XVII A.P.
DÜSSELDORF
21.5-1.6 1990
CCIR

XVIIth PLENARY ASSEMBLY
DÜSSELDORF, 1990

INTERNATIONAL TELECOMMUNICATION UNION

RECOMMENDATIONS OF THE CCIR, 1990

(ALSO RESOLUTIONS AND OPINIONS)

VOLUME XII

TELEVISION AND SOUND TRANSMISSION (CMTT)

CCIR INTERNATIONAL RADIO CONSULTATIVE COMMITTEE

Geneva, 1990

CCIR

1. The International Radio Consultative Committee (CCIR) is the permanent organ of the International Telecommunication Union responsible under the International Telecommunication Convention "... to study technical and operating questions relating specifically to radiocommunications without limit of frequency range, and to issue recommendations on them..." (International Telecommunication Convention, Nairobi 1982, First Part, Chapter I, Art. 11, No. 83).*

2. The objectives of the CCIR are in particular:

a) to provide the technical bases for use by administrative radio conferences and radiocommunication services for efficient utilization of the radio-frequency spectrum and the geostationary-satellite orbit, bearing in mind the needs of the various radio services;

b) to recommend performance standards for radio systems and technical arrangements which assure their effective and compatible interworking in international telecommunications;

c) to collect, exchange, analyze and disseminate technical information resulting from studies by the CCIR, and other information available, for the development, planning and operation of radio systems, including any necessary special measures required to facilitate the use of such information in developing countries.

* *See also the Constitution of the ITU, Nice, 1989, Chapter 1, Art. 11, No. 84.*

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CCIR INTERNATIONAL RADIO CONSULTATIVE COMMITTEE

92-61-04301-1

Geneva, 1990

PLAN OF VOLUMES I TO XV
XVIIth PLENARY ASSEMBLY OF THE CCIR

(Düsseldorf, 1990)

VOLUME I (Recommendations) *Annex to Vol. I* (Reports)	Spectrum utilization and monitoring
VOLUME II (Recommendations) *Annex to Vol. II* (Reports)	Space research and radioastronomy services
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VOLUME XV-3 (Questions)	Study Groups 10, 11, CMTT
VOLUME XV-4 (Questions)	Study Groups 4, 9

All references within the texts to CCIR Recommendations, Reports, Resolutions, Opinions, Decisions and Questions refer to the 1990 edition, unless otherwise noted; i.e., only the basic number is shown.

Printed in Belgium

DISTRIBUTION OF TEXTS OF THE XVIIth PLENARY ASSEMBLY OF THE CCIR IN VOLUMES I TO XV

Volumes and Annexes I to XV, XVIIth Plenary Assembly, contain all the valid texts of the CCIR and succeed those of the XVIth Plenary Assembly, Dubrovnik, 1986.

1. Recommendations, Resolutions, Opinions are given in Volumes I-XIV and Reports, Decisions in the Annexes to Volumes I-XII.

1.1 *Numbering of texts*

When a Recommendation, Report, Resolution or Opinion is modified, it retains its number to which is added a dash and a figure indicating how many revisions have been made. Within the text of Recommendations, Reports, Resolutions, Opinions and Decisions, however, reference is made only to the basic number (for example Recommendation 253). Such a reference should be interpreted as a reference to the latest version of the text, unless otherwise indicated.

The tables which follow show only the original numbering of the current texts, without any indication of successive modifications that may have occurred. For further information about this numbering scheme, please refer to Volume XIV.

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1.2 *Recommendations (cont.)*

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1.3 *Reports*

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* Not reprinted, see Dubrovnik, 1986.

(1) Published separately.

1.3 *Reports (cont.)*

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635-637	XII	803	X/XI-3	997-1004	IV-1
639	XII	804, 805	XI-1	1005, 1006	IV/IX-2
642, 643	XII	807-812	X/XI-2	1007-1010	V
646-648	XII	814	X/XI-2	1011, 1012	VI
651	I	815, 816	XII	1016, 1017	VII
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659	I	826-842	I	1026-1033	VIII-2
662-668	I	843-854	II	1035-1039	VIII-2
670, 671	I	857	III	1041-1044	VIII-2
672-674	II	859-865	III	1045	VIII-3
676-680	II	867-870	IV-1	1047-1051	VIII-3
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687	II	876, 877	IV/IX-2	1058-1061	X-1
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781-789	IX-1	965-970	XII	1234-1241	XII

* Not reprinted, see Dubrovnik, 1986.

(1) Published separately.

1.3.1 *Note concerning Reports*

The individual footnote “Adopted unanimously” has been dropped from each Report. Reports in Annexes to Volumes have been adopted unanimously except in cases where reservations have been made which will appear as individual footnotes.

