a second field (starting line 336).

The sequence in which the fields are transmitted by the address of the first luminance sample of the first line need not be specified because the decoder reconstructs the picture from the received addresses. A sequence which has been found satisfactory, giving no patterns with moving objects (e.g. a pointing finger), is as follows with the number in parentheses indicating a first or second field:

1 (2), 13 (1), 6 (2), 18 (1), 3 (2), 10 (1), 15 (2), 4 (1), 0 (2), 8 (1), 12 (2), 5 (1), 14 (2), 9 (1), 17 (2), 2 (1), 11 (2), 7 (1), 16 (2),

followed by:

1 (1), 13 (2), 6 (1), ... in the same sequence as above, but with the field number interchanged.

After 38 fields, the complete piture has been replenished and the sequence repeats itself from the beginning.

A.2.3.2 *Data structure*

In each transmitted field, the data comprises 8 field synchronization bytes followed by 9685 or 9686 bytes of picture data (the total number of picture elements per field, 639 × 288, is not divisible by 19). Fields where the address of the first element is in the range 0 to 16 have 9686 transmitted bytes, while those with the first element address 17 or 18, have 9685 transmitted bytes.

Each byte of picture data comprises 6 bits of luminance data plus 2 bits of colour-difference data. The most significant bits of the colour-difference sample are transmitted first and the pairs of bits of the colour-difference data are placed in the least significant positions of the picture data byte. The data is arranged so that the luminance sample for the first picture element of a line carries with it the two most significant bits of the 19th colour-difference element belonging to the following line. The centre bits and the least significant bits of this colour-difference element are attached to the 20th and 39th along-the-line addresses respectively, these being the next two transmitted samples.

There are no colour-difference data transmitted for the first line of the picture and, on the second line, the first 18 colour-difference elements of the second line cannot be reconstructed at the decoder.

A.2.3.3 *Data output*

The graphics data are generated at a nominal rate of 3.74 Mbit/s and are transmitted via a buffer store whose capacity is in excess of 160 kbits. The output to the transmission channel is at less than 2 Mbit/s, the actual value depending on the number of time slots allocated to video. When the buffer level at the end of a field exceeds 160 kbits, the sampling is suspended for two complete fields to allow the buffer to empty.

If the level is then still in excess of 160 kbits, sampling is suspended for a further two fields.

The octet structure of the output data must be aligned with the time slot structure of the primary rate interface.

The approximate transmission time for a complete picture is therefore in the range of 1.6 to 4.6 s.

A.2.4 *Decoder*

The received data are associated with addresses derived from the field synchronization code and assembled in a picture store having 639 × 576 addressable positions with 8-bit capacity. The data are stored in the multiplexed form (luminance and colour-difference) used for transmission. The contents of this store are read out sequentially, the luminance and colour-difference components de multiplexed, and the colour-difference components line-interpolated to give the ($E'_R - E'_Y$) and ($E'_B - E'_Y$) components simultaneously and coincident with their associated luminance.

A.3 *Graphics codec for videoconferencing-graphics – Mode 2*

A.3.1 *Facilities*

The graphics mode provides, for still pictures, the capability of full 625-line luminance and colour definition. It allows transmission for still pictures with studio quality, defined in CCIR Recommendation 601. The graphics codec can be operated in two modes. In the single shot mode, the face-to-face picture is frozen for about 4 s, while the graphics picture is being transmitted, to restart while the graphics picture is displayed on another monitor. In the continuous mode, the face-to-face picture is frozen during the graphics presentation. The graphics picture is continuously transmitted in order to reproduce slow movement, e.g. for blackboard presentations. When the graphics picture has stabilized or when the presentation is finished, the graphics picture is frozen and the face-to-face picture is restarted.

The definition of the graphics mode 2 is better than that of the PAL, SECAM and NTSC systems and is sufficient to permit good reproduction of one-half of an A4 page typescript.

A.3.2 *Coding*

The luminance signal (E'_Y) and colour-difference signals ($E'_R - E'_Y$, $E'_B - E'_Y$) are sampled at 13.5 MHz and 6.75 MHz respectively, according to the encoding parameters of digital television for studios given in CCIR Recommendation 601. The sampling frequencies are related in the ratio 4 : 2 : 2. The sampling structure is orthogonal, line, field and picture repetitive. The samples of the colour-difference signals are co-sited with the first, third, fifth, etc. luminance sample in each line. All samples are uniformly quantized PCM-values with 8 bits/sample.

Only the active picture area is sampled. There are 720 luminance samples along the line and two fields of 288 lines.

The luminance signal is amplitude-limited so that its PCM values are confined to the range:

Black level: 16

Peak level: 235.

Each colour-difference signal exhibits 225 quantization levels in the centre part of the quantization scale with zero signal corresponding to level 128.

Further details are given in CCIR Recommendation 601.

A.3.3 *Transmission and synchronization*

A.3.3.1 *General*

Luminance and colour-difference samples of each line are arranged as sets for four samples:

$$\left[(E'_B - E'_Y)_n, (E'_Y)_n, (E'_R - E'_Y)_n, (E'_Y)_{n+1}\right],$$

where

$n = 0, 2, 4, 6, \ldots, 718$. Each set consists of four words with 8 bit word length. There are 360 sets in each line of the picture.

The PCM sets are transmitted to give a continuous update of the picture store in the receiver. The update pattern which has been chosen transmits every 19th set. The sequence of every 19th set continues from one line to the next, as if the 360 sets in one active line are followed immediately (that is, without a gap for line blanking) by the 360 sets of the next line. The use of this continuous sequence of samples makes line addressing unnecessary. A field synchronization code, followed by the address of the first set of the first active line, provides all the necessary synchronizing information. The address is in the range 0 to 18.

The field synchronization code comprises eight bytes of the form 11110011 or 11111100. These two codewords are not allowed for the coded video signal. The order of the last two pairs of bits, 0011 or 1100, in each of the first seven bytes, representing 0 and 1 respectively, signals the address of the first set of the field. In the 8th byte, 1100 signals a first field (starting line 23) and 001 a second field (starting line 336).

The sequence in which the fields are transmitted is defined by the address of the first set of the first line and need not be specified, because the decoder reconstructs the picture from received addresses.

A.3.3.2 *Data structure*

In each transmitted field, the data comprises 8 field synchronization bytes followed by the sets of picture data. In each set, $(E'_B - E'_Y)_n$ is transmitted first, followed by $(E'_Y)_n$, $(E'_R - E'_Y)_n$ and $(E'_Y)_{n+1}$. For transmission, a parallel-to-serial conversion takes place at the encoder. The most significant bits in the transmitted bit stream are leading.

A.3.3.3 *Data output*

The output bit-rate to the transmission channel is at less than 2 Mbit/s, the actual value depending on the number of time slots allocated to video.

The octet structure of the ouput data must be aligned with the time slot structure of the primary rate interface.

The approximate transmission time for a complete picture is 4 s.

A.3.4 *Decoder*

The received data are associated with addressed derived from the field synchronization code and assembled in a picture store having a capacity of 6.6355 Mbits. The contents of this store are read out sequentially.

A.3.5 *Interface*

A.3.5.1 *Video interface*

i) *Analogue interface* – An RGB-interface rather than a composite signal interface (PAL, SECAM) is recommended to maintain a high quality video signal.

ii) *Digital interface* – The structure of sets defined in § A.3.3.2 allows for definition of a digital interface according to CCIR Recommendation 656 for E'_Y, $E'_R - E'_Y$ and $E'_B - E'_Y$.

A.3.5.2 *Digital interface for transmission signal*

The graphics mode may be internal or external to the face-to-face codec. An external device may have a digital interface according to Recommendations X.21 and V.11 (leased circuits). The picture data must be delayed at least 40 ms with respect to the control signal C specified in Recommendation X.21.

A.3.6 *Signalling of graphics mode 2*

The graphics mode 2 is signalled in the codec-to-codec information with bit 3.1.5 set to 1. For nomenclature of the bits, see Recommendation H.130.

A.3.7 *Compatibility with graphics mode 1*

In the graphics encoder and decoder, additional means are incorporated to make graphics mode 2 compatible with mode 1. They are signalled by setting bit 3.1.0 to 1 in the codec-to-codec information. If bit 3.1.5 of the codec-to-codec information is received set to 0 and bit 3.1.0 is received set to 1, then the graphics codec automatically switches into mode 1.

ANNEX B

(referring to § 1 of Recommendation H.120)

Encryption option – 625 line

Under study.

ANNEX C

(referring to § 2 of Recommendation H.120)

Graphics option – 525 line

C.1 *Introduction*

The 525-line version of this graphics mode is very similar to the 625-line version for mode 1 specified in Annex A. It uses the same systematic replenishment technique and since the receiver is totally asynchronous from the transmitter, no adjustment for the differing picture rates is necessary. Instead of any form of standards conversion, the interworking between the 525-line and 625-line versions is provided by creating a small change in picture size. In 525 to 525-line transmission, the displayed picture size is the same as produced by the transmitting camera. In 525 to 625-line transmission, the displayed picture is reduced in size and is surrounded by a small black border (about 8%). In 625 to 525-line transmission, the displayed picture is expanded (equivalent to an overscan of about 8.5% in each border) so that a small amount of the transmitted picture area is not displayed.

Most of the details of this graphics mode are identical to those of the 625-line version for mode 1 in Annex A, so that only the difference need be specified in this Annex.

C.2 *Facilities*

The facilities are essentially the same as for the 625-line version.

C.3 *Coding*

The luminance and colour-difference sampling frequencies are 10.08 and 10.08/3 MHz respectively, the sampling frequency being locked to the television line-scanning frequency.

The arrangements for PCM coding are identical with the 625-line version but an area greater than the active picture area is sampled. There are 639 samples per line, the same number as for the 625-line version; 494 or 516 lines per picture are sampled. When a 525-line signal is being sampled at 10.08 MHz, only about 537 samples are required for the active line. The excess 102 samples, set to black level, are placed evenly on each side of the active line samples.

For 525-line transmission, samples from the first active line of Field No. 1 (line 14) form the $(E'_B - E'_Y)$ component, while those from the first active line of Field No. 2 (line 277) form the $(E'_R - E'_Y)$ component. For 625-line transmission, samples from the first active line of Field No. 1 (line 9) form the $(E'_R - E_Y)$ component and those from the first active line of Field No. 2 (line 272) form the $(E'_B - E'_Y)$ component.

C.4 *Transmission and synchronization*

C.4.1 *General*

The systematic replenishment algorithm based on the consecutive transmission of every 19th sample is also used in the 525-line version. However, since the samples extend over almost the complete line period, the divide-by-nineteen clock is suspended for only one luminance sample period during line-blanking. During transmission to a 625-line decoder, 5 additional lines per field are included before the picture area starts together with 6 additional lines per field after the end of the picture area, increasing the lines per field from 247 to 258. The luminance and colour-difference values on the extra lines are set to black level. Also, the divide-by-nineteen clock is changed to divide-by-five while samples are being selected from the added lines. This deceives the 625-line decoder into believing that there are 19 lines (per field) of black above the picture and 22 lines below the picture, making the total number of lines per field equal to 288, the same value as in Annex A.

The field synchronization code and the method of identifying fields are identical to those described in Annex A (except that the first line of Field No. 1 may be either line 14 or 9 and that of Field No. 2 either line 277 or 272).

C.4.2 *Data structure*

In 525 to 525-line transmission, each transmitted field comprises 8 field-synchronization bytes followed by 8307 bytes of picture data.

In 525 to 625-line transmission, each transmitted field comprises 8 field-synchronization bytes followed by 9685 or 9686 bytes of picture data, exactly the same as in the 625-line version in Annex A. In the 525-line coder the picture data are assembled from:

5 lines of 639 samples, every 5th sample ... 639 bytes,

247 lines of 639 samples, every 19th sample ... 8207 bytes,

6 lines of 639 samples, every 5th sample ... 766 bytes.

The number of bytes required form the 6 lines at the bottom of the picture is 739 or 740. The excess bytes (all at black level), arising from non-integral results of division, are discarded.

Other details of the data structure are as in Annex A.

C.4.3 *Data output*

The graphics data are generated at a nominal rate of about 4 Mbit/s and fed into buffer store. The output from the buffer is at less than 2 Mbit/s (depending on the number of time slots allocated to video). When the buffer level at the end of a field exceeds 160 kbits, the sampling is suspended for two complete fields to allow the buffer to empty. If the level is then still in excess of 160 kbits, sampling is suspended for a further two fields.

The octet structure of the output data must be aligned with the time slot structure of the primary rate interface.

The resulting transmission time for a complete picture is of the order of 1.7 to 3 s.

C.5 *Decoder*

The received data are associated with addresses derived from the field synchronization code and assembled in a picture store having 639 × 494 addressable positions with 8-bit capacity. The data are stored in the multiplexed form (luminance and colour-difference) used for transmission. The contents of the store are read out sequentially, the luminance and colour-difference components line-interpolated to give the $(E'_R - E'_Y)$ and $(E'_B - E'_Y)$ components simultaneously and coincident with the associated luminance.

The length of the line in the picture store is 639 elements; for a 525-line picture sampled at 10.08 MHz, the active line requires only 537 elements. When television blanking is applied to the output signals, the 102 extra elements are suppressed and the standard 525-line signal results.

When receiving a signal from a 625-line terminal, 639 elements per line are received and stored. However, the first 19 active lines and the last 22 active lines of each field of the 625-line signal are not read into the store and are discarded. This, together with the effect of the line blanking on the horizontal output from the store, provides a 525-line display corresponding to the 625-line input picture after a border about 8% wide has been trimmed from all four edges.

ANNEX D

(referring to § 2 of Recommendation H.120)

Encryption option – 525 line

Under study.

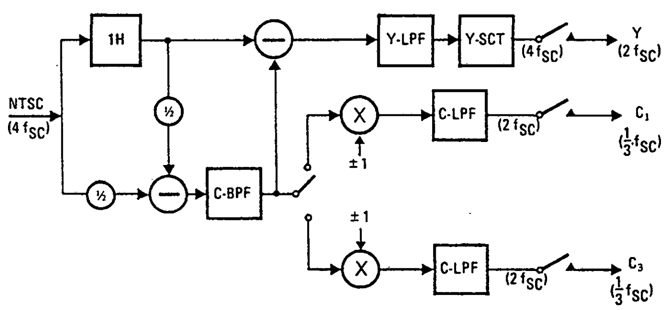
ANNEX E

(referring to § 3 of Recommendation H.120)

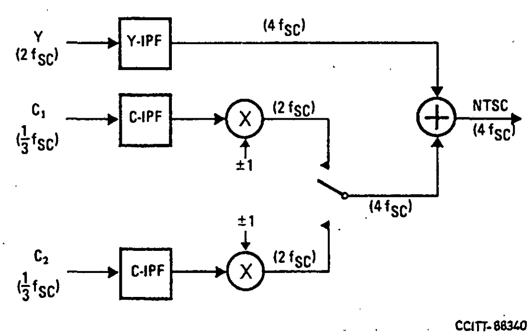
Colour decoding and coding filters

E.1 *Configuration*

See Figure E-1/H.120.



a) Digital colour separation circuit



b) Digital colour composition circuit

H Line delay
Y-LPF Low pass filter for Y signal
Y-SCT Subcarrier trapper
C-BPF Band pass filter for C signal
C-LPF Low pass filter for C signal
f_{SC} Colour subcarrier frequency
Y-IPF Interpolation filter for Y signal
C-IPF Interpolation filter for C signal

FIGURE E-1/H.120

E.2 *Basic filter characteristics*

See Table E-1/H.120.

TABLE E-1/H.120

Filter	Transfer function H(z)
C-BPF	$(-Z^{-2}+2-Z^{2})/4$
Y-LPF	$(-3Z^{-3}+19Z^{-1}+32+19Z-3Z^{3})/64$
Y-SCT	$(Z^{-5}-3Z^{-3}+10Z^{-1}+10Z-3Z^{3}+Z^{5})/16$
C-LPF	$(Z^{-4}+3Z^{-2}+4+3Z^{2}+Z^{4})/12$
Y-IPF	$(-3Z^{-3}+19Z^{-1}+32+19Z-3Z^{3})/64$
C-IPF	$(Z^{-2}+1+Z^{2})(Z^{-1}+2+Z)(-Z^{-8}-2Z^{-6}+2Z^{-4}+6Z^{-2}+6+6Z^{2}+2Z^{4}-2Z^{6}-Z^{8})/192$

E.3 *Advanced filter characteristics*

See Table E-2/H.120.

TABLE E-2/H.120

Filter	Transfer function H(z)
C-BPF	$(Z^{-8}-9Z^{-6}+17Z^{-4}-23Z^{-2}+28-23Z^{2}+17Z^{4}-9Z^{6}+Z^{8})/128$
Y-LPF	$(-Z^{-7}+4Z^{-5}-10Z^{-3}+39Z^{-1}+64+39Z-10Z^{3}+4Z^{5}-Z^{7})/128$
Y-SCT	$(Z^{-5}-3Z^{-3}+10Z^{-1}+10Z-3Z^{3}+Z^{5})/16$
C-LPF	$(Z^{-4}+3Z^{-2}+4+3Z^{2}+Z^{4})/12$
Y-IPF	$(-Z^{-7}+4Z^{-5}-10Z^{-3}+39Z^{-1}+64+39Z-10Z^{3}+4Z^{5}-Z^{7})/128$
C-IPF	$(Z^{-2}+1+Z^{2})(Z^{-1}+2+Z)(-Z^{-8}-2Z^{-6}+2Z^{-4}+6Z^{-2}+6+6Z^{2}+2Z^{4}-2Z^{6}-Z^{8})/192$

ANNEX F

(referring to § 3 of Recommendation H.120)

An example of coding control sequence

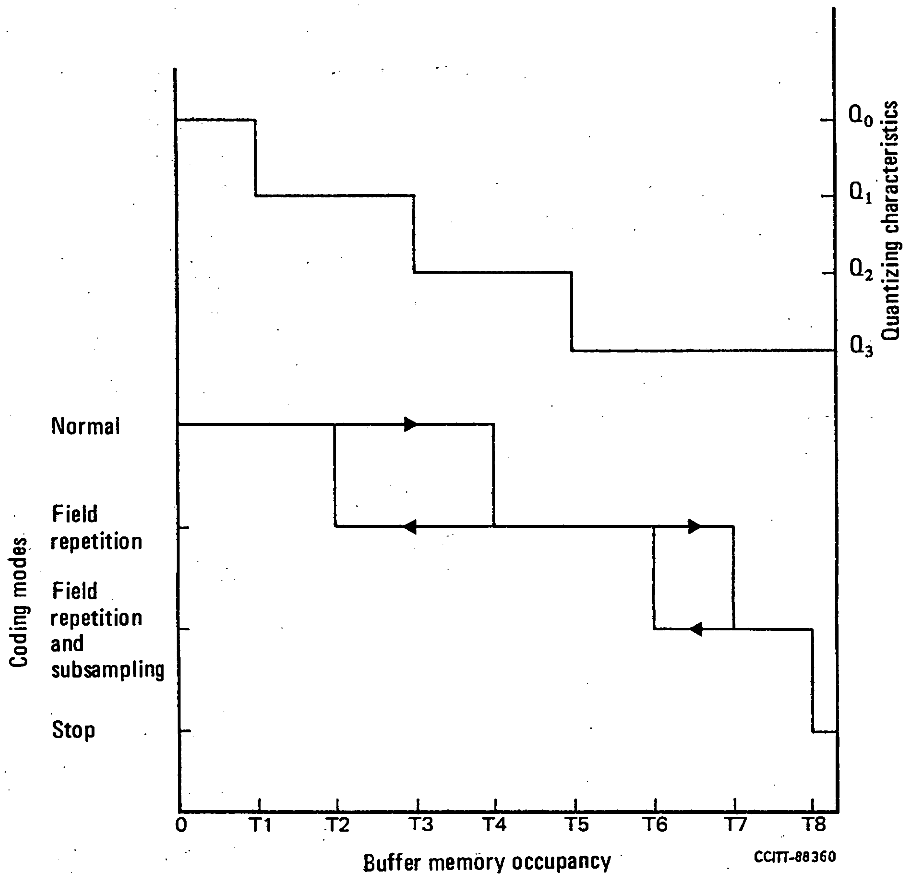


FIGURE F-1/H.120

ANNEX G

(referring to § 3 of Recommendation H.120)

Examples of entropy coding

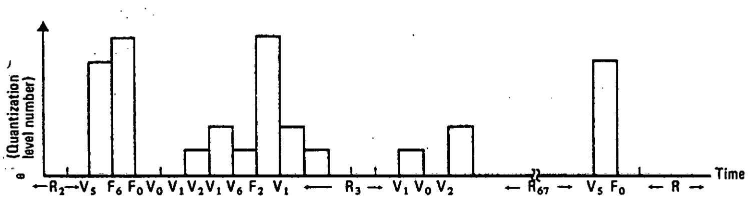


FIGURE G-1/H.120

Coding of prediction error e

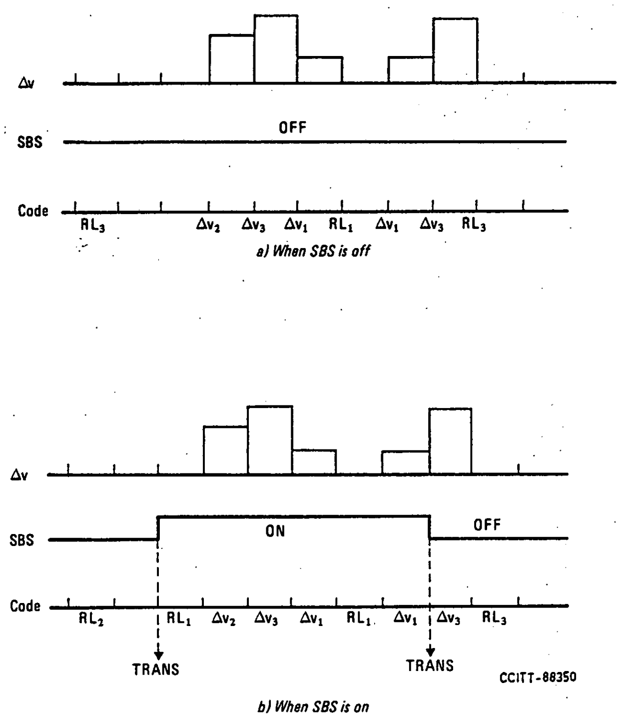


FIGURE G-2/H.120

Coding of motion vector v

APPENDIX I

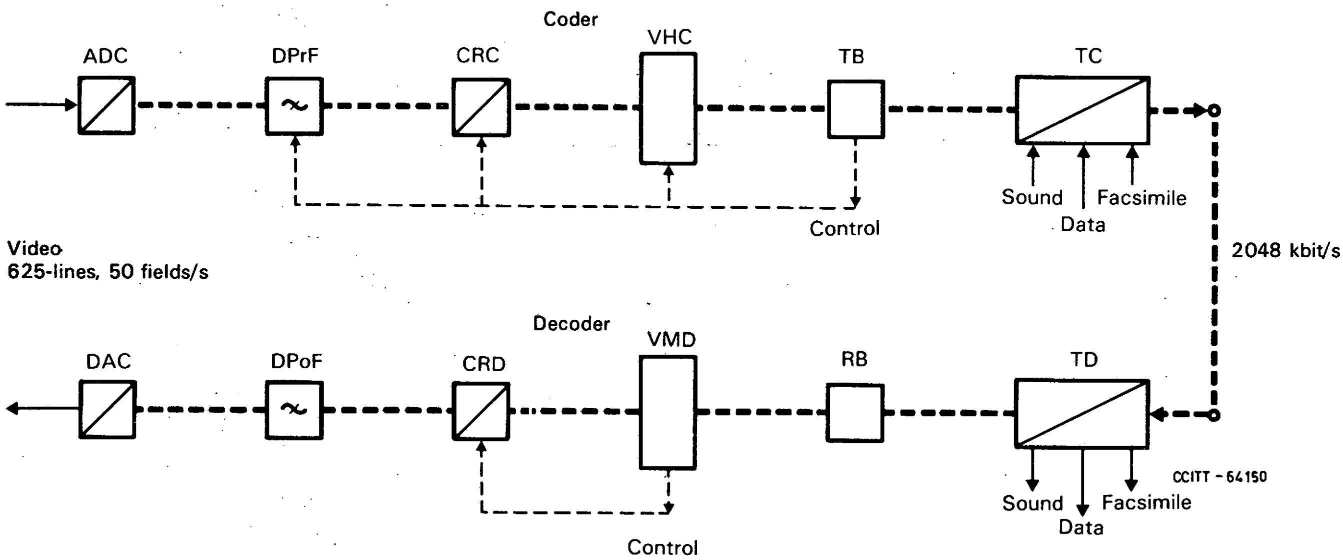
(to Recommendation H.120)

Outline description of operation of codecs in §§ 1 and 2

Since the conditional replenishment codec is a complex and unfamiliar item, this simplified outline of its method of operation is included to make the Recommendation more easily comprehensible. More complete descriptions are to be found in published papers [1], [2].

A conditional replenishment codec operates by transmitting only those parts of a picture which differ significantly from one television frame to the next. This normally gives rise to data being generated in spurts separated by gaps in which no data are being generated. To match the non-uniform data generation to a channel transmitting at a uniform rate, a buffer is used to smooth out short-term fluctuations while, for longer-term variations, the coding algorithm is adaptively modified to change the rate of generation. In the event of too much data, caused for example by a lot of of movement, the definition of the transmitted moving area is decreased, taking advantage of the reduced ability of the eye to perceive detail as the rate of movement increases. When little movement is present, the moving-area data are supplemented by data from non-moving areas in such a way that the whole picture is replenished over several picture periods. Picture stores are required at both transmitter and receiver and the objective is to make the content of the receiving store follow that of the transmitting store as closely as possible.

The codec can be regarded as comprising three basic sections: the source codec, the video multiplex codec and the transmission codec. Figure I-1/H.120 shows an outline of the arrangement.



Coder		Decoder	
ADC	Analogue-Digital Converter	DAC	Digital-Analogue Converter
DPrF	Digital Pre-Filter	DPoF	Digital Post-Filter
CRC	Conditional Replenishment Coder	CRD	Conditional Replenishment Decoder
VMC	Video Multiplex Coder	VMD	Video Multiplex Decoder
TB	Transmission Buffer	RB	Receiving Buffer
TC	Transmission Coder	TD	Transmission Decoder

FIGURE I-1/H.120

Outline block diagram of codec

In the source codec, the video signal is first digitized and optionally pre-filtered. When used, the pre-filter conditions the signal for further processing by reducing noise to improve the performance of the subsequent movement detector and to reduce the subjective effects of subsampling. The movement detector, in conjunction with the picture store, determines which areas in the picture are deemed to be moving. Noise introduces uncertainties in this decision and, when two or more groups of picture elements along a scanning line are deemed to be moving but are separated by small numbers of non-moving picture elements (probably caused by noise), the moving groups and separating elements are combined to form a single cluster, thus minimizing the addressing information which is required. Clusters of moving picture elements are then coded using DPCM followed by variable-length (entropy) coding where the shortest codes are allocated to the most frequently-occurring DPCM prediction errors.

The video multiplex codec adds to the video data the line- and field-synchronization signals together with addressing and other information (for example, whether PCM or DPCM is being transmitted) which must be transmitted in close association with the video to ensure that the decoder responds correctly.

The buffer, which strictly is part of the source coder, accepts the irregularly spaced bursts of data and delivers them at a uniform rate for transmission. The extent to which the buffer is filled is monitored and this is used to modify the rate of data generation by the source coder. It can reduce the data rate by modifying the pre-filter response and the thresholds in the movement detector, and by initiating element and field subsampling. On the other hand, if the buffer tends to empty, it may initiate the generation of complete PCM coded lines to provide systematic updating of the picture stores.

The transmission codec accepts the video data, adds a 64 kbit/s channel for sound, a 32 kbit/s channel for codec-codec signalling and optional additional data channels for facsimile, signalling or other data. It assembles the various signals into a frame structure, defined in Recommendation H.130 which is compatible with Recommendation G.732 and therefore suitable for transmission on 2048 kbit/s digital paths. In doing so, it provides the justification facilities to enable the clock for video processing to be independent of the network clock.

References

[1] DUFFY (T. S.) and NICOL (R. C.): A codec for visual teleconferencing, *Communications 82*, IEE Conference Publication No. 209, 1982.

[2] NICOL (R. C.), CHIARIGLIONE (L.) and SCHAEFER (P.): The development of the European Videoteleconference Codec, *Globecom 82*, IEEE global telecommunications conference, 1982.

Recommendation H.130

FRAME STRUCTURES FOR USE IN THE INTERNATIONAL INTERCONNECTION OF DIGITAL CODECS FOR VIDEOCONFERENCING OR VISUAL TELEPHONY

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

Introduction

Videoconferencing and visual telephony are new services which require greater bit rates than telephony. In the studies in CCITT on the ISDN and on international interworking, 384 kbit/s is emerging as an important channel capacity for wideband services. On this basis it is recommended that the videoconferencing and visual telephone services should be based on multiples of 384 kbit/s.

It is noted that both the 2048 kbit/s and 1544 kbit/s primary digital levels can be expressed by the formula $y + (n \times 384)$ kbit/s, where $n = 5$ or 4 and $y = 128$ or 8 kbit/s, respectively.

While this Recommendation covers only frame structures for transmission at the primary digital rates, it is not intended to suggest that transmissions using other frame structures or formats at primary rates or lower are precluded. In the future, frame structures based on other multiples and/or sub-multiples of 384 kbit/s may also be considered.

1 Characteristics of a 2048 kbit/s (n = 5) frame structure for use in codecs described in § 1 of Recommendation H.120

1.1 *General characteristics*

The multiplex structure described under § 1 is suitable for use on digital paths and connections which interconnect video codecs for videoconferencing or visual telephony using 2048 kbit/s transmission. The connections may either be direct or via higher-order digital multiplex equipment compatible with the primary PCM multiplex equipment defined in Recommendation G.732.

Some of the characteristics of this multiplex structure are identical to those in Recommendation G.704 and are covered by cross-references to that Recommendation.

The main features of the multiplex structure are that it provides:

- one 64 kbit/s channel for frame alignment, alarm signals and other signals as required;
- one 64 kbit/s channel reserved for the transmission of the sound signal;
- one 32 kbit/s channel for codec-to-codec information;
- the option of one or two 64 kbit/s channels and/or one 32 kbit/s channel for stereophonic sound, facsimile, data, etc.;
- the possibility of end-to-end and subscriber-to-network signalling;
- the remaining capacity (between 1664 and 1888 kbit/s) is used for the encoded video signal.

1.1.1 *Fundamental characteristics*

The multiplex structure contains 32 time slots, each of 64 kbit/s.

1.1.2 *Bit rate*

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

1.1.3 *Timing signal*

The timing signal is a 2048-kHz signal from which the bit rate is derived. It should be possible to derive the timing signal from an internal source or from the network.

1.1.4 *Interfaces*

The interfaces should comply with Recommendation G.703.

1.2 *Frame structure and time slot allocations*

The frame structure is in accordance with Recommendation G.704, § 3.3. The time slot (TS) allocations within the frame are given in Table 1/H.130, two options are shown according to whether or not the network is switched (under control of signals within the frame structure).

1.3 *Codec-to-codec information*

This information is transmitted in the 32 kbit/s channel corresponding to odd frames of TS2 (frame parity is gained from the multiframe alignment in the 8th bit of alternate TS2, the frames are consecutively numbered 0 to 15, forming a multiframe).

The 32 kbit/s channel is structured in a multiframe and supermultiframe derived from 128 consecutive 256 bit frames. The multiframe is composed of 8 octets numbered 1, 3, 5, . . ., 15, each from TS2 in an odd numbered 256 bit frame. The supermultiframe corresponds to 8 consecutive multiframes which are numbered 0, 1, 2, . . ., 7.

The use of the bits in each octet in the odd frames is as follows:

- Bit 1 for clock justification,
- Bit 2 for buffer state,
- Bit 3 for coding mode identification; the 8 consecutive bits 3 of TS2 in a multiframe will carry the following information:

Bit 3.1 [1]	Codec facilities	(see below)
Bit 3.3	Colour transmission	(1 if provided)
Bit 3.5	split-screen indicator	(if required)
Bit 3.7	Fast update request	(1 if required)
Bit 3.9	Advance warning of interruption	(1 if required)
Bit 3.11	Sound power signal, for use with encrypted multipoint	(under study)
Bit 3.13	Data distribution	(1 if required)
Bit 3.15	Detection of looped ports	(set to 1)

Bit 3.1 is used to signal the availability of certain facilities in the decoder at supermultiframe rate, as follows:

Bit 3.1.0	Graphics (mode 1)	(1 if provided)
Bit 3.1.1	High-quality speech	(1 if provided)
Bit 3.1.2	4 × 384 kbit/s capability (see Note 1)	(1 if provided)
Bit 3.1.3	Encryption	(1 if provided)
Bit 3.1.4	System M	(1 if 525-line signal being coded)
Bit 3.1.5	Graphics (mode 2)	(1 if provided)
Bit 3.1.6	Spare	(set to 0)
Bit 3.1.7	2 × 384 kbit/s capability (see Note 1)	(1 if provided)

[1] The notation used here should be interpreted as in the following examples: Bit 3.1 means Bit 3 (in TS2) of frame No. 1 in each multiframe: Bit 3.1.0 means Bit 3 (in TS2) of frame No. 1 in multiframe No. 0 of each supermultiframe.

TABLE 1/H.130

Timeslot allocation in 32 Timeslot frame structure of Recommendation G.704

	Bit rate (kbit/s)	Timeslot allocation (within the 256-bit frame) — Non switched (i)	Timeslot allocation (within the 256-bit frame) — Switched (ii)
Frame alignment, network alarms, etc.	as in G.704	0	0
Speech information	64	1	1
Codec-to-codec information	32	2	2
Signalling information (subscriber-network)	64	–	16
Fax, data, etc. (optional)	up to 2 × 64	17 and/or 18	17 and/or 18
Encoded video information (minimum)	(i) 27 × 64 (ii) 26 × 64	3 to 16 + 19 to 31	3 to 15 + 19 to 31

Note 1 – *Frame alignment, network alarms, etc.*

This information is transmitted in TS0 with the same rules and characteristics as recommended in Recommendation G.704. Additionally, bit 8 in odd frames is used as a synchronization bit which is required when the codec is used with synchronous digital networks. On receipt of this bit set to zero, the transmission clock for the encoder will be derived from the incoming data stream. This bit is always set to one in the encoder.

Note 2 – *Speech*

Speech is transmitted at 64 kbit/s in TS1. The coding law is the A-law of Recommendation G.711 or, for future applications, the law that will be recommended by CCITT for higher quality speech. In the case of stereophonic transmission, the second speech channel will be transmitted in TS17.

Note 3 – *Codec-to-codec information*

This information requires a capacity of 32 kbit/s and is transmitted on odd frames of TS2. The remaining 32 kbit/s capacity on the even frames of TS2 will be used for encoded video or data transmission. The detailed *use* and structure of the 32 kbit/s channel for codec-to-codec information is described in § 1.3.

Note 4 – *Signalling (subscriber-to-network)*

A capacity of 16 kbit/s is considered adequate for videoconference as for basic access. The methods of switched access to the ISDN at 2048 kbit/s have not yet been formulated. Option (ii) avoids any problems in this respect, by leaving the whole of TS16 (64 kbit/s) clear of video information and available for subscriber signalling and call set-up information when switched access is required. For non-switched access, option (i) should be used.

Note 5 – *Facsimile, data, etc.*

When required, this information will be transmitted in TS17 and/or 18.

Note 6 – *Encoded video*

A minimum of 26 × 64 kbit/s capacity is reserved for encoded video in TS3 to 15 and 19 to 31. In addition, depending on applications, TS2 (even frames), TS16, 17 and 18 may also be used for video, providing a maximum of 29.5 × 64 kbit/s capacity; the available video bit-rate therefore lies between 1664 and 1888 kbit/s.

- Bit 4 to identify the use of time slots; the 8 consecutive bits 4 of TS2 in a multiframe will carry the following information:

Bit 4.1	TS2 (even) is used for video (0) or other (1)	
Bit 4.3	TS16 is used for video (0) or other (1)	
Bit 4.5	TS17 is used for video (0) or other (1)	
Bit 4.7	TS18 is used for video (0) or other (1)	
Bit 4.9	TS16, 26 to 31 are not used for video	(see Note 2)
Bit 4.11	Graphics transmission	(1 if required)
Bit 4.13	Error correction	(1 if required) (see Note 3)
Bit 4.15	Use of time slots for video in conjuction with bit 4.9	(see Note 2)

- Bit 5 for multipoint conferencing; provides a 4 kbit/s message channel (transparent through the codec) from customer to multipoint control unit, between control units and from customer to customer. (The message format and protocols are under study.)

  When the codec is not equipped with a message channel, bit 5 is used to signal split-screen: 1 = split-screen active, 0 = split-screen inactive.

- Bit 6 free (for possible national use) (set to 0)
- Bit 7 free (for possible national use)
- Bit 8 for multiframe and supermultiframe alignment; the values of bit 8 in each frame of the multiframe (multiframe and supermultiframe alignment patterns) should be as detailed in Table 2/H.130.

Note 1 – Bits 3.1.2 and 3.1.7, taken together, signal the capability of the codec to operate at various bit rates, as follows:

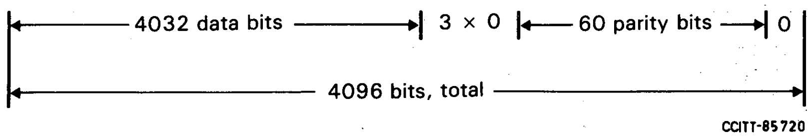
Bit 3.1.2	Bit 3.1.7	
0	0	2 Mbit/s only
1	0	2 Mbit/s and 4 × 384 kbit/s operation
0	1	2 Mbit/s and 2 × 384 kbit/s operation
1	1	2 Mbit/s and 4, 3, and 2 × 384 kbit/s operation

Note 2 – Bits 4.9 and 4.15, taken together, signal the time slots available (subject to the settings of bits 4.1, 4.3, 4.5 and 4.7) for video at various bit rates. The use of TS0, TS1 and TS2 (odd) is unaffected by these bits.

Bit 4.9	Bit 4.15	Bit rate	Time slot available for video
0	0	2 048 kbit/s	TS2 (even), TS3-31
1	0	4 × 384 kbit/s	TS2 (even), TS3-15 and 17-25
1	1	3 × 384 kbit/s	TS2 (even), TS3-9 and 17-25
0	1	2 × 384 kbit/s	TS2 (even), TS3-6 and 17-22

A 2 Mbit/s codec which allows $n \times 384$ kbit/s working, will set to zero time slots other than those mentioned above in its transmitter and ignore them in the receiver.

Note 3 – When set to 1, the last 64 bits of each multiframe contain the error corrector parity bits. The multiframe then appears as follows:



The conditions signalled in bits 3 and 4 can only change at supermultiframe rate. The change at the decoder will take place at the start of the first supermultiframe following the one where the change in signalling has been detected. This procedure can be used to improve the resistance to transmission errors.

TABLE 2/H.130

Multiframe and supermultiframe alignment on bit 8 of TS2 (odd)

Multiframe alignment pattern

Frame	0	1	2	3	4	5	6	7
1	1	1	1	1	1	1	1	1
3	1	1	1	1	1	1	1	1
5	1	1	1	1	1	1	1	1
7	0	0	0	0	0	0	0	0
9	0	0	0	0	0	0	0	0
11	1	1	1	1	1	1	1	1
13	0	0	0	0	0	0	0	0
15	1	1	1	0	0	1	0	(Note)
Multiframe	0	1	2	3	4	5	6	7

Supermultiframe alignment pattern

Note – Undefined (reserved for possible future use in a higher level framing structure).

2 Characteristics of a 1544 kbit/s (n = 4) frame structure for use with codecs described in § 2 of Recommendation H.120

2.1 *General characteristics*

The multiplex structure described under § 2 is suitable for use on digital paths and connections which interconnect video codecs for videoconferencing or visual telephony using 1544 kbit/s transmission. The connections may either be direct or via higher-order digital multiplex equipment compatible with the primary PCM multiplex equipment defined in Recommendation G.733.

Some of the characteristics of this multiplex structure are identical to those in Recommendation G.704 and/or in § 1 of this Recommendation; these are covered by cross-references to the appropriate documents.

The main features of the multiplex structure are that it provides:

- one 8 kbit/s channel for frame alignment, alarm signals and other signals as required;
- one 64 kbit/s channel for the sound signal;
- one 32 kbit/s channel for codec-to-codec information;
- the option of one or two 64 kbit/s channels and/or one 32 kbit/s channel for auxiliary data services;
- the remaining capacity (between 1280 and 1440 kbit/s) is used for the encoded video signal.

2.1.1 *Fundamental characteristics*

The multiplex structure contains 24 time slots per frame, each of 64 kbit/s, plus one bit per frame for frame alignment and signalling. The number of bits per frame is 193 and the nominal frame repetition rate is 8000 Hz.

2.1.2 *Bit rate*

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

2.1.3 *Timing signal*

The timing signal is a 1544 kHz signal from which the bit rate is derived. It should be possible to derive the timing signal from an internal source or from the network.

2.1.4 *Interfaces*

The interfaces should comply with Recommendation G.703; the option of AMI or B8ZS should be provided as the interface code. Which of the two codes is used should be determined by bilateral agreement.

2.1.5 *Format restrictions enforced by the network*

As indicated in Recommendation G.703, runs of more than 15 "zeros" are forbidden in some networks; also, there must be, on average, at least three "ones" in every 24 digits. Provision is made by means of a scrambling system to ensure that forbidden patterns cannot occur.