1.4 *Resolutions*

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15	I	64	X-1	89	XIII
20	VIII-1	71	I	95	XIV
23	XIII	72, 73	V	97-109	XIV
24	XIV	74	VI	110	I
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1.5 *Opinions*

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15	X-1	51	X-1	77	XIV
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35	I	65	XIV	85	VI
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9	VI	65	VII	93	X/XI-2
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18	X-1 + XI-1 + XII	69	VIII-1	95	X-1 + XI-1
		70	IV-1	96, 97	X-1
27	I	71	VIII-3	98	X-1 + XII
42	XI-1	72	X-1 + XI-1	99	X-1
43	X/XI-2	76	IV-1 + X-1 + XI-1 + XII	100	I
51	X/XI-2			101	II
53, 54	I	77	XII	102	V
56	I	78, 79	X-1	103	VIII-3
57	VI	80	XI-1	105	XIV
58	XI-1	81	VIII-3	106	XI-1
59	X/XI-3	83-86	VI		

2. **Questions** (Vols. XV-1, XV-2, XV-3, XV-4)

2.1 *Numbering of texts*

Questions are numbered in a different series for each Study Group: where applicable a dash and a figure added after the number of the Question indicate successive modifications. The number of a Question is completed by an *Arabic figure indicating the relevant Study Group.* For example:

– Question 1/10 would indicate a Question of Study Group 10 with its text in the original state;
– Question 1-1/10 would indicate a Question of Study Group 10, whose text has been once modified from the original; Question 1-2/10 would be a Question of Study Group 10, whose text has had two successive modifications.

Note – The numbers of the Questions of Study Groups 7, 9 and 12 start from 101. In the case of Study Groups 7 and 9, this was caused by the need to merge the Questions of former Study Groups 2 and 7 and Study Groups 3 and 9, respectively. In the case of Study Group 12, the renumbering was due to the requirement to transfer Questions from other Study Groups.

2.2 *Assignment of Questions*

In the plan shown on page II, the relevant Volume XV in which Questions of each Study Group can be found is indicated. A summary table of all Questions, with their titles, former and new numbers is to be found in Volume XIV.

2.3 *References to Questions*

As detailed in Resolution 109, the Plenary Assembly approved the Questions and assigned them to the Study Groups for consideration. The Plenary Assembly also decided to discontinue Study Programmes. Resolution 109 therefore identifies those Study Programmes which were approved for conversion into new Questions or for amalgamation with existing Questions. It should be noted that references to Questions and Study Programmes contained in the texts of Recommendations and Reports of Volumes I to XIII are still those which were in force during the study period 1986-1990.

Where appropriate, the Questions give references to the former Study Programmes or Questions from which they have been derived. New numbers have been given to those Questions which have been derived from Study Programmes or transferred to a different Study Group.

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VOLUME XII

TELEVISION AND SOUND TRANSMISSION

(CMTT)

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NUMERICAL INDEX OF TEXTS

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CMTT

Joint CCIR/CCITT Study Group for Television and Sound Transmissions

TRANSMISSION OF SOUND BROADCASTING AND TELEVISION SIGNALS OVER LONG DISTANCES

Terms of reference:

To study, in cooperation with the Study Groups of the CCIR and the CCITT, the specifications to be satisfied by telecommunication systems to permit the transmission of sound and television broadcasting programmes over long distances.

1986-1990	*Chairman:*	W. G. SIMPSON (United Kingdom)
	Vice-Chairman:	G. ZEDLER (Germany (Federal Republic of))

As from the next study period, in conformity with Resolution 61 adopted at the XVIIth Plenary Assembly, Düsseldorf (May-June 1990), the scope of the work which will be undertaken and the names of the Chairman and Vice-Chairman concerned are given below.

CMTT

CCIR/CCITT JOINT STUDY GROUP FOR TELEVISION AND SOUND TRANSMISSION

Scope:

Study, in cooperation with the Study Groups of the CCIR and CCITT, the specifications to be satisfied by telecommunication systems to permit the transmission of sound and television broadcasting programmes.

1990-1994	*Chairman:*	W. G. SIMPSON (United Kingdom)
	Vice-Chairman:	G. ZEDLER (Germany (Federal Republic of))

INTRODUCTION BY THE CHAIRMAN OF THE CMTT

1. CMTT activities from 1986 to 1990

1.1 During the study period 1986-1990, the CMTT held two meetings in Geneva:

- Interim Meeting from 2-13 November 1987;
- Final Meeting from 2-13 October 1989.