2.2 *Frame structure and time slot allocations*

The basic frame structure follows Recommendation G.704. The time slots are numbered from 1 to 24, with the 1st bit positioned between TS24 and TS1.

2.2.1 *Frame alignment*

The basic frame alignment is obtained at bit No. 1, as in Recommendation G.704, Method 2 (see § 2.1.3.2). The pattern transmitted is as shown in Table 3/H.130.

TABLE 3/H.130

Frame No.	Frame alignment signal	S-bit	Signalling bit
1	1	–	
2	–	0	
3	0	–	
4	–	0	
5	1	–	
6	–	1	A
7	0	–	
8	–	1	
9	1	–	
10	–	1	
11	0	–	
12	–	0	B

2.2.2 *Speech*

Speech is transmitted at 64 kbit/s in TS1. The coding law is the A-law of Recommendation G.711 or, for future applications, the law that will be recommended by CCITT for higher-quality speech. In the case of stereophonic transmission, the second speech channel will be transmitted in TS17.

2.2.3 *Codec-to-codec information*

This information is transmitted in the 32-kbit/s channel corresponding to the odd frames of TS2. The channel is structured in multiframes of 16 frames and supermultiframes of 8 multiframes in exactly the same way as in the 2-Mbit/s version in § 1. Multiframe and supermultiframe alignment are obtained from bit 8 of TS2 (odd) in the same way as in § 1.

The multiframe of TS2 for codec-to-codec signalling is quite independent of the basic 12-frame multiframe of Recommendation G.704.

2.2.4 *Signalling*

In the future, some 1.5 Mbit/s networks will allow the use of bits A and B for signalling. This facility is not available on all networks.

2.2.5 *Facsimile, data, etc.*

When required, this information will be transmitted in TS16 and TS17 and TS2 (even).

2.2.6 *Encoded video*

A minimum of 20 × 64 kbit/s capacity is reserved for encoded video in TS3-15 and 18-24; depending on applications, TS2 (even), TS16 and 17 may also be used for video, providing a maximum of 22.5 × 64 kbit/s capacity. The available bit rate for video therefore lies between 1280 and 1440 kbit/s.

2.3 *Codec-to-codec information*

The structure of the multiframe and supermultiframe are exactly the same as in § 1, except that each frame contains only 24 time slots as compared with 32 in the frames in § 1.

The bit allocations [in TS2 (odd)] are identical with § 1, with the following exceptions:

- Bit 1 for clock justification; required for interworking with 625-line codecs; disregarded in 525-line decoders.
- Bit 3.1.2 is permanently set to 1 (see Note 1)
- Bit 4.9 time slots are used for video (see Note 2)
- Bit 6 is reserved for the transmission of encryption data (see Annex D Recommendation H.120).
- Bit 7 is used for scrambler control (see § 2.4).

Note 1 – Bits 3.1.2 and 3.1.7, taken together, signal the capability of the codec to operate at various bit rates, as follows:

Bit 3.1.2	Bit 3.1.7	
0	0	Not used in 525-line codecs
1	0	4 × 384 kbit/s
0	1	2 × 384 kbit/s operation
1	1	4, 3, and 2 × 384 kbit/s operation

Note 2 – Bits 4.9 and 4.15, taken together, signal the time slots available (subject to the settings of bits 4.1, 4.3, 4.5 and 4.7) for video at various bit rates. The use of TS1 and TS2 (odd) is not affected by these bits.

Bit 4.9	Bit 4.15	Bit rate	Time slots available for video
0	0	This combinaison is not used in 525-line codecs	
1	0	4 × 384 kbit/s	TS2 (even), TS3-24
1	1	3 × 384 kbit/s	TS2 (even), TS3-9 and 16-24
0	1	2 × 384 kbit/s	TS2 (even), TS3-6 and 16-21

2.4 *Scrambling*

2.4.1 *General*

The bit sequence produced by a videoconference codec is not subject to any limitation on the bit patterns that are generated. Therefore, reversible processing has to be carried out at the output and input ports to ensure that the format restrictions specified for some 1544 kbit/s networks are not violated.

There are two typical constraints on the format:

1) There must not be runs of more than 15 consecutive "zeros".
2) The average density of "ones" must be at least 12.5%.

A classical self-synchronizing or reset scrambler, based on a maximum-length pseudo-random sequence, is incapable of guaranteeing that such a bit-sequence never occurs. It is however possible, by judicious choice of scrambler design, to minimize the number of violations of the above rules to such an extent that the residual violations can be removed by forcibly inserting "ones". The effect of this is to introduce transmission errors giving a residual bit-error-ratio of approximately 1×10^{-7}, which is imperceptible as far as the picture quality is concerned.

2.4.2 *Details of scrambling – first stage*

The scrambling sequence is applied to all 24 time slots but not to bit 193 nor to bit 7 of TS2 (odd).

Note – If data are inserted and/or extracted from TS2 (even), 16 or 17 within the network, the insertion/extraction equipments must ensure that the network constraints are not violated.

The 1544 kbit/s serial data from the codec are first applied to the following scrambling sequence:

I N I N N I,

where

I = inverted and

N = do not invert.

This sequence starts from the bit following bit 193, and is restarted every frame. Bit 193 and bit 7 of TS2 (odd) are not scrambled but the scrambling sequence is continuous through bit 7 of TS2 (odd).

2.4.3 *Details of scrambling – second stage*

Data scrambled by the above sequence are then checked for runs of more than 15 zeros. For signalling purposes, these data are considered to be in blocks of 385 bits. Each block starts with bit 8 of TS2 (odd) and ends with bit 6 of TS2 (odd). If a block of data preceding bit 7 of TS2 (odd) is found *not* to contain the string of data, 1 00000000 00000000 (i.e. no runs of 16 or more zeros), bit 7 of TS2 (odd) is set to one.

If a block of data preceding bit 7 of TS2 (odd) is found to contain the string of data, 1 00000000 00000001 (i.e. a run of 15 zeros), bit 7 of TS2 (odd) remains set to one, even if one or more subsequent runs of zeros within the same block reaches or exceeds 16. However, in such a case, the 16th zero(s) of the run(s) are set to one. As this is not signalled to the descrambler, it causes (a) single-bit transmission error(s).

Bit 7 of TS2 (odd) is set to zero only if the preceding block of data is found to contain the string, 1 00000000 00000000 (i.e. a run of 16 zeros or more), in which case the 16th zero is inverted to one and all subsequent strings of the form 1 00000000 0000000B within the same block have bit B inverted, except in the case where bit B = 1 before inversion, in which case it remains unchanged.

2.4.4 *Details of descrambler*

When bit 7 of TS2 (odd) is one, the preceding block of scrambled data is left unchanged. When bit 7 of TS2 (odd) is zero, the descrambler must detect all occurrences of the string 1 00000000 0000000B in the preceding block and invert the bit B. This can introduce transmission errors if the second or subsequent runs of zeros within the block (at the scrambler) contain 15 zeros.

The repetitive scrambling sequence, I N I N N I, is then applied to the data.

For the purpose of counting runs of zeros, at both the scrambler and descrambler, bit 7 of TS2 (odd) and bit 193 are both assumed to be zero. In the case where bit B would be on bit 193 or bit 7 of TS2 (odd), the string 1 00000000 000000B is used instead of 1 00000000 0000000B. Only bit B has to be within the block of data being considered. The preceding zeros may lie partially or completely within the preceding block.

When bit B is inverted, the "zeros" counter is reset to zero.

3 Characteristics of a 1544 kbit/s (n = 4) frame structure for use with codecs described in § 3 of Recommendation H.120

3.1 *General characteristics*

The multiplex structure describes under § 3 is suitable for use on digital paths and connections which interconnect video codecs for videoconferencing or visual telephony using 1544 kbit/s transmission. The connection may either be directly via the ISDN defined in Recommendation I.431 or via higher order digital multiplex equipment compatible with the primary PCM multiplex equipment defined in Recommendation G.733.

The main features of the multiplex structure are that it provides:

- one 8 kbit/s channel for frame alignment, alarm signals and other signals as required,
- one 64 kbit/s channel for the audio signal,
- one 32 kbit/s channel for codec-to-codec information,
- one optional 64 kbit/s for auxiliary data service, and
- the use of the remaining capacity (between 1376 and 1440 kbit/s) for the encoded video signal.

3.1.1 *Fundamental characteristics*

The multiplex structure contains 192 bits per frame plus one bit per frame for frame alignment and other purposes. The nominal frame repetition rate is 8000 Hz.

3.1.2 *Bit rate*

The nominal bit rate is 1544 kbit/s with a tolerance of ± 50 parts per million (ppm).

3.1.3 *Timing signal*

The timing signal is a 1544 kHz signal from which the bit rate is derived. It should be possible to derive the timing signal either from an internal source or from the network.

3.1.4 *Interfaces*

The interfaces should comply with Recommendation G.703. The interface code should be either of AMI/B8ZS described in Recommendation G.703, in addition to which CMI (Coded Mark Inversion) code is also applicable when the codec is installed as a part of terminal equipment. Which of the three codes is used should be determined by bilateral agreement.

3.1.5 *Format restrictions enforced by the network*

As indicated in Recommendation G.703, runs of more than 15 "zeros" are forbidden in some networks. Additionally, on the average, there must be at least three "ones" in every 24 digits. Provision is made by means of a stuffing system to ensure that forbidden patterns do not occur.

3.2 *Frame structure and bit allocation*

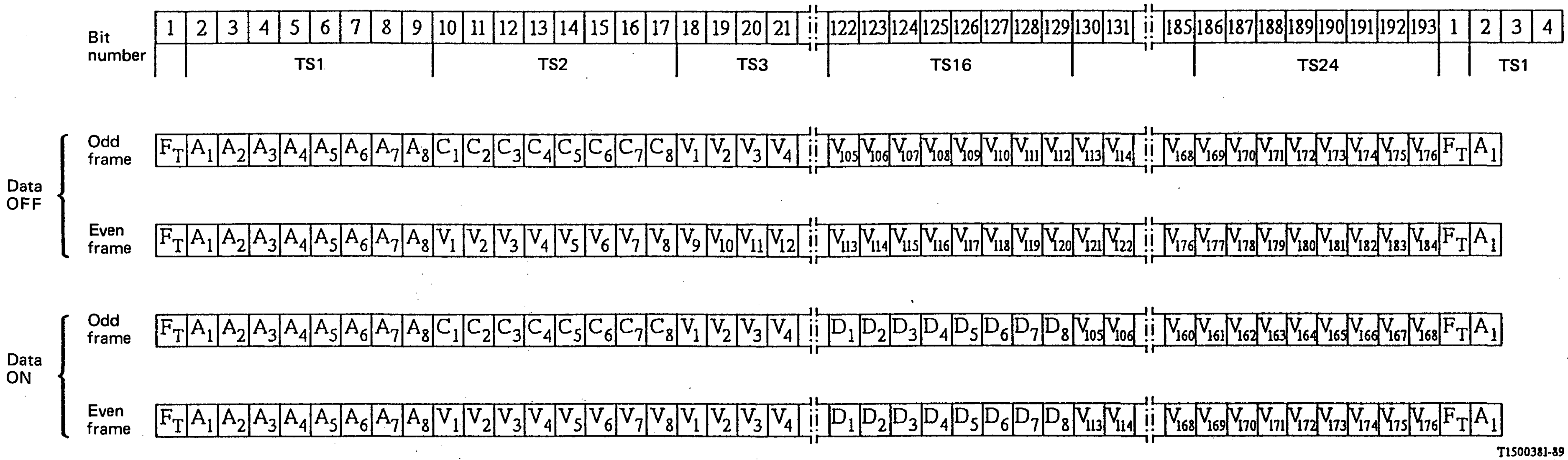
The basic frame structure follows Recommendation G.704 with changes in bit allocations. The bits in a frame are numbered from 1 to 193 with a transmission frame bit as numbered one. Remained 192 bits are divided into 24 time slots (TS) in which each time slot has 64 kbit/s rate. Time slot number is assigned to each TS in the way that the first slot is TS1 and the last slot is TS24. Bit allocation in a frame is shown in Figure 1/H.130.

3.2.1 *Frame alignment*

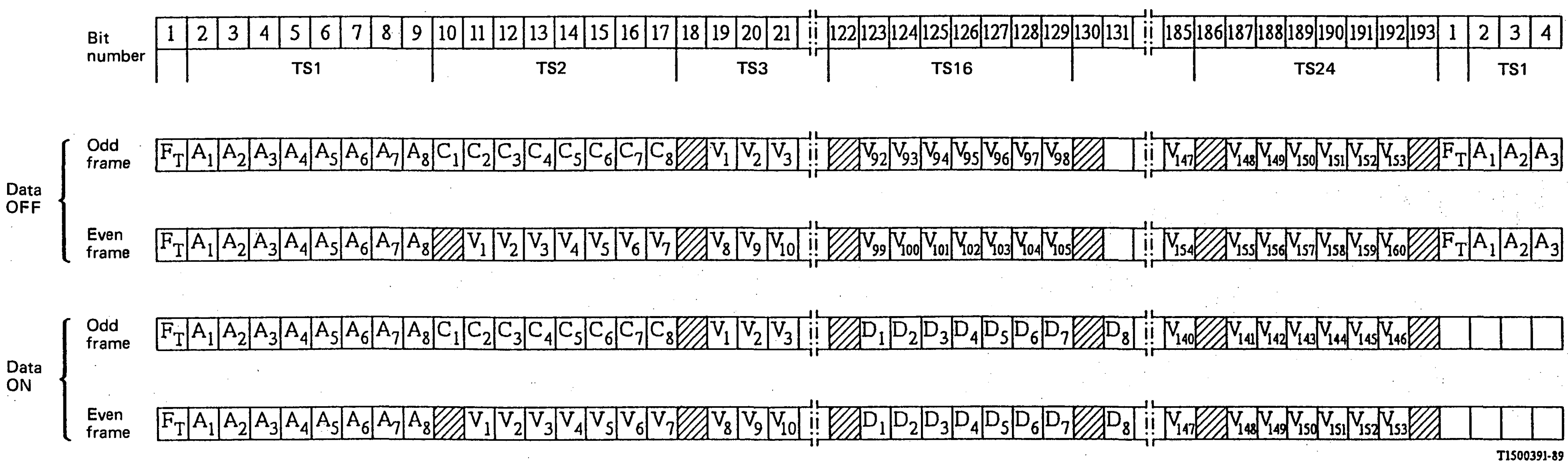
The basic frame alignment is obtained at bit No. 1 as in Recommendation G.704, Method 1 (§ 2.1.3.1).

3.2.2 *Audio signal*

The audio signal is transmitted at 64 kbit/s in TS1.



a) Without stuffing



b) With stuffing

FIGURE 1/H.130

Frame structure and bit allocation

3.2.3 *Codec-to-codec information*

This information is transmitted in odd numbered time slots 2 the 32 kbit/s channel. Identification of codec-to-codec information is made by detecting multiframe alignment which is inserted in the eighth bit of the odd numbered TS2.

The channel is structured in multiframes of 16 frames each (numbered from 1 to 16) and supermultiframes of 8 multiframes each (numbered from 1 to 8). Multiframe and supermultiframe alignment is obtained from bit No. 8 in TS2.

The multiframe of the codec-to-codec information channel is independent of the multiframe of the transmission frame generated by bit No. 0.

3.2.4 *Auxiliary data information*

When required, this information is transmitted basically in TS16 which is used for the encoded video signal when no optional auxiliary equipment is connected. If stuffing is performed due to some channel restrictions, data alignment is as given in § 3.4.2.

3.2.5 *Encoded video*

A minimum of 64 × 21.5 kbit/s capacity is primarily reserved for encoded video in even numbered TS2, TS3 through TS15 and TS17 through TS24. When the auxiliary data information channel is not set up, the capacity is increased to 64 × 22.5 kbit/s with TS16 added. The available bit rate for the encoded video signal therefore lies between 1376 and 1440 kbit/s. If stuffing is performed, data alignment is as given in § 3.4.2.

3.3 *Codec-to-codec information channel*

The use of the bits in the codec-to-codec information channel is as follows (see Table 4/H.130). In the following Sections, the notation, "m.n.l", is used for a bit position expressing the *n*th multiframe and the *l*th supermultiframe of bit No. m.

3.3.1 *C_1 Bit*

Bits 1.1, 1.5, 1.9, 1.13	Permanently set to 1
Bits 1.3, 1.7, 1.11	FC (sampling frequency control)

The lower 8 bits of the binary count for the two supermultiframe periods, i.e. 32 ms, are measured with the video sampling frequency clock, the MSB first. These same 8-bit words are transmitted in the three bits (1.3, 1.7 and 1.11) as well as in the two consecutive multiframes.

Bit 1.15	Spare (Note)

Note – Spare bits are set to 1.

3.3.2 *C_2 bit: stuffing flag*

Bits 2.1-2.15 (odd number)	0 if not stuffed

The stuffing flag consists of four bits including C_2 and C_7 in each violation detection block (four frame length) which is defined in § 3.4.2. The first three bits are used for majority decision logic at the decoder. When the result indicates "stuffing", the decoder undergoes destuffing.

TABLE 4/H.130

Codec-to-codec information

<table>
<tr><th>Multiframe frame number</th><th>C_1</th><th>C_2</th><th>C_3</th><th>C_4</th><th>C_5</th><th>C_6</th><th>C_7</th><th>C_8</th></tr>
<tr><td>1</td><td>1</td><td rowspan="2">Stuffing flag</td><td>Codec facility</td><td rowspan="4">Data channel flag</td><td rowspan="8">Message channel
1</td><td rowspan="8">Message channel
2</td><td rowspan="2">Stuffing flag</td><td>MAS (1)</td></tr>
<tr><td>3</td><td>Spare</td><td rowspan="6">Spare</td><td>MAS (1)</td></tr>
<tr><td>5</td><td>1</td><td rowspan="2">Stuffing flag</td><td rowspan="2">Stuffing flag</td><td>MAS (1)</td></tr>
<tr><td>7</td><td>Spare</td><td>MAS (0)</td></tr>
<tr><td>9</td><td>1</td><td rowspan="2">Stuffing flag</td><td rowspan="4">Graphics mode flag</td><td rowspan="2">Stuffing flag</td><td>MAS (0)</td></tr>
<tr><td>11</td><td>Spare</td><td>MAS (1)</td></tr>
<tr><td>13</td><td>1</td><td rowspan="2">Stuffing flag</td><td rowspan="2">Stuffing flag</td><td>MAS (0)</td></tr>
<tr><td>15</td><td>Spare</td><td>Coding mode</td><td>SAS</td></tr>
</table>

MAS Multiframe alignment signal

SAS Supermultiframe alignment signal (1110010 * : * is for future use)

3.3.3 *C_3 bit: codec facility/coding mode*

Bit 3.1	Codec facilities	
Bit 3.1.1	Graphics mode 1 (high resolution)	(0 if provided)
Bit 3.1.2	Bit sequence independence	(0 if secured)
Bit 3.1.3	Monochrome mode	(0 if provided)
Bit 3.1.4	Video encryption	(0 if provided)
Bit 3.1.5	Audio encryption	(0 if provided)
Bit 3.1.6	Pointing function	(0 if provided)
Bit 3.1.7	Graphics (mode 2, standard resolution)	(0 if provided)
Bit 3.1.8	Spare (Note 1)	
Bit 3.3	Spare (Note)	
Bit 3.5	Spare (Note)	
Bit 3.7	Spare (Note)	
Bit 3.9	Spare (Note)	
Bit 3.11	Spare (Note)	
Bit 3.13	Spare (Note)	
Bit 3.15	Coding mode	
Bit 3.15.1	Video encryption	(0 if used)
Bit 3.15.2	Audio encryption	(0 if used)
Bit 3.15.3	Frame memory refresh request	(0 if requested)
Bit 3.15.4	Backward path	(0 if available)
Bit 3.15.5-3.15.8	Spare (Note)	

Note – Spare bits are set to 1.

3.3.4 *C_4 bit: channel assignment flag*

Bits 4.1, 4.3, 4.5, 4.7	Auxiliary data channel flag	(0 if used)
Bits 4.9, 4.11, 4.13, 4.15	Graphics mode flag	(0 if used)

In graphics mode, video data are inhibited and their bit positions are used for graphics data transmission.

These two flags consist of four bits as stuffing flag. Both auxiliary data and graphics data can be inserted or removed in a unit of multiframe (16 frames). Flags should precede the data by a multiframe.

3.3.5 *C_5 bit: Message channel 1*

Bits 5.1-5.15 (odd number) Message channel 1 (Note)

Note – Protocols for these message channels are under study.

3.3.6 *C_6 bit: Message channel 2*

Bits 6.1-6.15 (odd number)	Message channel 2 (Note)

Note – Protocols for these message channels are under study.

3.3.7 *C_7 bit: Stuffing flag*

Bit 7.1-7.15 (odd number)	0 if stuffed

3.3.8 *C_8 bit: Multiframe alignment*

Bits 8.1, 8.3, 8.7, 8.9, 8.11. 8.13	Multiframe alignment signal (1110010)
Bit 8.15	Supermultiframe alignment signal (1110010*) (see Note)

Note – The bit * is used for future higher order multiframing.

3.4 *Stuffing*

3.4.1 *General*

The bit sequence produced by a videoconferencing codec is not subject to any limitation on the bit patterns that are generated. Therefore, reversible processing has to be carried out at the output and input ports to ensure that the format restrictions specified for some 1544 kbit/s networks (described in § 3.1.5 above) are not violated.

To ensure this, the stuffing method shuld be employed in which necessary "ones" are inserted, or stuffed, if any violations are found in a block of bit streams to be transmitted. A flag is attached to the block to identify whether or not the block is stuffed.

3.4.2 *Details of stuffing*

Each block, which consists of four transmission frame lengths, i.e. $4 \times 193 = 772$ bits starting from the C_1 bit of the codec-to-codec information in the $(4n-3)$th frame, is checked. If any violations occur with respect to the rules:

– no more than 15 consecutive zeros, and

– at least 3 ones in any 24 bits,

ones are stuffed as follows:

TS1 not stuffed,

TS2 not stuffed in odd numbered frames, stuffed at the top bit of TS in even numbered frames,

TS3-23 stuffed at the top bit of each time slot,

TS24 stuffed at the top bit and bottom bit of the time slot.

The stuffing position is shown in Figure 1/H.130.

Note – When stuffing pulses are inserted, the transmisison bit rate for encoded video is reduced to 1252 kbit/s without auxiliary data transmission and to 1188 kbit/s with auxiliary data transmission.

In order to ease processing at block boundaries, the C_1 bit at the start of any block is assigned to be always as described in § 3.3.1 above as shown in Table 4/H.130.

To prevent 8 consecutive zeros in the codec-to-codec information when stuffing is carried out, the stuffing flag transmitted in the (C_2, C_7) bits are assigned as (1,0) for stuffing and (0,1) for no stuffing.

Violations are checked assuming that all transmission framing bits in bit No. 0 and stuffing flag bits in C_2 and C_7 are zero.

Note – If audio data are processed in the network, corresponding bits should also be assumed as zero for violation checking. However, as this may increase the probability of stuffing, measures are necessary to prevent such stuffing from becoming excessive.

3.4.3 *Stuffing mode operation*

Stuffing should be operated only when necessary. To identify the network restrictions, the bit sequence independence (BSI) in the codec-to-codec information channel is used. A coder usually operates without stuffing, but shifts to the stuffing mode if the received BSI is "one".

Recommendation H.140

A MULTIPOINT INTERNATIONAL VIDEOCONFERENCE SYSTEM

(Melbourne, 1988)

1 Scope

This Recommendation defines a multipoint videoconference system which enables three or more videoconference sites to intercommunicate simultaneously, provided that the codecs conform to Recommendations H.120 and H.130 (§ 1, Note).

Note – Codecs conforming to § 2 of Recommendations H.120 and H.130 are in principle also applicable.

2 General requirements

A multipoint control unit (MCU) is a piece of equipment located in a node of the network (terrestrial or satellite) which receives several (maximum seven) 2 Mbit/s channels from access ports (each access port corresponding either to a local or a remote codec, or to another MCU) and, according to a certain criterion, causes some of them called selected channels to be distributed towards the connected studios (see Figure 1/H.140).

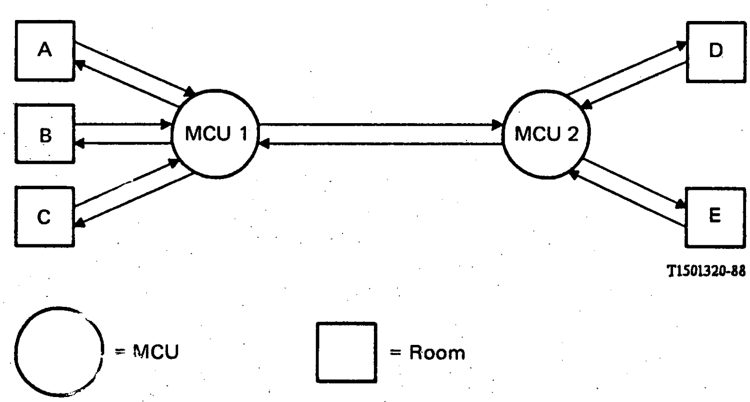


FIGURE 1/H.140

Use of MCU in a terrestrial network

The basic functions of the MCU for a terrestrial or a satellite network are identical. The MCU shall have the capability:

- To synchronise the incoming streams to a single 2048 kHz pilot clock.
- To extract frame alignment from TS0 in order to synchronise the different streams to the frame clock, to extract frame parity, multiframe and supermultiframe alignment from TS2 in order to access in each incoming stream the codec-to-codec signalling channel.
- To process this signalling channel.
- To process the sound channels in order to create an open sound system, in the case of an unencrypted system.
- To decide image switching and dispatching according to a selection criterion (automatic or on request).
- To signal in advance the switching decisions to the codecs so that degradation during and after the switching can be avoided.
- To multiplex the selected video channels with the open sound channel and the effective data channels.
- To distribute the reconstructed streams to the corresponding access ports.

3 Synchronisation of bit streams

3.1 *Clock synchronisation*

All incoming bit streams to the MCU must be derived from the same basis 2048 kbit/s clock. If no codec involved in a multipoint conference resides in a synchronous network, i.e. no signal is received with bit 8 in TS0 of odd frames set to zero, then the MCU acts as a master clock source. Such an MCU should have a reference clock with a short term accuracy of 1 in 10^9, in order to avoid frame slips during a conference session. If one or more codecs are in synchronous networks (bit 8 = 0), then their clocks are taken as the master.

In both cases the MCU sets on all outgoing channels bit 8 in odd TS0's to zero.

3.2 *Frame synchronisation*

The MCU has the following functions:

i) Extract from TS0 frame alignment and generate the frame clock. Frame parity should not be extracted from TS0 as it is not transparently transmitted through some networks.

ii) Extract from TS2 multiframe and supermultiframe alignment and generate: frame parity, multiframe clock, supermultiframe clock.

iii) Synchronise the bit streams at the PCM frame rate, such that switching can be performed without interrupting the Recommendation G.704 frame structure.

4 MCU and codec use of TS2 odd for multipoint conference applications

The bits are encoded according to Recommendation H.130, (§ 1). A majority decision of 5 out of 8 is used to provide resilience to channel errors with respect to the signals in bits 3 and 4.

4.1 Bits 1, 2, 6, 7 are transparently transmitted by the MCU.

4.2 Bit 8 gives multiframe and supermultiframe alignment and recovery of frame parity.

4.3 Bit 3 is for coding mode identification.

Bit 3.1.c indicates the facilities offered by the codec (set to 1 if provided) and are fixed for each codec. The MCU should take account of these bits in order to set up a minimum operating mode for all the codecs involved in the conference. For each individual port on the MCU a logical AND is made between the incoming signals from all other ports. The resultant signal is then used as the outgoing signal for that specific port, the rule being that an individual ports facilities bits should not be echoed back.

Bit 3.1.0	Graphics (mode 1)
Bit 3.1.1	High quality speech
Bit 3.1.3	Encryption
Bit 3.1.4	System M
Bit 3.1.5	Graphics (mode 2)
Bit 3.1.6	Spare – set to zero

Note 1 – MCUs not equipped to mix Recommendation G.722 audio will set bit 3.1.1 to zero.

Note 2 – The use of bit 3.1.3 for encryption is under study.

Bit 3.1.2	Bit 3.1.7	
0	0	2 Mbit/s working only
1	0	2 Mbit/s and 4 × 384 kbit/s working only
0	1	2 Mbit/s and 2 × 384 kbit/s working only
1	1	2 Mbit/s and 4, 3, 2 × 384 kbit/s working

Note – If the bit rate signalled by bits 3.1.2 and 3.1.7 exceeds that available at the codec digital interface, then the meaning of the facilities bits is as follows:

– with codecs having a 1.5 Mbit/s serial interface

0	0	Never occurs
1	0	Means 4 × 384 kbit/s working only
0	1	Means 2 × 384 kbit/s working only
1	1	Means 4, 3, 2 × 384 kbit/s working only

– with codecs having a 2 Mbit/s serial interface, but effective rate of 768 kbit/s

0	0	Never occurs
1	0	Never occurs
0	1	Means 2 × 384 kbit/s working only
1	1	Means 2 × 384 kbit/s working only

Bits 3.3 (colour transmission) and 3.5 (split-screen display) are transparently transmitted by the MCU.

4.3.1 *Bit 3.7 – Fast update request (FUR)*

When set to 1, the transmitter buffer occupancy is forced to decrease and stabilise to a state of less than 6 K by preventing coded picture elements entering the buffer.

4.3.2 *Bit 3.9 – Freeze frame request (FFR)*

Used to warn a decoder that its received signal may be interrupted after the start of the next supermultiframe for a period of no more than 2 s. On receipt of bit 3.9 set to 1 a decoder will normally "freeze" the contents of its frame store for 2 s or until a field start code is received with bit A set to 1 (see Recommendation H.120 § 1).

Both bits 3.7 and 3.9 should pass transparently across an MCU if set on an incoming signal: this is to allow for multipoint conferencing using distributed MCU's.

Bit 3.11.c indicates the power of the sound channel, integrated during 16 ms (period of the supermultiframe) and encoded with 8 bits. It is only used with encrypted multipoint, otherwise it is set to zero. The MCU can use this bit to select the New and Previous Speaker's channels (see § 6).

4.3.3 *Bit 3.13 – Data distribution*

When a codec received this bit set to one, it must vacate in its transmission channel the same time slots which are vacated with respect to the video signal in its receive channel and which are signalled by bits 4.1, 4.3, 4.5, 4.7.

The MCU uses this bit to ensure data continuity during a conference (see § 9).

4.3.4 *Bit 3.15 – Loop detection*

Bit 3.15 may be used by the MCU to detect whether one of its bidirectional 2 Mbit/s ports has been externally looped. It is necessary to monitor this condition since instability may result from such a configuration. The definition of bit 3.15 is as follows:

Codecs set bit 3.15 to 1 in their outgoing paths. MCUs use a number of consecutive bit 3.15s to transmit a serial random bit stream of length *n*, repeatedly. If the received bit sequence is the same as the transmitted random serial sequence then a loop has been detected. It should be noted that the received bit sequence may exhibit a phase delay compared with the transmitted sequence.

The details of the random sequence need not be rigidly specified as the sequence is only relevant when an individual MCU is in a looped configuration. However precautions must be taken to avoid false loop detection. This is likely to occur when two or more MCUs are connected together or when the transmission medium is subject to errors. A number of recommendations are given below.

The length of the transmitted random sequence *n* should be sufficiently large to avoid duplication when two or more MCUs are connected together. It is suggested that the total length be in excess of 15 bits, thus the possibility of duplication is less than 1 in 65536. The sequence transmission and detection mechanism should be sufficiently resilient to channel errors. This can be achieved in a number of ways, two simple methods are suggested here.

First, considering the sequence as a number of individual bits, each bit may be transmitted for 8 consecutive bit 3.15s. The receiver takes a majority 5 from 8 vote as the received sequence bit. Thus to transmit a single sequence takes $8 \times n$ bits. This is similar to the method adopted for bits $4 \cdot x$.

Alternatively, as the random sequence is transmitted repetitively, the decision as to whether the port is in the looped state or not is taken only when a number of sequences has been received.

4.4 Bit 4 is for time slot allocation.

If the following bits are set to 1:

Bit 4.1	TS2 even is not used for video
Bit 4.3	TS16 is not used for video
Bit 4.5	TS17 is not used for video
Bit 4.7	TS18 is not used for video
Bit 4.11	Graphics transmission
Bit 4.13	Use of error correcting code

When a codec receives any of the bits 4.3/5/7 set to one, and bit 3.13 is set to 1 (see § 4.3), it also vacates the corresponding time slots in its transmitted stream and sets to one the corresponding bits 4.b in its transmission channel.

Bit 4.1 is transmitted transparently by the MCU as the MCU cannot switch half time slots, i.e. the MCU takes no action.

Bits 4.9 and 4.15 are used for bit rate signalling.

Bit 4.9	Bit 4.15	
0	0	2 Mbit/s operation
1	0	4 × 384 kbit/s
1	1	3 × 384 kbit/s
0	1	2 × 384 kbit/s

At 5 × 384 kbit/s	Time slots 1-15 and 17-31 active
At 4 × 384 kbit/s	Time slots 1-15 and 17-25 active
At 3 × 384 kbit/s	Time slots 1-9 and 17-25 active
At 2 × 384 kbit/s	Time slots 1-6 and 17-22 active

The MCU should take account of bits 4.9 and 4.15 in order to set up a minimum operating mode for all the codecs involved in the conference. For each individual port, bits 4.9 and 4.15 from every other port on the MCU are analysed to determine which is the lowest requested bit rate permitted by the facilities bits 3.1.2 and 3.1.7. The code for this bit rate is then used as the outgoing signal in bits 4.9 and 4.15 for that specific port. Again the rule is that an individual port's bit rate facilities bits should not be echoed back.

To avoid a lock up situation, the codec should not return its received bits 4.9 and 4.15 in its transmit path, but should generate them independently.

4.5 Bit 5 carries a 4/kbits message channel

This bit is used to convey an asynchronous message channel at 4 kbit/s for signalling between the room and the MCU or between rooms or between MCUs.

The protocol of this message channel is under study.

5 Audio processing

Each terminal connected to an MCU must receive a mix of the audio from all other terminals. The audio signals should be summed at the MCU without normalisation, i.e. unity gain on each channel. The introduction of dynamic mixing for the suppression of ambient noise may be included but speakers would still enjoy unity gain.

Note – Not applicable with encrypted multipoint.

6 Switching decision criteria

The criteria for switching to some extent depends on the philosophy of the multiconference service in each Administration. Any solution, automatic or manual, can be implemented without altering the basic arrangement of the MCU.

The minimum working mode or "automatic" mode is as follows: the MCU, by comparison of the incoming sound channels or, in case of encrypted sound channels, by the means of the sound power bit (bit 3.11 in TS2 odd), selects the loudest speaker (called new speaker or NS). A second channel is selected by the MCU, being the previous loudest speaker (called previous speaker or PS). The NS is sent the PS channel and the other rooms are sent the NS channel. This mode is always used when the multiconference is established. Details of the switching criterion with respect to sound levels, hangover time, etc., are under study.

Five manual overrides are currently identified:

a) The system remains automatic but one location is considered to be the chairman of the conference. Participants are able to transmit a "request for the floor" to either the chairman or all rooms. At an appropriate time, the chairman orally gives the floor to the requesting conferee who, as he begins to speak, is automatically selected as the NS.

b) One location (e.g. NS or chairman or VIP) is able to choose the allocation of the second selected channel (normally the PS channel) by transmitting a request to the MCU.

c) Each location has the choice between the channels, which can be made available by the MCU connected to that location without affecting the displays of other locations.

d) Complete manual chairman control with no voice detection.

e) Manual forcing, where one location may force the MCU to regard his port as the NS.

This override is known as visualisation forcing. It may be used in one of two cases:

i) where a chairman or VIP wishes to be seen without interrruption,

ii) where a terminal is using a graphics camera, but is not equipped with a graphics capable codec.

Only the "automatic" mode does not require the use of the message channel in bit 5.

Modes a), b), c), d) imply the use of the message channel and extra control equipment (push-buttons, lights, signalling and data connections to the codec, etc.) at the conference room. Mode e) normally makes use of the message channel, but an interim solution is available on a National basis (see § 8.1).

7 MCU procedure for source switching

Once the switching decision is taken, (either by monitoring the audio levels or by the message channel, the MCU must prepare the connected codecs and operate as following:

i) it sends a FFR (bit 3.9) to all the codecs which will be affected by the switch, via the selected transmission channels connected to them.

ii) It performs image switching whilst maintaining the basic Recommendation G.704 frame structure continuity in the selected channel(s).

iii) It waits for at least 32 ms to allow sync recovery in all decoders.

iv) It sends a FUR (bit 3.7) to the codec(s) which are about to be used as a new picture source.

A FUR or a FFR must be set to one for at least one supermultiframe (SMF), or 256 frames in the case of non SMF-synchronous MCUs.

If the newly selected channels are connected to the MCU via a terrestrial link, the whole operation is likely to take no more than 100 ms. If it is via a satellite link, switch times of 500 ms are typical.

8 Protocols for "who is seen" in a multipoint conference

8.1 *Automatic mode*

This is described in § 6.

During automatic operation, it is convenient for the NS and PS to have some local indication that their picture is being transmitted. This facility is known as "visualisation status" or "on-air".

When defined, the message channel will have the capability of signalling such information together with many other useful facilities. In the short term, for existing codecs, an alternative means of signalling using TS2 odd bit 5 which is currently reserved for the message channel may be used for transmission of the visualisation and forcing bit by those countries who wish to implement such a simplified system. Under these circumstances, in a multiple MCU conference, the inter MCU link should be inhibited from sending the visualisation signal to avoid problems of contention. As a long term solution, the message channel is required to ensure compatibility with audioconferencing (this point is under study). In the mean time, the method of transmission should be bilateral agreement.

8.2 *Control using message channel*

Initialization and addressing procedure including the following items are under study;

- request for floor,
- local selection by request to view,
- chairman control.

9 Transmission of graphics during multipoint

This concerns the use of graphics modes 1 and 2 in the codec, not separate SPTV systems.

9.1 *Automatic mode*

The general principle is that all participants see the graphics information except the sender who sees the loudest speaker (other than himself).

The MCU first needs to establish whether all codecs in the conference have a graphics facility. If both the graphics facilities bits (bits 3.1.0 and 3.1.5) in any of the incoming channels to the MCU are set to zero, then the MCU sets both bits to zero in all its outgoing paths. This forces all codecs to use face-to-face type coding for graphics transmission.

When the MCU receives the graphics transmission bit set to 1 (bit 4.11), then the voice detector is overridden and the originating port (say port A) is made the new speaker, and hence transmitted to all other participants. Port A is sent the loudest speaker from the remaining ports (in the PS channel).

9.2 *Manual Mode*

Under study.

10 Transmission of data during multipoint

If a participant wishes to transmit data to all other terminals then data continuity must be ensured by the simultaneous vacation of the data channels by all codecs. This involves some delay (800 ms maximum if double satellite hops occur).

Time slot 2 even is not used for data distribution so it does not need to be separately switched by the MCU.

10.1 *Fully automatic mode* (i.e. no message channel)

The terminal "A" wishing to broadcast data sets to 1 the relevant bit 4 of TS2 for the data channel to be used. The MCU sets bit 3.13 to 1 in all outgoing streams except A and overrides the speaker detection procedure to make "A" the present speaker.

Other terminals on receipt of bit 3.13 and the relevant bit 4 in TS2 set to 1, vacate their outgoing equivalent data ports and set the corresponding bit 4's to 1.

The MCU then allows voice switching by the other ports after 2 s. When A concludes the data transmission it sets its relevant outgoing bit 4 in TS2 to zero. The MCU in turn sets bit 3.13 to zero. Normal voice switched operation then continues.

10.2 *Operation with message channel*

Under study.

11 Outline of output from MCU

Each location is sent a 2 Mbit/s channel reconstructed from one of the selected video channels, the corresponding TS2 odd with the possible modifications issued from the MCU in bits 3 or 5, the sound channel resulting of the mixing of the other sound channels and the effective data channels.

12 Multipoint conference configurations

12.1 *Terrestrial configuration*

Figure 1/H.140 shows a terrestrial multiconference using multiple MCU's. Many multiconferences may need only one MCU in a star formation.

12.2 *Possible satellite configurations*

Figure 2/H.140 shows a multiconference where rooms are connected through the same earth station to a single MCU. This situation is similar to that of § 12.1 but with a double hop between the rooms X and Y.

Other possibilities for satellite working are currently under study.

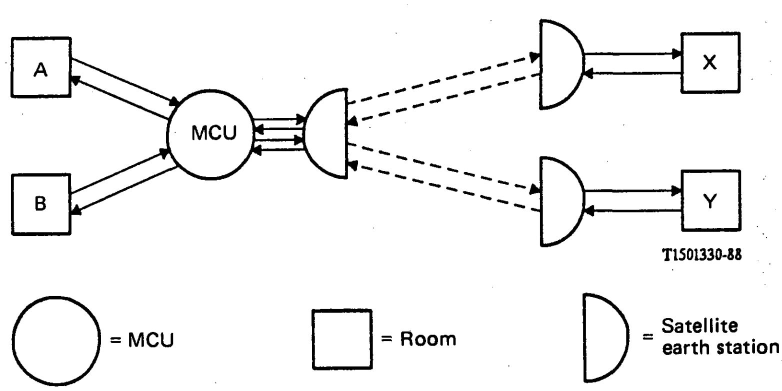


FIGURE 2/H.140

Use of a single MCU in a satellite configuration

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SECTION 3

INFRASTRUCTURE FOR AUDIOVISUAL SERVICES

Recommendation H.200

FRAMEWORK FOR RECOMMENDATIONS FOR AUDIOVISUAL SERVICES

(Melbourne, 1988)

1 Audiovisual services

A number of services are, or will be, defined in CCITT having as their common characteristic the transmission of speech together with other information reaching the eventual user in visual form. This Recommendation concerns a set of such services which should be treated in a harmonised way; it is convenient to refer to the members of this set as "audiovisual services" (abbreviated to AV services).