A total of 200 technical contributions were considered during the two meetings. Sixty nine new or modified texts were adopted by the Final Meeting, including 8 new and 10 revised Recommendations and 8 new and 19 revised Reports.

The CMTT wishes to keep the classification of subjects as adopted for the previous period, as follows:

- Section A (Replies to Questions 13 and 14)
  Television transmission standards and performance objectives.
- Section B (Replies to Questions 15 and 16)
  Methods of operation and assessment of performance of television transmissions.
- Section C (Replies to Questions 17 and 18)
  Transmission standards and performance objectives for sound channels.
- Section D (Replies to Questions 19 and 20)
  Methods of operation and assessment of performance of sound channel transmissions.
- Section E (Replies to Questions 21, 22, 23 and 24)
  Transmission of signals with multiplexing of video, sound and data signals of new systems.

1.2 During its Interim and Final Meetings, the CMTT set up Working Groups operating as follows:

Working Group	*Terms of reference*	*Chairman*
CMTT-A	Q. 13, 15, 21, 22, 24	Mr. L. Gooddy (Canada)
CMTT-B	Q. 14, 16, 23	Mr. J. M. Corbett (United Kingdom)
CMTT-C	Q. 17, 18, 19, 20	Mr. G. Zedler (Germany (Federal Republic of))

1.3 The CMTT also set up an Editorial Group at each of its meetings, with the following membership:

Interim Meeting	*Final Meeting*
Mr. C. Dorkins (United Kingdom)	Mr. C. Dorkins (United Kingdom)
Mr. M. Bosch (France)	M. C. Bremenson (France)
Mr. L. Bascuñana (Spain)	

1.4 During both the Interim Meeting and the Final Meeting, Interim Working Party CMTT/1 met at the request of its Chairman, Mr. W. G. Simpson (United Kingdom).

At these meetings, statements were made by the following:

- on television transmissions:

  Dr. Yamamoto (Japan), Chairman of Working Party 11-D;
  Mr. Corbett (United Kingdom), Chairman of Working Party CMTT-B,

- on sound-programme transmissions:

  Mr. Steinke (OIRT), Chairman of Working Party 10-C;
  Mr. Zedler (Germany (Federal Republic of)), Chairman of Working Party CMTT-C.

These speakers informed the members of the IWP on the work of Study Groups 10, 11 and the CMTT concerning digital signals.

The results obtained by IWP CMTT/1 are summarized in § 2.2.3.

1.5 During the previous Final Meeting held in 1985, Decisions were adopted by the CMTT which established two new IWPs, CMTT/2 and CMTT/3.

1.5.1 IWP CMTT/2, chaired by Dr. L. Stenger (Germany (Federal Republic of)) with Vice-Chairmen Dr. H. Murakami (Japan) and Mr. K. Davies (Canada) was charged with establishing draft new Recommendations on the digital transmission of component-coded television signals. This IWP cooperated closely with IWP 11/7 by arranging that their meetings would be held consecutively at the same place. Five meetings were held during the study period with the effective result that two new Recommendations and two associated Reports were presented to the Final Meeting of CMTT for its approval*. Additional work on all four of these texts remained to be completed before the XVIIth Plenary Assembly. A meeting of IWP CMTT/2 was held in Grenada (Spain) early in March 1990 with the objective of completing the remaining work.

Recommendation 721 was satisfactorily completed during that meeting. A draft of the revisions proposed by IWP CMTT/2 and consequent revisions proposed for Report 1234 were submitted to and approved by the XVIIth Plenary Assembly.

Agreement could not be obtained for a single system proposed for the third hierarchical level of CCITT Recommendation G.702. As a consequence, CMTT/2 proposed to continue the work on an urgent basis into the next study period, with the objective of obtaining agreement as soon as possible. In the interim, the IWP proposed necessary revisions to Recommendation 723 and associated Report 1235 which were also submitted to and approved by the XVIIth Plenary Assembly.

1.5.2 IWP CMTT/3, under the chairmanship of Mr. A. Brown (EBU), had the task of dealing with the problems of the transmission of television and sound-programme signals in the broadband ISDN. It was essential that information concerning the ISDN services, that broadcasters may wish to use, be communicated to CCITT Study Group XVIII in a timely manner. To this end the IWP was authorized to submit the results of its work directly to Study Group XVIII, subject to their subsequent approval by the CMTT. IWP CMTT/3 held five meetings during the study period (including one held in Geneva in February 1990, which prepared liaison statements to Study Group XVIII and IWP CMTT/2) and the results were presented to Study Group XVIII by the Chairman of IWP CMTT/3. This method of working was considered to be very effective by both Study Group XVIII and the CMTT.

1.5.3 At its Interim Meeting in 1987, the CMTT adopted a Decision to establish an IWP CMTT/4 with the task of preparing a draft new Recommendation on the digital transmission of sound-programme signals of studio quality, this draft to be presented to the Final Meeting of the CMTT. The IWP, under the chairmanship of Mr. A. Weisser (Télédiffusion de France) held two meetings in the period between the Interim and Final Meetings and successfully completed its originally-assigned task.

1.5.4 The Special Rapporteurs of the CMTT for liaison with CCITT Study Groups during the study period 1986-1990 were as follows:

– CCITT Study Group IV: Mr. G. Knowlson (United States of America).
– CCITT Study Group XV: Mr. W. Walter (Germany (Republic Federal of)).
– CCITT Study Group XVIII: Mr. P. Wery (Canada).

2. Results obtained

The Working Groups prepared a number of documents during the two meetings, with the following results:

2.1 *Texts submitted to the Plenary Assembly for approval*

2.1.1 *Recommendations*

The CMTT proposed the modification of 10 of the 21 Recommendations appearing in Volume XII (Dubrovnik, 1986). These are: Recommendation 567-2 (MOD F) (Doc. CMTT/1004), Recommendation 604-1 (MOD I) (Doc. CMTT/1012), Recommendation 658 (MOD F) (Doc. CMTT/1013), Recommendation 473-4 (MOD I) (Doc. CMTT/1023), Recommendation 503-3 (MOD F) (Doc. CMTT/1027), Recommendation 505-3 (MOD F) (Doc. CMTT/1028), Recommendation 606 (MOD F) (Doc. CMTT/1034), Recommendation 571-1 (MOD F) (Doc. CMTT/1038), Recommendation 645 (MOD F) (Doc. CMTT/1039) and Recommendation 661 (MOD I) (Doc. CMTT/1040). Recommendation 642, which also concerns Study Group 10, will no longer appear in the CMTT Volume.

* – These are: Recommendation 723 "Transmission of component-coded digital television signals for contribution-quality applications at the third hierarchical level of CCITT Recommendation G.702";
– associated Report 1235 "Digital transmission of component-coded television signals at 30-34 Mbit/s and 45 Mbit/s";
– Recommendation 721 "Transmission of component-coded digital television signals for contribution-quality applications at bit rates near 140 Mbit/s"; and
– associated Report 1234 "Digital transmission of component-coded television at bit rates near 68 Mbit/s and 140 Mbit/s".

The CMTT also proposed the adoption of 8 new draft Recommendations:

Section A

722 Uniform technical standards and uniform operational procedures for satellite news gathering (SNG);

723 Transmission of component-coded digital video signals for contribution quality applications at the third hierarchical level of CCITT Recommendation G.702;

721 Transmission of component-coded digital television signals for contribution quality application at bit rates near 140 Mbit/s.

Section B

720 Measurement methods and test procedures for teletext signals.

Section C

724 Transmission of digital studio quality sound signals over H1 channels;

718 Digital transmission of high-quality sound-programme signals on distribution circuits using 480 kbit/s (496 kbit/s) per audio channel;

719 Transmission of high-quality sound-programme analogue signals over mixed analogue/digital circuits at 320 kbit/s.

Section D

717 Tolerances for transmission time differences between the vision and sound components of a television signal.

2.1.2 *Questions*

No changes of substance were made to the 12 Questions appearing in the 1986 CMTT Volume. New Question [25], concerning secondary distribution was proposed for adoption, along with associated new Study Programme AQ/CMTT (Question [26]).

2.1.3 *Opinions*

The CMTT does not have any proposals concerning Opinions.

2.2 *Texts submitted to the Plenary Assembly*

2.2.1 *Reports*

The CMTT adopted modifications to 18 of the 39 Reports appearing in Volume XII (Dubrovnik, 1986). Two Reports were deleted (Report 1093 and Report 817-2. The latter was replaced by new Report 1241).

The CMTT also adopted 8 new Reports: 1234, 1235, 1236, 1237, 1238, 1239, 1240 and 1241.

2.2.2 *Study Programmes*

The CMTT adopted substantive modifications to 10 of the 45 Study Programmes appearing in Volume XII (Dubrovnik, 1986): (13G-1, 14A-3, 14E-1, 15D-2, 17D-2, 18A-3, 18F-1, 18G-1, 19D-2, 21A-1). One Study Programme was deleted (Study Programme 18H).

The CMTT adopted 7 new Study Programmes against existing Questions (13H, 14F, 22A, 22B, 22C, 22D, 24A). One new Study Programme is proposed for adoption against a new Question (see § 2.1.2).