2 Harmonisation of audiovisual services

While the various audiovisual services may easily be distinguished in terms of their user-application, common methods are used for the transport of signals representing speech, moving or still pictures, and associated controls/indications, and also telematic auxiliary facilities. The standardisation process seeks the greatest possible harmonisation of these common features, confining the distinction to the application layers wherever possible, in order to:

a) Maximise the possibilities for intercommunication between terminals intended for different applications;

b) Maximise the commonality of hardware and software in the interests of economies of scale. The scope for commonality includes: audio and video input/output parameters, audio and video codecs, the control/indication set, frame structures and multiplexing, call control procedures (including multipoint).

The embodiment of this harmonisation policy will be a consistent set of Recommendations, consistent in the sense that all members of the set take into account all other members.

3 Purpose of this Recommendation

The purpose of this Recommendation H.200 is to define the set that shall be consistent. In fulfilling this function it is important to distinguish, at a given time, between Recommendations and draft Recommendations.

Recommendations are members of the set by virtue of their consistency with other adopted members of the set: these are listed in Annex A to this Recommendation. It is of course necessary to ensure continued consistency when amendments are introduced.

Draft Recommendations range from mere titles or outline contents through varying stages of maturity to a stable final draft. As many different intended members of the Recommendation H.200 set are developed in parallel to ensure consistency they should be treated as "provisional" members of the set. The list of set members including provisional items does not form part of Recommendation H.200, but this Recommendation H.200 should be updated in the future to include new members of the set formally adopted.

4 Framework

Recommendations in the H.200 set are arranged in three main sections:

Service definitions: These specify the service as seen by the user, including basic service, optional enhancements, quality, and intercommunication requirements, together with operational aspects; technical implementation methods are taken into account but not defined herein.

Infrastructure: This section includes all the Recommendations which are applicable to two or more distinct services: these encompass network configuration, frame structures, control/indications, communication/intercommunication, and audio/video coding. The "infrastructure" includes this generality of signals which flow on unrestricted digital bearers on established network connections – it does not include the methods of call establishment and control, orchestrated by signals outside these bearers.

Systems and terminal equipment: This section deals with the technical implementation of specific services: it therefore includes service-specific equipment for the application layer, and draws upon the infrastructure Recommendations to identify the detailed processes required for the particular service.

A *network aspects* section is also proposed, to cover those matters which are particular to AV services but, involving out-of-band signals, do not come within the scope of the infrastructure section above.

5 List of audiovisual services covered

The following audiovisual services shall be included in the harmonized set:

- narrowband videophone (1 and 2 × 64 kbit/s under study);
- broadband videophone (a teleservice for broadband ISDN);
- narrowband videoconferencing (n × 384 kbit/s and m × 64 kbit/s under study);
- broadband videoconferencing (a teleservice for broadband ISDN);
- audiographic teleconferencing;
- telephony (a degenerate case of an AV service, included for intercommunication purposes);
- telesurveillance.

The following audiovisual services are in the process of being defined, and consideration should be given to their inclusion in the set for either of the reasons given in § 2:

- video mail;
- videotex (including pictures and sound);
- video retrieval;
- high resolution image retrieval;
- distribution services.

ANNEX A

(to Recommendation H.200)

Framework for Recommendations for audiovisual services

				CCITT Rec. No.
A.1	*Service definition*			
	AV100	General Recommendation for AV services		F.700
		AV110	Teleconference services	F.710
			AV111	
			AV112	
		AV120	(Videophone services)	
			AV121 Basic narrow-band videophone service in the ISDN	F.721
A.2	*Infrastructure*			
	AV200	(General Recommendation for AV service infrastructure)		
		AV210	(Reference network configuration)	
		AV220	(General Recommendation for frame structures)	
			AV221 Frame structure for a 64 kbit/s channel in audiovisual teleservices	H.221
			AV222 Frame structure for 384-1920 kbit/s channels audiovisual teleservices	H.222
		AV230	(AV system control and indications)	
		AV240	(Principles for communiction between AV terminals)	
			AV241 System aspects for the use of the 7 kHz audio codec within 64 kbit/s	G.725
			AV242	
		AV250	(Audio coding)	
			AV251 Narrow-band audio coding at 64 kbit/s	G.711
			AV252 Wideband audio coding in 64 kbit/s	G.722
			AV253	
			AV254	
		AV260	(Video coding)	
			AV261 $n \times 384$ kbit/s video codec	H.261
			AV262	
A.3	*Systems and terminal equipment*			
	AV300	(General Recommendations for AV systems and terminals)		
		AV310	(Requirements for teleconferencing)	
			AV311	
			AV312	
			AV313 (Teleconference protocol)	
		AV320	(Requirements for videophone services)	
A.4	*Network aspects*			
	AV400			
		AV410	(Reservation systems)	
		AV420	(HLC for use in audiovisual calls)	
		AV430	(Call control command & indication)	
		AV440	(Multipoint call set-up)	

Note 1 – It is intended to merge the substance of existing Recommendations H.100 and H.110 into this framework in the next study period.

Note 2 – Entries in parentheses are indicative of the purpose of the various positions in the framework.

Note 3 – Further Recommendations will be added to the list as they are formally adopted.

Recommendation H.221

FRAME STRUCTURE FOR A 64 kbit/s CHANNEL IN AUDIOVISUAL TELESERVICES

(Melbourne, 1988)

Introduction

The purpose of this Recommendation is to define a frame structure for audiovisual teleservices in a single 64 kbit/s channel which makes the best use of the characteristics and properties of the audio/video encoding algorithms, of the transmission framing structure and of the existing CCITT Recommendations. It offers several advantages:

- It takes into account Recommendations such as G.704, X.30/I.461, etc. It may allow the use of existing hardware or software.
- It is simple, economic and flexible. It may be implemented on a simple microprocessor, using well known hardware principles.
- It is a synchronous procedure. The exact time of a configuration change is the same in the transmitter and the receiver. Configurations can be changed at 20 ms intervals.
- It needs no return link, since a configuration is signalled by a repeatedly transmitted codeword.
- It is very secure in case of transmission errors, since the BAS is protected by a double error correcting code.
- It allows the control of a higher multiplex configuration, into which the basis 64 kbit/s channel is inserted (in the case of $n \times 64$ kbit/s multimedia services such as videoconference).
- It can be used to derive octet synchronization in networks where this is not provided by other means.
- It can be used in multipoint configurations, where no dialogue is needed to negotiate the use of a data channel.
- It provides a variety of data bit-rates (from 6.25 bit/s up to 64 kbit/s) to the user.

1 Basic principle

The 64 kbit/s channel is structured into octets transmitted at 8 kHz. The eighth bit of each octet conveys a subchannel of 8 kbit/s. This subchannel, called service channel (SC), provides end-to-end signalling and consists of three parts (see Figure 1/H.221):

- *Frame alignment signal* (FAS): This signal structures the 64 kbit/s channel into frames of 80 octets each and multiframes (MF) of 16 frames each. Each multiframe is divided into eight 2-frame submultiframes (SMF): In addition to framing and multiframing information, control and alarm information may be inserted, as well as error check information to control end-to-end error performance and to check frame alignment validity. The FAS can be used to derive octet timing when it is not provided by the network.

- *Bit-rate alloction signal* (BAS): This signal allows the transmission of codewords to describe *the capability* of a terminal to structure the residual 62.4 kbit/s capacity in various ways, and to *command* a receiver to demultiplex and make use of the constituent signals in such structures; if other 64 kbit/s channels are associated, as in the case of $n \times 64$ kbit/s services (e.g. videoconference, videophone), this association may also be defined.

  Note – For some countries having 56 kbit/s channels, the net available bit rates will be 8 kbit/s less.

- *Application channel* (AC): This channel allows transmission of binary information of the insertion of message type data channel(s) (e.g. for telematic information) at up to 6400 bit/s. A minimum required command and indication channel should be provided and defined as part of the application channel (for further study). The remaining bit rate for the application channel may be added to the sound data or video channel. In this context, compatibility problems among audiovisual services should be considered.

Bit number

1	2	3	4	5	6	7	8	Octet number
Sub-channel #1	Sub-channel #2	Sub-channel #3	Sub-channel #4	Sub-channel #5	Sub-channel #6	Sub-channel #7	FAS	1 . 8
							BAS	9 . 16
							AC	17 80

FAS Frame Alignment Signal (Note)
BAS Bit-rate Allocation Signal
AC Application Channel

Note – The block termed as FAS also contains information other than that used for frame alignment.

FIGURE 1/H.221

Frame structure

The remaining 56 kbit/s capacity (with fully reserved application channel), carried in bits 1 to 7 of each octet, may convey a variety of signals within the framework of a multimedia service, under the control of the BAS and possibly the AC. Some examples follow:

- Voice, encoded at 56 kbit/s using a truncated form of the PCM of Recommendation G.711 (A-law or μ-law).
- Voice, encoded at 32 kbit/s and data at 24 kbit/s or less.
- Voice, encoded at 56 kbit/s with a bandwidth 50 to 7000 Hz (sub-band ADPCM according to Recommendation G.722). The coding algorithm is also able to work at 48 kbit/s. Data can then be dynamically inserted at up to 14.4 kbit/s.
- Still pictures coded at 56 kbit/s.
- Data at 56 kbit/s inside an audiovisual session (e.g. file transfer for communicating between personal computers).
- Sound and video sharing the 56 kbit/s capacity.

2 Frame alignment

2.1 *General*

An 80-octet frame length produces an 80-bit word in the service channel. These 80 bits are numbered 1 to 80. Bits 2-8 of the service channel in every even frame contain the frame alignment word (FAW) 0011011. These bits are completed by bit 2 in the succeeding odd frame to form the complete frame alignment signal (FAS).

So a pattern similar to the one in Recommendation G.704 is used (see Figure 2/H.221).

<table>
<tr><td>Successive frames</td><td>Bit No.</td><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 colspan="2" rowspan="2">Even frames (those containing FAW)</td><td rowspan="2">(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 word</td></tr>
<tr><td colspan="2">Odd frames</td><td>(Note 1)</td><td>1
(Note 2)</td><td>A
(Note 3)</td><td>E</td><td>C1</td><td>C2</td><td>C3</td><td>C4</td></tr>
<tr><td colspan="2"></td><td></td><td></td><td></td><td></td><td colspan="2">(Note 4)</td><td></td><td></td></tr>
</table>

Note 1 – See § 2.2 and Figure 3/H.221.

Note 2 – Bit used to avoid simulation of FAW by a frame-repetitive pattern.

Note 3 – A – Loss of either frame or multiframe alignment indication (0 = alignment, 1 = loss).

Note 4 – The use of bits E and C1-C4 is described in § 2.6.

FIGURE 2/H.221

Assignment of bits 1-8 of the service channel in each frame

2.2 *Multiframe structure*

Each multiframe contains 16 consecutive frames numbered 0 to 15 divided into eight submultiframes of 2 frames each (Figure 3/H.221). The multiframe alignment signal is located in bit 1 of frames 1-3-5-7-9-11 and has the form 001011. Bits 1 of frames 8-10-12-13-14-15 are reserved for future use. Their value is provisionally fixed at 0.

Bits 1 of frames 0-2-4-6 may be used for a modulo 16 counter to number multiframes in descending order. The least significant bit is transmitted in frame 0, and the most significant bit in frame 6. The receiver may use the mutiframe numbering to determine the differential delay of separate 64 kbit/s connections, and to synchronize the received signals. The use of an additional reserved bit in frame 8 to turn on and off the counting procedure is for further study.

2.3 *Loss and recovery of frame alignment*

Frame alignment is defined to have been lost when three consecutive frame alignment signals have been received with an error.

Frame alignment is defined to have been recovered when the following sequence is detected:

– for the first time, the presence of the correct frame alignment word;

– the absence of the frame alignment signal in the following frame detected by verifying that bit 2 is a 1;

– for the second time, the presence of the correct frame alignment word in the next frame.

When the frame alignment is lost, bit 3 (A) of the next odd frame is set to 1 in the transmit direction.

If frame alignment is achieved, but multiframe alignment cannot be achieved, then frame alignment should be sought at another position.

	Sub multiframe	Frame	Bits 1 to 8 of the service channel in every frame							
			1	2	3	4	5	6	7	8
Multiframe	SMF 1	0	N1	0	O	1	1	0	1	1
		1	0	1	A	E	C1	C2	C3	C4
	SMF 2	2	N2	0	O	1	1	0	1	1
		3	0	1	A	E	C1	C2	C3	C4
	SMF 3	4	N3	0	O	1	1	0	1	1
		5	1	1	A	E	C1	C2	C3	C4
	SMF 4	6	N4	0	O	1	1	0	1	1
		7	0	1	A	E	C1	C2	C3	C4
	SMF 5	8	N5	0	O	1	1	0	1	1
		9	1	1	A	E	C1	C2	C3	C4
	SMF 6	10	R1	0	O	1	1	0	1	1
		11	1	1	A	E	C1	C2	C3	C4
	SMF 7	12	R2	0	O	1	1	0	1	1
		13	R3	1	A	E	C1	C2	C3	C4
	SMF 8	14	TEA	0	O	1	1	0	1	1
		15	R4	1	A	E	C1	C2	C3	C4

R1-R4 Reserved for future use (provisionally set to 0).

A, E, C1-C4 As in Figure 2/H.221.

N1-N4 Used for multiframe numbering as described in § 2.2. Set to 0 while numbering is inactive.

N5 Reserved for an indicator of whether multiframe numbering is active or inactive. Currently set to 0.

TEA The terminal equipment alarm is set to 1 while an internal terminal equipment fault exists such that it cannot receive and act on the incoming signal. Otherwise it is set to 0.

FIGURE 3/H.221

Assignment of bits 1-8 of the service channel in each frame in a multiframe

2.4 *Loss and recovery of multiframe alignment*

Multiframe alignment is needed to validate the bit-rate allocation signal (see § 3). The criteria for loss and recovery of multiframe alignment described below are provisional.

Multiframe alignment is defined to have been lost when three consecutive multiframe alignment signals have been received with an error. It is defined to have recovered when the multiframe alignment signal has been received with no error in the next multiframe. When multiframe alignment is lost, even when an unframed mode is received, bit 3 (A) of the next odd frame is set to 1 in the transmit direction. It is reset to 0 when multiframe alignment is regained again.

2.5 *Procedure to recover octet timing from frame alignment*

When the network does not provide octet timing, the terminal may recover octet timing in the receive direction from bit timing and from the frame alignment. The octet timing in the transmit direction may be derived from the network bit timing and an internal octet timing.

2.5.1 *General rule*

The receive octet timing is normally determined from the FAS position. But at the start of the call and before the frame alignment is gained, the receive octet timing may be taken to be the same as the internal transmit octet timing. As soon as a first frame alignment is gained, the receive octet timing is initialized as the new bit position, but it is not yet validated. It will be validated only when frame alignment is not lost during the next 16 frames.

2.5.2 *Particular cases*

a) When, at the initiation of a call, the terminal is in a forced reception mode, or when the frame alignment has not yet been gained, the terminal may temporarily use the transmit octet timing.

b) When frame alignment is lost after being gained, the receive octet timing should not change until frame alignment is recovered.

c) As soon as frame and multiframe alignment have been gained once, the octet timing is considered as valid for the rest of the call, unless frame alignment is lost and a new frame alignment is gained at another bit position.

d) When the terminal switches from a framed mode to an unframed mode (by means of the BAS), the octet timing, previously gained, must be kept.

e) When a new frame alignment is gained on a new position, different from that previously validated, the receive octet timing is reinitialized to the new position but not yet validated and the previous bit position is stored. If no loss of frame alignment occurs in the next 16 frames, the new position is validated; otherwise the stored old bit position is reutilized.

2.5.3 *Search for frame alignment signal (FAS)*

Two methods may be used: sequential or parallel. In the sequential method, each of the eight possible bit positions for the FAS is tried. When FAS is lost after being validated, the search must resume starting from the previously validated bit position. In the parallel method, a sliding window, shifting one bit for each bit period, may be used. In that case, when frame alignment is lost, the search must resume starting from the bit position next to the previously validated one.

2.6 *Description of the CRC4 procedure*

In order to provide an end-to-end quality monitoring of the 64 kbit/s connection, a CRC4 procedure may be used and the four bits C1, C2, C3 and C4 computed at the source location are inserted in bit positions 5 to 8 of the odd frames. In addition, bit 4 of the odd frames, noted E, is used to transmit an indication about the received signal in the opposite direction whether the most recent CRC block has been received with errors or not.

When the CRC4 procedure is not used, bit E shall be set to 0, and bits C1, C2, C3 and C4 shall be set to 1 by the transmitter. Provisionally, the receiver may disable reporting of CRC errors after receiving eight consecutive CRCs set to all 1s, and it may enable reporting of CRC errors after receiving two consecutive CRCs each containing a 0 bit. (This method of eanbling and disabling CRC error reporting must be verified and is for further study.)

2.6.1 *Computation of the CRC4 bits*

The CRC4 bits C1, C2, C3 and C4 are computed from the whole 64 kbit/s channel, for a block made of two frames: one even frame (containing the FAW) followed by one odd frame (not containing the FAW). The CRC4 block size is then 160 octets, i.e. 1280 bits, and the computation is performed 50 times per second.

2.6.1.1 *Multiplication division process*

A given C1-C4 word located in block N is the remainder after multiplication by x^4 and then division (modulo 2) by the generator polynominal $x^4 + x + 1$ of the polynomial representation of block $(N-1)$.

When representing contents of a block as a polynominal the first bit in the block should be taken as being the most significant bit. Similarly C1 is defined to be the most significant bit of the remainder and C4 the least significant bit of the remainder.

This process can be realized with a four-stage register and two exclusive-ors.

2.6.1.2 *Encoding procedure*

i) The CRC bit positions in the odd frame are initially set at zero, i.e. C1 = C2 = C3 = C4 = 0.

ii) The block is then acted upon by the multiplication-division process referred to above in § 2.6.1.1.

iii) The remainder resulting from the multiplication-division process is stored ready for insertion into the respective CRC locations of the next odd frame.

Note – These CRC bits do not affect the computation of the CRC bits of the next block, since the corresponding locations are set at zero before the computation.

2.6.1.3 *Decoding procedure*

i) A received block is acted upon by the multiplication division process, referred above in § 2.6.1.1, after having its CRC bits extracted and replaced by zeros.

ii) The remainder resulting from this multiplication-division process is then stored and subsequently compared on a bit-by-bit basis with the CRC bits received in the next block.

iii) If the decoded calculated remainder exactly corresponds to the CRC bits sent from the encoder, it is assumed that the checked block is error-free.

2.6.2 *Consequent actions*

2.6.2.1 *Action on bit E*

Bit E of block N is set to 1 in the transmitting direction of bits C1-C4 detected in the most recent block in the opposite direction have been found in error (at least one bit in error). In the opposite case, it is at zero.

2.6.2.2 *Monitoring for incorrect frame alignment*

In case of a long simulation of the FAW, the CRC4 information can be used to re-invite a search for frame alignment. For such a purpose, it is possible to count the number of blocks CRC in error within 2 s (100 blocks) and to compare this number with 89. If the number of CRC blocks in error is greater than or equal to 89, a search for frame alignment should be re-initiated.

These values of 100 and 89 have been chosen in order that:

- for a random transmission error rate of 10^{-3}, the probability of incorrectly re-initiating a search for frame alignment because of 89 or more blocks in error, be less than 10^{-4};
- in case of simulation of frame alignment, the probability of not re-initiating a search of frame alignment after a 2 s period be less than 2.5%.

2.6.2.3 *Monitoring for error performance*

The quality of the 64 kbit/s connection can be monitored by counting the number of CRC blocks in error within a period of one second (50 blocks). For instance, a good evaluation of the proportion of seconds without errors as defined in Recommendation G.821 can be provided.

For information purposes, the following proportions of CRC block in error can be computed for randomly distributed errors of error rate Pe, as shown in Table 1/H.221.

TABLE 1/H.221

Pe	10^{-3}	10^{-4}	10^{-5}	10^{-6}	10^{-7}
Proportion of CRC blocks in error	70%	12%	1.2%	0.12%	0.012%

By counting the received E bits, it is possible to monitor the quality of the connection in the opposite direction.

3 Bit-rate allocation signal (BAS) and switching between configurations

The bit-rate allocation signal (BAS) occupies bits 9-16 of the service channel in every frame. An eight bit BAS code (b_0, b_1, b_2, b_3, b_4, b_5, b_6, b_7) is complemented by eight error correction bits (p_0, p_1, p_2, p_3, p_4, p_5, p_6, p_7) to implement a (16,8) double error correcting code. This error correcting code is obtained by shortening the (17,9) cyclic code with generator polynomial:

$$g(x) = x^8 + x^7 + x^6 + x^4 + x^2 + x + 1$$

The error correction bits are calculated as coefficients of the remainder polynomial in the following equation:

$$\begin{aligned} & p_0x^7 + p_1x^6 + p_2x^5 + p_3x^4 + p_4x^3 + p_5x^2 + p_6x + p_7 \\ = {} & RES_{g(x)}\,[b_0x^{15} + b_1x^{14} + b_2x^{13} + b_3x^{12} + b_4x^{11} + b_5x^{10} + b_6x^9 + b_7x^8] \end{aligned}$$

where $RES_{g(x)}\,[f(x)]$ represents the residue obtained by dividing $f(x)$ by $g(x)$.

The BAS code is sent in the even-numbered frame, while the associated error correction bits are sent in the subsequent odd-numbered frame. Each bit of BAS code or the error correction is transmitted in the order shown in Table 2/H.221, to avoid emulation of the frame alignment signal.

TABLE 2/H.221

Bit position	Even frame	Odd frame
9	b_0	P_2
10	b_3	P_1
11	b_2	P_0
12	b_1	P_4
13	b_5	P_3
14	b_4	P_5
15	b_6	P_6
16	b_7	P_7

The decoded BAS value is valid if:

- the receiver is in frame and multiframe alignment, and
- the FAS in the same submultiframe was received with 2 or fewer bits in error.

Otherwise, the decoded BAS value is ignored. When the receiver actually loses frame alignment, it should undo any changes caused by the three previously decoded BAS values and revert to the state determined by the fourth previously decoded BAS value.

The encoding of BAS is made in accordance with the attribute method.

The first three bits (b_0, b_1, b_2) represent the attribute number, which describes the general command or capability, and the next five bits (b_3, b_4, b_5, b_6, b_7) identify the specific command or capability. The following attributes are defined:

000	Audio coding command: values defined in Annex A
001	Transfer rate command: values defined in Annex B
010	Video and other command: values defined in Annex D
011	Data command: values defined in Annex E
100	Terminal capability: values defined in Annex C

Annex A defined a number of modes, according to the audio coding type and bit rate. Since a validated value of BAS command code applies to the next submultiframe, a change in configuration can occur every 20 ms. This applies equally to the use of video and data command BAS, controlling sub-modes of various configurations of the remaining capacity.

When the incoming bit A (see § 2.3) is set to 1, the distant rceiver is not in multiframe alignment and will not immediately validate a new BAS value.

Capability BAS require a response from the distant terminal and should not be sent unnecessarily when the incoming signal is unframed.

See Recommendations G.725 for further information on signalling procedures.

4 Application channel (AC)

It occupies bits 17-80 of the service channel in each frame, providing a user available bit rate of 6.4 kbit/s. According to the application, different kinds of information may be inserted herein. In particular, information concerning forward error correction or end-to-end encryption which both depend on the application, could take place in the application channel.

The AC may be used to convey a message channel conforming to the OSI protocols where appropriate. With this message channel, a transport and a session protocol may be used to control the use of audio and data channels. For example, once the command/response procedure has agreed to open a connection, if necessary, the BAS is used to adjust the capability available for data.

Examples for the use of AC are given in Appendix I.

5 Access to non-audio information within bits 1-7

Use of attribute (000) according to Annex A provides for the static or dynamic allocation of "data channels" of up to 56 kbit/s capacity; in some applications, it may be desirable to combine the application channel with the data channel in order to have a single user-data path, of capacity up to 62.4 kbit/s.

Unless BAS codes (010), (011) are used to direct otherwise, the "data channel" is treated as a single stream of non-video information; in this case access may be realised according to standardised procedures (e.g. Recommendations I.461, I.462, I.463). Data is transmitted in the order received from the data terminal equipment or data terminal adaptator.

In the presence of a non-zero video command BAS (010) the data channel is assigned to moving picture information, except that some part may be subtracted for other data purposes by application of a non-zero data command BAS (011).

ANNEX A

(to Recommendation H.221)

Attribute 000 used for BAS encoding

Attribute Bits b_0 - b_2	Attribute value Bits b_3 - b_7	Meaning	
000 Audio coding	00000	Neutralised channel (the 62.4 kbit/s user date are unused) PCM [G.711] (truncated to 7 bits)	(Note 1) (Note 2)
	S0010	A-law; data at 0 or 6.4 kbit/s	Mode OF
	S0011	μ-law; data at 0 or 6.4 kbit/s	Mode OF
	S0001	32 kbit/s ADPCM data at 24 or 30.4 kbit/s	(Note 3)
		64 kbit/s unframed mode	(Note 4)
	00100	PCM A-law	Mode 0
	00101	PCM μ-law	Mode 0
	00110	SB-ADPCM [G.722]	Mode 1 (Note 5)
	00111	0 kbit/s; data at 64 kbit/s	Mode 10
		Variable bit-rate audio coding	
	S1000	G.722 56 kbit/s; data at 0 or 6.4 kbit/s	Mode 2
	S1001	G.722 48 kbit/s; data at 8 or 14.4 kbit/s	Mode 3
	S1010 ... S1110	Reserved for audio coding at bit rates less than 48 kbit/s	(Note 6)
	S1111	0 kbit/s; data at 56 or 62.4 kbit/s	Mode 9 (Note 7)
	10000	Free	
	101xx	Free	

Note 1 – The 8th bit is fixed to 0 in the audio PCM decoder.

Note 2 – The S bit set to 1 indicates that the application channel is merged with the data channel to form a single user-data path. The method for merging the two channels is shown in Figure A-1/H.221 for the 14.4 kbit/s case.

Note 3 – The coding law and respective place of data and audio in each byte of the 64 kbit/s channel is under study.

Note 4 – Attribute values 001xx imply the switching to an unframed mode. In the receive direction, reverting to a framed mode can only be achieved by recovering frame and multiframe alignment, which might take up to 2 multiframes (i.e. 320 ms).

Note 5 – The allocation of bits in each byte of the 64 kbit/s channel is as follows:

Audio bit-rate	1	2	3	4	5	6	7	8
64 kbit/s	H	H	L	L	L	L	L	L
56 kbit/s	H	H	L	L	L	L	L	S
48 kbit/s	H	H	L	L	L	L	D	S

S Service channel
D Data channel
H High band audio
B Low band audio

Bit-rates of 56 and 48 kbit/s are respectively modes 2 and 3 of Recommendation G.722.

Note 6 – Audio coding bit-rates 40-32-24-16-8 kbit/s require further study.

Note 7 – The whole of the 56 (or 62.4) kbit/s is used for data and the audio channel is not available.

Bit number 7	Bit number 8	Octet number
1	FAS	1
2		2
.		.
.		.
8		8
9	BAS	9
.		.
.		.
16		16
17	18	17
19	20	18
.	.	.
.	.	.
143	144	80

FIGURE A-1/H.221

Bit number for a merged 14.4 kbit/s data

ANNEX B

(to Recommendation H.221)

Attribute 001 used for BAS encoding

Attribute Bits b_0 - b_2	Attribute value Bits b_3 - b_7	Meaning
001 Transfert rate	00000	64 kbit/s
	00001	64 kbit/s (audio) + 64 kbit/s (data/video)
	00010	64 kbit/s (audio) + 64 kbit/s (data/video) treated as a single 128 kbit/s channel
	01010 01011	384 kbit/s: 64 (audio) + 320 (video) 64 (audio) + 256 (video) + 64 (data)
	01100 01101	768 kbit/s: 64 (audio) + 704 (video) 64 (audio) + 640 (video) + 64 (data)
	01110 01111	1152 kbit/s: 64 (audio) + 1088 (video) 64 (audio) + 1024 (video) + 64 (data)
	10000 10001	1536 kbit/s: 64 (audio) + 1472 (video) 64 (audio) + 1408 (video) + 64 (data)
	10010 10011	1920 kbit/s: 64 (audio) + 1856 (video) 64 (audio) + 1792 (video) + 64 (data)

ANNEX C

(to Recommendation H.221)

Attribute 100 used for BAS encoding

Attribute Bits b_0 - b_2	Attribute value Bits b_3 - b_7	Meaning
100 Terminal capability	00000	Neutral (Note 1)
	00001	G.725 Type 0 – A-law (Note 2)
	00010	G.725 Type 0 – μ-law
	00011	G.725 Type 1 – G.722
	00100	G.725 Type 2 – G.722 + data
	00101 ... 00110	Reserved for audio capabilities
	00111	Reserved for national use
	01000	Non-standard video capability (Note 3)
	01001 ... 01110	Reserved for video capabilities
	01111	Reserved for national use
	10000	Non-standard system capability (Note 3)
	10001	2B transfer rate capability (Note 4)
	10010	3B transfer rate capability (Note 4)
	10011	4B transfer rate capability (Note 4)
	10100	5B transfer rate capability (Note 4)
	10101	6B transfer rate capability (Note 4)
	10110	Reserved for transfer rate capability
	10111	Reserved for national use
	11000	300 bit/s data capability (Note 5)
	11001	1200 bit/s data capability (Note 5)
	11010	2400 bit/s data capability (Note 5)
	11011	4800 bit/s data capability (Note 5)
	11100	6400 bit/s data capability (Note 5)
	11101	8000 bit/s data capability (Note 5)
	11110	9600 bit/s data capability (Note 5)
	11111	14 400 bit/s data capability (Note 5)

Note 1 – The neutral value indicates no change in the current capabilities of the terminal.

Note 2 – Types 0, 1 and 2 are defined according to Recommendation G.725 § 2.

– Type 0 terminal can work in mode 0 (PCM) only.

– Type 1 terminal preferably works in mode 1 (G.722) but is able to work in mode 0.

– Type 2 terminal preferably works in mode 2 (G.722 + H.221) but is able to work in modes 1 and 0.

Note 3 – If sent (additional), an improved video algorithm decoding or whole system capability is indicated; it is specified elsewhere.

Note 4 – A capability to use several B channels implies the capability to use fewer channels.

Note 5 – A data capability specifies only one rate; if multiple rates are possible the capabilities are sent individually.

ANNEX D

(to Recommendation H.221)

Attribute 010 used for BAS encoding

Attribute Bits b_0 - b_2	Attribute value Bits b_3 - b_7	Meaning
010 Video and other command	00000	No video; video switched OFF
	00001	Standard video for m × 64 kbit/s
	00010	Video ON, using improved algorithm
	00011	Standard video to Recommendation H.261
	...	
	11111	Transfer to non-standard system mode

ANNEX E

(to Recommendation H.221)

Attribute 011 used for BAS encoding

Attribute Bits b_0-b_2	Attribute value Bits b_3-b_7	Meaning
011 Data command	00000	No data; data switched OFF
	00001	300 bit/s in AC assigned to data (bit 8 of last three octets in each frame)
	00010	1200 bit/s in AC assigned to data (bit 8 of last 12 octets in each frame)
	00011	4800 bit/s in AC assigned to data (bit 8 of last 48 octets in each frame)
	00100	6400 bit/s in AC assigned to data (whole of AC)
	00101	8000 bit/s assigned to data (bit 7)
	00110	9600 bit/s assigned to data (bit 7 + bit 8 of last 16 octets in each frame)
	00111	14.4 kbit/s assigned to data (bit 7 + AC)
	...	
	10000 to 10111	Reserved for communicating the status of the data terminal equipment interfaces
	...	
	11111	Variable rate data; data switched ON (Note)

Note – When video is switched on, the entire variable data capacity is used for video.

APPENDIX I

(to Recommendation H.221)

Examples for the use of the application channel

I.1 *Binary information*

Each bit of the application channel may be used to convey the information of a 100 kbit/s channel, repeated 100 times per second. If odd and even frames are identified, each bit may carry the 150 Hz bit/s channels. If multiframing is used, each bit may carry the information of 16 channels, each at 6.25 bit/s.

An example of this kind of information is, in teleconference, the use of a bit to synchronize the encoder clock on the receive clock, or to indicate the microphone number, or to signal the use of the grahics mode, etc.

I.2 *Synchronous message-type channel*

As each bit of the application channel represents a bit-rate of 100 bit/s, any synchronous channel working at $n \times 100$ bit/s may be inserted in the application channel. An example is, in videoconference, the message channel at 4 kbit/s which is used for multipoint management.

Another possibility is the insertion of data channels at one of the bit rates defined in Recommendation X.1, according to Recommendations X.30/I.461: "Support of X.21 and X.21*bis* based DTEs by an ISDN". The present frame structure is consistent with the Recommendations X.30/I.461 frame structure in a double way:

- it has the same length (80 bits by bearer channel at 8 kbit/s);
- it needs 63 bits per frame (17 bits are used for framing information not to be transmitted), which fits into the 64 bits available in this frame structure.

I.3 *Asynchronous message-type channel*

In case of asynchronous terminals, Recommendation X.1 bit-rates are relevant, too. The applicable standard is that specified in [1]. This standard also uses the same 80-bit frame structure as Recommendations X.30/I.461 mentioned above. The application channel will therefore allow adoption of this ECMA standard if needed.

I.4 *Error correction and encryption*

When needed, forward error correction and encryption information may be transmitted in the application channel. The bit-rate and the protocol to be used will depend on the application.

Reference

[1] ECMA-TAxx *Bit-rate adaption for the support of synchronous and asynchronous terminal equipment using the V-Series interfaces on a PSTN.*

Recommendation H.222

FRAME STRUCTURE FOR 384-1920 kbit/s CHANNELS IN AUDIOVISUAL TELESERVICES

(Melbourne, 1988)

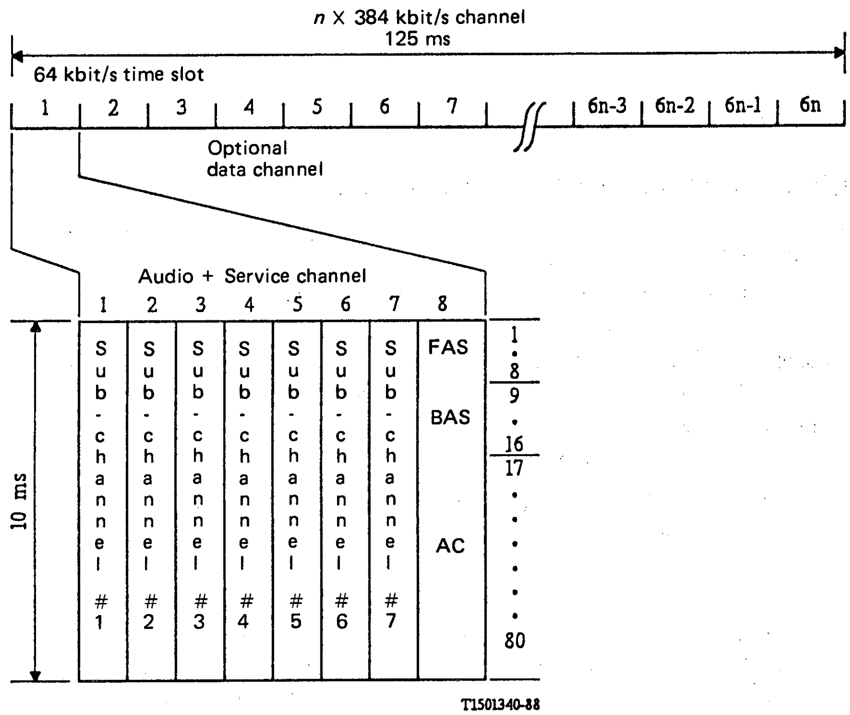
1 Scope

This Recommendation provides a mechanism to multiplex multimedia signals such as audio, video, data, control and indication, etc., for audiovisual teleservices using an $n \times 384$ kbit/s ($n = 1\text{-}5$) channel.

2 Basic structure

The multiplex structure is based upon multiple octets transmitted at 8 kHz as in Recommendation I.431.

As $n \times 384$ kbit/s channel consists of $6 \times n$ time slots of 64 kbit/s (see Figure 1/H.222). The first 64 kbit/s time slot has a frame structure conforming to Recommendation H.221, containing frame alignment signal (FAS), bit rate allocation signal (BAS) and application channel (AC).



FAS Frame alignment signal (Note)
BAS Bit rate allocation signal
AC Application channel

Note – The block termed as FAS also contains information other than for frame alignment purposes.

FIGURE 1/H.222

Frame structure for $n \times$ 384 kbit/s audiovisual teleservices

3 BAS codes

Particular codes for allocating audio, video and data signals in an $n \times 384$ kbit/s channel are given in Annex B to Recommendation H.221 for Attribute "001".

4 Data transmission

A 64 kbit/s data channel can be allocated to the fourth time slot in the $n \times 384$ kbit/s channel if controlled by the corresponding BAS code.

Provision of more than one 64 kbit/s data channels is under study.

5 Bit assignment in application channel

Application channel conveys control and indication signals, message channel, etc., for audiovisual teleservices using $n \times 384$ kbit/s transmission. Bit assignment is under study.

CODEC FOR AUDIOVISUAL SERVICES AT n × 384 kbit/s

(Melbourne, 1988)

The CCITT,

considering

(a) that there is significant customer demand for videoconference service;

(b) that circuits to meet this demand can be provided by digital transmission using the H_0 rate or its multiples up to the primary rate;

(c) that ISDNs are likely to be available in some countries that provide a switched transmission service at the H_0 rate;

(d) that the existence of different digital hierarchies and different television standards in different parts of the world complicates the problems of specifying coding and transmission standards for international connections;

(e) that videophone services are likely to appear using basic ISDN access and that some means of interconnection of videophone and videoconference terminals should be possible;

(f) that Recommendation H.120 for videoconferencing using primary digital group transmission was the first in an evolving series of Recommendations,

appreciating

that advances are being made in research and development of video coding and bit rate reduction techniques which will lead to further Recommendations for videophone and videoconferencing at multiples of 64 kbit/s during subsequent Study Periods, so that this may be considered as the second in the evolving series of Recommendations.

and noting

that it is the basic objective of CCITT to recommend unique solutions for international connections,

recommends

that in addition to those codecs complying with Recommendation H.120, codecs having signal processing and interface characteristics described below should be used for international videoconference connections.

Note 1 – Codecs of this type are also suitable for some television services where full broadcast quality is not required.

Note 2 – Equipment for transcoding from and to codecs according to Recommendation H.120 is under study.

Note 3 – It is recognised that the objective is to provide interworking between $n \times 384$ kbit/s codecs and $m \times 64$ kbit/s codecs as defined in the H-Series Recommendations. Interworking will be on the basis of $m \times 64$ kbit/s, where the values of m are under study.

1 Scope

This Recommendation describes the coding and decoding methods for audiovisual services at the rates of $n \times 384$ kbit/s, where n is 1 to 5. Possible extension of this scope to meet the objective in Note 3 above is under study.

2 Brief specification

An outline block diagram of the codec is given in Figure 1/H.261.

2.1 *Video input and output*

To permit a single Recommendation to cover use in and between 625 and 525-line regions, pictures are coded in one common intermediate format. The standards of the input and output television signals, which may, for example, be composite or component, analogue or digital and the methods of performing any necessary conversion to and from the intermediate coding format are not subject to recommendation.

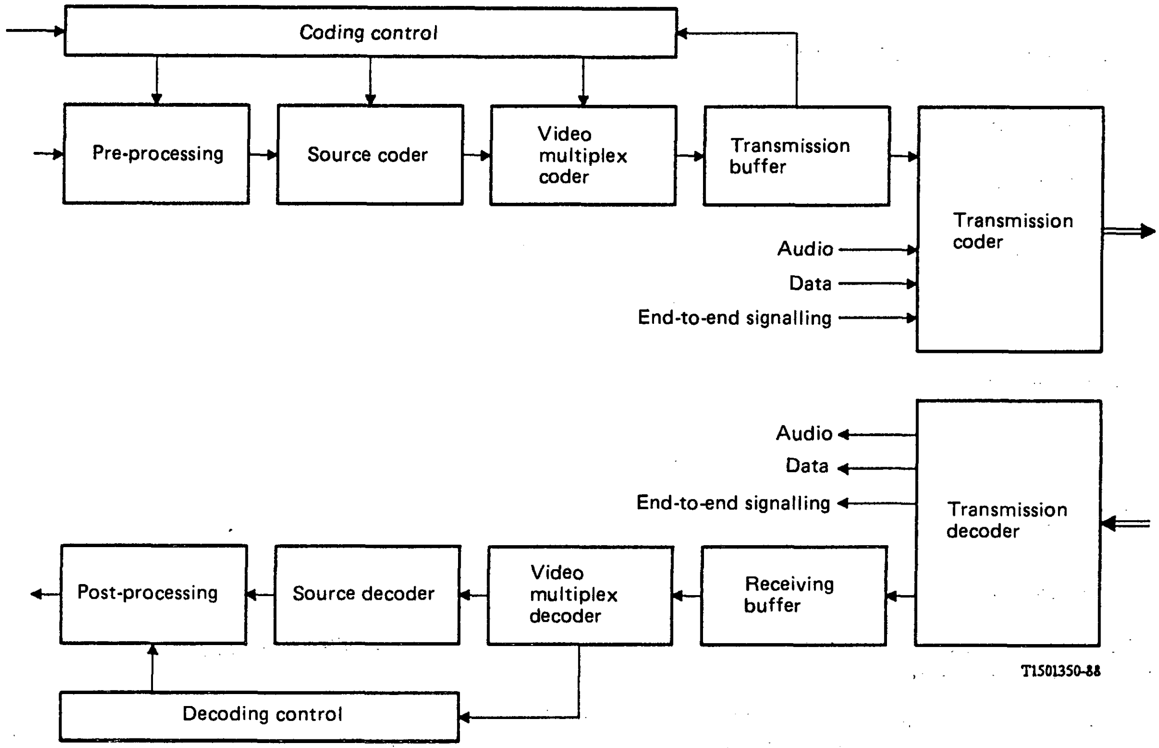


FIGURE 1/H.261

Outline block diagram of the codec

2.2 *Digital output and input*

Digital access at the primary rate of 1544 or 2048 kbit/s is with vacated time slots in accordance with Recommendation I.431.