2.2.3 *Decisions*

The CMTT made substantive modifications to all of its existing Decisions and added one new Decision:

Decision 18-6 Digital systems for the transmission of sound-programme and television signals

A major revision to Decision 18 expands the participation of former IWP CMTT/1 to include Study Groups 10 and 11. The result is new JIWP CMTT-10-11/2.

Decision 67-2 Digital transmission of component-coded television signals

The terms of reference of IWP CMTT/2 are expanded to include component and multiplexed analogue component HDTV systems.

Decision 68-2 Television and sound-programme signals in the broadband ISDN

Representatives of CCIR Study Groups 10 and 11 are added to the membership, and Special Rapporteurs from the CCITT are invited to participate in IWP CMTT/3.

Decision 76-1 Satellite news gathering (SNG)

The study of overall transmission and performance objectives for HDTV transmission by portable satellite earth stations for SNG is added to the terms of reference of JIWP CMTT-4-10-11/1.

Decision 77-1 Transmission of digital sound programme signals of digital studio quality on circuits using the H1 channel

IWP CMTT/4 is extended into the 1990-1994 study period.

Decision 98 Low bit-rate digital-audio coding systems

This Decision, prepared by Study Group 10, includes the participation of specialists from the CMTT.

2.2.4 *Proposed modification to the terms of reference of the CMTT*

At the Interim Meeting of the CMTT in 1987 the CMTT adopted a proposal to change its terms of reference to read as follows:

To study, in cooperation with the Study Groups of the CCIR and the CCITT, the specifications to be satisfied by telecommunication systems to permit the transmission of sound and television broadcasting programmes.

This change has the effect of extending the terms of reference to embrace all telecommunication systems, irrespective of length.

This proposal was submitted to the CCITT Plenary Assembly in 1988 where it was endorsed by the CCITT. As the CMTT is a joint CCIR/CCITT Study Group this proposal was also submitted to the CCIR Plenary Assembly for its endorsement.

3. Acknowledgements

On behalf of the Vice-Chairman and himself, the Chairman wishes to express his gratitude to participating administrations and organizations for their contributions and to the delegates for their considerable work in an excellent spirit of cooperation which has resulted in the substantial output described above.

He especially thanks the Chairmen of Working Groups and Sub-Groups and the Chairmen of the IWPs who, through their skill in the guiding of the discussions, have enabled the CMTT texts to be produced.

SECTION CMTT A: TELEVISION TRANSMISSION STANDARDS AND PERFORMANCE OBJECTIVES

RECOMMENDATION 567-3

TRANSMISSION PERFORMANCE OF TELEVISION CIRCUITS DESIGNED FOR USE IN INTERNATIONAL CONNECTIONS

(Question 13/CMTT)

(1978-1982-1986-1990)

The CCIR,

CONSIDERING

the need for a Recommendation concerning analogue television transmissions over long distances, common to the CCIR and CCITT,

UNANIMOUSLY RECOMMENDS

that, taking account of the definitions in Parts A and B and the measurement methods in Part C and its Annexes, the transmission performance of international television circuits should satisfy the objectives for design given in Parts D and E.

Introduction

The Joint CCIR/CCITT Study Group for Television and Sound Transmission (CMTT) has studied problems which occur when transmitting television signals of various standards over long distances.

The CMTT decided to study unified test methods and transmission performance which can be recommended for circuits intended for transmission of signals conforming to the majority of television standards.

This Recommendation is intended for use where circuits will be required at various times to transmit television signals of the 525-line and the 625-line standards.

However, in view of the comprehensive nature of the Recommendation, it is appropriate that it also be applicable to circuits which are required to transmit television signals of only one standard. Accordingly, the document addresses the different requirements of 525-line, 625-line and multi-standard needs where necessary.

The assumption is made that the circuit does not contain satellite systems using line-rate energy dispersal or systems which employ digital transmission techniques. If it does, it is probable that additional objectives will be required.

The Recommendation contains five parts listed below:

Part A: Definitions of a connection and circuits

Part B: Definition of parameters

Part C: Measurement methods and test signals

Part D: Design objectives and tolerances applicable to the hypothetical reference circuit

Part E: Performance of circuits shorter or longer than the hypothetical reference circuit.

Note – References and Bibliography are detailed at the end of the particular Part or Annex to which they are relevant.

PART A – DEFINITION OF AN INTERNATIONAL TELEVISION CONNECTION AND DEFINITION OF THE TERRESTRIAL AND COMMUNICATION SATELLITE HYPOTHETICAL REFERENCE CIRCUITS

A.1 Definitions

A.1.1 *Definition of an international television connection* (Fig. 1)

– Point A, to be considered as the sending end of the international television connection, may be the point at which the programme originates (studio or outside location), a switching centre or the location of a standards converter.

– Point D, to be considered as the receiving end of the international television connection, may be a programme-mixing or recording centre, a broadcasting station, a switching centre or the location of a standards converter.

– The local television circuit AB connects point A to the sending terminal station, point B, of the international television circuit.

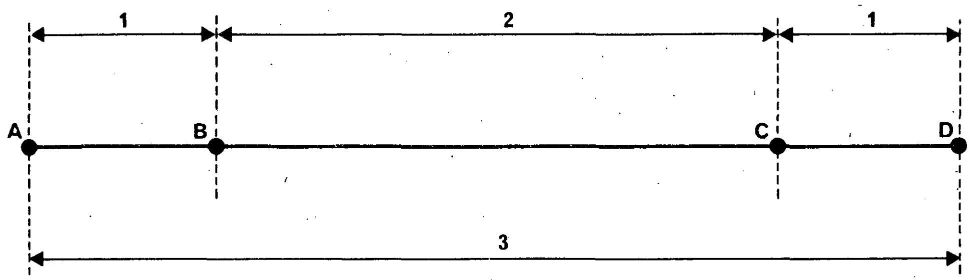


FIGURE 1 – *An international television connection*

1 : Local television circuit

2 : International television circuit

3 : International television connection

– The international television circuit, BC, comprises a chain of national and international television links. The precise locations (e.g. within buildings), to be regarded as the points B and C, will be designated by the authorities concerned.
– The local television circuit CD connects point C, the receiving terminal station of the international television circuit, to the point D.
– The combination AD, of the international television circuit, BC, and the local television circuits AB and CD, constitutes the international television connection.

The requirements given in subsequent parts of this Recommendation refer to the performance of international television circuits only; no requirements have been laid down for the local television circuits, AB and CD.

A.1.2 *Definition of the terrestrial hypothetical reference circuit* (Fig. 2a)

The main features of the television terrestrial hypothetical reference circuit, which is an example of an international television circuit (BC in Fig. 1) and which may be of either radio or cable type, are:
– the overall length between video terminal points is 2500 km;
– two intermediate video points (M and M′) divide the circuit into three sections of equal length;
– the three sections are lined up individually and then interconnected without any form of overall adjustment or correction;
– the circuit does not contain a standards converter or a synchronizing pulse regenerator, or an equipment for the insertion of signals in the line/field blanking interval.

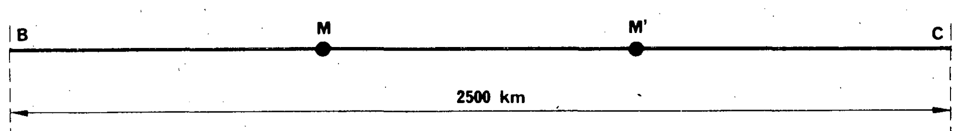


FIGURE 2a – *Hypothetical reference circuit for television transmissions over a terrestrial system*

A.1.3 *Definition of the hypothetical reference circuit in the fixed satellite service* (Fig. 2b)

A hypothetical reference circuit for a system in the fixed satellite service which may form part of an international television circuit (BC in Fig. 1) is defined as follows:
– it consists of one Earth station-satellite-Earth station system;
– it includes one pair of modulation and demodulation equipments for translation from the baseband to the radio-frequency carrier, and from the radio-frequency carrier to the baseband, respectively;
– it does not include a standards converter or a synchronizing-pulse regenerator, or equipment for the insertion of signals in the line/field blanking interval.

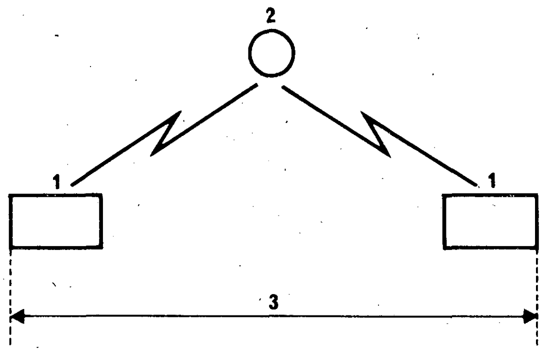


FIGURE 2b – *Hypothetical reference circuit for television transmissions over a system in the fixed satellite service*

1: Earth station

2: Satellite space station

3: Hypothetical reference circuit

PART B – DEFINITIONS OF PARAMETERS

This part defines terms which are necessary for the understanding of the Recommendation itself. The definitions are necessary either because many of them do not appear in any recognized technical vocabulary or because they are general definitions which have been given meanings which are particular to television transmission. Their use in this limited context should not be taken as a restriction on their use in the International Electrotechnical Vocabulary (IEV) or other vocabularies where they may be given wider definitions, i.e. not limited to television.