Interfaces using ISDN basic accesses are under study (see Recommendation I.420).

2.3 *Sampling frequency*

Pictures are sampled at an integer multiple of the video line rate. This sampling clock and the digital network clock are asynchronous.

2.4 *Source coding algorithm*

A hybrid of inter-picture prediction to utilize temporal redundancy and transform coding of the remaining signal to reduce spatial redundancy is adopted. The decoder has motion compensation capability, allowing optional incorporation of this technique in the coder.

2.5 *Audio channel*

Audio is coded according to mode 2 of Recommendation G.722. This is combined with control and indication information and conveyed in one 64 kbit/s time slot which conforms to Recommendation H.221.

2.6 *Data channels*

Recommendation H.221 permits part of the 64 kbit/s time slot carrying the audio to be used for auxiliary data transmission.

Additionally, one of the time slots normally used for video may be reassigned as a 64 kbit/s data channel. The possibility of further such channels is under study.

2.7 *Symmetry of transmission*

The codec may be used for bidirectional or unidirectional audiovisual communication.

2.8 *Error handling*

Under study.

2.9 *Propagation delay*

Under study.

2.10 *Additional facilities*

Under study.

3 Source coder

3.1 *Source format*

The source coder operates on non-interlaced pictures occurring 30000/1001 (approximately 29.97) times per second. The tolerance on picture frequency is ± 50 ppm.

Pictures are coded as luminance and two colour difference components (Y, C_R et C_B). These components and the codes representing their sampled values are as defined in CCIR Recommendation 601.

Black = 16

White = 235

Zero colour difference = 128

Peak colour difference = 16 and 240.

These values are nominal ones and the coding algorithm functions with input values of 0 through to 255.

For coding, the luminance sampling structure is 288 lines per picture, 352 pels per line in an orthogonal arrangement. Sampling of each of the two colour difference components is at 144 lines, 176 pels per line, orthogonal. Colour difference samples are sited such that their block boundaries coincide with luminance block boundaries as shown in Figure 2/H.261. The picture area covered by these numbers of pels and lines has an aspect ratio of 4 : 3 and corresponds to the active portion of the local standard video input.

Note – The number of pels per line is compatible with sampling the active portions of the luminance and colour difference signals from 525 to 625-line sources at 6.75 and 3.375 MHz, respectively. These frequencies have a simple relationship to those in CCIR Recommendation 601.

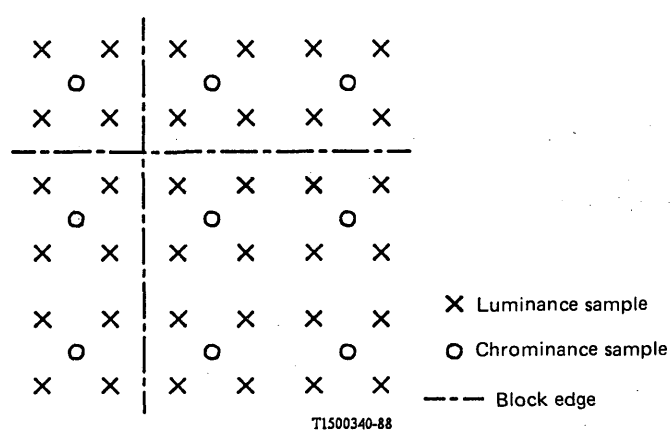
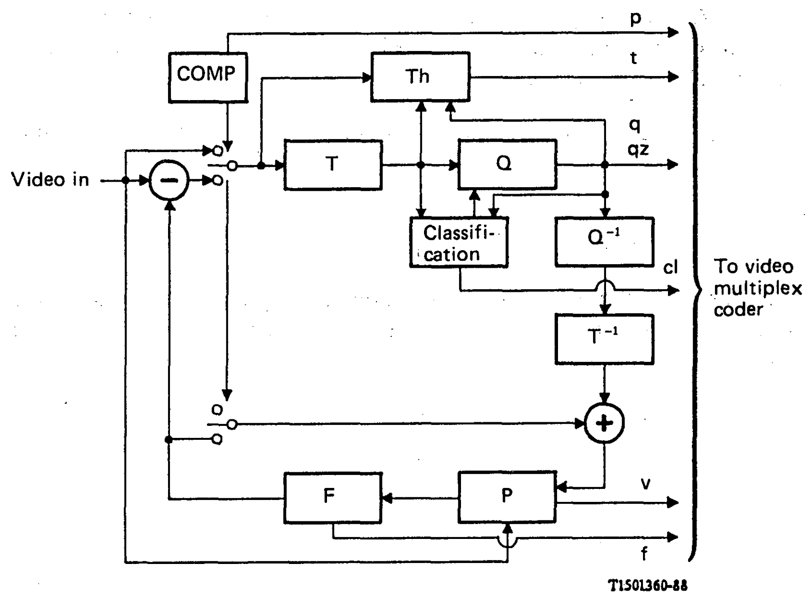


FIGURE 2/H.261

Positioning of luminance and chrominance samples

3.2 *Video source coding algorithm*

The video coding algorithm is shown in generalised form in Figure 3/H.261. The main elements are prediction, block transformation, quantization and classification.



COMP Comparator for intra/inter
Th Threshold
T Transform
Q Quantizer
P Picture memory with motion compensated variable delay
F Loop filter
p Flag for intra/inter
t Flag for transmitted or not
q Quantizing index for transform coefficients
qz Quantizer indication
v Motion vector
cl Classification index
f Switching on/off of the loop filter

FIGURE 3/H.261

Video coding algorithm

The prediction error (INTER mode) or the input picture (INTRA mode) is subdivided into 8 pel by 8 line blocks which are segmented as transmitted or non-transmitted. The criteria for choice of mode and transmitting a block are not subject to recommendation and may be varied dynamically as part of the data rate control strategy. Transmitted blocks are transformed and resulting coefficients are quantized and variable length coded.

3.2.1 *Prediction*

The prediction is inter-picture and may be augmented by motion compensation (§ 3.2.2) and a spatial filter (§ 3.2.3).

3.2.2 *Motion compensation*

Motion compensation is optional in the encoder. The decoder will accept one vector for each block of 8 pels by 8 lines. The range of permitted vectors is under study.

A positive value of the horizontal or vertical component of the motion vector signifies that the prediction is formed from pels in the previous picture which are spatially to the right or below the pels being predicted.

Motion vectors are restricted such that all pels referenced by them are within the coded picture area.

3.2.3 *Loop filter*

The prediction process may be modified by a two-dimensional spatial filter which operates on pels within a predicted block.

The filter is separable into one dimensional hironzontal and vertical functions. Both are non-recursive with coefficients of 1/4, 1/2, 1/4. At block edges, where one of the taps would fall outside the block, the peripheral pel is used for two taps. Full arithmetic precision is retained with rounding to 8 bit integer values at the 2-D filter output. Values whose fractional part is one half are rounded up.

The filter may be switched on or off on a block by block basis. The method of signalling is under study.

3.2.4 *Transformer*

Transmitted blocks are coded with a separable 2-dimensional discrete cosine transform of size 8 by 8. The input to the forward transform and output from the inverse transform have 9 bits. The arithmetic procedures for computing the transforms are under study.

Note – The output from the forward and input to the inverse are likely to be 12 bits.

3.2.5 *Quantization*

The number of quantizers, their characteristics and their assignment are under study.

3.2.6 *Clipping*

To prevent quantization distortion of transform coefficient amplitudes causing arithmetic overflow in the encoder and decoder loops, clipping functions are inserted. In addition to those in the inverse transform, a clipping function is applied at both encoder and decoder to the reconstructed picture which is formed by summing the prediction and the prediction error as modified by the coding process. This clipper operates on resulting pel values less than 0 or greater than 255, changing them to 0 and 255 respectively.

3.3 *Data rate control*

Sections where parameters which may be varied to control the rate of generation of coded video data include processing prior to the source coder, the quantizer, block significance criterion and temporal subsampling. The proportions of such measures in the overall control strategy are not subject to recommendation.

When invoked, temporal subsampling is performed by discarding complete pictures. Interpolated pictures are not placed in the picture memory.

3.4 *Forced updating*

This function is achieved by forcing the use of the INTRA mode of the coding algorithm. The update interval and pattern are under study.

4 Video multiplex coder

4.1 *Data structure*

Note 1 – Unless specified otherwise, the most significant bit is transmitted first.

Note 2 – Unless specified otherwise, bit 1 is transmitted first.

Note 3 – Unless specified otherwise, all unused or spare bits are set to '1'.

4.2 *Video multiplex arrangement*

4.2.1 *Picture header*

The structure of the picture header is shown in Figure 4/H.261. Picture headers for dropped pictures are not transmitted.

PSC	TR	TYPE1	PEI	PARITY	PSPARE

FIGURE 4/H.261

Structure of picture header

4.2.1.1 *Picture start code (PSC)*

A unique word of 21 bits which cannot be emulated by error-free data. Its value is under study.

4.2.1.2 *Temporal reference (TR)*

A five bit number derived using modulo-32 counting of pictures at 29.97 Hz.

4.2.1.3 *Type information (TYPE1)*

Information about the complete picture:

Bit 1	Split screen indicator. '0' off, '1' on.
Bit 2	Document camera. '0' off, '1' on.
Bit 3	Freeze picture release. Under study.
Bit 4	Under study. Possible uses include signalling of the use of motion compensation and the method of switching the loop filter.
Bit 5	Number of classes. '0' one, '1' four.
Bits 6 to 12	Under study.

4.2.1.4 *Extra insertion information (PEI)*

Two bits which signal the presence of the following two optional data fields.

4.2.1.5 *Parity information (PARITY)*

For optional use and present only if the first PEI bit is set to '1'. Eight parity bits each representing odd parity of the aggregate of the corresponding bit planes of the locally decoded PCM values of Y, C_R and C_B in the previous picture period.

4.2.1.6 *Spare information (PSPARE)*

Sixteen bits are present when the second PEI bit is set to '1'. The use of these bits is under study.

4.2.2 *Group of blocks header*

A group of blocks consists of 2k lines of 44 luminance blocks each, k lines of 22 C_R blocks and k lines of 22 C_B blocks. The value of k is under study.

The structure of the group of blocks header is shown in Figure 5/H.261. All GOB headers are transmitted except those in dropped pictures.

GBSC	GN	TYPE2	QUANT1	GEI	GGMV	GSPARE

FIGURE 5/H.261

Structure of group of blocks header

4.2.2.1 *Group of blocks start code (GBSC)*

A word of 16 bits, 0000 0000 0000 0001.

4.2.2.2 *Group number (GN)*

An m bit number indicating the vertical position of the group of blocks. The value of m is the smallest integer greater than or equal to $\log_2 (18/k)$. GN is 1 at the top of the picture.

Note – GBSC plus the following GN is not emulated by error-free video data.

4.2.2.3 *Type information (TYPE2)*

TYPE2 is p bits which give information about all the transmitted blocks in a group of blocks. The value of p is under study.

Bit 1 When set to '1' indicates that all the transmitted blocks in the GOB are coded in INTRA mode and without block addressing data.

Bits 2 to p Spare, under study.

4.2.2.4 *Quantizer information (QUANT1)*

A j bit code word which indicates the blocks in the group of blocks where QUANT2 code words are present. These blocks, their code words and the value of j are under study.

Whether QUANT1 is in the GOB header or the picture header is under study.

4.2.2.5 *Extra insertion information (GEI)*

Under study.

4.2.2.6 *Group of blocks global motion vector (GGMV)*

Under study.

4.2.2.7 *Spare information (GSPARE)*

Under study.

4.2.3 *Block data alignment*

The structure of the data for n transmitted blocks is shown in Figure 6/H.261. The values of n and the order are under study. Elements are omitted when not required.

BA	TYPE3	QUANT2	CLASS	MVD	TCOEFF1	EOB	--	TCOEFFn	EOB

FIGURE 6/H.261

Data structure of transmitted block

4.2.3.1 *Block address (BA)*

A variable length code word indicating the position of n blocks within a group of blocks. VLC code words using a combination of relative and absolute addressing are under study.

The transmission order and addressing of blocks are under study.

When bit 1 of TYPE2 is '1', BA is not included and up to 132k blocks beginning with and continuing in the above transmission order are transmitted before the next GOB header.

4.2.3.2 *Block type information (TYPE3)*

Variable length code words indicating the types of blocks and which data elements are present. Block types and VLC code words are under study.

4.2.3.3 *Quantizer (QUANT2)*

A code word of up to q bits signifying the table(s) used to quantize transform coefficients. The value of q and the code words are under study. QUANT2 is present in the first transmitted block after the position indicated by QUANT1.

4.2.3.4 *Classification index (CLASS)*

CLASS is present if bit 5 of TYPE1 is set to '1' and indicates which of the four available transmission sequence orders is used for luminance block coefficients. If bit 5 of TYPE1 is set to '0' then luminance block coefficients are transmitted in the default sequence order.

Chrominance block coefficients are transmitted in one sequence order.

The CLASS code words and sequence orders are under study.

4.2.3.5 *Motion vector data (MVD)*

Calculation of the vector data is under study.

When the vector data is zero, this is signalled by TYPE3 and MVD is not present.

When the vector data is non-zero, MVD is present consisting of a variable length code word for the horizontal component followed by a variable length code word for the vertical component.

Variable length coding of the vector components is under study.

4.2.3.6 *Transform coefficients (TCOEFF)*

The quantized transform coefficients are sequentially transmitted according to the sequence defined by CLASS. The DC component is always first. Coefficients after the last non-zero one are not transmitted.

The coding method and tables are under study.

4.2.3.7 *End of block marker (EOB)*

Use of and code word for EOB are under study. An EOB without any transform coefficients for a block is allowed.

4.3 *Multipoint considerations*

4.3.1 *Freeze picture request*

Causes the decoder to freeze its received picture until a picture freeze release signal is received. The transmission method for this control signal is under study.

4.3.2 *Fast update request*

Causes the encoder to empty its transmission buffer and encode its next picture in INTRA mode with coding parameters such as to avoid buffer overflow. The transmission method for this control signal is under study.

4.3.3 *Data continuity*

The prototocl adopted for ensuring continuity of data channels in a switched multipoint connection is handled by the message channel. Under study.

5 **Vide data buffering**

The size of the transmission buffer at the encoder and its relationship to the transmittion rate are under study.

Transmission buffer overflow and underflow are not permitted. Measures to prevent underflow are under study.

6 Transmission coder

6.1 *Bit rate*

The net bit rate including audio and optional data channels is an integer multiple of 384 kbit/s up to and including 1920 kbit/s.

The source and stability of the encoder output clock are under study.

6.2 *Video clock justification*

Video clock justification is not provided.

6.3 *Frame structure*

6.3.1 *Frame structure for 384-1920 kbit/s channels*

The frame structure is defined in Recommendation H.222.

6.3.2 *Bit assignment in application channel*

Under study.

6.3.3 *Time slot positioning*

According to Recommendation I.431.

6.4 *Audio coding*

Recommendation G.722 56/48 kbit/s audio, 0/8 kbit/s data and 8 kbit/s service channel in the first time slot.

The delay of the encoded audio relative to the encoded video at the channel output is under study.

6.5 *Data transmission*

One or more time slots may be allocated as data channels of 64 kbit/s each. The first channel uses the fourth time slot.

Positioning of the other channels, and possible restrictions on availability at lower overall bit rates are under study. The BAS codes used to signal that these data channels are in use are specified in Recommendation H.221.

6.6 *Error handling*

Under study.

6.7 *Encryption*

Under study.

6.8 *Bit sequence independence restrictions*

Under stydy.

6.9 *Network interface*

Access at the primary rate is with vacated time slots as per Recommendation I.431.

For 1544 kbit/s interfaces the default H_0 channel is time slots 1 to 6.

For 2048 kbit/s interfaces the default H_0 channel is time slots 1-2-3-17-18-19.

Interfaces using ISDN basic accesses are under study (see Recommendation I.420).

PART II

Series J Recommendations

SOUND-PROGRAMME AND TELEVISION TRANSMISSIONS

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SECTION 1

GENERAL RECOMMENDATIONS CONCERNING SOUND-PROGRAMME TRANSMISSIONS

Recommendation J.11

HYPOTHETICAL REFERENCE CIRCUITS FOR SOUND-PROGRAMME TRANSMISSIONS[1), 2), 3)]

(Geneva, 1972; amended at Geneva, 1976, and at Melbourne, 1988)

Terrestrial systems and systems in the fixed-satellite service

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,

unanimously recommends

(1) that the main features of 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:

- the overall length between audio points (B and C) is 2500 km,
- two intermediate audio points (M and M') which divide the circuit into three sections of equal lengths,
- the three sections which are lined up individually and then inter-connected without any form of overall adjustment or correction;

(2) that the main features of the hypothetical reference circuit for sound-programme transmissions over a system in the fixed-satellite service (shown in Figure 2/J.11) should be:

- one link: earth station – satellite – earth station,
- one pair of modulation and demodulation equipments for translation from baseband to radio frequency, and from radio frequency to baseband, respectively.

1) This Recommendation corresponds to CCIR Recommendation 502.

2) The hypothetical reference circuits defined in this Recommendation should apply for both analogue and digital systems.

3) For maintenance purposes there may be a need to define other circuits of which an illustration is shown in Annex A of this Recommendation.

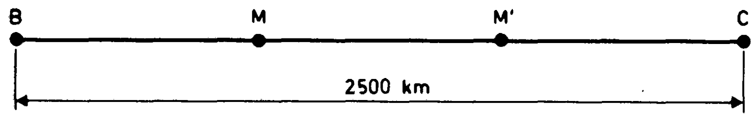


FIGURE 1/J.11

Hypothetical reference circuit for sound-programme transmissions over a terrestrial system

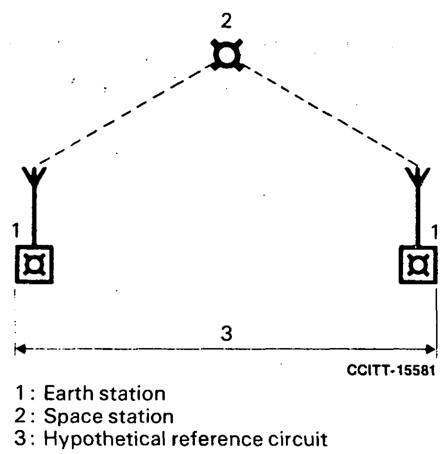


FIGURE 2/J.11

Hypothetical reference circuit for sound-programme transmissions over a system in the Fixed-Satellite Service

ANNEX A

(to Recommendation J.11)

Illustration of an international sound-programme connection

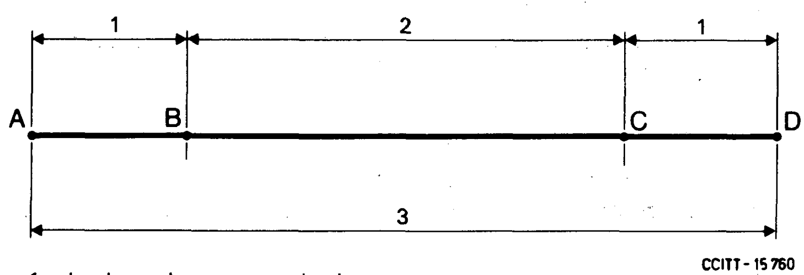
Figure A-1/J.11 illustrates a typical international sound-programme connection in which:

- 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 above or to those defined for maintenance purposes in [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 circuits
2 — international sound-programme circuit
3 — international sound-programme connection

FIGURE A-1/J.11

An international sound-programme connection

Reference

[1] *Maintenance; international sound-programme and television transmission circuits.* Recommendations of the N Series. Fascicle IV.3.

Recommendation J.12

TYPES OF SOUND-PROGRAMME CIRCUITS ESTABLISHED OVER THE INTERNATIONAL TELEPHONE NETWORK

(former Recommendation J.11; amended at Geneva, 1972 and 1980, and at Melbourne, 1988)

The CCITT recognizes the types of sound-programme circuits defined below.

Note – For the purposes of this Recommendation and other Recommendations in the Series J, 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 D.180 [1]) is given under each type of equipment in the following paragraphs.

1 15 kHz-type 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 D.180 [1].

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 equipment is specified in Recommendation J.31, for analogue transmission and in Recommendations J.41, G.735 and G.737 for digital transmission.

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. This type of circuit corresponds to the "wideband circuit" referred to in Recommendation D.180 [1]. 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.

Note – Recommendations J.22 and J.32 are reproduced in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985.

3 Narrow bandwidth sound-programme circuit (7 and 5 kHz-type sound-programme circuit)

These types of circuits are recommended:

- for setting up a large number of temporary sound-programme circuits for the transmission of commentaries and reports on events of large interest (e.g. sporting events); and
- for permanent sound-programme circuits which are used primarily for speech transmission or as connection between studio outputs and long-, medium- or short-wave broadcast-transmitter inputs.

The performance characteristics of narrow bandwidth sound-programme circuits are defined in Recommendation J.23, and as suitable equipment for 7 kHz-type circuit is specified in Recommendation J.34, for analogue transmission.

Note – These types of circuits fall within the category of "medium-band circuits" referred to in Recommendation D.180 [1] for tariff purposes.

4 Use of ordinary telephone circuits

For this type of transmission of special programmes such as speech, some operational aspects are given in Recommendation N.15 [2].

References

[1] CCITT Recommendation *Occasional provision of circuits for international sound- and television-programme transmissions*, Vol. II, Rec. D.180.

[2] CCITT Recommendation *Maximum permissible power during an international sound-programme transmission*, Vol. IV, Rec. N.15.

Recommendation J.13

DEFINITIONS FOR INTERNATIONAL SOUND-PROGRAMME CIRCUITS

(former Recommendation J.12; amended at Geneva, 1972 and 1980)

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.).

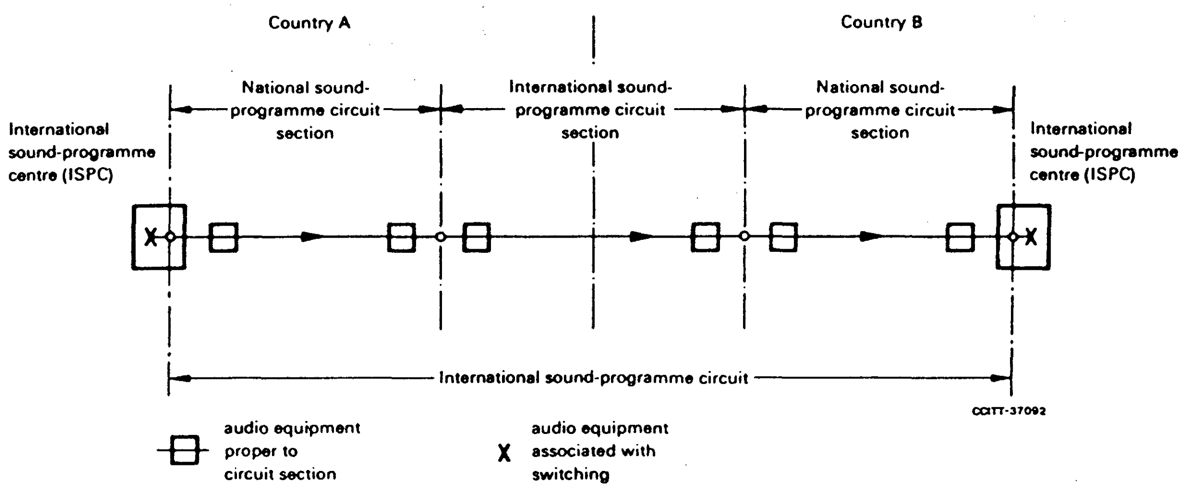


FIGURE 1/J.13

An international sound-programme circuit composed of two national and one international sound-programme circuit-section

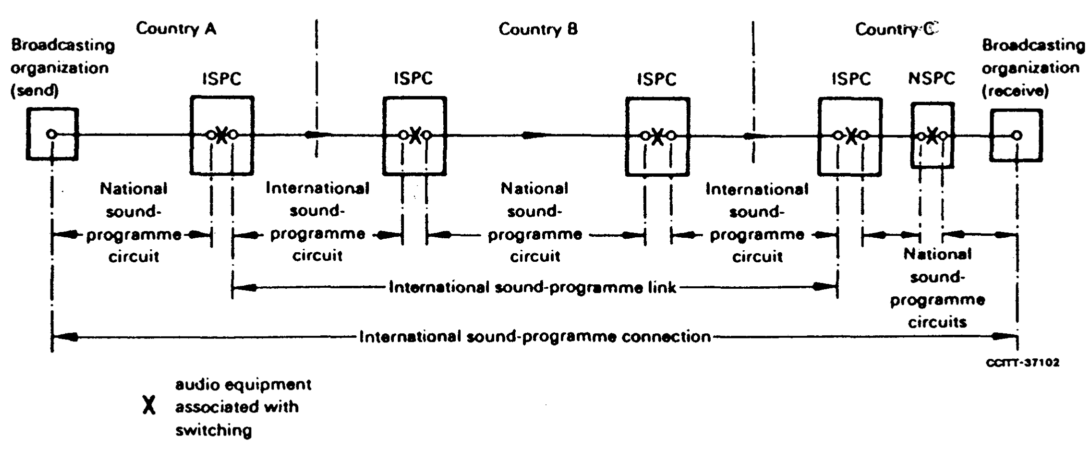
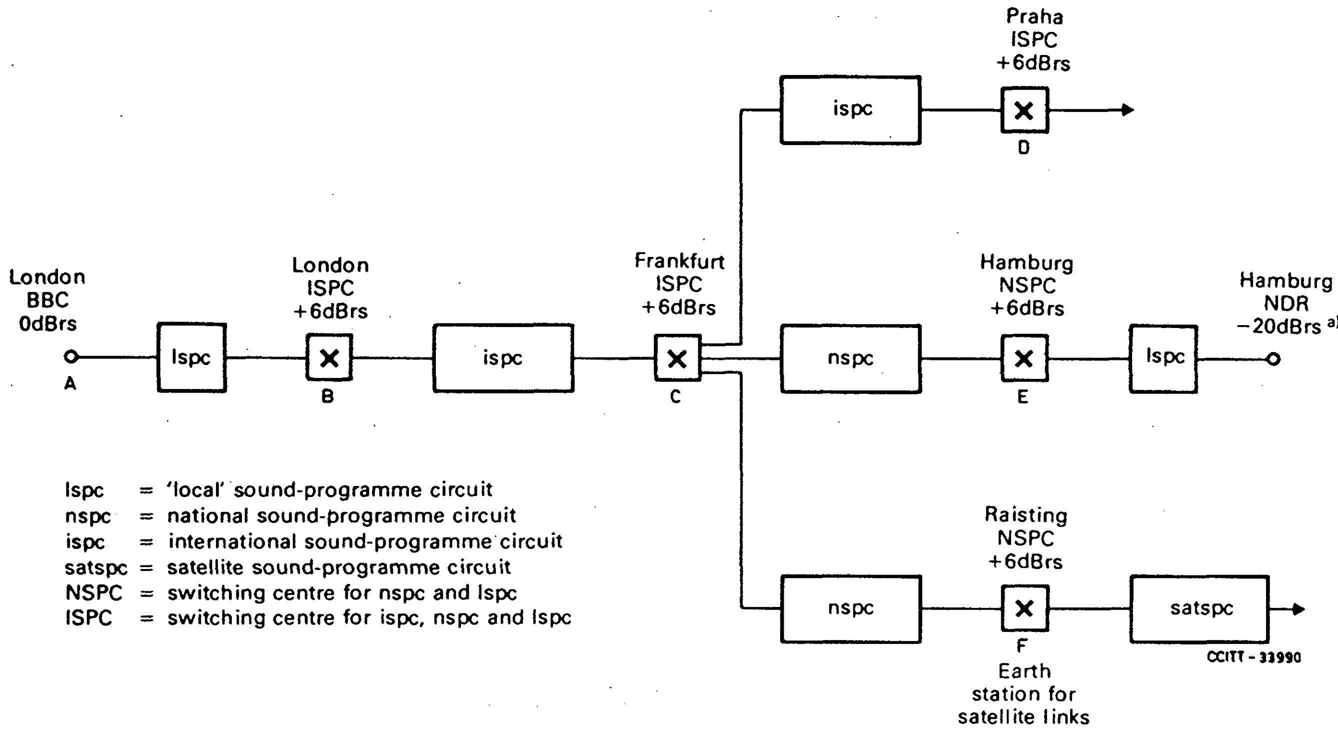


FIGURE 2/J.13

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



Note – Maximum level of sound programme signals: +9dBm0s (this means +9dBms at a 0dBrs relative level point and +15dBms at a +6dBrs relative level point respectively). The value of +9dBms corresponds to a peak voltage of 3.1 V which is the maximum value of a sine-wave signal of 2.2 V r.m.s.

a) Other values can be chosen by the relevant Administration on a national basis.

FIGURE 3/J.13

Diagram of an international sound-programme circuit

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.

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 [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.

Reference

[1] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151, § 1, Note 1.

RELATIVE LEVELS AND IMPEDANCES ON AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

(former Recommendation J.13; amended at Geneva, 1972, 1976 and 1980, and at Melbourne, 1988)

1 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 the reference frequency 0.8 or 1 kHz, the absolute voltage level at the output of each sound-programme circuit (Points B, C, D ... F of Figure 3/J.13) should be +6 dB (i.e. 1.55 volts r.m.s.). Therefore these points have to be regarded as relative level points of +6 dBrs according to Recommendations J.21, J.22 and J.23.

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 connection. At a point of zero relative levels, 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).

In Recommendation 645, CCIR has defined test signals to be used on international sound-programme connections based on existing CCITT Recommendations.

2 Diagram of signal levels on an international sound-programme connection

All signal levels 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 will not 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 ... F of Figure 3/J.13).

Considering that rare excursions of the permitted maximum signal level may occur, and that adjustment errors and maintenance tolerance have to be taken into account, sound-programme circuits need a definite overload margin. The amount of this margin is still under study.

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 Recommendations J.31 for 15 kHz-type circuits, J.34 for 7 kHz-type circuits and in the Annex to Recommendation J.22 for 10 kHz-type circuits.

3 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.

3.1 dBm0 [1)]

The absolute signal power level, in decibels, referred to a point of zero relative level.

3.2 dBr [1)]

The relative power level, in decibels.

3.3 dBm0s

The absolute signal power level, in decibels, referred to a point of zero relative sound-programme level.

3.4 dBrs

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.)

Note – The use of level definitions is given in CCIR Recommendation 574.

Recommendation J.15

LINING-UP AND MONITORING AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

(former Recommendation J.14; amended at Geneva, 1972 and 1980, and at Melbourne, 1988)

For the alignment of international sound-programme connections the CCIR, in Recommendation 661, recommends a *three-level test signal.*

This test signal is based on the test signal definitions given in CCIR Recommendation 645 and specifies a test signal which should be used on sound-programme circuits generally. A common alignment procedure for peak programme meters and VU-meters using the three-level test signal can be found in Annex I of CCIR Recommendation 645. From this information it can be seen what indicators will be produced by the three-level test signal on the different types of peak programme meters and volume meters.

1) These symbols traditionally relate to telephony relative levels.

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 (see references [1] to [8]).

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. The unified use of the peak indicator specified in reference [9] is under study.

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 transmissions

Type of instrument	Rectifier characteristic (Note 1)	Time to reach 99% of final reading (milliseconds)	Integration time (milliseconds) (Note 2)	Time to return to zero (value and definition)
(1) Vu meter (United States of America)	1.0 to 1.4	300	165 (approx.)	Equal to the integration time
(2) Vu meter (France)	1.0 to 1.4	300 ± 10%	207 ± 30	300 ms ± 10% from the reference deviation
(3) Peak programme meter, used by the Netherlands	1	Not specified	10 ms for −1 dB 5 ms for −2 dB 0-4 ms for −15 dB	0 to −20 dB: 1-5 s 0 to −40 dB: 2-5 s
(4) Programme level meter (Italy)	1	Approx. 20 ms	Approx. 1.5 ms	Approx. 1.5 s from 100% to 10% of the reading in the steady state
(5) Peak indicator for sound-programme transmissions used by the British Broadcasting Corporation (BBC peak programme meter)	1		10 (Note 3)	3 s for the pointer to fall 26 dB
(6) Maximum amplitude indicator used by the Federal Republic of Germany (type U 21)	1	Around 80	5 (approx.)	1 or 2 s from 100% to 10% of the reading in the steady state
(7) 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	10 ± 5 60 ± 10	For both types: 1.5 to 2 s from the 0 dB point which is at 30% of the length of the operational section of the scale
(8) E.B.U. standard peak programme meter (Note 4)	1	–	10	2.8 s for the pointer to fall 24 dB

Note 1 – The number given in the column is the index n in the formula $V_{(output)} = [V_{(input)}]^n$ applicable for each half-cycle.

Note 2 – 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 3 – The figure of 4 ms, 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 arbitray, quasi-d.c. test signal, but the integration time, as defined in Note 2, is about 20% greater at the higher meter readings.

Note 4 – This meter is intented specifically for use in monitoring sound signals transmitted internationally, and therefore incorporates a scale conforming to CCITT Recommendation N.15 [5], calibrated in dB from −12 to +12 relative to a level marked "TEST" corresponding to 0 dBm at a zero relative level point. In addition to the normal mode of opeation having the characteristics shown above, the meter may be operated temporarily in a "slow" mode faciliting the comparison of observations made at widely separate points. The peak values indicated in this mode have no absolute significance, and may only be used for such comparisons.

References

[1] CCITT Recommendation *Limits for the lining-up of international sound-programme links and connections*, Vol. IV, Rec. N.10.

[2] CCITT Recommendation *Essential transmission performance objectives for international sound-programme centres (ISPC)*, Vol. IV, Rec. N.11.

[3] CCITT Recommendation *Measurements to be made during the line-up period that precedes a sound-programme transmission*, Vol. IV, Rec. N.12.

[4] CCITT Recommendation *Measurements to be made by the broadcasting organizations during the preparatory period*, Vol. IV, Rec. N.13.

[5] CCITT Recommendation *Maximum permissible power during an international sound-programme transmission*, Vol. IV, Rec. N.15.

[6] CCITT Recommendation *Identification signal*, Vol. IV, Rec. N.16.

[7] CCITT Recommendation *Monitoring the transmission*, Vol. IV, Rec. N.17.

[8] CCITT Recommendation *Monitoring for charging purposes, releasing*, Vol. IV, Rec. N.18.

[9] IEC Publication 268-10A.

Recommendation J.16

MEASUREMENT OF WEIGHTED NOISE IN SOUND-PROGRAMME CIRCUITS

(Geneva, 1972; amended at Geneva, 1976 and 1980)

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 on 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 be used which conforms to the characteristics laid down in CCIR Recommendation 468 which is reproduced at the end of this Recommendation.

Annex A gives symbols and definitions used in noise measurements.

ANNEX A

(to Recommendation J.16)

Symbols and definitions used in noise measurements

A clear distinction should be made between measurements performed with equipment conforming to the Recommendation cited in [1] and those with equipment conforming to CCIR Recommendation 468.

It is recommended that the definitions and symbols in Table A-1/J.16 be used.

TABLE A-1/J.16

Definitions and symbols for the specification of noise measured on sound-programme circuits

Definitions	Symbols
Unweighted noise level, measured with a quasi-peak measuring instrument complying with CCIR Recommendation 468 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 and referred to a point of zero relative sound-programme level	dBq0ps

Reference

[1] CCITT Recommendation *Psophometers (apparatus for the objective measurement of circuit noise)*, Green Book, Vol. V, Rec. P.53, Part B, ITU, Geneva, 1973.

CCIR RECOMMENDATION 468-4*

MEASUREMENT OF AUDIO-FREQUENCY NOISE VOLTAGE LEVEL IN SOUND BROADCASTING

(Question 50/10)

(1970 - 1974 - 1978 - 1982 - 1986)

The CCIR,

CONSIDERING

(a) that it is desirable to standardize the methods of measurement of audio-frequency noise in broadcasting, in sound-recording systems and on sound-programme circuits;

(b) that such measurements of noise should provide satisfactory agreement with subjective assessments,

UNANIMOUSLY RECOMMENDS

that the noise voltage level be measured in a quasi-peak and weighted manner, using the measurement system defined below:

1. Weighting network

The nominal response curve of the weighting network is given in Fig. 1 b which is the theoretical response of the passive network shown in Fig. 1 a. Table I gives the values of this response at various frequencies.

* This Recommendation should be brought to the attention of the CMTT.

The permissible differences between this nominal curve and the response curve of the measuring equipment, comprising the amplifier and the network, are shown in the last column of Table I and in Fig. 2.

Note 1. – When a weighting filter conforming to § 1 is used to measure audio-frequency noise, the measuring device should be a quasi-peak meter conforming to § 2. Indeed, the use of any other meter (e.g. an r.m.s. meter) for such a measurement would lead to figures for the signal-to-noise ratio that are not directly comparable with those obtained by using the characteristics that are described in the present Recommendation.

Note 2. – The whole instrument is calibrated at 1 kHz (see § 2.6).

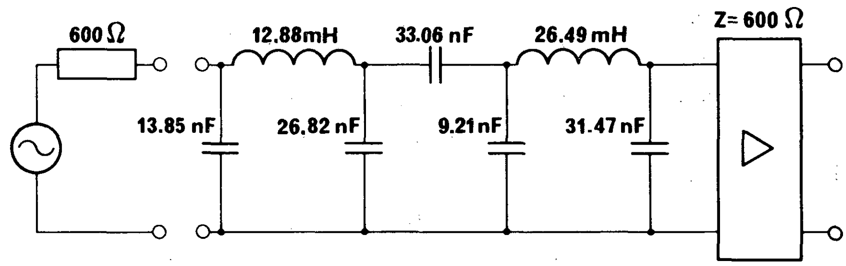


FIGURE 1a — *Weighting network, simple form*

(A constant-resistance realization is described in Annex I)

(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.)
(The difference between the responses at 1000 Hz and 6300 Hz may be adjusted more precisely by a small adjustment of the 33.06 nF capacitor, or by a different approach using an active filter [CCIR, 1982-86a].)

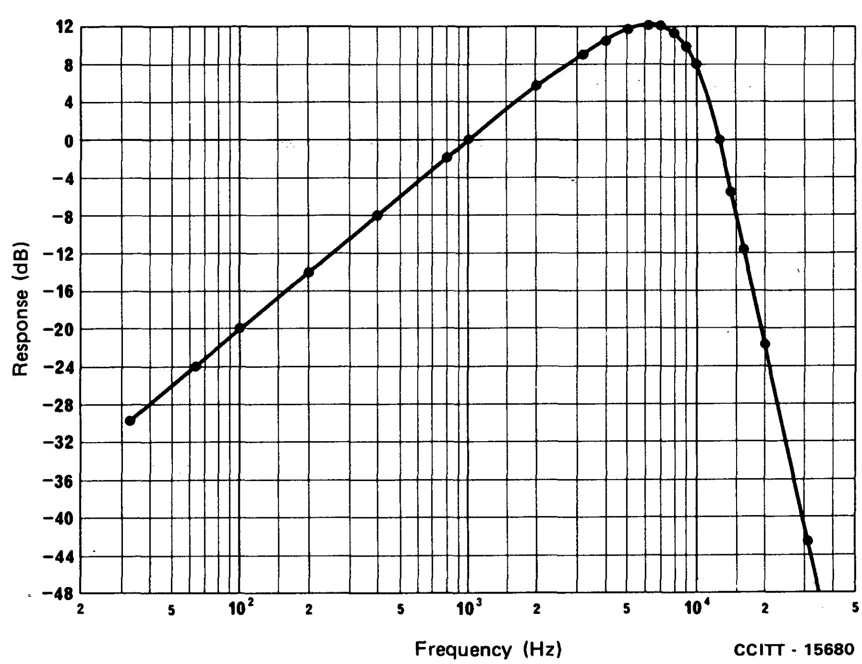


FIGURE 1*b* – *Frequency response of the weighting network shown in Fig. 1 a*

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
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	± 1.6 (1)
20 000	−22.2	± 2.0
31 500	−42.7	+ 2.8 (1) − ∞

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.