B.1 Waveform terminology

The following terms concerning the components and values of a composite colour video signal are illustrated in Fig. 3:

- A: the non-useful d.c. component,
- B: the useful d.c. component, integrated over a complete frame period,
- C: the picture d.c. component, integrated over the active line period (T_u),
- D: the instantaneous value of the luminance component,
- E: the instantaneous signal value with respect to the bottom of the synchronizing pulses,
- F: the peak signal amplitude (positive or negative with respect to blanking level),
- G: the peak amplitudes of chrominance components,
- H: the peak-to-peak signal amplitude,
- J: the difference between black level and blanking level (set-up),
- K: the peak-to-peak amplitude of the colour burst,
- L: the nominal value of the luminance component,
- M: the peak-to-peak amplitude of a monochrome composite video signal ($M = L + S$),
- S: the amplitude of the synchronizing pulses,
- T_{sy}: duration of line synchronizing pulse,
- T_{lb}: duration of line blanking period,
- T_u: duration of active line period,
- T_b: duration of breezeway,
- T_{fp}: duration of front porch,
- T_{bp}: duration of back porch.

The amplitudes L, S and M are used as reference amplitudes for the video signal. The amplitudes defined by B, C, D, E, F, G, H and J above, may be expressed as percentages of the value L.

Average picture level (APL) is the mean value of C over a complete frame period (excluding blanking periods) expressed as a percentage of L.