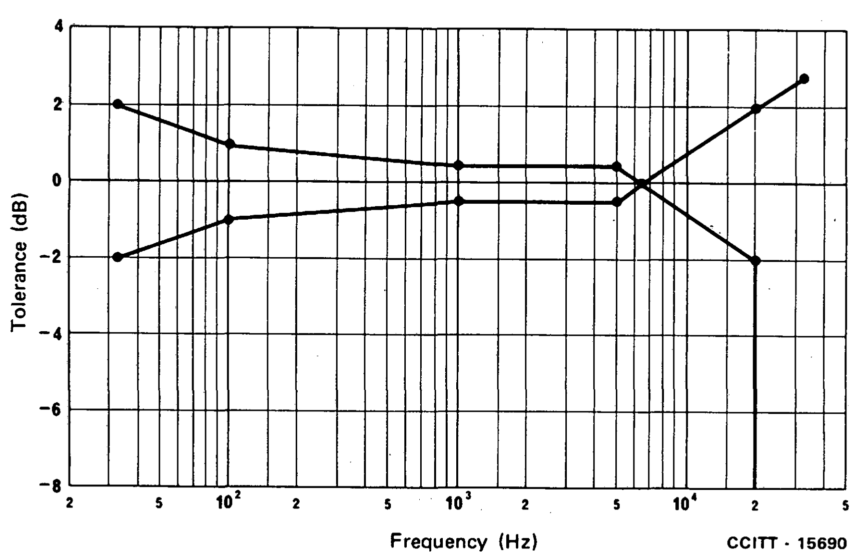


FIGURE 2 — *Maximum tolerances for the frequency response of the weighting network and the amplifier*

2. Characteristics of the measuring device

A quasi-peak value method of measurement shall be used. The required dynamic performance of the measuring set may be realized in a variety of ways (see Note). It is defined in the following sections. Tests of the measuring equipment, except those for § 2.4, should be made through the weighting network.

Note – After full wave rectification of the input signal, a possible arrangement would consist of two peak rectifier circuits of different time constants connected in tandem [CCIR, 1974-78].

2.1 *Dynamic characteristic in response to single tone-bursts*

Method of measurement

Single bursts of 5 kHz tone are applied to the input at an amplitude such that the steady signal would give a reading of 80% of full scale. The burst should start at the zero-crossing of the 5 kHz tone and should consist of an integral number of full periods. The limits of reading corresponding to each duration of tone burst are given in Table II.

The tests should be performed both without adjustment of the attenuators, the readings being observed directly from the instrument scale, and also with the attenuators adjusted for each burst duration to maintain the reading as nearly constant at 80% of full scale as the attenuator steps will permit.

TABLE II

Burst duration (ms)		1 [1]	2	5	10	20	50	100	200
Amplitude reference steady signal reading	(%)	17.0	26.6	40	48	52	59	68	80
	(dB)	−15.4	−11.5	−8.0	−6.4	−5.7	−4.6	−3.3	−1.9
Limiting values:									
– lower limit	(%)	13.5	22.4	34	41	44	50	58	68
	(dB)	−17.4	−13.0	−9.3	−7.7	−7.1	−6.0	−4.7	−3.3
– upper limit	(%)	21.4	31.6	46	55	60	68	78	92
	(dB)	−13.4	−10.0	−6.6	−5.2	−4.4	−3.3	−2.2	−0.7

[1] The Administration of the USSR intends to use burst durations ⩾ 5 ms.

2.2 *Dynamic characteristic in response to repetitive tone-bursts*

Method of measurement

A series of 5 ms bursts of 5 kHz tone starting at zero-crossing is applied to the input at an amplitude such that the steady signal would give a reading of 80% of full scale. The limits of the reading corresponding to each repetition frequency are given in Table III.

The tests should be performed without adjustment of the attenuators but the characteristic should be within tolerance on all ranges.

TABLE III

Number of bursts per second		2	10	100
Amplitude reference steady signal reading	(%) (dB)	48 −6.4	77 −2.3	97 −0.25
Limiting values:				
– lower limit	(%) (dB)	43 −7.3	72 −2.9	94 −0.5
– upper limit	(%) (dB)	53 −5.5	82 −1.7	100 −0.0

2.3 *Overload characteristics*

The overload capacity of the measuring set should be more than 20 dB with respect to the maximum indication of the scale at all settings of the attenuators. The term "overload capacity" refers both to absence of clipping in linear stages and to retention of the law of any logarithmic or similar stage which may be incorporated.

Method of measurement

Isolated 5 kHz tone-bursts of 0.6 ms duration starting at zero-crossing are applied to the input at an amplitude giving full scale reading using the most sensitive range of the instrument. The amplitude of the tone-bursts is decreased in steps by a total of 20 dB while the readings are observed to check that they decrease by corresponding steps within an overall tolerance of ± 1 dB. The test is repeated for each range.

2.4 *Reversibility error*

The difference in reading when the polarity of an asymmetrical signal is reversed shall not be greater than 0.5 dB.

Method of measurement

1 ms rectangular d.c. pulses with a pulse repetition rate of 100 pulses per second or less are applied to the input in the unweighted mode, at an amplitude giving an indication of 80% of full scale. The polarity of the input signal is reversed and the difference in indication is noted.

2.5 *Overswing*

The reading device shall be free from excessive overswing.

Method of measurement

1 kHz tone is applied to the input at an amplitude giving a steady reading of 0.775 V or 0 dB (see § 2.6). When this signal is suddenly applied there shall be less than 0.3 dB momentary excess reading.

2.6 *Calibration*

The instrument shall be calibrated such that a steady input signal of 1 kHz sine-wave at 0.775 V r.m.s., having less than 1% total harmonic distortion, shall give a reading of 0.775 V, 0 dB. The scale should have a calibrated range of at least 20 dB with the indication corresponding to 0.775 V (or 0 dB) between 2 and 10 dB below full scale.

2.7 *Input impedance*

The instrument should have an input impedance $\geqslant$ 20 kΩ and if an input termination is provided then this should be 600 Ω ± 1%.

3. Presentation of results

Noise voltage levels measured according to this Recommendation are expressed in units of dBqps.

Note 1 – If, for technical reasons, it is desirable to measure unweighted noise, the method described in Annex II should be used.

Note 2 – The influence of the weighting network on readings obtained with different spectra of random noise is discussed in Report 496.

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[1978-82]: 10/9 (EBU); 10/31 (L M Ericsson); 10/38 (OIRT); 10/225 (German Democratic Republic).

ANNEX I

CONSTANT RESISTANCE REALIZATION OF WEIGHTING NETWORK

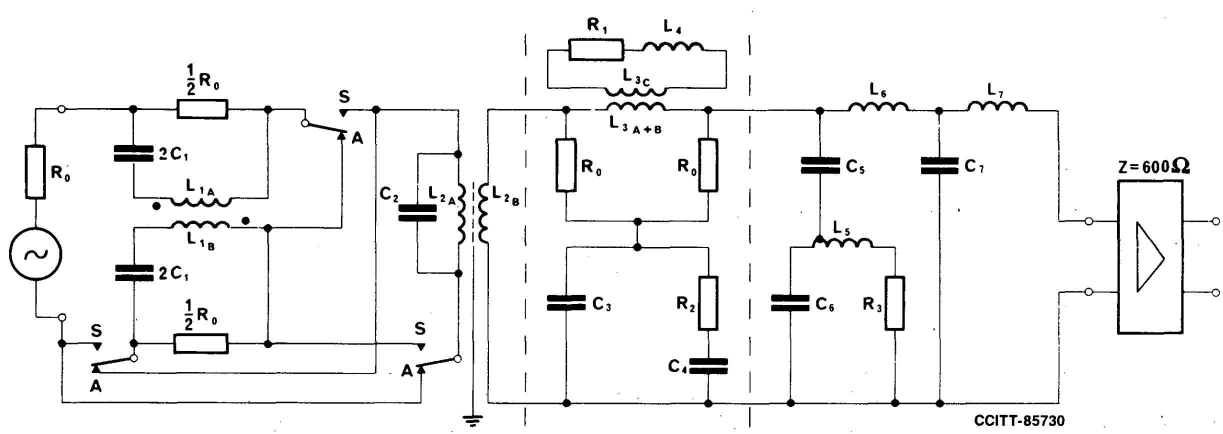

FIGURE 3

Constant resistance realization of weighting network

R (Ω)	C (nF)		L (mH)
R_0: 600	$2C_1$: 83.7	L_1:	12.70 (for both windings in series)
½ R_0: 300	C_2: 35.28	L_2:	15.06 (for each of two windings separated by electrostatic shield)
R_1: 912	C_3: 38.4	L_{3A+B}:	16.73 (two equal windings in series)
R_2: 3340	C_4: 7.99	L_{3C}:	4.18 (one winding, turns half L_{3A+B}, can have large d.c. resistance, absorbed in R_3)
R_3: 941	C_5: 23.8	L_4:	20.1 (can have large d.c. resistance, absorbed in R_3)
	C_6: 13.94	L_5:	31.5 (with tap 20.1 at 0.798 of total turns)
	C_7: 35.4		
A: unbalanced		L_6:	13.29
S: balanced		L_7:	8.00

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AUSTRALIAN BROADCASTING COMMISSION Engineering Development Report No. 106 – Constant resistance realization of CCIR noise weighting network, Recommendation 468.

ANNEX II

UNWEIGHTED MEASUREMENT

It is recognized that unweighted measurements outside the scope of this Recommendation may be required for specific purposes. A standard response for unweighted measurements is included here for guidance.

Frequency response

The frequency response shall be within the limits given in Fig. 4.

This response serves to standardize the measurement and ensure consistent readings of noise distributed across the useful spectrum. When out-of-band signals, e.g. carrier leaks, are present at a sufficient amplitude, they may produce readings that are inconsistent between measuring equipments whose responses are different but still fall within the tolerance template of Fig. 4.

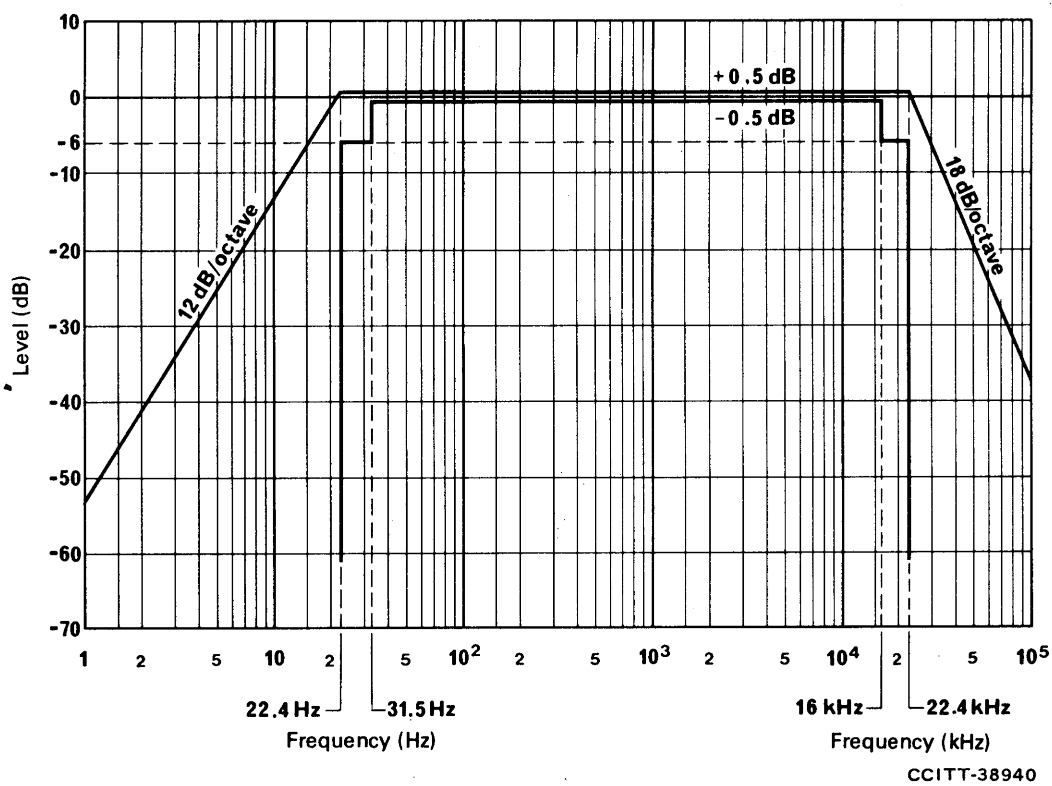

FIGURE 4

BIBLIOGRAPHY

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PRE-EMPHASIS USED ON SOUND-PROGRAMME CIRCUITS

(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 further 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. Some values are given in Table 1/J.17.

TABLE 1/J.17

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

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 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.31, J.34, J.41).

Recommendation J.18

CROSSTALK IN SOUND-PROGRAMME CIRCUITS SET UP ON CARRIER SYSTEMS

(Geneva, 1972; amended at Geneva, 1980)

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 of the Series J Recommendations, Section 3); the limits are such that at least the objectives for intelligible crosstalk ratio between *telephone* circuits (generally 65 dB, Recommendation G.151 [1]) 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 [2] 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 methods 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.233 [3]), 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 so that these precautions may be relaxed somewhat. This study is in progress in the CCITT with respect to 60 MHz systems.

ANNEX A

(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

References

[1] CCITT Recommendation *General performance objectives applicable to all modern international circuits and national extension circuits*, Vol. III, Rec. G.151.

[2] CCITT Recommendation *Through-connection of groups, supergroups, etc.*, Vol. III, Rec. G.242.

[3] CCITT Recommendation *Recommendations concerning translating equipments*, Vol. III, Rec. G.233.

A CONVENTIONAL TEST SIGNAL SIMULATING SOUND-PROGRAMME SIGNALS FOR MEASURING INTERFERENCE IN OTHER CHANNELS[2]

(Geneva, 1980)

The CCITT,

considering

(a) that on FDM systems non-linear crosstalk may cause mutual interference between the several types of transmission channels;

(b) that the interference depends on the total loading of the FDM system;

(c) that the interference in a channel can be measured as a noticeable deterioration of the signal-to-noise ratio;

(d) that for setting realistic performance limits of interference, a conventional test signal imitating the sound-programme channel loading is desirable,

unanimously recommends

that for simulating sound-programme signals a conventional test signal with the following parameters should be used:

(1) a uniform spectrum energizing signal covering the frequency band up to at least 15 kHz shall be shaped according to the nominal insertion loss/frequency shown in Table 1/J.19 and Figure 1/J.19;

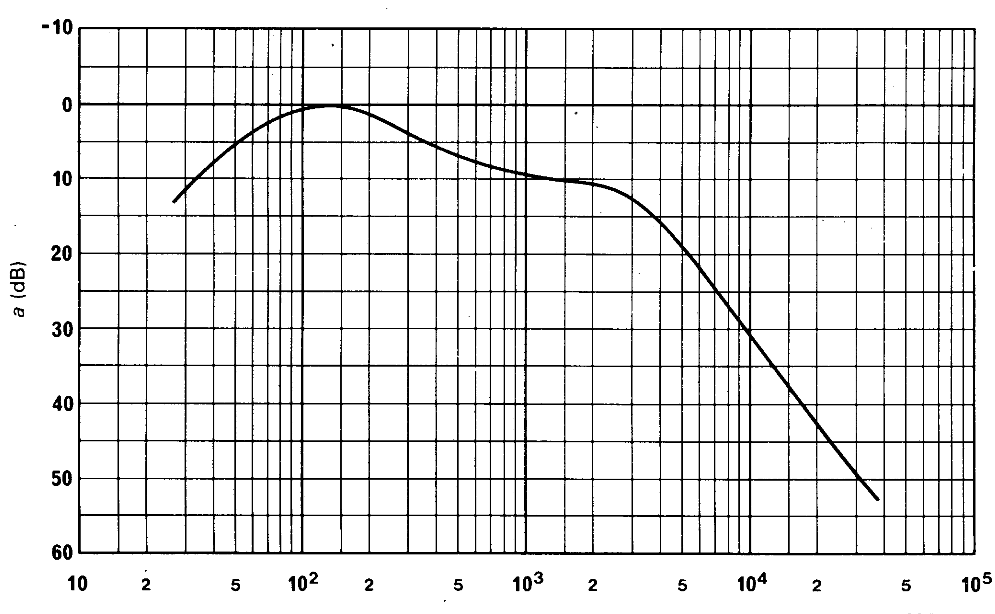

FIGURE 1/J.19
Insertion local frequency

[1] This Recommendation corresponds to CCIR Recommendation 571.

[2] For the definitions of absolute power, relative power and noise levels, see CCIR Recommendation 574.

(2) the conventional test signal can be produced from a Gaussian white noise generator associated with a shaping network conforming with Figure 2/J.19;

(3) the total test signal power applied to a sound-programme circuit under test shall be cyclically changed in level according to Table 2/J.19.

Note – This Recommendation is derived from studies given in Report 497.

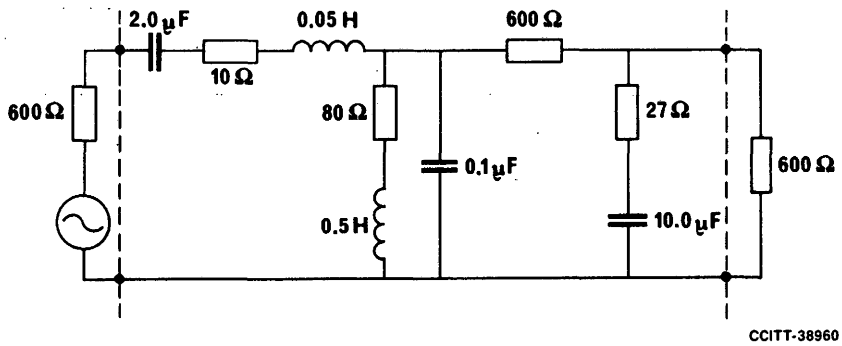

FIGURE 2/J.19

TABLE 1/J.19

Frequency (Hz)	Relative insertion-loss (dB)	Tolerance (± dB)
31.5	10.9	0.5
63	3.4	0.3
100	0.4	0.2
(122)	(0.0)	(0)
200	1.5	0.2
400	5.7	0.3
800	8.7	0.3
1 000	9.2	0.3
2 000	10.6	0.5
3 150	13.0	0.5
4 000	15.7	0.5
5 000	18.8	0.5
6 300	22.5	0.5
7 100	24.6	0.5
8 000	26.6	0.5
9 000	28.6	0.5
10 000	30.4	1.0
12 500	34.3	1.0
14 000	36.3	1.0
16 000	38.6	1.0
20 000	42.5	1.0
31 500	50.4	1.0

TABLE 2/J.19

Step	Level	Time for which signal is applied
1	−4 dBm0s	4 s
2	+3 dBm0s	2 s
3	no signal	2 s

ANNEX A

(to Recommendation J.19)

Study Group XV of the CCITT had put some questions as regards CCIR Recommendation 571 and the CMTT has worked out their answers. As those questions and the answers may be helpful for anyone who applies the conventional test signal for carrying out measurements of any kind, they are given below:

Question

a) For the measurement of crosstalk from a sound-programme circuit to a telephone circuit, could the signal described in CCIR Recommendation 571 be used, considering the different bandwidth and possible frequency shift?

Reply:

– The intelligible crosstalk ratio is based on selective measurements in the telephone circuit when the sinusoidal signals are transmitted in the sound-programme circuit within the frequency range of 0.3 to 3.4 kHz. In Recommendation J.21 a minimum ratio of 65 dB is defined.

– The unintelligible crosstalk ratio should be ascertained by measuring the increase of noise in the telephone circuit by loading the disturbing sound-programme with the simulated test signal defined in CCIR Recommendation 571. As for this increase no tolerable values are recommended up to now, the CMTT proposes such values based on a maximum noise contribution produced by interference of −65 dBm0p. Depending on the basic noise level in the telephone circuit the following increased values can be tolerated:

TABLE A-1/J.19

Basic noise level (dBm0p)	−75	−70	−65	−60	−55	−50
Tolerable increase of noise level (dB)	10.4	6.2	3	1.2	0.4	0.1

Question

b) What is the equivalent value for 65 dB ratio (given in Recommendations J.21, J.22 and J.23) using sinusoidal tones, when measuring with the recommended new test signal?

Reply:

The answer to this question is included in the proposal for the measurement of the ratio for the total crosstalk caused by intermodulation given in the answer to Question a).

Question

c) Can the signal defined in Table 2/J.19, from the point of view of the mean loading it would impose on transmission systems and in the light of Recommendations N.12 and N.13, be regarded as acceptable for unrestricted use over complete sound-programme circuits of any constitution?

Reply:

The conventional test signal simulating sound-programme signals defined in CCIR Recommendation 571/Recommendation J.19 in all aspects can be regarded as acceptable for unrestricted use over sound-programme circuits of any constitution.

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SECTION 2

PERFORMANCE CHARACTERISTICS OF SOUND-PROGRAMME CIRCUITS

Recommendation J.21 [1)]

PERFORMANCE CHARACTERISTICS OF 15 kHz TYPE SOUND-PROGRAMME CIRCUITS [2)]

Circuits for high-quality monophonic and stereophonic transmissions

(Geneva, 1972; amended at Geneva, 1976 and 1980, and at Melbourne, 1988)

The CCITT,

considering

(a) that it is necessary to set transmission standards for sound-programme circuits;

(b) that quality requirements for the hypothetical reference circuit are established for analogue sound programmes;

(c) that advantage should be taken of the technical evolution made possible by the introduction of digital techniques, particularly for mixed analogue and digital circuits,

recommends

that, with due regard to the application constraints, equipment for new circuits meet the requirements laid out below.

1 Application

The Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation J.11.

For estimation of the performance of circuits shorter or longer than the HRC, see CCIR Recommendation 605.

Note 1 – For all-digital circuits a separate Recommendation might be envisaged after further study.

Note 2 – For further work, CCIR Report 496 may be consulted. This Report also draws attention to certain differences between CCIR and OIRT Recommendations.

1) This Recommendation corresponds to CCIR Recommendation 505.

2) For the definition of absolute power, relative power and noise levels, see CCIR Recommendation 574.

2 Interface characteristics

2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 Ω resistive.

2.2 *Impedance*

System input impedance	600 Ω, balanced[3)]
System output impedance, provisionally	Low, balanced

The open-circuit output level shall not decrease more than 0.3 dB within the nominal frequency range, if the output is terminated by the specified test load.

The reactive part of the source impedance must be restricted to 100 Ω max. (provisional value) within the nominal frequency range.

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 § 3.2.2. This aspect needs further study.

2.3 *Levels*

Input maximum programme level	+9 dBm0s
Insertion gain (1 kHz at −12 dBm0)	0 dB
Adjustment error, within	± 0.5 dB
Variation over 24 hours not to exceed	± 0.5 dB
Relative level (see Recommendation J.14)	+6 dBrs

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organizations to insert additional trimming attenuators.

3 Overall performance

3.1 *Common parameters*

3.1.1 *Gain/frequency response*

Reference frequency	1 kHz (nominal value)
The response shall be measured at	−12 dBm0s

The gain/frequency response is given in Table 1/J.21.

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

TABLE 1/J.21

Frequency (kHz)	Response (dB)
$0.04 \leq f < 0.125$	+0.5 to −2.0
$0.125 \leq f \leq 10$	+0.5 to −0.5
$10 < f \leq 14$	+0.5 to −2.0
$14 < f \leq 15$	+0.5 to −3.0

3) The tolerance, permitted reactance and degree of unbalance need further study.

3.1.2 *Group delay variation*

Difference $\Delta\tau$, between the value of group delay at certain frequencies and the minimum value is given in Table 2/J.21. Between the points defined in Table 2/J.21, the tolerance limit varies linearly on a linear-delay/logarithmic frequency diagram.

TABLE 2/J.21

kHz	$\Delta\tau$ (ms)
0.04	55
0.075	24
14	8
15	12

3.1.3 *Noise*

The measurement to be made with an instrument conforming to CCIR Recommendation 468.

For radio-relay systems the requirements of Table 3/J.21 shall be met for at least 80% of the total time of any 30-day period. For 1% of the time an additional impairment of 4 dB, and for 0.1% of the time an additional impairment of 12 dB is acceptable.

Programme-modulated noise can only occur on sound-programme circuits which are equipped with compandors (for example types of circuits corresponding to Recommendation J.31).

This noise value may be measured with the aid of an auxiliary sinusoidal test signal +9 dBm0s/60 Hz which has to be suppressed by a high-pass filter ($f_0 \leqslant$ 400 Hz, $a \geqslant$ 60 dB/60 Hz) before the measuring set.

CCIR Report 493 indicates that if a compandor is used, an improved signal-to-noise ratio is necessary to avoid objectionable effects with some programme material[4].

Note – For digital systems appropriate values are under study. For further information see CCIR Report 647.

TABLE 3/J.21

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
Idle channel noise (dBq0ps), max	−42	−51
Programme-modulated noise (dBq0ps), max	−30	−39

3.1.4 *Single tone interference*

Level of any individual tone:

$$\leqslant (-73 + \psi) \text{ dBm0s}$$

where ψ is the weighting factor (positive or negative) as per CCIR Recommendation 468 at the particular frequency.

[4] Administrations are urged to supply additional information on an appropriate value.

For 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 mid-frequency, is recommended. The use of stop filters influencing frequencies below 8 kHz should be avoided.

3.1.5 *Disturbing modulation by power supply*

The level of the strongest unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains shall be less than −45 dBm0s with a test signal of 1 kHz at alignment level 0 dBm0s.

3.1.6 *Non-linear distortion*

3.1.6.1 *Harmonic distortion*

The total harmonic distortion (THD) shall be measured with the input signal at +9 dBm0s for frequencies up to 2 kHz at +6 dBm0s for frequencies above 2 kHz up to 4 kHz.

The duration for which a single-tone is to be transmitted at these levels should be restricted in accordance with Recommendations N.21 and N.23.

The THD when measured with a true-RMS meter shall not be less than the following requirements shown in Table 4/5.21.

TABLE 4/J.21

Input frequency (kHz)	Total harmonic distorsion	Second and third harmonic measured selectively
$0.04 \leqslant f < 0.125$	1% (−31 dBm0s)	0.7% (−34 dBm0s)
$0.125 \leqslant f \leqslant 2.0$	0.5% (−37 dBm0s)	0.35% (−40 dBm0s)
$2.0 < f \leqslant 4.0$	0.5% (−40 dBm0s)	0.35% (−43 dBm0s)

3.1.6.2 *Intermodulation*

With input signals at 0.8 kHz and 1.42 kHz, each at a level of +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than 0.5% (−43 dBm0s).

Note – 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.

For 15 kHz systems intended for baseband transmissions on physical circuits only, and on modulation equipment in local loops, assuming no pre-emphasis, the additional requirements of Table 5/J.21 apply.

TABLE 5/J.21

Input signals at +3 dBm0s each	Maximum difference-tone level at 1.6 kHz
5.6 kHz and 7.2 kHz	0.5% (−43 dBm0s) (second order)
4.2 kHz and 6.8 kHz	0.5% (−43 dBm0s) (third order)

3.1.6.3 *Distortion products measured by shaped noise*

Under study. See CCIR Report 640 (Kyoto, 1978).

3.1.7 *Error in reconstituted frequency* (applies only to FDM systems)

Not to be greater than 1 Hz.

Note – A maximum error of 1 Hz is in principle acceptable where there is only a single transmission path between the signal source and the listener.

Where the broadcast network can involve two or more parallel paths, e.g. the left and right channels of a stereo signal, commentary and separate sound channels, or radio broadcast from different transmitters on the same frequency, unacceptable beats may occur unless zero error can be assured. This is under study.

3.1.8 *Intelligible cross-talk ratio*

3.1.8.1 The intelligible near-end and far-end cross-talk ratios between sound-programme circuits, or from a telephone circuit (disturbing) into a sound-programme circuit (disturbed) shall be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and shall not be less than the values of Table 6/J.21.

TABLE 6/J.21

Frequency (kHz)	Crosstalk attenuation (dB)
$f = 0.04$	50
$0.04 < f < 0.05$	Oblique straight-line segment on linear-decibel and logarithmic-frequency scales
$0.05 \leq f \leq 5$	74
$5 < f < 15$	Oblique straight-line segment on linear-decibel and logarithmic-frequency scales
$f = 15$	60

3.1.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1 – It is understood that this value is defined between the relative levels applicable to telephone circuits. (Administrations are invited to submit contributions on methods for measuring this parameter.)

Note 2 – The attention of Administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or at low frequencies (e.g. below about 100 kHz) on certain coaxial cable carrier systems. If sub-standard performance is to be avoided, such systems or parts of systems, must not be used for setting up programme channels.

Note 3 – When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

Note 4 – The attention of Administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk 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).

Note 5 – The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6 – The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

3.1.9 *Amplitude linearity*

When a 1 kHz input signal is stepped from –6 dBm0s to +6 dBm0s, or vice versa, the output level shall change accordingly by 12 ± 0.5 dB.

3.2 *Additional parameters for stereophonic programme transmission*

3.2.1 The difference in gain between A and B channels shall not exceed the values in Table 7/J.21.

TABLE 7/J.21

Frequency (kHz)	Gain difference (dB)
$0.04 \leq f < 0.125$	1.5
$0.125 \leq f \leq 10$	0.8
$10 < f \leq 14$	1.5
$14 < f \leq 15$	3.0

3.2.2 The phase difference between the A and B channels shall not exceed the values in Table 8/J.21.

TABLE 8/J.21

Frequency (kHz)	Phase difference (degrees)
$f = 0.04$	30
$0.04 < f < 0.2$	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
$0.2 \leq f \leq 4$	15
$4 < f < 14$	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
$f = 14$	30
$14 < f < 15$	Oblique straight-line segment on linear-degree and logarithmic-frequency scale
$f = 15$	40

3.2.3 The cross-talk ratio between the A and B channels shall not be less than the following limits:

3.2.3.1 Intelligible cross-talk ratio, measured with sinusoidal test signal 0.04 to 15 kHz: 50 dB.

3.2.3.2 Total cross-talk ratio predominantly caused by intermodulation: 60 dB.

This value is ascertained by loading one of the two channels with the sound-programme simulating signal defined in CCIR Recommendation 571. In the other channel, the noise contribution due to intermodulation shall not be higher than −51 dBq0ps.

This leads to an increase of noise depending on the idle channel noise value. The tolerable increase is given Table 9/J.21.

TABLE 9/J.21

Idle channel noise (dBq0ps)	−60	−57	−54	−51	−48	−45	−42
Tolerable increase of noise (dB)	9.5	7	4.8	3	1.8	1.0	0.5

3.3 *Additional requirements for digital systems*

3.3.1 If a test signal is harmonically related to the sampling frequency, measuring difficulties may arise. In that case the nominally 1 kHz test signal must be offset. Recommendation O.33 recommends 1020 Hz.

3.3.2 *Unbalance of the limitation level*

The difference between those levels which lead to a limitation of the positive or negative half-wave of the test signal shall not exceed 1 dB.

3.3.3 *Intermodulation with the sampling signal*

Intermodulation products (f_d) caused by non-linearities may occur in the sound-channel when the sampling signal signal (f_o) is combined with the inband audio signals (f_i) or out-of-band interfering signals (f_a).

3.3.3.1 *Inband intermodulation*

The following combination rule applies: $f_d = f_o - nf_i$.

Only values with $n = 2$ or 3 are of importance.

The level difference between a 0 dBm0s signal (f_i) and the intermodulation products (f_d) shall not be less than 40 dB.

A restriction to the f_i/f_d values in Table 10/J.21 is sufficient.

TABLE 10/J.21

	$n = 2$		$n = 3$	
f_i (kHz)	9	13	7	11
f_d (kHz)	14	6	11	1

3.3.3.2 *Out-of-band intermodulation*

The following combination rule applies: $f_d = nf_o \pm f_a$.

Only values with $n = 1$ or 2 are of importance.

The level difference between a 0 dBm0s signal (f_a) and the intermodulation products (f_d) shall not be less than 60 dB.

A restriction to the f_a/f_d values in Table 11/J.21 is sufficient.

TABLE 11/J.21

	$n = 1$		$n = 2$	
f_a (kHz)	31	33	63	65
f_d (kHz)	1			

3.3.4 *Further parameters*

Characteristics for bit errors, clicks, jitter, etc. are under study. (See Study Programme 18A/CMTT and CCIR Report 647.)

Note – The CCIR has issued Recommendation 572 which deals with the transmission of one sound-programme associated with an analogue television signal by means of time-division multiplex in the line synchronizing pulse. The system recommended is a digital one, using pulse code modulation. A sound-programme bandwidth of 14 kHz is provided.

Biblography

CCIR Document (1978-1982): CMTT/68 (OIRT).

Recommendation J.22

PERFORMANCE CHARACTERISTICS OF 10 kHz TYPE SOUND-PROGRAMME CIRCUITS

(The text of this Recommendation can be found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

PERFORMANCE CHARACTERISTICS OF 7 kHz TYPE (NARROW-BANDWIDTH) SOUND-PROGRAMME CIRCUITS [1), 2), 3), 4)]

Circuits of medium quality for monophonic transmission

(amended at Geneva, 1980 and at Melbourne, 1988)

The CCITT,

considering

(a) that it is necessary to set transmission standards for sound-programme circuits;

(b) that quality requirements for the hypothetical reference circuit are established for analogue sound programmes;

(c) that advantage should be taken of the technical evolution made possible by the introduction of digital techniques, particularly for mixed analogue and digital circuits,

recommends

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out below.

1 Application

The Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation J.11.

For estimation of the performance of circuits shorter or longer than the HRC, see CCIR Recommendation 605.

Note 1 – For all-digital circuits a separate Recommendation might be envisaged after further study.

Note 2 – For further work, CCIR Report 496 may be consulted. This Report also draws attention to certain differences between CCIR and OIRT Recommendations.

2 Interface characteristics

2.1 *Test conditions*

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 Ω resistive.

2.2 *Impedance*

System input impedance	600 Ω, balanced [5)]
System output impedance, provisionally	Low, balanced

The open-circuit output level shall not decrease more than 0.3 dB within the nominal frequency range, if the output is terminated by the specified test load.

The reactive part of the source impedance must be restricted to 100 Ω max. (provisional value) within the nominal frequency range.

1) This Recommendation corresponds to CCIR Recommendation 503. CCIR has agreed, at its XVIth Plenary Assembly, Dubrovnik, 1986, that CCIR Recommendation 504-2 will not be published in the next CCIR book.

2) For the definition of absolute power, relative power and noise levels, see CCIR Recommendation 574.

3) Sound-programme circuits of the 5 kHz type are widely used in North America.

4) 6.4 kHz-type narrow bandwidth sound-programme circuits are still being used in some countries.

5) The tolerance, permitted reactance and degree of unbalance need further study.

2.3 *Levels*

Input maximum programme level	+9 dBm0s
Insertion gain (1 kHz at −12 dBm0)	0 dB
Adjustment error, within	± 0.5 dB
Variation over 24 hours not to exceed	± 0.5 dB
Relative level (see Recommendation J.14)	+6 dBrs

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organizations to insert additional trimming attenuators.

3 **Overall performance**

3.1 *Common parameters*

3.1.1 *Gain/frequency response*

Reference frequency	1 kHz (nominal value)
The response shall be measured at	−12 dBm0s

The gain/frequency response is given in Table 1/J.23.

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

TABLE 1/J.23

Frequency (kHz)	Response (dB)
$0.05 \leqslant f < 0.1$	+1 to −3
$0.1 \leqslant f \leqslant 6.4$	+1 to −1
$6.4 < f \leqslant 7$	+1 to −3

3.1.2 *Group delay variation*

Difference $\Delta\tau$, between the value of group delay at certain frequencies and the minimum value is given in Table 2/J.23. Between the points defined in Table 2/J.23, the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

TABLE 2/J.23

Frequency (kHz)	$\Delta\tau$ (ms)
0.05	80
0.1	20
6.4	5
7	10

3.1.3 *Noise*

The measurement is to be made with an instrument conforming to CCIR Recommendation 468.

For radio-relay systems the requirements of Table 3/J.23 shall be met for at least 80% of the total time of any 30-day period. For 1% of the time an additional impairment of 4 dB, and for 0.1% of the time an additional impairment of 12 dB is acceptable.

TABLE 3/J.23

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
Idle channel noise, max (dBq0ps)	−44	−49
Programme-modulated noise, max (dBq0ps)	−32	−37

Programme-modulated noise can only occur on sound-programme circuits which are equipped with compandors (for example types of circuits corresponding to Recommendation J.31).

This noise value may be measured with the aid of an auxiliary sinusoidal test signal +9 dBm0s/60 Hz which has to be suppressed by a high-pass filter ($f_0 \leqslant 400$ Hz, $a \geqslant 60$ dB/60 Hz) before the measuring set.

CCIR Report 493 indicates that if a compandor is used, an improved signal-to-noise is necessary to avoid objectionable effects with some programme material.[6)]

Note – For digital systems appropriate values are under study. For further information see CCIR Report 647.

3.1.4 *Single tone interference*

Level of any individual tone:

$$\leqslant (-73 + \psi) \text{ dBm0s}$$

where ψ is the weighting factor (positive or negative) as per CCIR Recommendation 468 at the particular frequency.

For 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 mid-frequency, is recommended. The use of stop filters influencing frequencies below 8 kHz should be avoided.

3.1.5 *Disturbing modulation by power supply*

The level of the strongest unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains shall be less than −45 dBm0s with a test signal of 1 kHz at alignment level 0 dBm0s.

3.1.6 *Non-linear distortion*

3.1.6.1 *Harmonic distortion*

The total harmonic distortion (THD) shall be measured with the input signal at +9 dBm0s.

The duration for which a single-tone is to be transmitted at this level should be restricted in accordance with Recommendations N.21 and N.23.

6) Administrations are urged to supply additional information on an appropriate value.

The THD when measured with a true-RMS meter shall not be less than the following requirements shown in Table 4/J.23.

TABLE 4/J.23

Input frequency (kHz)	Total harmonic distorsion
$0.05 \leq f < 0.1$	2% (−25 dBm0s)
$0.1 \leq f \leq 2.0$	1.4% (−28 dBm0s)

Note – If THD cannot be measured directly, compliance is considered to be fulfilled if the second or third harmonics are measured selectively and a calculated value of k meets the requirement:

$$k = \sqrt{k_2^2 + k_3^2}$$

where k_2 is the second harmonic coefficient and k_3 is the third harmonic coefficient.

3.1.6.2 *Intermodulation*

With input signals of 0.8 kHz and 1.42 kHz, each at a level of +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than 1.4% (−34 dBm0s).

3.1.6.3 *Distortion products measured by shaped noise*

Under study. See CCIR Report 640 (Kyoto 1978).

3.1.7 *Error in reconstituted frequency* (applies only to FDM systems)

Not to be greater than 1 Hz.

Note – A maximum error of 1 Hz is in principle acceptable where there is only a single transmission path between the signal source and the listener.

Where the broadcast network can involve two or more parallel paths, e.g. commentary and separate sound channels, or radio broadcast from different transmitters on the same frequency, unacceptable beats may occur unless zero error can be assured. The CCITT is studying methods of effecting this in all recommended systems.

3.1.8 *Intelligible cross-talk ratio*

3.1.8.1 The intelligible near-end and far-end cross-talk ratios between sound-programme circuits, or from a telephone circuit (disturbing) into a sound-programme circuit (disturbed) shall be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and shall not be less than the values of Table 5/J.23.

TABLE 5/J.23

Frequency (kHz)	Crosstalk attenuation (dB)
$f < 0.5$	Slope 6 dB/octave
$0.5 \leqslant f \leqslant 3.2$	74
$f > 3.2$	Slope -6 dB/octave

3.1.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1 – It is understood that this value is defined between the relative levels applicable to telephone circuits. (Administrations are invited to submit contributions on methods for measuring this parameter.)

Note 2 – The attention of Administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or at low frequencies (e.g. below about 100 kHz) on certain coaxial cable carrier systems. If sub-standard performance is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

Note 3 – When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

Note 4 – The attention of Administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk 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).

Note 5 – The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6 – The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

3.1.9 *Amplitude linearity*

When a 1 kHz input signal is stepped from -6 dBm0s to $+6$ dBm0s, or vice versa, the output level shall change accordingly by 12 ± 0.5 dB.

3.2 *Additional parameters for stereophonic programme transmission*

Not applicable, this section concerns 15 kHz type sound-programme circuits (see Recommendation T.21).

3.3 *Additional requirements for digital systems*

3.3.1 If a test signal is harmonically related to the sampling frequency, measuring difficulties may arise. In this case the nominal 1 kHz test signal must be offset. The Recommendation O.33 recommends 1020 Hz.

3.3.2 *Unbalance of the limitation level*

The difference between those levels which lead to a limitation of the positive or negative half-wave of the test signal shall not exceed 1 dB.

3.3.3 *Intermodulation with the sampling signal*

Intermodulation products (f_d) caused by non-linearities may occur in the sound-channel when the sampling signal signal (f_o) is combined with the inband audio signals (f_i) or out-of-band interfering signals (f_a).

3.3.3.1 *Inband intermodulation*

The following combination rule applies: $f_d = f_o - nf_i$.

Only values with $n = 2$ or 3 are of importance.

The level difference between a 0 dBm0s signal (f_i) and the intermodulation products (f_d) shall not be less than 40 dB.

A restriction to the f_i/f_d values in Table 6/J.23 is sufficient.

TABLE 6/J.23

	$n = 2$		$n = 3$	
f_i (kHz)	5	7	3	5
f_d (kHz)	6	2	7	1

3.3.3.2 *Out-of-band intermodulation*

The following combination rule applies: $f_d = nf_o \pm f_a$.

Only values with $n = 1$ or 2 are of importance.

The level difference between a 0 dBm0s signal (f_a) and the intermodulation products (f_d) shall not be less than 60 dB.

A restriction to the f_a/f_d values in Table 7/J.23 is sufficient.

TABLE 7/J.23

	$n = 1$		$n = 2$	
f_a (kHz)	15	17	31	33
f_d (kHz)	1			

3.3.4 *Further parameters*

Characteristics for bit errors, clicks, jitter, etc., are under study. (See Study Programme 18A/CMTT and CCIR Report 647.)

Bibliography

CCIR Document [1978-1982]: CMTT/68 (OIRT).

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 and 1980)

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 § 1 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 A, B and C.

The characteristics of the group links, which have to be used in any case, are given in § 2 below.

1 Characteristics of an equipment allowing two 15 kHz type carrier-frequency sound-programme circuits to be established on a group

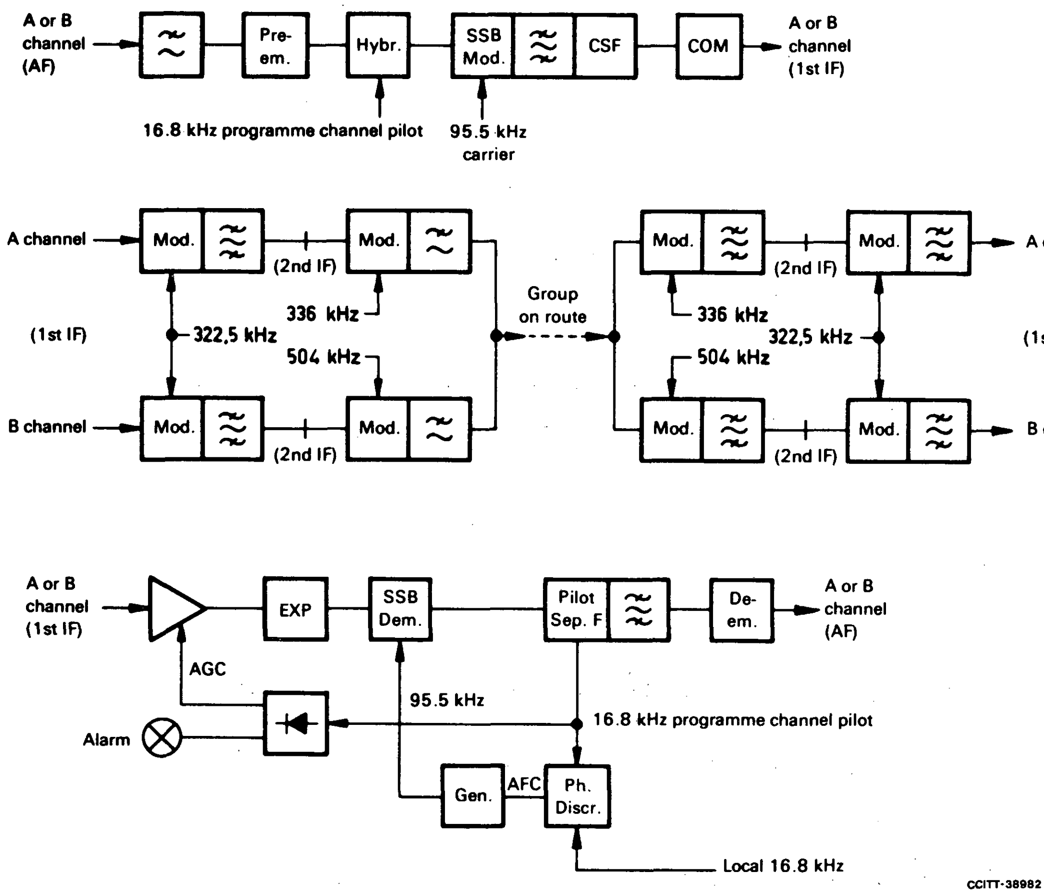
Introduction

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 [1] 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.

[1] This is the objective given in Recommendation J.14 for new design of equipment.

The block schematic of a suitable equipment is given in Figure 1/J.31.



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

1.1 *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.

1.2 *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.

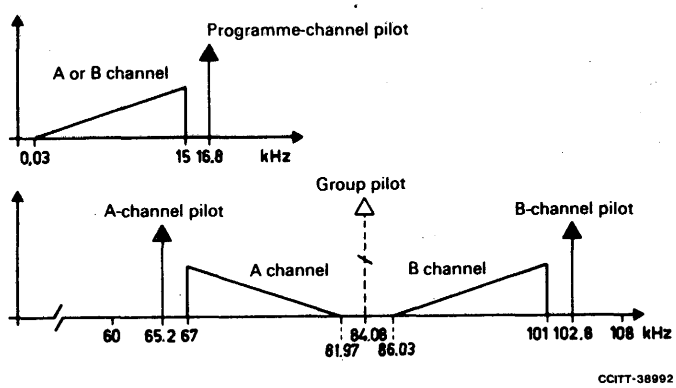


FIGURE 2/J.31

Line-frequency positions of the two-programme channels in the group

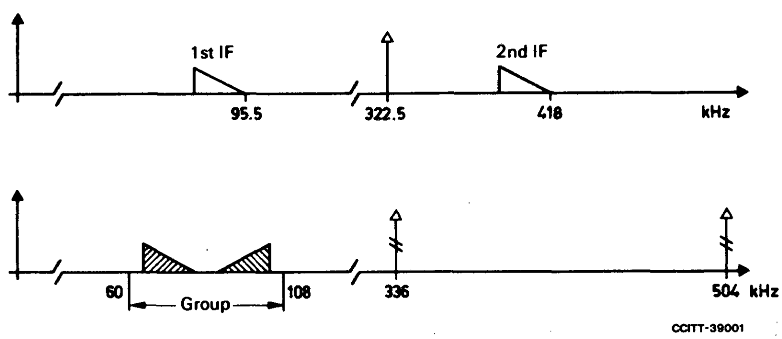


FIGURE 3/J.31

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.

1.3 *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.

1.4 *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.

1.5 *Compandor*

1.5.1 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.

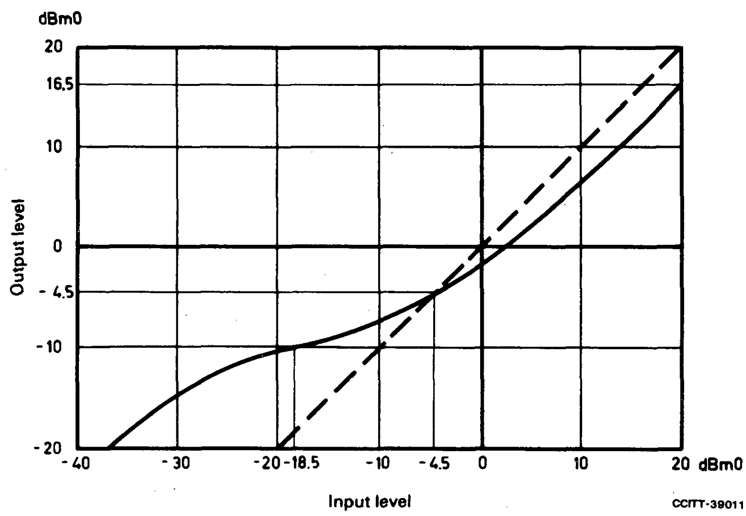


FIGURE 4/J.31

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.

1.5.2 The attack and recovery times of the compressor are measured in 12 dB steps (see Recommendations G.162 [2] and O.31 [3]) 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

2) dBm0(t) denotes that the level quoted is referred to a zero relative level point in a telephone channel.

that is approximately in the middle of the IF band. The attack and recovery times of the compressor are, as in Recommendation G.162 [2], 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.

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

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 here are not 1.5 and 0.75, respectively, as in the case of the telephone compandor.

1.6 *Impedance at audio points*

The audio input-impedance should be 600 ohms balanced with a minimum return loss of 26 dB.

1.7 *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.

1.8 *Suppression of carrier leaks at 10 kHz and 14 kHz*

Since, according to Recommendation H.14 [4], 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: $\leq \pm$ 150 Hz

at 14 kHz: $\leq \pm$ 210 Hz

Attenuation for the midfrequencies

at 10 kHz: $\geq$ 36 dB

at 14 kHz: $\geq$ 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.

1.9 *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 § 1.2 above, interconnection in these positions will exclude unnecessary compandor stages from the through connection.

1.10 *Equalizers for gain and phase difference*

In order to be able to meet the quality parameters specified in Recommendation J.21, § 3.1.3, for monophonic and §§ 3.2.1 and 3.2.2 for stereophonic sound-programme transmissions, gain and phase-difference equalizers in the group-frequency position have to be assigned to the sound-programme channel equipment before the hybrid at the receiving end. These equalizers can be switched in steps and their characteristics are adapted to the typical distortions by making them fan-shaped.

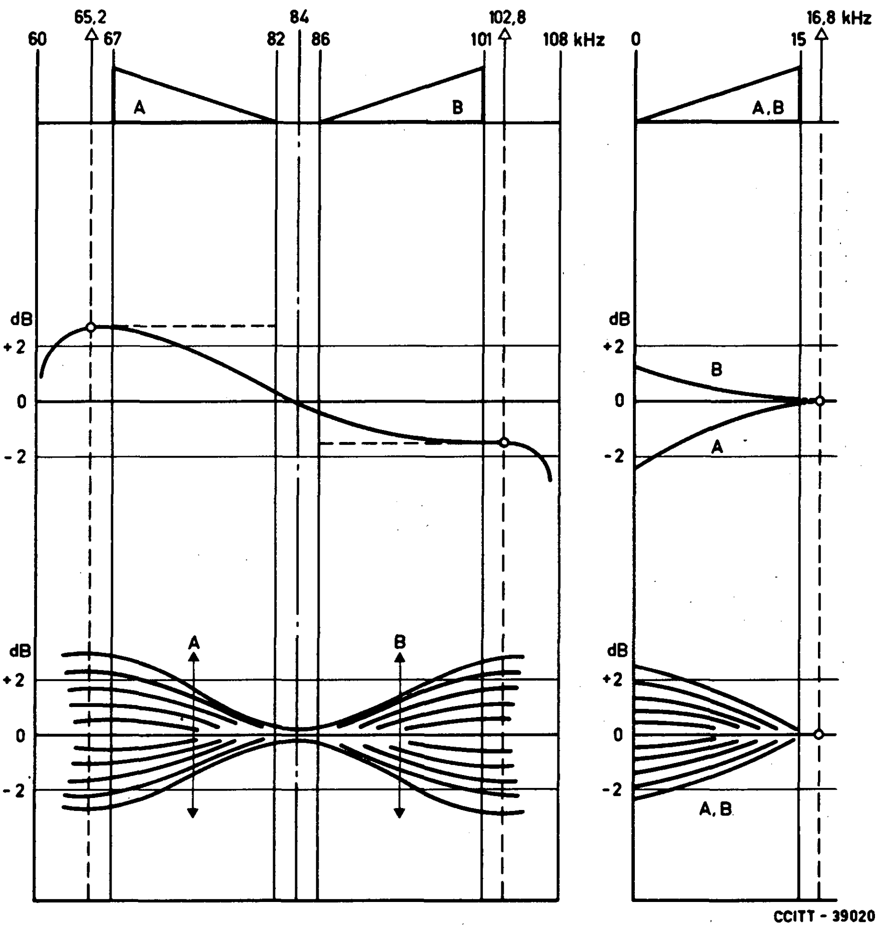
The gain equalizers are required to compensate for the frequency-dependent gain distortions in the lower and upper frequency ranges of the group on which the sound-programme channels are established. By means of the phase-difference equalizers, the phase distortion occurring in the group is increased in the upper or lower half of the group-frequency band to such an extent that a characteristic which is skew-symmetric about the centre frequency of the group is obtained, i.e. phase coincidence between the sound-programme channel positions.

Figures 5/J.31 and 6/J.31 show the effectiveness of the gain and phase-difference equalizers within the frequency band of the group and their effects on gain and phase-difference of the sound-programme channels in the AF position. Here, allowance is made for the fact that deviations at the pilot frequency of 16.8 kHz in the AF position are always automatically adjusted to zero by means of the pilot regulation.

In order to facilitate international cooperation in determining the optimum equalizer setting within a very short time, the lining-up procedure and arrangement of measuring equipment detailed below is recommended.

At the sending end, this arrangement consists of a signal generator with a high level accuracy and a very low output impedance, which produces the measuring frequencies of 0.525 kHz (= 1/32) and 8.4 kHz (= 1/2) derived from the pilot frequency of 16.8 kHz. The two measuring frequencies should be transmitted simultaneously over both sound-programme channels, individually or at automatically alternating 3.9-second intervals. In the latter case the clock is obtained by a further division of 0.525 kHz by 2^{12}.

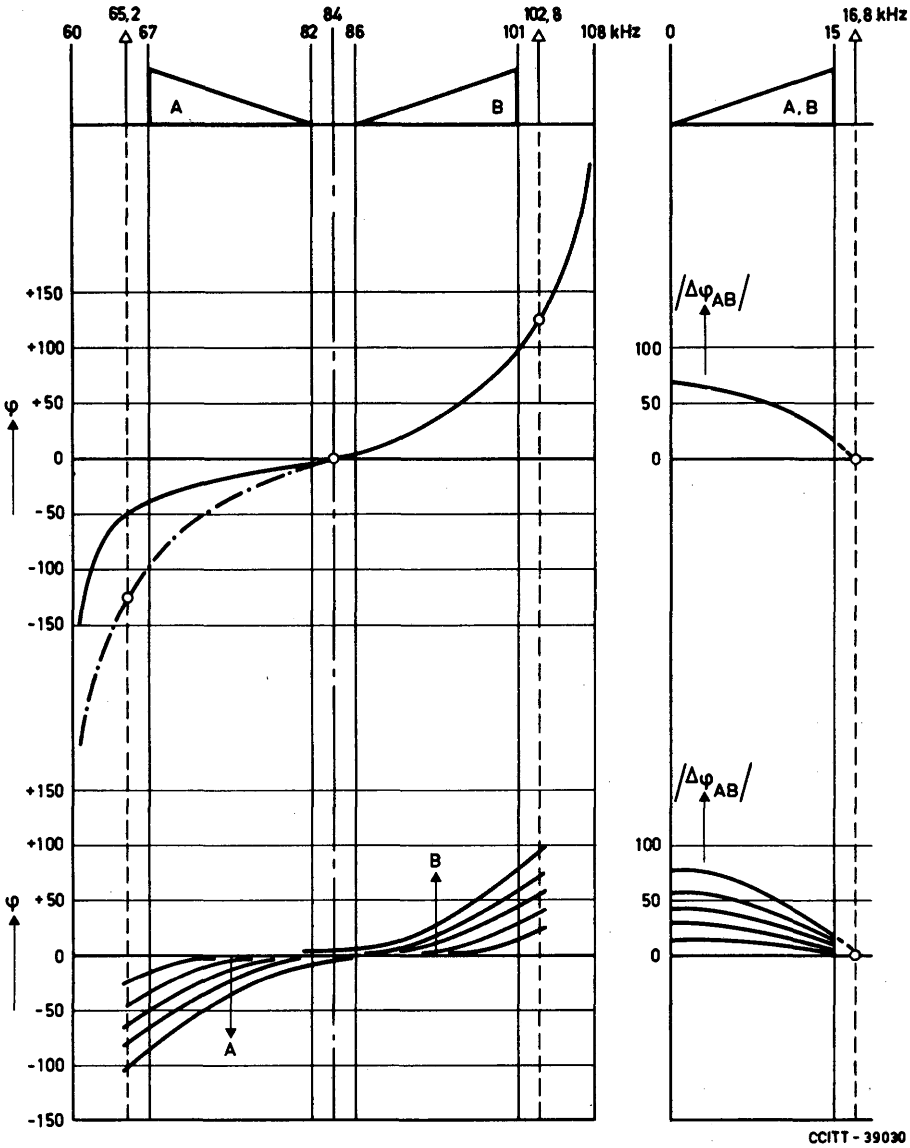
At the receiving end, use is made of a receiver having a calibrated measuring instrument which indicates the level in each of the two sound-programme channels and the phase-difference between them derived from the level of the voltage difference in the two channels. The received measuring frequency is indicated by a lamp. Since the frequency-dependent characteristic of the so-called fan equalizer used for gain and phase-difference equalization is defined for the individual steps, it is possible to confine oneself to the two measuring frequencies considered to be sufficiently representative when determining the optimum equalizer setting.



Top: Example of a gain distortion.
Bottom: Fan-shaped characteristics of the two gain equalizers.

FIGURE 5/J.31

Principle of gain equalization in the group-frequency position and its effect on the sound-programme channels in the AF position, allowance being made for the pilot regulation



Top: Example of phase symmetry distortion. Ideal skew-symmetric phase characteristic shown.
Bottom: Fan-shaped characteristics of the phase-symmetry equalizers.

FIGURE 6/J.31

Principle of phase-symmetry equalization in the group-frequency position and its effect on the phase difference between the sound-programme channels in the AF position, allowance being made for the pilot phase regulation

1.11 *Usable power reserve*

1.11.1 *Audio-frequency parts of the equipment* (before pre-emphasis and after de-emphasis):

1.11.1.1 *Peak power level*

The equivalent power level of the peak of sound-programme signals, when they are controlled in accordance with Recommendations J.14 and J.15 so as to have a quasi-peak power of +9 dBm0s, exceeds a level of about +12 dBm0s with a probability of 10^{-5}, as is documented by several Administrations (see CCIR Report 491 [5]). For the telephone service, the level with a probability of 10^{-5}, i.e. the level of +12 dBm0s, should be respected in any case.

1.11.1.2 *Margin against saturation*

A margin of 3 dB should be maintained between the peak power level in § 1.11.1.1 and the overload point, to allow for level variations.

1.11.1.3 *Overload point, definitions*

First definition – The **overload point** or overload level of an amplifier is at that value of absolute power level at the output, at which the absolute power level of the third harmonic increases by 20 dB when the input signal to the amplifier is increased by 1 dB.

This first definition does not apply when the test frequency is so high that the third harmonic frequency falls outside the useful bandwidth of the amplifier. The following definition may then be used:

Second definition – The overload point or overload level of an amplifier is 6 dB higher than the absolute power level in dBm, at the output of the amplifier, of each of two sinusoidal signals of equal amplitude and of frequencies A and B respectively, when these absolute power levels are so adjusted that an increase of 1 dB in both of their separate levels at the input to the amplifier causes an increase, at the output of the amplifier, of 20 dB in the intermodulation product of frequency 2A-B.

1.11.1.4 *Value of the overload point*

The overload point of these audio-frequency parts, therefore, should be higher than +15 dBm0s.

1.11.2 *Carrier-frequency parts of the programme modulating equipment (between compressor and telephone multiplex and between telephone multiplex and expander)*

The overload point, as defined in § 1.11.1.3 should have a margin of 2 dB against the equivalent peak power value of a group channel (+19 dBm0). The overload point of these carrier-frequency parts, therefore, should be higher than +21 dBm0.

1.11.3 *Complete equipment, back to back*

Test measurements should be possible without degradation visible on an oscilloscope:

– with one or two sine-wave test signals of any frequency with peak power levels up to +12 dBm0s,
– with tone pulses of any frequency with levels up to 0 dBm0s.

1.12 *Loading of groups and supergroups*

Table 2/J.31 gives some observed figures for the loading of groups and supergroups in the most essential cases.

2 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 [9] 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.

TABLE 2/J.31

Loading of groups and supergroups in the case of sound-programme transmission with the carrier programme system recommended in CCITT Recommendation J.31, §1

	n_m (dBm0)	n_p (dBm0)	
Group			
12 telephone channels (as in Recommendation G.223 [6])	−4	+19	
1 programme channel only	−6	+12	
1 programme channel + 6 telephone channels	−3.5	+12	programme channel only
2 programme channels (different monophonic programmes)	−3	+13	
1 stereophonic pair [a)]	−3	+17	
2 programme channels (identical monophonic programmes)	−3	+17	
Supergroup			
60 telephone channels (as in Recommendation G.223 [6])	+3	+21	
4 programme channels in 2 groups + 36 telephone channels:			
4 different programmes	+3.5	+14	programme channels only
2 different stereophonic programmes	+3.5	+18	programme channels only
2 equal stereophonic programmes	+3.5	+22	programme channels only
10 programme channels			
10 different programmes	+4	+15	
5 different stereophonic programmes	+4	+19	
2 equal stereophonic programmes + 6 different monophonic programmes	+4	+22	

n_m Long-term mean power level [7].

n_p Equivalent peak power level [8] (= level of equivalent sine-wave whose amplitude is exceeded by the peak voltage of the multiplex signal only with a bilateral probability of 10^{-5}).

a) Loading by one stereophonic programme is treated as loading by two identical monophonic programmes (worst case).

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 − Δps) 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:

a) 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 − Δps) dBm0s taking account of the amount of the narrow-band crystal stop-filter attenuation.

3) This value has been specified in Recommendation J.21 by CMTT. CCIR Report 493 [10] 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.

b) 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:

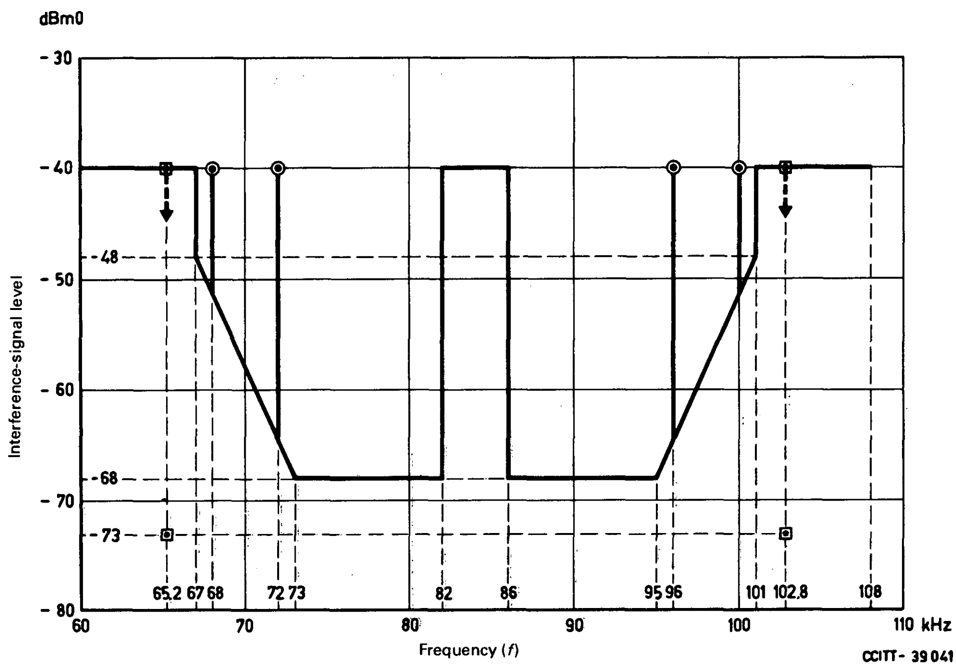
- for frequencies between 73 kHz and 95 kHz: −68 dBm0,
- for frequencies at 67 kHz and 101 kHz: −48 dBm0.

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 are needed beyond those covered in Recommendation M.460 [9] (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 7/J.31.

Note – Figure 8/J.31 gives the permissible level of single-tone interference for the systems described in Annexes A, B and C, such that the basic requirement of (−73 – Δps) dBm0s mentioned above is met.



The continuous curve represents the general requirements for single-frequency interfering tones, with the following exceptions:

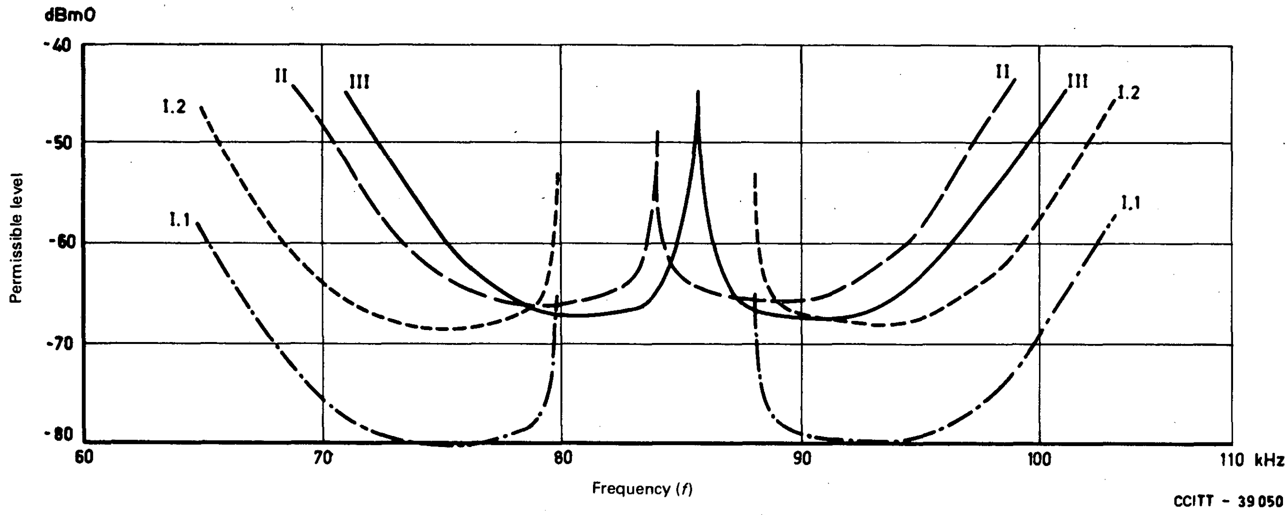
◉ carrier-leak frequencies at which the requirements are relaxed to −40 dBm0 are shown thus.

▣ at frequencies of A- and B-channel pilots, 65.2 and 102.8 kHz ±300 Hz interfering signals should be at least 40 dB below the lowest possible level of the pilots (i.e. −29 dBm0 −3.5 dB when compressor input signal is large).

FIGURE 7/J.31

Mask for the carrier leaks and any other tone-interference signal falling within the group band

[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.



Curve I.1: requirement for the system of Annex A, *without* compandor gain.
Curve I.2: requirement for the system of Annex A, *with* compandor gain.
Curve II: requirement for the DSB system of Annex B.
Curve III: requirement for the system of Annex C.

FIGURE 8/J.31
Permissible level of a single-frequency interference on the group link

ANNEX A

(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.1 *Frequency allocation in the group*

TABLE A-1/J.31

	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 (see Table A-1/J.31) 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.

A.2 *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.

A.3 *Compandor*

A.3.1 *Steady-state characteristics*

The compandor has a separate frequency-modulated control channel containing the information on the degree of compression, as indicated in Table A.2/J.31.

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 the Recommendation cited in [11]).

TABLE A-2/J.31

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 $\pm$ 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.

A.3.2 *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.

A.3.3 *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.

A.4 *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 B

(to Recommendation J.31)

Double-sideband system

(Contribution of L.M. Ericsson, ITT and Telettra)

B.1 *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.2 *Pre-emphasis*

The pre-emphasis curve given in Recommendation J.17 should be used.

B.3 *Compandors*

Compandors are not an integral part of these systems.

B.4 *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.

B.5 *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 the Recommendation cited in [12].

B.6 *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 C

(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 Contribution COM XV-No. 151 (Study Period 1973-1976) 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.

References

[1] CCITT Recommendation *Noise objectives for design of carrier-transmission systems of 2500 km*, Vol. III, Rec. G.222.

[2] CCITT Recommendation *Characteristics of compandors for telephony*, Vol. III, Rec. G.162.

[3] CCITT Recommendation *Specification for an automatic measuring equipment for sound-programme circuits*, Vol. IV, Rec. O.31.

[4] CCITT Recommendation *Characteristics of group links for the transmission of wide-spectrum signals*, Vol. III, Rec. H.14.

[5] CCIR Report *Characteristics of signals sent over sound-programme circuits*, Vol. XII, Report 491, ITU, Geneva, 1982.

[6] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*, Vol. III, Rec. G.223.

[7] *Ibid.*, § 1.

[8] *Ibid.*, § 6.2.

[9] CCITT Recommendation *Bringing international group, supergroup, etc., links into service*, Vol. IV, Rec. M.460.

[10] CCIR Report *Compandors for sound-programme circuits*, Vol. XII, Report 493, ITU, Geneva, 1982.

[11] CCITT Recommendation *Psophometers (apparatus for the objective measurement of circuit noise)*, Green Book, Vol. V, Rec. P.53, Part B, ITU, Geneva, 1973.

[12] CCITT Recommendation *Pilots on groups, supergroups, etc.*, Vol. III, Rec. G.241, §§ 2 and 3.

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 10 kHz TYPE SOUND-PROGRAMME CIRCUITS

(The text of this Recommendation can se found in Fascicle III.4 of the *Red Book*, ITU, Geneva, 1985)

Recommendation J.33

CHARACTERISTICS OF EQUIPMENT AND LINES USED FOR SETTING UP 6.4 kHz TYPE SOUND-PROGRAMME CIRCUITS[1)]

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[2)].

If there is an arrangement between interested Administrations, including if necessary the Administration of transit countries, a solution allowing the establishment of up to four 6.4 kHz-type sound-programme circuits in a basic group, as described in Annex A, may be used.

ANNEX A

(to Recommendation J.33)

Four 6.4 kHz type sound-programme circuits in a basic group

(Contribution by the PTT of China)

A.1 *Frequency position and modulation scheme*

In order that the requirements of the performance characteristics of adjacent basic groups, supergroups, etc., through-connection equipments are not more stringent than those for the 15 kHz type sound-programme circuits, the band of four 6.4 kHz programme frequencies in a group should be within the range of 65.3 to 102.7 kHz band.

In order that modulation procedure is the same as that of 15 kHz type sound-programme circuits, three level modulations are adopted. Modulation procedure and frequency position are shown in Figure A-1/J.33. All the carriers and pilots are derived from 12 kHz basic frequency.

A.2 *Emphasis network and compandor*

In order that the signal mean load of four 6.4 kHz type sound-programme circuits in telephone circuits is less than -3 dBm0, and the peak value load less than $+19$ dBm0, it is necessary that the programme relative level (dBrs) be lower than that of telephone relative level (dBr) by 6.5 dB and emphasis network be applied.

In order to meet the requirement of -39 dBm0s noise level of 2500 km hypothetical reference circuits defined in Recommendation J.23 (Yellow Book, 1980) in addition to the emphasis network, compandor should also be applied.

6.4 kHz system applies emphasis network as described in Recommendation J.17. At 0.8 kHz, the insertion loss of pre-emphasis is 6.5 dB, while the insertion gain of de-emphasis is 6.5 dB.

6.4 kHz system applies the same compandors as 15 kHz system does. (See Figure 4/J.31, Recommendation J.31.)

1) The performance characteristics of 6.4 kHz type sound-programme circuits are given in Recommendation J.23 (Yellow Book, 1980).

2) For the choice of groups and supergroups used, see Recommendation J.32.

A.3 *Pilot*

To ensure the stability of instertion loss and deviation of frequency required in programme circuits, a 7.5 kHz pilot at a level of -29 dBm0 $\pm$ 0.1 dB is inserted after pre-emphasis and before modulator in transmission path.

The pilot, after demodulator in receiving path, is derived so as to regulate frequencies and levels.

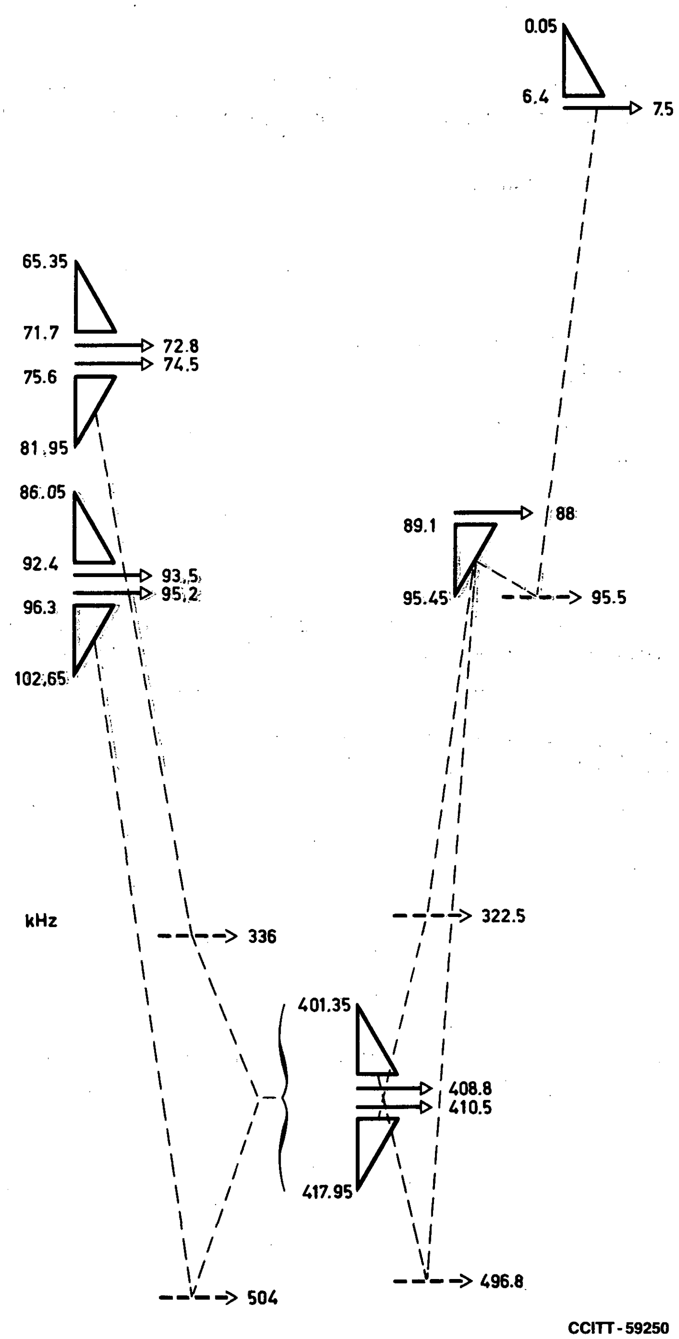


FIGURE A-1/J.33

Frequency position of four 6.4 kHz-type sound-programme channels in a group

A.4 *Noise*

Weighted noise of telephone channel hypothetical reference circuits	−50 dBm0p
Due to:	
telephone weighting network loss	2.5 dB
bandwidth expanding from 3.1 kHz to 6.4 kHz	3.2 dB
CCIR Recommendation 468 sound-programme weighted network (0.05 to 6.4 kHz)	9.0 dB
CCIR Recommendation 468 quasi-peak value measurement	5 dB
Sum (noise of hypothetical reference circuit without emphasis and compandor)	−30.3 dBq0ps
Variation of weighted noise level within the range of 0.05 to 6.4 kHz band due to de-emphasis (6.5 dB/800 Hz)	− 3 dB
Variation of noise level due to expandor	−12 dB
Noise of weighted hypothetical reference circuit of 6.4 kHz type programme channels (with emphasis and compandor)	−45.3 dBq0ps

There is about 6 dB safety margin compared with −39 dBq0ps for 6.4 kHz type programme circuits described in Recommendation J.23.

A.5 *Summary*

In a group, four 6.4 kHz sound-programme channels (A, B, C and D) can be established, and A (or D) can be replaced by three telephone channels, A + B (or C + D) can be replaced by one 15 kHz sound-programme channel or by six telephone channels.

This system meets every requirement of 6.4 kHz type sound-programme circuits described in Recommendation J.23 (Yellow Book, 1980). There is no risk of overload in a group even when four programme channels transmit the same programme simultaneously.

Recommendation J.34

CHARACTERISTICS OF EQUIPMENT USED FOR SETTING UP 7 kHz TYPE SOUND-PROGRAMME CIRCUITS

(Geneva, 1980)

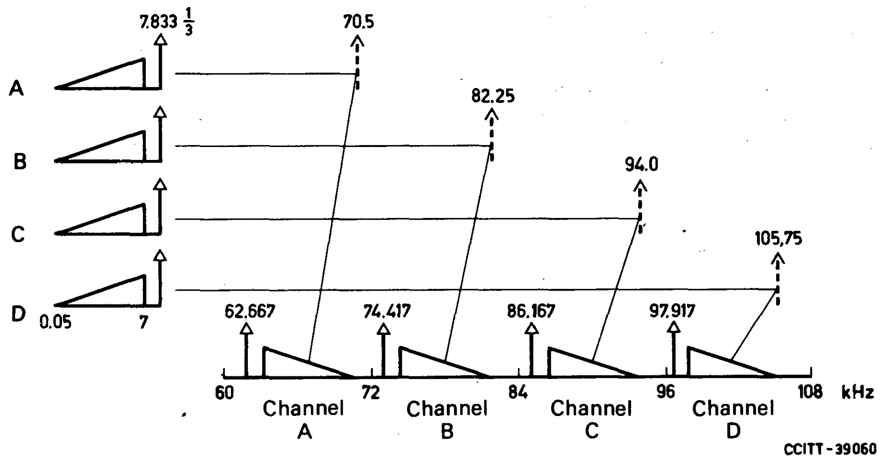
Introduction

An equipment allowing the establishment of 7 kHz type sound-programme circuits (in accordance with CCIR Recommendation 503 [1]) on carrier telephone systems which conform to the noise objectives in Recommendation G.222 [2] 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. The sound-programme circuits set up on one group can be used only as monophonic circuits.

The following recommendations, 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.

1 Frequency position in the basic group 60-108 kHz

The frequency position in the basic group is shown in Figure 1/J.34. For the programme channels, the stability of the virtual carrier frequency is $\pm 10^{-5}$ and the programme-channel pilot is fed in as 7833 1/3 Hz (stability better than $\pm 10^{-5}$) in the audio-frequency position.



Note – The carrier frequencies are multiples of 11.75 kHz and can be derived from a common generator frequency.

FIGURE 1/J.34

Frequency allocation for four 7 kHz type sound-programme channels set up on one group

Note 1 – Programme channel D can be replaced by telephone channels 1 to 3; programme channel C by telephone channels 4 to 6; programme channel B by telephone channels 7 to 9; programme channel A by telephone channels 10 to 12.

Note 2 – The use of programme channel D is only compatible with group pilots at 84.14 and 84.08 kHz, but not at 104.08 kHz. Moreover, this channel cannot be used in Group 3 of a supergroup with a 411.92 kHz pilot or a 411.86 kHz pilot.

The frequency positions are as shown in Table 1/J.34.

TABLE 1/J.34

Channel range (kHz)	Virtual carrier frequency [a] (kHz)
60 to 72	Inverted position 70.5
72 to 84	Inverted position 82.25
84 to 96	Inverted position 94
96 to 108	Inverted position 105.75

[a] The carrier frequencies are multiples of 11.75 kHz and can be derived from a common generator frequency.

2 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.

3 7833 1/3-Hz pilot signal

At the sending end, the 7833 1/3-Hz 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 (the relative level at this point being defined under the assumption that the compressor is switched off and replaced by 0 dB loss). In the absence of a programme signal, this pilot level is increased by 14 dB by the compressor to −15 dBm0 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 7833 1/3-Hz bandpass filter and is then suppressed in the transmission channel.

The control functions of the pilot are frequency regeneration of the demodulator and compensation of the transmission loss deviations between compressor and expander. The frequency regeneration of the demodulator should be sufficiently accurate so that the frequency offset between the audio-frequency (AF) programmes at the transmit end and at the receive end is less than 0.6 Hz even if the frequency offset of the group connection is 2 Hz.

4 Compandor

The characteristic of the compressor is the same as in Recommendation J.31, § 1.5.1 with the only exception that the output level is decreased by 3 dB. The maximum compressor gain is 14 dB, the minimum compressor gain is −6.5 dB. With an input level of −18.5 dBm0, its output level is −13 dBm0.

The tolerance of the compressor gain is ± 0.5 dB, but it is ± 0.1 dB at programme signal levels at the compressor input of −∞, −15 and +3 dBm0 (in agreement with Table 1/J.31).

The amplification of the expander is 3 dB larger than that given in Recommendation J.31, § 1.5.1.

5 Attenuation/frequency distortion due to the sending and receiving equipments

The total attenuation/frequency distortion introduced by a sending and a receiving equipment should not exceed the following preliminarily recommended ranges:

0.05 to 0.1 kHz: +0.7 to −1.0 dB

0.1 to 6.4 kHz: +0.5 to −0.5 dB

6.4 to 7 kHz: +0.7 to −1.0 dB

relative to the gain at 800 or 1000 Hz.

Note – These values are still under study. Three carrier sections with two intermediate audio points according to the hypothetical reference circuit (h.r.c.), (Recommendation J.11), should comply with the CCIR Recommendation cited in [3].

6 Suppression of carrier leaks

Carrier leaks which, after demodulation, fall into the AF programme band should have a level lower than −68 dBm0 in the carrier frequency position.

A carrier leak at, and residuals from pilots in the vicinity of, 64 kHz with a level above −68 dBm0 will generate an intolerable single-tone interference at 6.5 kHz in channel A. If required, it may be suppressed sufficiently with a lowpass filter at the AF output of channel A. Then this channel can be used for a 5 kHz type sound-programme circuit.

References

[1] CCIR Recommendation *Performance characteristics of narrow-bandwidth sound-programme circuits*, Vol. XII, Rec. 503, ITU, Geneva, 1978.

[2] CCITT Recommendation *Noise objectives for design of carrier-transmission systems of 2500 km*, Vol. III, Rec. G.222.

[3] CCIR Recommendation *Performance characteristics of narrow-bandwidth sound-programme circuits*, Vol. XII, Rec. 503, § 3.3.1, ITU, Geneva, 1978.

SECTION 4

CHARACTERISTICS OF EQUIPMENTS FOR CODING ANALOGUE SOUND PROGRAMME SIGNALS

Recommendation J.41

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE HIGH QUALITY SOUND PROGRAMME SIGNALS FOR TRANSMISSION ON 384 kbit/s CHANNELS

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

1 General

1.1 This Recommendation gives the characteristics of equipment for the coding of 15 kHz monophonic analogue sound-programme signals into a digital signal of 384 kbit/s. For stereophonic operation, two monophonic digital codecs can be utilized. Two monophonic digital signals that form a stereophonic signal should be routed together over the same transmission systems (path) to avoid any difference in transmission delay.

1.2 Equipment for coding of analogue sound-programme signals, as specified in this Recommendation, can be:

a) A stand-alone encoder/decoder with a digital interface at 384 kbit/s. The encoder operation and the decoder operation may be performed in two separate equipments or in the same equipment.

b) A combined encoder-multiplex/decoder-demultiplex with a digital interface at 1544 or 2048 kbit/s. The encoder-multiplex operation and the decoder-demultiplex operation may be performed in two separate equipments or in the same equipment.

In case b), it is not mandatory to provide an external digital sound programme access port at 384 kbit/s.

1.3 Two methods of encoding have been specified by the CMTT [1] and these form the basis for this Recommendation.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.21 (CCIR Recommendation 505) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

Note – When transmitting stereophonic sound programme signals, it is necessary that the encoder and decoder are designed such that they will meet the specified requirements for phase difference.

In order to avoid any unnecessary complexity, the sampling of channels A and B should be performed simultaneously.

3 Method of encoding

3.1 The recommended encoding laws are as specified in [1].

3.2 These encoding laws are based on a uniformly quantized 14-bit per sample PCM technique with companding and employ either:

a) eleven-segment 14- to 11-bit instantaneous A-law companding, or

b) five-range 14- to 10-bit near instantaneous companding.

For provisional rules for through connection between the two companding methods, see Note 4 in [1].

3.3 Other coding techniques which may be used by bilateral agreement of the Administrations concerned are also listed in Annex A. However, these techniques do not form part of this Recommendation.

3.4 Equipment characteristics common to both methods of encoding are:

Nominal audio bandwidth:	0.04 to 15 kHz.
Audio interface:	see Recommendation J.21, § 2.
Sampling frequency (CCIR Recommendation 606):	$32 (1 \pm 5 \times 10^{-5})$ kHz.
Pre/de-emphasis:	Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

Note – Pre-emphasis and de-emphasis are not used by the Administrations of Canada, Japan and the United States on their national circuits and on international circuits between each other, but are used on international circuits to other countries.

4 Equipment using instantaneous companding

4.1 *Coding table*

4.1.1 The coding law is specified in Table 1/J.41.

4.1.2 The allocation of character signals (PCM code words) is also given in Table 1/J.41. Two variants (A and B) of character signals are allowed.

Note – In the case of digital interconnection between variants A and B, the conversion from one set of character signals to the other in Table 1/J.41 can be implemented without any performance degradation. In the case of analogue interconnection, a small reduction in the S/N ratio, in the order of 3 dB, is expected.

4.2 *Bit rates*

Nominal source coding bit rate (32 kHz × 11 bits/sample)	352 kbit/s
Error protection	32 kbit/s
Transmission bit rate	384 kbit/s

4.3 *Overload level*

The overload level for a sine-wave signal at zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis is +15 dBm0s.

4.4 *Digital signal format*

The character signal bit sequences for variants A and B are shown in Figure 1/J.41.

4.5 *Bit error protection*

One parity bit is added to each 11-bit character signal.

TABLE 1/J.41

11 segment, 14 to 11 bit instantaneous companding A-law PCM for sound-programme signals (positive half only) [a]

Normalized analogue input	Normalized analogue output	Compressed digital code	Segment No.	Effective resolution (bits)	11 bit coding Allocation of character signals Variant A [a)] 1	 2 3 4	 5 6 7 8 9 10	 11	 Variant B [b)] S	 X Y Z	 A B C D E F G
8160 to 8192 4096 to 4128	8176 4112	895 768	1	9	0	1 1 1	1 1 1 1 1 1 0 0 0 0 0 0	1 0	0	1 1 0	1 1 1 1 1 1 1 0 0 0 0 0 0 0
4080 to 4096 2048 to 2064	4088 2056	767 640	2	10	0	1 1 0	1 1 1 1 1 1 0 0 0 0 0 0	1 0	0	1 0 1	1 1 1 1 1 1 1 0 0 0 0 0 0 0
2040 to 2048 1024 to 1032	2044 1028	639 512	3	11	0	1 0 1	1 1 1 1 1 1 0 0 0 0 0 0	1 0	0	1 0 0	1 1 1 1 1 1 1 0 0 0 0 0 0 0
1020 to 1024 512 to 516	1022 514	511 384	4	12	0	1 0 0	1 1 1 1 1 1 0 0 0 0 0 0	1 0	0	0 1 1	1 1 1 1 1 1 1 0 0 0 0 0 0 0
510 to 512 256 to 258	511 257	383 256	5	13	0	0 1 1	1 1 1 1 1 1 0 0 0 0 0 0	1 0	0	0 1 0	1 1 1 1 1 1 1 0 0 0 0 0 0 0
255 to 256 128 to 129	255.5 128.5	255 128	6	14	0	0 1 0	1 1 1 1 1 1 0 0 0 0 0 0	1 0	0	0 0 1	1 1 1 1 1 1 1 0 0 0 0 0 0 0
127 to 128 0 to 1	127.5 0.5	127 0			0	0 0 1 0 0 0	1 1 1 1 1 1 0 0 0 0 0 0	X 0	0	0 0 0	1 1 1 1 1 1 1 0 0 0 0 0 0 0

X 11th bit freely available in variant A.

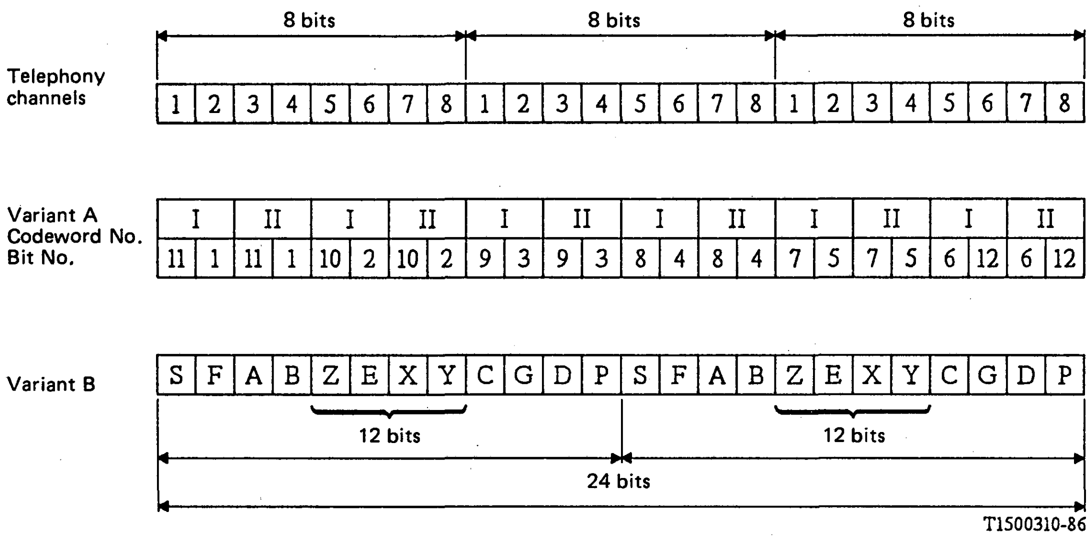
a) Character signals for the negative half are the same as those for the positive half except that the sign bits (bit 1 and S for variants A and B respectively) are inverted.

b) Variant A is presently used with digital equipment based on a 2048 kbit/s hierarchy. After coding and before the parity bit is inserted, bits 1 to 5 are inverted.

c) Variant B is presently used with digital equipment based on a 1544 kbit/s digital hierarchy. All bits, including the parity bit, are inverted and reformatted before transmission (see Figure 1/J.41).

4.5.1 *Variant A*

The five most significant bits of each sample are protected against errors by means of a parity bit. In the converter of the transmitting part, the parity bit is added as the 12th bit to each code word. Its value is fixed so that the 6 bit parity block always contains only an odd number of "one" values. In order that even bit error structures can also result in parity violations, the protected and unprotected bits of each code word are interleaved in ascending and descending sequence, as shown in Figure 1/J.41.



Variant A: BIT definitions:

1 Sign bit
2, 3, 4 Chord bits
5 to 11 Step bits
12 Parity bits

Variant B: BIT definitions:

S Sign bit
X, Y, Z Chord (since chord 1 1 1 is not used, and bits are inverted on the line, one of these bits will always be a one)
A to G Step
P Parity bit
One of these 4 bits will always be a one (see chord above)

FIGURE 1/J.41

15 kHz sound programme channel bit sequences for transmission on A-law companded systems

4.5.2 *Variant B*

The added parity bit shall be based on the 7 most significant bits of the 11-bit PCM word. These are bits S, X, Y, Z, A, B, C. The parity of "ones" bit shall be *even.* Since the chord bits (X, Y, Z) always contain a one, the minimum number of ones per sample is 2, resulting in a minimum ones density of 1/6.

4.5.3 *Error concealment*

If a parity violation is detected, an error concealment technique should be applied (for instance, replacement by interpolation, extrapolation or repetition. For multiple parity violation (error bursts), a muting technique should be applied.

4.6 *Digital interface at 384 kbit/s*

Under study (see Recommendations G.735 and G.737).

4.7 *Synchronization*

The coding equipment operates in synchronism with the clock of subsequent multiplex equipment or the network clock. In cases where the digital interface is provided, bit and byte (24 bit, as shown in Figure 1/J.41) timing information is required.

Variant A: A solution for synchronous access is given in Recommandations G.735 and G.737.

Variant B: The solution for synchronous access is under study.

4.8 *Fault condition and consequent actions*

4.8.1 *Variant A*

Where a 384 kbit/s digital interface is provided, the same principles for fault conditions and subsequent actions as those outlined in Recommendation G.732 should be followed.

4.8.2 *Variant B*

Under study.

5 Equipment using near-instantaneous companding

5.1 *Introduction*

The equipment described in this section uses the near-instantaneous method of companding in the coding of high quality sound-programme signals into digital form.

A two-stage process is used in the encoding equipment:

a) Conversion of a 15 kHz channel into a 338 kbit/s stream.

Note – The value of 338 kbit/s has been chosen to allow for the possible multiplexing of 6 channels into a 2048 kbit/s dedicated frame format.

b) Asynchronous insertion of the 338 kbit/s stream into a 384 kbit/s stream.

Note – The asynchronous insertion of the 338 kbit/s stream into a 384 kbit/s stream allows the use, at the encoder location, of a clock not necessarily synchronous to the network clock. It can be advantageous when the encoder equipment and the insertion equipment (see Recommendations G.735 and G.737) are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

5.2 *Conversion from 15 kHz to 338 kbit/s*

5.2.1 *Overload level*

The overload level for a sine-wave signal at the zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis circuit, is +12 dBm0s.

5.2.2 *Companding*

Near-instantaneous companding is used to achieve a data rate reduction from 14 bits/sample to 10 bits/sample. The system codes a block of 32 samples into one of 5 gain ranges, according to the highest value sample in the block. The companding characteristic is shown diagramatically in Figure 2/J.41 and the parameters are specified in Table 2/J.41.

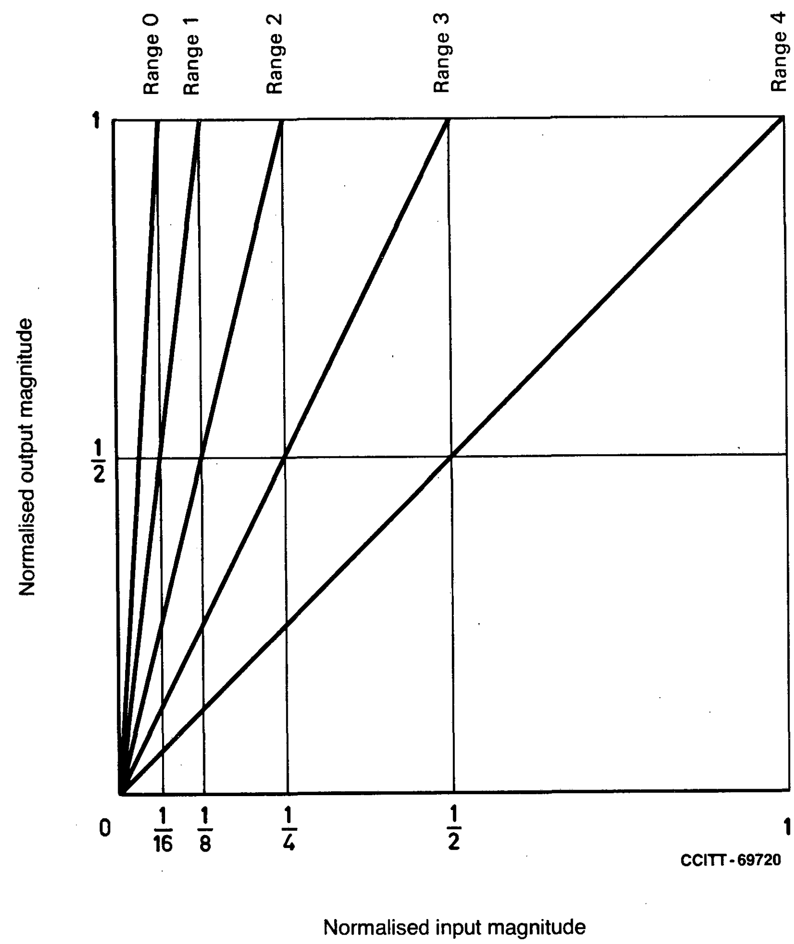


FIGURE 2/J.41
Near-instantaneous companding characteristic

5.2.3 *Range coding and protection*

Information defining the range used is transmitted over 3 successive blocks as a 7-bit word, increasing to 11 bits in a Hamming 7, 11 single error correcting code and distributed throughout the 3 blocks as follows:

The five possible values for each of the 3 range codes (one range code for each block in the 3 ms frame; see Figure 3/J.41), are:

Range 4 highest signal level

Range 3

Range 2

Range 1

Range 0 lowest signal level

Range codes generated in this way from three successive blocks are designated Ra, Rb and Rc. They are then used to compute a single 7-bit range code, R, as follows:

$$R = 25Ra + 5Rb + Rc + 1$$

R1 to R7 form the unsigned binary representation of this code which is transmitted LSB first (R1 to R7), followed by 4 protection bits R8 to R11 made up as follows:

$$R8 = (R3+R2+R1) \text{ MOD } 2$$

$$R9 = (R6+R5+R4) \text{ MOD } 2$$

$$R10 = (R7+R5+R4+R2+R1) \text{ MOD } 2$$

$$R11 = (R7+R6+R4+R3+R1) \text{ MOD } 2$$

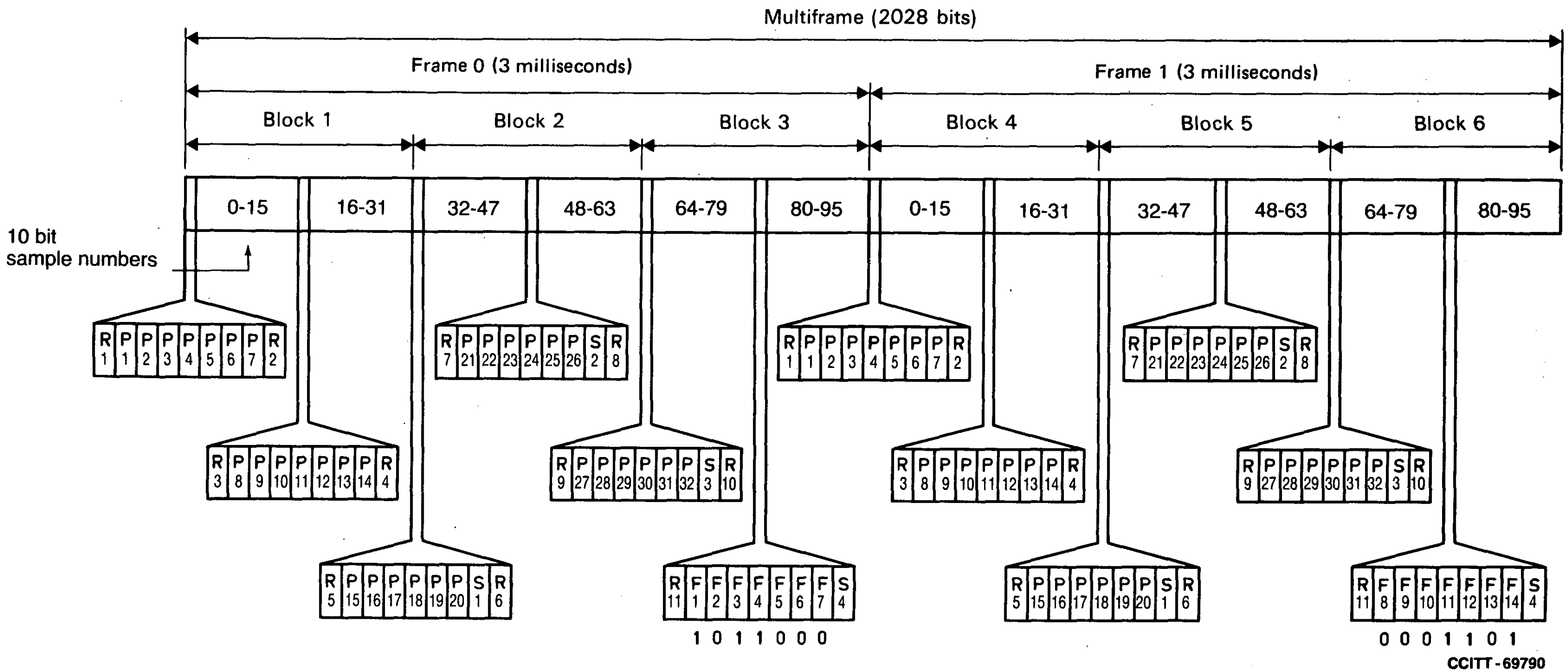
TABLE 2/J.41

Companding law – Two's complement coding

Range	Normalized analogue input	Normalized analogue output	Compressed digital code MSB LSB	Effective Resolution
4	+8176 to +8192 0 to +16 −16 to 0 −8192 to −8176	+8184 +8 −8 −8184	+511 (0111111111) 0 (0000000000) −1 (1111111111) −512 (1000000000)	10 bits
3	+4088 to +4096 0 to +8 −8 to 0 −4096 to −4088	+4092 +4 −4 −4092	+511 (0111111111) 0 (0000000000) −1 (1111111111) −512 (1000000000)	11 bits
2	+2044 to +2048 0 to +4 −4 to 0 −2048 to −2044	+2046 +2 −2 −2046	+511 (0111111111) 0 (0000000000) −1 (1111111111) −512 (1000000000)	12 bits
1	+1022 to +1024 0 to +2 −2 to 0 −1024 to −1022	+1023 +1 −1 −1023	+511 (0111111111) 0 (0000000000) −1 (1111111111) −512 (1000000000)	13 bits
0	+511 to +512 0 to +1 −1 to 0 −512 to −511	+511.5 +0.5 −0.5 −511.5	+511 (0111111111) 0 (0000000000) −1 (1111111111) −512 (1000000000)	14 bits

MSB Most significant bits.

LSB Less significant bits.



F Framing bit
P Sample parity bit
R Range code bit
S Signalling bit

FIGURE 3/J.41

Single channel frame format

5.2.4 *Sample error protection*

32 bits per frame are used for sample error detection on the basis of 1 parity bit per 3 samples. Odd parity is employed, i.e., the total number of data bits set to state 1, in the protected samples, plus the parity bit is always an odd number. The distribution of the parity bits within the frame and the allocation of the parity bits to the samples is shown in Figure 3/J.41 and Table 3/J.41, respectively. Only the 5 most significant bits of the samples are protected. In order to ensure that, if two sequential bits are corrupted, the error can still be detected by the parity checking process, the protected and unprotected bits of each sample are interleaved in descending and ascending order, respectively: 1, *10*, 2, *9*, 3, *8*, 4, *7*, 5, *6*. LSB is transmitted first and the bits underlined are those protected by the parity check. Error concealment should be used and can be achieved, for example, by replacing an erroneous sample value by a sample value calculated by linear interpolation between adjacent correct samples, or by extrapolation of the previous sample if the following sample is itself in error.

TABLE 3/J.41

Allocation of parity bits to the samples

Parity bit	Protects samples	Parity bit	Protects samples
1	3, 35, 66	17	14, 47, 78
2	8, 39, 71	18	18, 52, 83
3	12, 44, 75	19	23, 58, 89
4	17, 48, 79	20	27, 63, 95
5	21, 53, 84	21	15, 50, 80
6	26, 57, 88	22	22, 56, 85
7	31, 62, 92	23	29, 61, 91
8	19, 51, 82	24	0, 34, 65
9	24, 55, 86	25	5, 40, 70
10	28, 60, 90	26	10, 45, 74
11	32, 64, 94	27	7, 33, 68
12	2, 37, 69	28	13, 38, 76
13	6, 42, 73	29	16, 43, 81
14	11, 46, 77	30	20, 49, 87
15	4, 36, 67	31	25, 54, 93
16	9, 41, 72	32	1, 30, 59

This order has been chosen:

a) to spread each group of 3 protected samples as widely as possible;

b) to spread the 18 or 21 samples protected by each housekeeping word, with the maximum number of other samples between them.

5.2.5 *Single channel frame format*

Three 32 sample blocks, together with various housekeeping bits, form a single channel frame having a bit rate of 338 kbit/s and a duration of 3 ms. The number of bits per frame is therefore 3338 = 1014 bits, and these have been allocated as shown in Table 4/J.41. Figure 3/J.41 illustrates the frame arrangement for a single channel. Two frames are shown in Figure 3/J.41 and this format is referred to as a multiframe. Framing information is reversed, i.e. alternate bits in each frame of the multiframe.

5.2.6 *Two channels (stereo-pair) format*

Two separate 338 kbit/s streams are used to form a stereo-pair. Each of these bit streams is arranged as shown in Figure 3/J.41. The coders of the stereo-pair must be in synchronization. Care must be taken at the receiving end to compensate for any phase difference between the 2 channels.

5.2.7 *Synchronization of the 338-kbit/s stream*

The 338 kbit/s stream is synchronized to the coder sampling frequency.

TABLE 4/J.41

Bit allocation in the frame

	Frame allocation (bits/frame)	Bit rate per channel (kbit/s)
Sample words	960	320.0
Range coding (including error protection)	11	3.6
Sample word error protection	32	10.6
Signalling	4	1.3
Frame alignment	7	2.3
Total	1014	338.0

5.2.8 *Loss and recovery of frame alignment*

One of the following strategies is used:

a) Loss of single channel frame alignment shall occur if two or more consecutive frame alignment words are received incorrectly (for this purpose, bits F1 to F7, Frame 0, and bits F8 to F14, Frame 1, are both considered as frame alignment words: see Figure 3/J.41). An incorrect frame alignment signal is defined as one in which two or more bits are in error. Realignment shall be achieved when a single frame alignment signal is received correctly. If this word is a spurious code, a second attempt at realignment shall be made.

b) Only bits 1 to 10 of the 14 bit frame alignment word, derived from Frame 0 and Frame 1 (see Figure 3/J.41), are taken into account at the receiving end. Loss of frame alignment is assumed to have occurred when three consecutive frame alignment signals are received incorrectly in their predicted position. When frame alignment is assumed to have been lost, the automatic frame alignment recovery device will decide that alignment has been recovered when it registers two consecutive correct frame alignment signals.

5.3 *Conversion from 338 kbit/s to 384 kbit/s*

5.3.1 *Frame structure*

The frame structure (see Figure 4/J.41) with a nominal bit rate of 384 kbit/s and 613 bits in length is composed of:

– data input of 338 kbit/s;
– 63 redundancy bits for single error correction;
– bits for justification (J) and for identifiction of justification (IJ);
– the frame alignment (FA) signal.

The frame is arranged in 4 sections.

5.3.2 *Justification strategy*

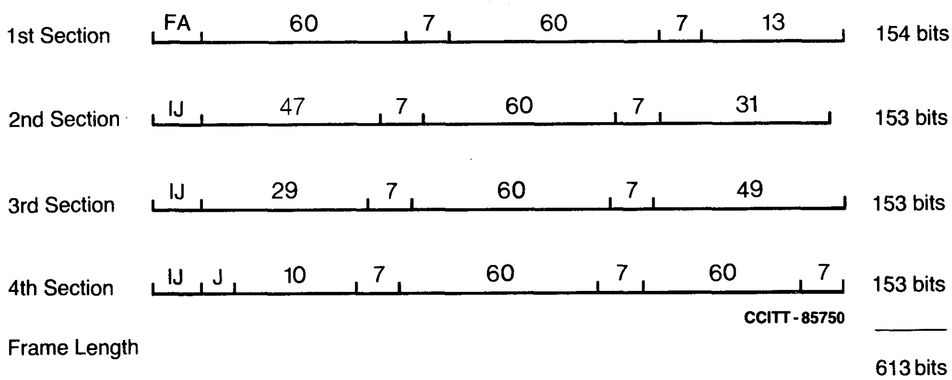
The first bits of sections 2, 3 and 4 are used to identify justification.

The 462nd bit of the frame (second bit of the fourth section) is the justification bit.

In cases of justification, the justification bit may assume any value.

Where there is no justification, the position of the justification bit is occupied by an information bit.

On the basis of a majority criterion, the demultiplexer recognizes that justification has taken place, if two out of three justification identification bits are in state 1.



FA Frame Alignment Word: 1 1 1 0 1 0 0
IJ Bit for identification of justification
J Bit for justification

FIGURE 4/J.41
338 kbit/s to 384 kbit/s frame structure

5.3.3 *Error protection for the 338-kbit/s stream*

A redundancy of 7 bits is calculated every 60 bits (see Figure 4/J.41), to allow for the correction of a single error (Hamming code 67, 60) on reception of each group of 67 bits. The first bit transmitted in a group of 60 bits is considered as the most significant bit of the group for the computation of the redundancy. The first bit transmitted among the 7 redundancy bits represents the most significant bit of the remainder.

The polynomial generator is equal to $x^7 + x + 1$.

5.3.4 *Synchronization of the 384 kbit/s stream*

At the output of the coder, the 384 kbit/s stream is synchronously locked to the subsequent primary hierarchical level digital stream.

5.3.5 *Loss and recovery of frame alignment*

Loss of frame alignment is assumed to have occurred when three consecutive frame alignment signals are incorrectly received in their predicted position. When frame alignment is assumed to have been lost, the automatic frame alignment recovery device will decide that alignment has been recovered when it registers two consecutive correct frame alignment signals.

5.4 *Digital interface at 384 kbit/s*

Under study.

5.5 *Fault conditions and consequent action*

Under study.

6 Digital interface between equipments using different coding standards

Under study.

References

[1] CCIR Recommendation *Transmission of analogue high-quality sound-programme signals on mixed analogue and digital circuits using 384 kbit/s channels*, Vol. XII, Rec. 660, UIT, Geneva, 1986.

ANNEX A

(to Recommendation J.41)

Coding methods for use by bilateral agreement

(see § 3.3 of this Recommendation)

TABLE A-1/J.41

Nominal bandwidth Pre/de-emphasis Overload point (Note 3) Sampling frequency	0.04-15 (Note 1) (Note 2) +12 32	0.04-15 (Note 1) (None) +12 32	kHz – dBm0s kHz
Companding law Bit rate reduction	13 segments 14/10	7 segments 13/11	– bits
Finest resolution and corresponding noise	14 –66	13 –55	bits/sample dBq0ps
Coarsest resolution at +9 dBm0s/f_0 [a] and corresponding noise	8 –30	10 –37	bits/sample dBq0ps
Resolution at +9 dBm0s/60 Hz and corresponding noise	10 –42	10 –37	bits/sample dBq0ps
Source coding Error protection Framing and signalling Service bit rate Transmission bit rate	320 16 0.66 336.66 336.66 [b] 384	352 32 0 384 384	kbit/s kbit/s kbit/s kbit/s kbit/s
Proposed by	Italy	Japan	

a) f_0 = zero loss frequency of pre-emphasis.

b) Dedicated frame.

Note 1 – Performance characteristics for analogue 15 kHz type sound-programme circuits are given in Recommendation J.21 and the proposals are assumed to meet these requirements with at least three codecs in tandem.

Note 2 – The pre-emphasis used is:

$$\text{insertion loss} = 10 \log \frac{8.5 + \left(\frac{f}{1900}\right)^2}{1 + \left(\frac{f}{650}\right)^2} \quad (f \text{ in Hz with } f_0 = 1900 \text{ Hz}).$$

Note 3 – This is defined as the maximum r.m.s. level of sinusoidal signal which does not cause clipping: this value is independent of frequency if analogue peak limiter and pre-emphasis are removed and replaced by zero dB loss; with pre-emphasis the overload level is defined at the zero dB loss frequency (f_0).

For detailed information, see Table I in CCIR Report 647.

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE MEDIUM QUALITY SOUND-PROGRAMME SIGNALS FOR TRANSMISSION ON 384-kbit/s CHANNELS

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

1 General

1.1 This Recommendation gives the characteristics of equipment for the coding of 7 kHz monophonic analogue sound-programme signals into a digital signal. Two monophonic digital signals can be combined to form a 384-kbit/s signal already specified in Recommendation J.41.

1.2 Equipment for coding of analogue sound-programme signals, as specified in this Recommendation, can be:

a) A stand alone encoder/decoder with a digital interface at 384 kbit/s. The encoder operation and the decoder operation may be performed in two separate equipments or in the same equipment.

b) A combined encoder-multiplex decoder-demultiplex with a digital interface at 1544 or 2048 kbit/s. The encoder-multiplex operation and the decoder-multiplex operation may be performed in two separate equipments or in the same equipment.

In case b) it is not mandatory to provide an external digital sound-programme access port at 384 kbit/s.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.23 (CCIR Recommendation 503) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

3 Method of encoding

3.1 The recommended encoding laws are as specified in [1].

3.2 These encoding laws are based on a uniformly quantized, 14-bit per sample PCM technique with companding and employ either:

a) eleven-segment 14 to 11 bit instantaneous A-law companding, or

b) five-range 14 to 10 bit near-instantaneous companding.

3.3 Equipment characteristics common to both methods of encoding are:

Nominal audio bandwidth:	0.05 to 7 kHz.
Audio interface:	see Recommendation J.23, § 2.
Sampling frequency:	16 ($1 \pm 5 \times 10^{-5}$) kHz.
Pre/de-emphasis:	Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

Note – Pre-emphasis and de-emphasis are not used by the Administrations of Canada, Japan and the United States of America on their national circuits and on international circuits between each other, but are used on international circuits to other countries.

4 Equipment using instantaneous companding

4.1 *Coding table*

4.1.1 The coding law is specified in Table 1/J.41.

4.1.2 The allocation of character signals (PCM code words) is also given in Table 1/J.41. Two variants (A and B) of character signals are allowed.

Note – In the case of digital interconnection between variants A and B, the conversion from one set of character signals to the other set in Table 1/J.41, can be done without any performance degradation. In the case of analogue interconnection, a reduction in the S/N ratio, in the order of 3 dB, is expected.

4.2 *Bit rates*

Nominal source coding bit rate (16 kHz × 11 bit/sample)	176 kbit/s
Error protection (16 kHz × 1 bit/sample)	16 kbit/s
Transmission bit rate per sound-programme signal	192 kbit/s
Channel bit rate for 2 sound-programme signals	384 kbit/s

4.3 *Overload level*

The overload level for a sine-wave signal at zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis is +15 dBm0s.

4.4 *Digital signal format*

The character signal bit sequences for variants A and B, are shown in Figure 1/J.41.

4.4.1 *Variant A*

When transmitting two monophonic digital signals as one 384 kbit/s signal, with respect to the code word interleaving shown in Figure 1/J.41, the first two 12 bit code words are allocated to 7 kHz channel No. 1 and the second two 12 bit code words are allocated to 7 kHz channel No. 2.

4.4.2 *Variant B*

The 12 bit code word assignments when transmitting two monophonic digital signals as one 384-kbit/s signal is under study.

4.5 *Bit error protection*

One parity bit is added to each 11-bit character signal.

4.5.1 *Variant A*

The five most important bits of each sample are protected against errors by means of a parity bit. In the converter of the transmitting part, the parity bit is added as the 12th bit to each code word. Its value is fixed so that the 6 bit parity block always contains only an odd number of one values. In order that even bit error structures can also result in parity violations, the protected and unprotected bits of each code word are interleaved in ascending and descending sequence, as shown in Figure 1/J.41.

4.5.2 *Variant B*

The added parity bit shall be based on the 7 most significant bits of the 11-bit PCM word. These are bits S, X, Y, Z, A, B, C. The parity of "ones" bit shall be *even.* Since the chord bits (X, Y, Z) always contain a one, the minimum number of ones per sample is 2, resulting in a minimum ones density of 1/6.

4.5.3 *Error concealment*

If a parity violation is detected, an error concealment technique should be applied (for instance, replacement by interpolation, extrapolation or repetition). For multiple parity violation (error bursts), a muting technique should be applied.

4.6 *Digital interface at 384 kbit/s*

Under study (see Recommendations G.735 and G.737).

4.7 *Synchronization*

The coding equipment operates in synchronism with the clock of subsequent multiplex equipment or the network clock. In cases where the digital interface is provided, bit and byte (24 bit, as shown in Figure 1/J.41) timing information is required.

Variant A: A solution for synchronous access is given in the Recommandations G.735 and G.737.

Variant B: The solution for synchronous access is under study.

4.8 *Fault condition and consequent actions*

4.8.1 *Variant A*

Where a 384-kbit/s digital interface is provided, the same principles for fault conditions and subsequent actions as those outlined in Recommendation G.732, should be followed.

4.8.2 *Variant B*

Under study.

5 **Equipment using near-instantaneous companding**

5.1 *Introduction*

The equipment described in this section uses the near-instantaneous method of companding in the coding of medium quality sound-programme signals into digital form.

A two-stage process is used in the encoding equipment:

a) Conversion of a 7 kHz channel into a 169 kbit/s stream.

   Note – The value of 169 kbit/s has been chosen to allow for the possible multiplexing of 12 channels into a 2048 kbit/s dedicated frame format.

b) Asynchronous insertion of two synchronous 169 kbit/s streams into a 384 kbit/s stream.

   Note – The asynchronous insertion of two synchronous 169 kbit/s streams into a 384 kbit/s stream allows the use, at the encoder location, of a clock not necessarily synchronous to the network clock. It can be advantageous when the encoder equipment and the insertion equipment (see Recommendations G.735 and G.737) are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

5.2 *Conversion from 7 kHz to 169 kbit/s and constitution of the 338-kbit/s signal*

5.2.1 *Overload level*

The overload level for a sine-wave signal at the zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis circuit is +12 dBm0s.

5.2.2 *Companding*

The same near-instantaneous companding procedure with a block of 32 samples (2 ms) as described in § 5.2.2 of Recommendation J.41, is used. The character signal is coded in 2's complement form.

5.2.3 *Constitution of the 338-kbit/s signal*

Two 7-kHz channels (C1 and C2) are contained in one 338-kbit/s stream. The frame structure of the 338 kbit/s stream is defined in § 5.2.5 and in Figure 3/J.41. The following numbering of the samples within a given multiframe is defined as follows (see Figure 3/J.41):

Sample n of the multiframe is sample $(n - 96i)$ of frame i

$$0 \leqslant n \leqslant 191 \qquad i = 0 \text{ or } 1$$

Using the above notation, the following relationship between the bits of the 338 kbit/s multiframe and channels C1 and C2 can be defined:

Sample $2n$ of the multiframe corresponds to sample n of channel C1

Sample $(2n + 1)$ of the multiframe corresponds to sample n of channel C2

$$0 \leqslant n \leqslant 95$$

Range coding information associated with block $(2n - 1)$ of the multiframe is allocated to block n of channel C1 (derived from C1 samples in blocks $(2n - 1)$ and $(2n)$ of the multiframe).

Range coding information associated with block $(2n)$ of the multiframe is allocated to block n of channel C2 (derived from C2 samples in blocks $(2n - 1)$ and $(2n)$ of the multiframe).

$$1 \leqslant n \leqslant 3$$

The range coding information and its protection, the sample format and the sample error protection are defined and transmitted as specified in this Recommendation and in §§ 5.2.3 to 5.2.5 of Recommendation J.41.

The criteria for loss and recovery of frame alignment at 338 kbit/s is defined in § 5.2.8 of Recommendation J.41.

5.3 *Conversion from 338 kbit/s to 383 kbit/s*

See Recommendation J.41, § 5.3.

5.4 *Digital interface at 384 kbit/s*

Under study.

5.5 *Fault conditions and consequent action*

Under study.

6 Digital interface between equipments using different coding standards

Under study.

References

[1] CCIR Recommandation *Transmission of analogue high-quality sound-programme signals on mixed analogue and digital circuits using 384 kbit/s channels*, Vol. XII, Rec. 660, ITU, Geneva, 1986.

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE HIGH QUALITY SOUND PROGRAMME SIGNALS FOR TRANSMISSION ON 320 kbit/s CHANNELS[1)]

(Melbourne, 1988)

1 General

1.1 This Recommendation gives the characteristics of equipment for the coding of 15 kHz monophonic analogue sound-programme signals into a digital signal of 320 kbit/s. For stereophonic operation, two monophonic digital codecs can be utilized. Two monophonic digital signals that form a stereophonic signal should be routed together over the same transmission systems (path) to avoid difference in transmission delay.

1.2 Equipment for coding of analogue sound-programme signals can be:

a) A stand-alone encoder/decoder with a digital interface at 320 kbit/s. The encoder operation and the decoder operation may be performed in two separate equipments or in the same equipment.

b) A combined encoder-multiplex/decoder-demultiplex with a digital interface at 1544 or 2048 kbit/s. The encoder-multiplex operation and the decoder-demultiplex operation may be performed in two separate equipments or in the same equipment.

In case b), it is not mandatory to provide an external access at 320 kbit/s.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.21 (CCIR Recommendation 505) are not exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

3 Method of encoding

3.1 The encoding method is based on a uniformly quantized 14-bit per sample PCM technique with differential 14- to 9.5-bit near instantaneous companding.

3.2 Fundamental characteristics of the equipment are:

Nominal audio bandwidth:	0.04 to 15 kHz.
Audio interface:	see Recommendation J.21, § 2.
Sampling frequency (CCIR Recommendation 606):	$32\ (1 + 5 \times 10^{-5})$ kHz,
Pre/de-emphasis:	Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

4 Characteristics of the equipment

4.1 *Introduction*

The equipment being described uses the differential near-instantaneous method of companding in the coding of high-quality sound-programme signals into digital form.

A two-stage process is used in the encoding equipment:

a) conversion of a 15 kHz channel into a 316 kbit/s stream;

b) asynchronous insertion of the 316 kbit/s stream into a 320 kbit/s stream;

---

1) Digital interfaces between Administrations which have adopted different systems should, if a bilateral agreement is not reached, operate at 384 kbit/s (H_0 channel) and carry signals encoded, according to Recommendation J.41, § 4. Any necessary transcoding will be carried out by Administrations using the system specified in this Recommendation.

Note – The asynchronous insertion of the 316 kbit/s stream into a 320 kbit/s stream allows the use, at the encoder location, of a clock not necessarily synchronous to the network clock. It can be advantageous when the encoder equipment and the insertion equipment are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

4.2 *Conversion from 15 kHz to 316 kbit/s*

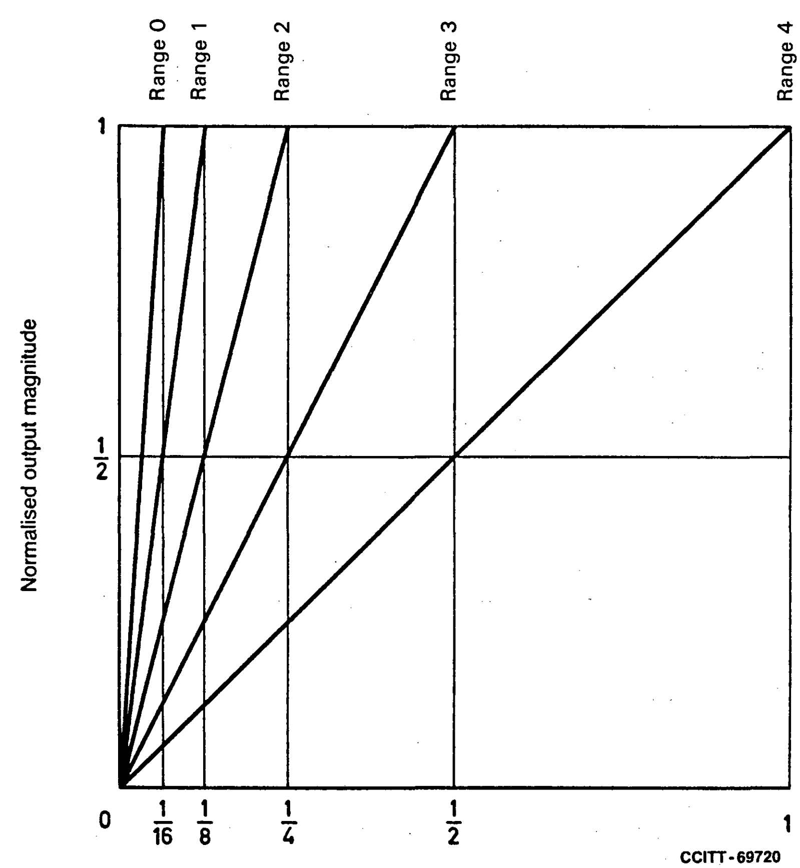
4.2.1 *Overload level*

The overload level for a sine-wave signal at the zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis circuit is +12 or +15 dBm0s.

4.2.2 *Companding*

Differential near-instantaneous companding is used to achieve a data rate reduction from 14 bits/sample to 9.5 bit/sample. The process of differential near-instantaneous companding is subdivided into the following stages:

a) near-instantaneous companding to achieve a data rate reduction from 14 bits/sample to 10 bits/sample as in § 5 of Recommendation J.41. The system coded a bloc of 32 samples into one of 5 gain ranges, according to the highest value sample in the block. The companding characteristic is shown diagramatically in Figure 1/J.43 and the parameters are specified in Table 1/J.43;



Normalised input magnitude

FIGURE 1/J.43

Companding characteristic

TABLE 1/J.43

14 to 10 bit near-instantaneous companding law

Range	Normalized analogue input	Normalized analogue output	Compressed digital code MSB LSB		Effective resolution
4	+8176 to +8192	+8184	+511	(0111111111)	10 bits
	0 to +16	+8	0	(0000000000)	
	−16 to 0	−8	−1	(1000000000)	
	−8192 to −8176	−8184	−512	(1111111111)	
3	+4088 to +4096	+4092	+511	(0111111111)	11 bits
	0 to +8	+4	0	(0000000000)	
	−8 to 0	−4	−1	(1000000000)	
	−4096 to −4088	−4092	−512	(1111111111)	
2	+2044 to +2048	+2046	+511	(0111111111)	12 bits
	0 to +4	+2	0	(0000000000)	
	−4 to 0	−2	−1	(1000000000)	
	−2048 to −2044	−2046	−512	(1111111111)	
1	+1022 to +1024	+1023	+511	(0111111111)	13 bits
	0 to +2	+1	0	(0000000000)	
	−2 to 0	−1	−1	(1000000000)	
	−1024 to −1022	−1023	−512	(1111111111)	
0	+511 to +512	+511.5	+511	(0111111111)	14 bits
	0 to +1	+0.5	0	(0000000000)	
	−1 to 0	−0.5	−1	(1000000000)	
	−512 to −511	−511.5	−512	(1111111111)	

MSB Most significant bit.

LSB Least significant bit.

b) division of a sequence of samples $x(n)$ into two sequences one of which is a sequence of odd samples $x(2 - 1)$ and the other is a sequence of even samples $x(2n)$. Calculation of differential even samples $\Delta(2n)$ by the formula

$$\Delta(2n) = x(2n) - \frac{x(2n + 1) + x(2n - 1)}{2} \tag{1}$$

c) additional near-instantaneous companding of the differential samples $\Delta(2n)$ to achieve a data rate reduction from 14 bits/sample to 9 bits/sample. The system codes a block of 16 even samples into one of 3 additional gain ranges, according to the highest value sample in the block. The companding characteristic is shown diagrammatically in Figure 2/J.43 and the parameters are specified in Table 2/J.43.

After multiplexing the odd samples $x(2n - 1)$ represented by a compressed code of 10 bits per sample and the differential even samples $\Delta(2n)$ additionally represented by a compressed code of 9 bits per sample, an average of 9.5 bits per sample is obtained.

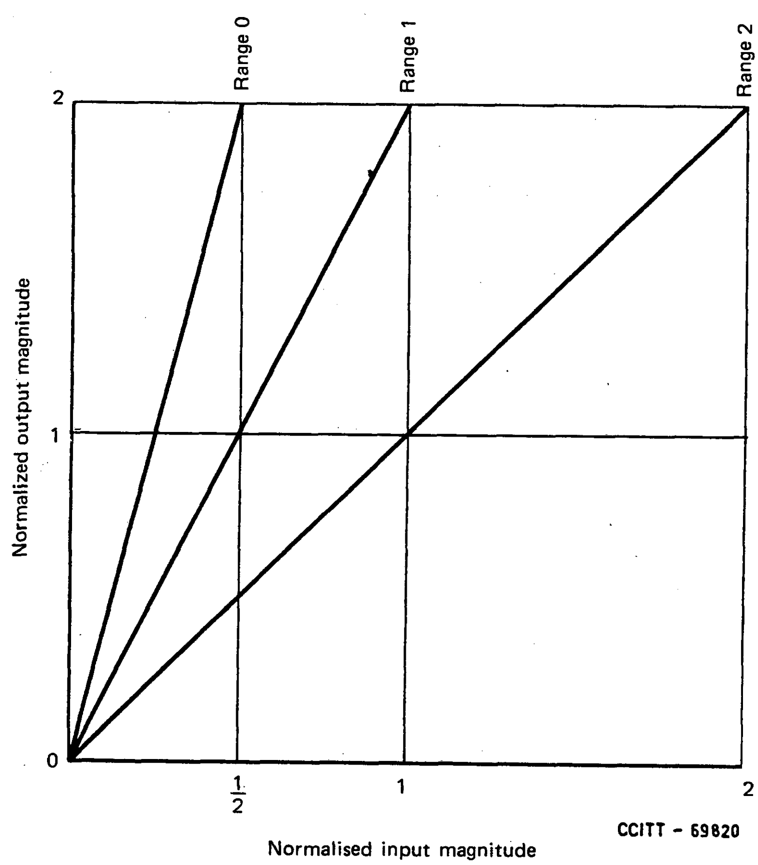


FIGURE 2/J.43

Companding characteristic

TABLE 2/J.43

14 to 9.0 bit near-instantaneous companding law

Range		Normalized input	Normalized output	Compressed digital code MSB LSB		Effective resolution
	2	+16 320 to +16 384	+16 352	+255	(011111111)	8 bits
		0 to +64	+32	0	(000000000)	
		−64 to 0	−32	−1	(100000000)	
		−16 384 to −16 320	−16 352	−256	(111111111)	
4	1	+8160 to +8192	+8176	+255	(011111111)	9 bits
		0 to +32	+16	0	(000000000)	
		−32 to 0	−16	−1	(100000000)	
		−8190 to −8160	−8176	−256	(111111111)	
	0	+4080 to +4096	+4088	+255	(011111111)	10 bits
		0 to +16	+8	0	(000000000)	
		−16 to 0	−8	−1	(100000000)	
		−4096 to −4080	−4088	−256	(111111111)	
	2	+8160 to +8192	+8176	+255	(011111111)	9 bits
		0 to +32	+16	0	(000000000)	
		−32 to 0	−16	−1	(100000000)	
		−8192 to −8160	−8176	−256	(111111111)	
3	1	+4080 to +4096	+4088	+255	(011111111)	10 bits
		0 to +16	+8	0	(000000000)	
		−16 to 0	−8	−1	(100000000)	
		−4096 to −4080	−4088	−256	(111111111)	
	0	+2040 to +2048	+2044	+255	(011111111)	11 bits
		0 to +8	+4	0	(000000000)	
		−8 to 0	−4	−1	(100000000)	
		−2048 to −2040	−2044	−256	(111111111)	
	2	+4080 to +4096	+4088	+255	(011111111)	10 bits
		0 to +16	+8	0	(000000000)	
		−16 to 0	−8	−1	(100000000)	
		−4096 to −4080	−4088	−256	(111111111)	
2	1	+2040 to +2048	+2044	+255	(011111111)	11 bits
		0 to +8	+4	0	(000000000)	
		−8 to 0	−4	−1	(100000000)	
		−2048 to −2040	−2044	−256	(111111111)	
	0	+1020 to +1024	+1022	+255	(011111111)	12 bits
		0 to +4	+2	0	(000000000)	
		−4 to 0	−2	−1	(100000000)	
		−1024 to −1020	−1022	−256	(111111111)	

TABLE 2/J.43 *(continuation)*

Range		Normalized input	Normalized output	Compressed digital code MSB LSB		Effective resolution
1	2	+2040 to +2048	+2044	+255	(011111111)	11 bits
		0 to +8	+4	0	(000000000)	
		−8 to 0	−4	−1	(100000000)	
		−2048 to −2040	−2044	−256	(111111111)	
	1	+1020 to +1024	+1022	+255	(011111111)	12 bits
		0 to +4	+2	0	(000000000)	
		−4 to 0	−2	−1	(100000000)	
		−1024 to −1020	−1022	−256	(111111111)	
	0	+510 to +512	+511	+255	(011111111)	13 bits
		0 to +2	+1	0	(000000000)	
		−2 to 0	−1	−1	(100000000)	
		−512 to −510	−511	−256	(111111111)	
0	2	+1020 to +1024	+1022	+255	(011111111)	12 bits
		0 to +4	+2	0	(000000000)	
		−4 to 0	−2	−1	(100000000)	
		−1024 to −1020	−1022	−256	(111111111)	
	1	+510 to +512	+511	+255	(011111111)	13 bits
		0 to +2	+1	0	(000000000)	
		−2 to 0	−1	−1	(100000000)	
		−512 to −510	−511	−256	(111111111)	
	0	+255 to +256	+255.5	+255	(011111111)	14 bits
		0 to +1	+0.5	0	(000000000)	
		−1 to 0	−0.5	−1	(100000000)	
		−256 to −255	−255.5	−256	(111111111)	

MSB Most significant bit.

LSB Least significant bit.

4.2.3 *Range coding*

The five possible values of a gain range for a block of 32 samples and three possible values of an additional gain range for differential even samples of this block produce 15 possible values of a complex gain range which is represented by a four-bit code word. Complex range codes are shown in Table 3/J.43.

TABLE 3/J.43

Additional \ Basic	0	1	2	3	4
0	1110	1101	1100	1011	1010
1	1001	1000	0111	0110	0101
2	0100	0011	0010	0001	0000

For error-protected transmission, two code words of the complex gain range (which correspond to two blocks) are combined into one 8-bit code word which is coded by a Hamming code (12,8). This code makes it possible to correct all singla errors in the code word of the complex gain range.

A code word of 12 bits comprising 8 bits of the gain range of two blocks and 4 check bits is transmitted in a cycle having a duration of 2 ms (ee Figure 3/J.43). The first 8 bits R1 to R8 correspond to two complex code words. The last four bits (R9 to R12) are check bits. They are determined as follows:

$$
\begin{aligned}
\overline{R}_9 &= R_1 \oplus R_2 \oplus R_3 \oplus R_7 \\
\overline{R}_{10} &= R_1 \oplus R_4 \oplus R_5 \oplus R_7 \oplus R_8 \\
\overline{R}_{11} &= R_2 \oplus R_4 \oplus R_6 \oplus R_7 \oplus R_8 \\
\overline{R}_{12} &= R_3 \oplus R_5 \oplus R_6 \oplus R_8
\end{aligned}
\qquad (2)
$$

Modulo 2 addition is designated by $\oplus$ and inversion of bit R is designated by $\overline{R}$.

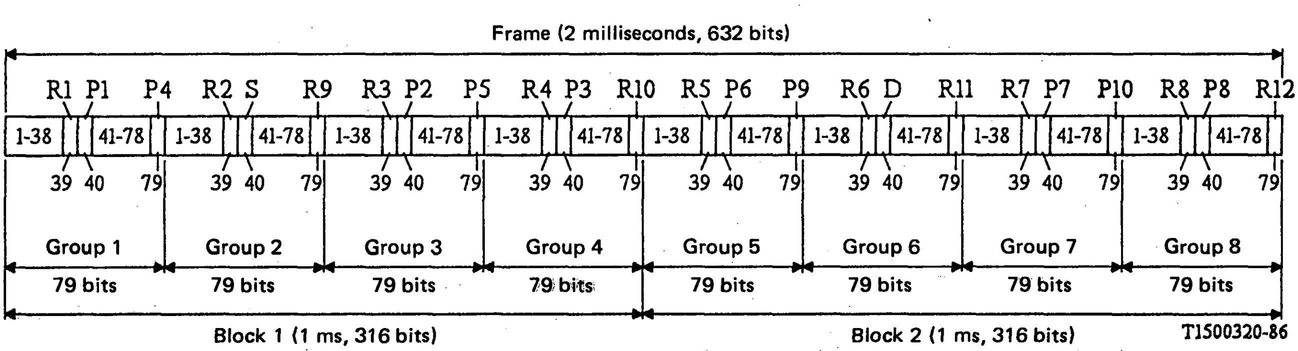


FIGURE 3/J.43

Single channel frame format

4.2.4 *Sample error protection*

The 5 most significant bits of 10-bit samples and 4 most significant bits of 9-digit samples are protected. One parity bit is generated for 5 most significant bits of each 10-digit sample. A parity bit is also generated for 4 most significant bits of each pair of 9-digit samples. A total of 24 bits are thus generated for a block of 32 samples. These 24 parity bits undergo error protection by means of a cyclic code (29,24). The code (29,24) is a shortened Hamming code (31,26). The polynomial generator of the code (29,24) is:

$$F(x) = x^5 + x^2 + 1 \tag{3}$$

To the receiving end only the check bits of the cyclic code (29,24) are sent, since 24 parity bits are reproduced according to the received sample. Thus, 5 protection bits correspond to a block of 32 samples, 10 protection bits for two blocks are transmitted in a cycle having a duration of 2 ms (see Figure 3/J.43).

In order to correct 8-bit error bursts, samples from four blocks are interleaved. Interleaving of samples from four blocks is shown in Table 6/J.43.

Note – Interleaving of samples from four adjacent blocks is an effective measure of error protection. Samples of a sound-programme signal are transmitted over the primary digital path in octets (8-bit words). Such samples interleaving ensures correction of erroneous octets.

4.2.5 *316 kbit/s channel frame*

The frame has a duration of 2 ms which corresponds to two 32-sample blocks. The frame duration of 2 ms equals to the multiframe duration of the primary digital multiplex equipment. Due to this coincidence of durations a possibility is provided to use the multiframe alignment signal of the primary digital multiplex equipment. With a digital rate of 316 kbit/s and a duration of 2 ms, the frame comprises 632 bits divided into 8 groups of 79 bits each. Bit allocation in the frame is shown in Table 4/J.43.

TABLE 4/J.43

Bit allocation in the frame

	Frame allocation (bits/frame)	Bit rate per channel (kbit/s)
Samples	608	304
Range code	8	4
Check bits of a range code	4	2
Check bits of samples	10	5
Signalling and data bits	2	1
Total	632	316

The frame structure is shown in Figure 3/J.43 and Table 5/J.43. Table 6/J.43 shows the allocation of sample bits in a group, which provides for interleaving of samples from four blocks (see § 4.2.4 above) and interleaving of bits from different samples.

Note – As can ben seen from Table 6/J.43, an 8-bit error burst disintegrates into isolated single errors. For example, when errors occur in bits 1 to 8 of the first group (1 = 1) of the N-th frame, errors appear in the next four samples: the first sample of the first block frame N – 1 (n = 1, k = 1), the second sample of the second block of frame N – 1 (n = 2, k = 2), the second sample of the first block of frame N – 2 (n = 2, k = 1), the first sample of the second block of frame N – 2 (n = 1, k = 2). These isolated errors are corrected by means of interpolation.

TABLE 5/J.43

316 kbit/s frame structure

Data type	Bit number in a group	Group number in a cycle
Sample bits	1-38; 41 to 78	1 to 8
Bits of the code words of the complex gain range of the 1st block (R1 to R4)	39	1 to 4
Bits of the code words of the complex gain range of the 2nd block (R5 to R8)	39	5 to 8
Check bits of two complex gain ranges (R9 to R12)	79	2, 4, 6, 8
Check bits of the samples of the 1st block (R1 to R5)	40 79	1, 3, 4 1, 3
Check bits of the samples of the 2nd block (R6 to R10)	40 79	5, 7, 8 5, 7
Signalling and check bits (S)	40	2
Data bits (D)	40	6

TABLE 6/J.43

Bit number in sample n of block k								Bit number in group 1 of frame N
N − 1				N − 2				
k = 1		k = 2		k = 1		k = 2		
n = 41 − 3	n = 41 − 1	n = 41 − 2	n = 41	n = 41 − 2	n = 41	n = 41 − 3	n = 41 − 1	
1.6		1.6		1.6		1.6		1 to 8
2.7		2.7		2.7		2.7		9 to 16
3.8		3.8		3.8		3.8		17 to 24
4.9		4.9		4.9		4.9		25 to 32
5.10		5		5		5.10		33 to 38
	1.6		1.6		1.6		1.6	41 to 48
	2.7		2.7		2.7		2.7	49 to 56
	3.8		3.8		3.8		3.8	57 to 64
	4.9		4.9		4.9		4.9	65 to 72
	5.10		5		5		5.10	73 to 78

N Number of the current frame: N = 0, ± 1, ± 2, . . .

1 Number of the group in the frame: 1 = 1, 2, . . ., 8

k Number of the block in the frame: k = 1, 2

n Number of the sample in the block: n = 1, 2, . . ., 32

4.2.6 *Synchronization of the 316 kbit/s stream*

The 316 kbit/s stream is synchronized to the coder sampling frequency.

4.2.7 *Frame alignment of the 316/s stream*

For the frame alignment the synchronizing properties of the Hamming code (12,8) are utilized and a special frame alignment signal is not employed. The signal R1-R12 is used as a frame alignment signal. In the frame alignment signal receiver the relationships (2) from § 4.2.3 are checked. The lock-in time of such a frame alignment signal is equal to the lock-in time of an 4-bit frame alignment signal.

4.3 *Asynchronous insertion of the 316 kbit/s signal into a 320 kbit/s stream*

4.3.1 *Frame structure of the 320 kbit/s signal*

The 320 kbit/s signal is composed of a data signal fo 316 kbit/s and a justification signal of 4 kbit/s. The 320 kbit/s stream is divided into groups of 80 bits, 79 bits being data bits and the 80th bit being the bit of the justification signal.

4.3.2 *Justification method*

A method of positive-negative justification with two-command control is used for the rate justification. The justification signal consists of justification commands and a data signal transmitted in the case of negative justification. The frame of the justification signal consists of 4 bits. The justification commands are transmitted by three bits 111 or 000. The same commands are used for frame alignment of the justification signal. The 4th bit in the frame is used to transmit a data signal in the case of negative justification.

4.3.3 *Allocation of the justification signal in the frame of the primary digital multiplex equipment*

Bits of the justification signal are allocated in the frames of the primary digital multiplex equipment, which comprise the frame alignment signal in the channel time slot 0.

In the frame of the primary digital multiplex equipment, which comprises the justification bit, this bit is the last of all bits of the 320 kbit/s signal which are allocated in the given frame, that is, the justification bit is the most remote bit from the frame alignment signal of the primary digital multiplex equipment.

4.4 *Digital interface between the encoder equipment and the insertion equipment*

Under study.

4.5 *Fault conditions and consequent actions*

Under study.

5 Digital interface between equipments using different coding standards

Under study.

Recommendation J.44

CHARACTERISTICS OF EQUIPMENT FOR THE CODING OF ANALOGUE MEDIUM QUALITY SOUND-PROGRAMME SIGNALS FOR TRANSMISSION ON 320 kbit/s CHANNELS[1)]

(Melbourne, 1988)

1 General

1.1 This Recommendation gives the characteristics of equipment for the coding of 7 kHz monophonic analogue sound-programme signals into a digital signal. Two monophonic digital signals can be combined to form a 320 kbit/s signal having a structure specified in Recommendation J.43.

1.2 Equipment for coding of analogue sound-programme signals, as specified in this Recommendation, can be:

a) A stand-alone encoder/decoder with a digital interface at 320 kbit/s. The encoder operation and the decoder operation may be performed in two separate equipments or in the same equipment.

b) A combined encoder-multiplex/decoder-demultiplex with a digital interface at 1544 or 2048 kbit/s. The encoder-multiplex operation and the decoder-demultiplex operation may be performed in two separate equipments or in the same equipment.

In case b) it is not mandatory to provide an external access at 320 kbit/s.

2 Transmission performance

The transmission performance per encoder/decoder pair shall be such that the limits specified in Recommendation J.23 (CCIR Recommendation 503) are exceeded by three encoder/decoder pairs connected in tandem at audio frequencies.

3 Method of encoding

3.1 The encoding method is based on a uniformly quantized 14-bit per sample technique with differential 14 to 9.5-bit near instantaneous companding.

3.2 Fundamental characteristics of the equipment are:

Nominal audio bandwidth:	0.05 to 7 kHz.
Audio interface:	see Recommendation J.23, § 2.
Sampling frequency:	$16 (1 \pm 5 \times 10^{-5})$ kHz.
Pre/de-emphasis:	Recommendation J.17 with 6.5 dB attenuation at 800 Hz.

4 Characteristics of the equipment

4.1 *Introduction*

The equipment described in this section uses the differential near-instantaneous method of companding in the coding of medium quality sound-programme signals into digital form.

A two-stage process is used in the encoding equipment:

a) conversion of a 7 kHz channel into a 158 kbit/s stream;

b) asynchronous insertion of two synchronous in-phase 158 kbit/s streams into a 320 kbit/s stream.

1) Digital interface between Administrations which have adopted different systems should, if a bilateral agreement is not reached, operate at 384 kbit/s (H_0 channel) and carry signals encoded according to Recommendation J.42, § 4. Any necessary transcoding will be carried out by Administrations using the system specified in this Recommendation.

Note – The asynchronous insertion of two asynchronous in-phase 158 kbit/s streams into a 320 kbit/s stream allows the use, at the encoder location, of a clock not necessarily synchronous to the network clock. It can be advantageous when the encoder equipment and the insertion equipment are located in different places, and when the transmission link between them is unidirectional,

and the reverse processes in the decoding equipment.

4.2 *Conversion from 7 kHz to 158 kbit/s and constitution of the 316 kbit/s signal*

4.2.1 *Overload level*

The overload level for a sine-wave signal at the zero dB insertion loss frequency (2.1 kHz) of the pre-emphasis circuit is +12 or +15 dBm0s.

4.2.2 *Companding*

The same differential near-instantaneous companding procedure with a block of 32 samples (2 ms), as described in § 4.2.2 of Recommendation J.43, is used.

4.2.3 *Range coding*

The same range coding for a block of 32 samples (2 ms), as described in § 4.2.3 of Recommendation J.43, is used.

4.2.4 *Sample error protection*

The same sample error protection for a block of 32 samples (2 ms), as described in § 4.2.4 of Recommendation J.43, is used.

4.2.5 *316 kbit/s channel frame*

Two 7 kHz channels (C1 and C2) are contained in one 316 kbit/s stream. The frame structure of the 316 kbit/s stream is described in § 4.2.5 of Recommendation J.43. The first block (k = 1) of each frame corresponds to channel C1 and the second block (k = 2) of each frame corresponds to channel C2.

4.3 *Asynchronous insertion of the 316 kbit/s signal into a 320 kbit/s stream*

See § 4.3 of Recommendation J.43.

4.4 *Digital interface between the encoder equipment and the insertion equipment*

Under study.

4.5 *Fault conditions and consequent actions*

Under study.

5 Digital interface between equipments using different coding standards

Under study.

SECTION 5

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SECTION 6

CHARACTERISTICS OF CIRCUITS FOR TELEVISION TRANSMISSIONS

Former Recommendations J.61 and J.62 of Volume III-2 of the *Orange Book* have been cancelled. The corresponding CCIR Recommendations have been combined into CCIR Recommendation 567, which refers to all television standards and colour systems. This Recommendation 567 and some other texts from CCIR may be very useful for television transmissions via cable, and reference is given to the following CCIR Recommendations, published in Volume XII (of the XV Plenary Assembly of the CCIR), ITU, Geneva, 1982.

Recommendation J.61

TRANSMISSION PERFORMANCE OF TELEVISION CIRCUITS DESIGNED FOR USE IN INTERNATIONAL CONNECTIONS

(Geneva, 1982)

(See CCIR Recommendation 567)

Recommendation J.62

SINGLE VALUE OF THE SIGNAL-TO-NOISE RATIO FOR ALL TELEVISION SYSTEMS

(Geneva, 1982)

(See CCIR Recommendation 568)

Recommendation J.63

INSERTION OF TEST SIGNALS IN THE FIELD-BLANKING INTERVAL OF MONOCHROME AND COLOUR TELEVISION SIGNALS

(Geneva, 1982)

(See CCIR Recommendation 473)

Recommendation J.64

DEFINITIONS OF PARAMETERS FOR SIMPLIFIED AUTOMATIC MEASUREMENT OF TELEVISION INSERTION TEST SIGNALS

(Geneva, 1982)

(See CCIR Recommendation 569)

Recommendation J.65

STANDARD TEST SIGNAL FOR CONVENTIONAL LOADING OF A TELEVISION CHANNEL

(Geneva, 1982)

(See CCIR Recommendation 570)

Recommendation J.66

TRANSMISSION OF ONE SOUND PROGRAMME ASSOCIATED WITH ANALOGUE TELEVISION SIGNAL BY MEANS OF TIME DIVISION MULTIPLEX IN THE LINE SYNCHRONIZING PULSE

(Geneva, 1982)

(See CCIR Recommendation 572)

SECTION 7

GENERAL CHARACTERISTICS OF SYSTEMS FOR TELEVISION TRANSMISSION OVER METALLIC LINES AND INTERCONNECTION WITH RADIO-RELAY LINKS

Recommendation J.73 [1)]

USE OF A 12-MHz SYSTEM FOR THE SIMULTANEOUS TRANSMISSION OF TELEPHONY AND TELEVISION

(amended at Geneva, 1964 and 1980)

The 12-MHz system on 2.6/9.5-mm coaxial cable pairs and the 12-MHz system on 1.2/4.4-mm coaxial pairs are defined in Recommendations G.332 [1] and G.345 [2] respectively.

Any 12-MHz system equipped for television transmission should be capable of transmitting the signals used in all the television systems defined in CCIR having a video bandwidth up to 5.5 MHz if necessary, by means of the switching (in terminal equipments only) of certain components.

1 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 5.5 MHz wide, whatever television system is to be used. The level 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 to these figures).

2 Modulation ratio

Amplitude modulation has to be used. The modulation ratio has to be higher than 100% (as indicated in Figure 3/J.73), so that, when the carrier is modulated by a signal corresponding to blanking level, its amplitude be 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 a luminance bar (see CCIR Recommendation 567, Annex 1 to Part C, test signal element B2) 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).

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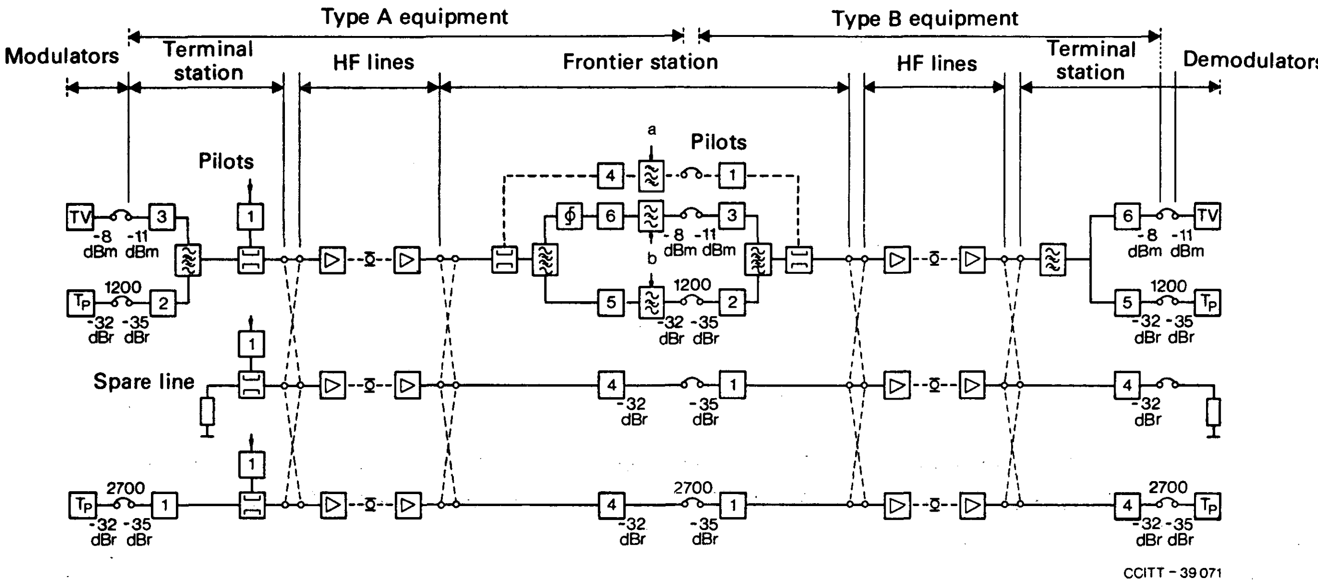
1) Recommendations J.71 and J.72 of Volume III-2 of the *Orange Book* have been deleted.



1 4 Pre-emphasis (respectively de-emphasis) for telephony 2700 channels

2 5 Pre-emphasis (respectively de-emphasis) for telephony 1200 channels } Simultaneous transmission

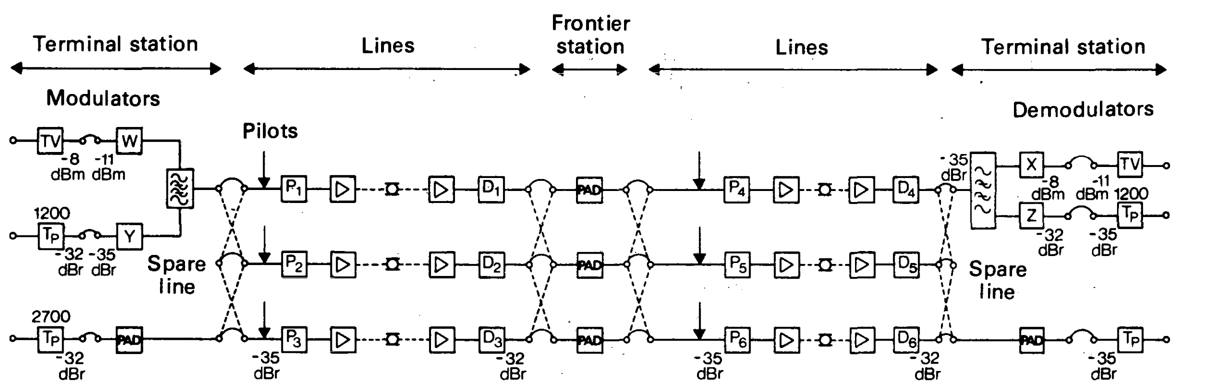
3 6 Pre-emphasis (respectively de-emphasis) for television

a Pilot pass-filter

b Pilot stop-filter

FIGURE 1/J.73

General case of interconnection of 12-MHz lines



Complement of line

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

Y Z Differential pre-emphasis and de-emphasis for telephony 1200 channels } Common for all types of 12 MHz line

P1 D1 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

Note 1 – Interconnection of pilots, e.g. blocking and re-injecting or by-passing, should be agreed between Administrations.

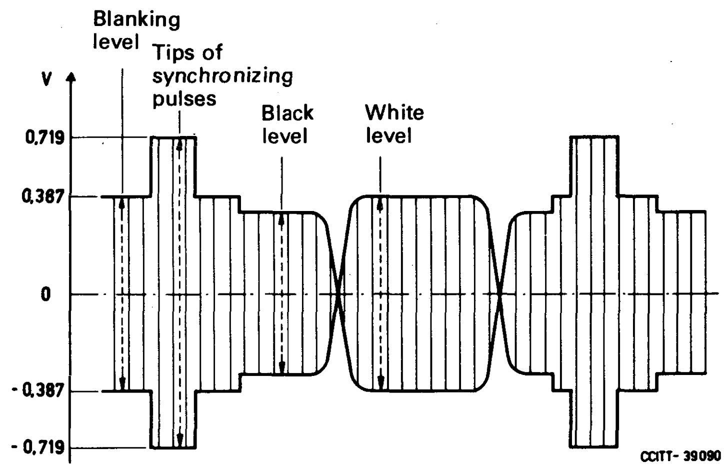
Note 2 – The level of the line pilots is fixed at – 10 dBm0 for the all-telephony case. When the line is used to transmit telephony and television simultaneously, different values of pre-emphasis may be rquired; although the absolute levels of the pilots will remain the same, they may no longer be at – 10 dBm0.

Note 3 – The television levels shown, are those of the modulated carrier relative to the white or blanking level (0 dBm) of the idealized reference signal described in § 2 of this Recommendation. This means that the television levels are indicated in dBm values.

Note 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



Note – The voltages shown are the values measured at a zero relative level point for television transmission in the 12 MHz system.

FIGURE 3/J.73

Envelope of carrier modulated by Test Signal No. 2

3 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 4/J.73 shows the frequency arrangement recommended for television transmission over the 12 MHz system.

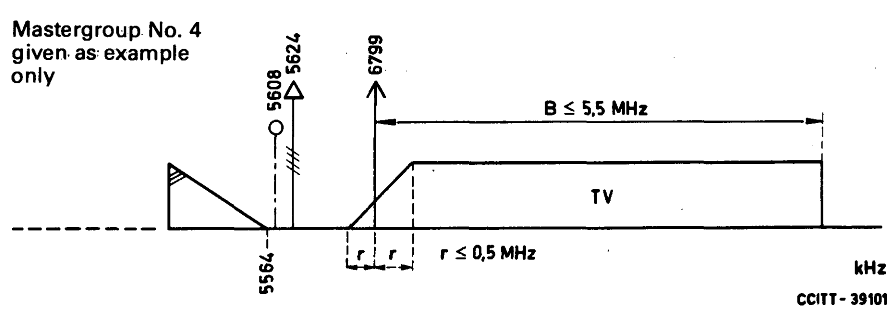


FIGURE 4/J.73

Frequency allocation for television on a 12 MHz system

4 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 [3], 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.

5 Interference

Recommendation J.61 (equal to CCIR-Recommendation 567, Part D), 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.

References

[1] CCITT Recommendation *12-MHz systems on standardized 2.6/9.5-mm coaxial cable pairs*, Vol. III, Rec. G.332.

[2] CCITT Recommendation *12-MHz systems on standardized 1.2/4.4-mm coaxial cable pairs*, Vol. III, Rec. G.345.

[3] CCITT Recommendation *Interconnection of coaxial carrier systems of different designs*, Vol. III, Rec. G.352.

Recommendation J.74

METHODS FOR MEASURING THE TRANSMISSION CHARACTERISTICS OF TRANSLATING EQUIPMENTS

1 No special measuring method is necessary for the carrier.

2 An oscilloscope can be used, for example, to measure the modulation ratio.

3 No special method is recommended for measuring pre-emphasis.

4 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.

5 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

1 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.

2 Telephony and television transmission, alternatively or simultaneously, on coaxial pairs or radio-relay links

2.1 *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 [1].

– For television transmission, interconnection should generally be made at video frequencies. Levels and impedances at interconnection points should then conform to Recommendation J.61.

2.2 *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, mastergroup 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.

Reference

[1] CCITT Recommendation *Interconnection at the baseband frequencies of frequency-division multiplex radio-relay systems*, Vol. III, Fascicle III.2, Rec. G.423.

Recommendation J.77 [1)]

CHARACTERISTICS OF THE TELEVISION SIGNALS TRANSMITTED OVER 18 MHz AND 60-MHz SYSTEMS

(Geneva, 1980)

For television transmission on 18 MHz and 60 MHz systems, a modulation procedure has to be used which is independent of the structure of the signal to be transmitted. This is achieved by a reference carrier which defines the phase relationship between the transmit and receive side.

The transmission channel is capable of transmitting the signals used in all those television systems defined by the CCIR, in accordance with Report 624 [1].

The requirements to be met by the 18 MHz and 60 MHz transmission systems are to be found in Recommendations G.334 [2] and G.333 [3].

It is recommended that the following conditions be met:

1 Vestigial sideband shaping

The shaping of the vestigial sideband signal has to be carried out entirely at the transmit side. The vestigial sideband shall not exceed a width of 1 MHz, i.e. the width of the Nyquist slope shall not exceed 2 MHz.

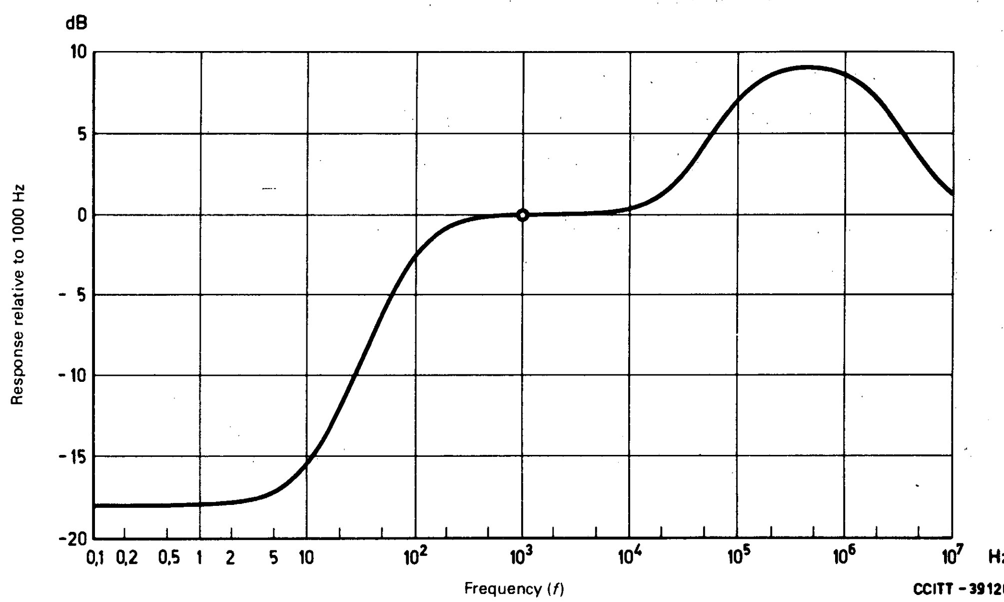
2 Video pre-emphasis

With regard to a more uniform loading of the coaxial line systems, it is recommended to use a video pre-emphasis network. The video pre-emphasis curve and the corresponding formula are shown in Figure 1/J.77. The video pre-emphasis amounts to 9 dB.

3 Nominal reference level of the modulated video signal

As a consequence of using a video pre-emphasis network, it is necessary to define a reference level at a suitable video frequency. It is recommended that this reference level be derived from the level of a single sideband measured after the Nyquist filter when a 1 kHz sine wave is transmitted, having a peak-to-peak amplitude of 0.7 volt at the video interconnection point. The reference level is this measured level plus 6 dB. The reference level is recommended to be +11 dBm0.

1) Recommendation J.76 of Volume III-2 of the *Orange Book* has been deleted.



Video pre-emphasis: $10 \log_{10} (1 + a) + 10 \log_{10} \left[1 + \frac{a}{\left(\frac{Q}{V}\right)^2 + 1} \right]$

where

$V = \frac{f}{f_0} - \frac{f_0}{f}$ $\qquad Q = 14.5$

$a = 7$ $\qquad f_0 = 450$ kHz

Low frequency suppression: $-10 \log_{10} \frac{b^2 + (2\pi\tau f)^2}{1 + (2\pi\tau f)^2}$

where

$b = 8$
$\tau = 14$ ms

FIGURE 1/J.77

Frequency response of video pre-emphasis and low frequency suppression relative to the value at 1 kHz

4 Accuracy of carrier frequencies

The carrier frequency of the first modulation stage should have a tolerance not exceeding 11 Hz. Tolerances of the carrier frequencies for the higher modulation stages can be ignored if either Recommendation G.225 [4] is met, or if the carriers are derived from the relevant TV channel-pair pilots (see [5] and [6]).

5 Reference carrier

In order to enable accurate demodulation of the signal at the receive side, it is necessary to transmit a reference carrier.

The following characteristics are recommended:

- carrier frequency of the first modulation stage corresponding to the video frequency of 0 Hz;
- polarity negative, i.e. such that the amplitude of the modulated video signal is greater at black than at white;
- nominal power level: +10 dBm0, independent of signal level.

6 Low frequency suppression

In order to prevent disturbance of the reference carrier by the low frequency components of the video signal, it is necessary to reduce the level of the low frequency components. A low frequency suppression of 18 dB is recommended. The low frequency suppression curve and the corresponding formula are shown in Figure 1/J.77.

References

[1] CCIR Report *Characteristics of television systems*, Vol. XI, Report 624, ITU, Geneva, 1982.

[2] CCITT Recommendation *18-MHz systems on standardized 2.6/9.5-mm coaxial pairs*, Vol. III, Rec. G.334.

[3] CCITT Recommendation *60-MHz systems on standardized 2.6/9.5-mm coaxial cable pairs*, Vol. III, Rec. G.333.

[4] CCITT Recommendation *Recommendations relating to the accuracy of carrier frequencies*, Vol. III, Rec. G.225.

[5] CCITT Recommendation *60-MHz systems on standardized 2.6/9.5-mm coaxial cable pairs*, Vol. III, Rec. G.333, § 8.4, Note 2.

[6] CCITT Recommendation, *18-MHz systems on standardized 2.6/9.5-mm coaxial pairs*, Vol. III, Rec. G.334, § 9.4.2, Note.

PART III

SUPPLEMENTS TO H AND J SERIES RECOMMENDATIONS

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Supplement No. 5

MEASUREMENT OF THE LOAD OF TELEPHONE CIRCUITS UNDER FIELD CONDITIONS

(Referred to in Recommendations G.223 and H.51 this supplement is to be found on page 295 of Fascicle III.2 of the *Red Book*, Geneva, 1985)

Supplement No. 12

INTELLIGIBILITY OF CROSSTALK BETWEEN TELEPHONE AND SOUND-PROGRAMME CIRCUITS

(Referred to in Recommendation J.32; this supplement is to be found on page 610 of Fascicle III.2 of the *Green Book*, Geneva, 1972.)

Supplement No. 16

OUT-OF-BAND CHARACTERISTICS OF SIGNALS APPLIED TO LEASED TELEPHONE-TYPE CIRCUITS

(Referred to in Recommendation H.51; this Supplement is to be found on page 191 of Fascicle III.4 of the *Red Book*, Geneva, 1985)

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