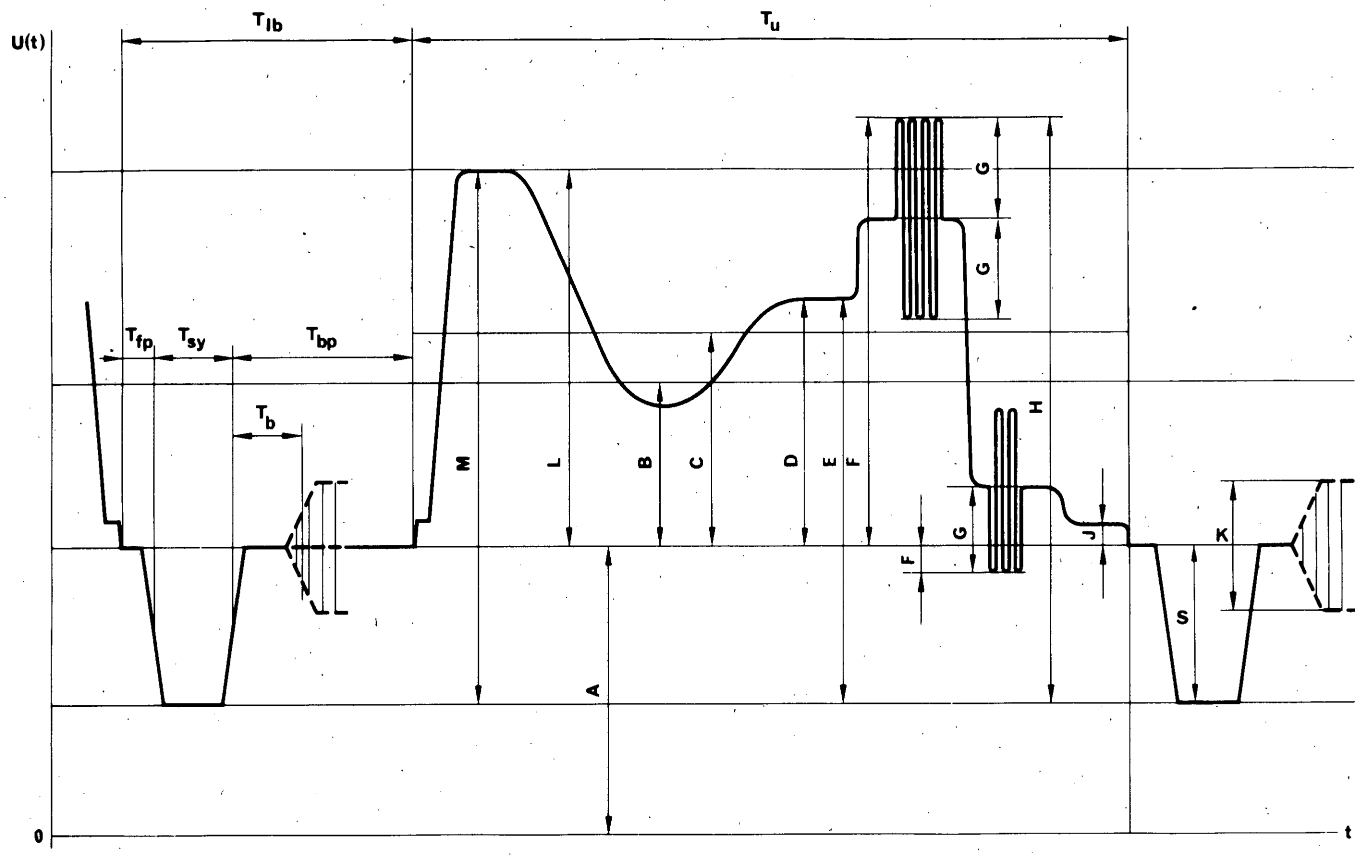


FIGURE 3 – *One line of a composite colour video signal*

B.2 Requirements at points of video interconnection

B.2.1 *Nominal impedance* (Z_0)

At points of video interconnection, the input and output impedance (Z_0) of each section should be specified, subject to bilateral agreement, as either unbalanced or balanced with respect to earth.

B.2.2 *Return loss*

The return loss, relative to Z_0, of an impedance Z is, in the frequency domain:

$$20 \log \left| \frac{Z_0 + Z(f)}{Z_0 - Z(f)} \right| \qquad \text{dB}$$

In the time domain, it is expressed by the symbolic formula:

$$20 \log \left| \frac{A_1}{A_2} \right| \qquad \text{dB}$$

where A_1 is the peak-to-peak amplitude of the incident signal and A_2 is the peak-to-peak amplitude of the reflected signal. Numerically, the result is the same as that obtained by the frequency domain method if the return loss is independent of frequency.

B.2.3 *Polarity and d.c. component*

The polarity of the signal should be "positive", that is to say, such that black-to-white transitions are positive-going.

The useful d.c. component, B in Fig. 3, which is related to the average luminance of the picture, may or may not be contained in the signal and need not be transmitted or delivered at the output.

A non-useful d.c. component, A in Fig. 3, may be present in the signal (for example, due to d.c. supplies). Limits for this component need to be specified for the terminated and unterminated conditions.

B.2.4 *Nominal signal amplitude*

The nominal signal amplitude is the peak-to-peak amplitude of the monochrome video signal that includes the synchronizing signal and luminance signal component set to peak-white (M in Fig. 3).

B.3 Transmission performance requirements

The definitions in § B.3.2 and the subsequent sub-sections assume that the circuit has nominal insertion gain as defined in § B.3.1 below.

B.3.1 *Insertion gain*

Insertion gain is defined as the ratio, expressed in decibels, of the peak-to-peak amplitude of a specified test signal at the receiving end to the nominal amplitude of that signal at the sending end, the peak-to-peak amplitude being defined as the difference between the amplitudes measured at defined points of the signal used.

B.3.2 *Noise*

B.3.2.1 *Continuous random noise*

The signal-to-noise ratio for continuous random noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal, L in Fig. 3, to the r.m.s. amplitude of the noise measured after band limiting. A signal-to-weighted-noise ratio is defined as a ratio, expressed in decibels, of the nominal amplitude of the luminance signal, L in Fig. 3, to the r.m.s. amplitude of the noise measured after band limiting and weighting with a specified network.

The measurement should be made with an instrument having, in terms of power, a defined time constant or integrating time.

B.3.2.2 *Low-frequency noise*

The signal-to-noise ratio for low-frequency noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal, L, in Fig. 3, to the peak-to-peak amplitude of the noise after band limiting to include only the spectrum 500 Hz to 10 kHz.

B.3.2.3 *Periodic noise*

The signal-to-noise ratio for periodic noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal, L in Fig. 3, to the peak-to-peak amplitude of the noise. Different values are specified for noise at a single frequency between 1 kHz and the upper limit of the video frequency band and for power-supply hum including lower-order harmonics.

B.3.2.4 *Impulsive noise*

The signal-to-noise ratio for impulsive noise is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal, L in Fig. 3, to the peak-to-peak amplitude of the impulsive noise.

B.3.3 *Cross-talk from another television channel*

The signal-to-cross-talk ratio is defined as the ratio, expressed in decibels, of the nominal amplitude of the luminance signal (L in Fig. 3) to the peak-to-peak amplitude of the interfering signal.

B.3.4 *Non-linear distortion*

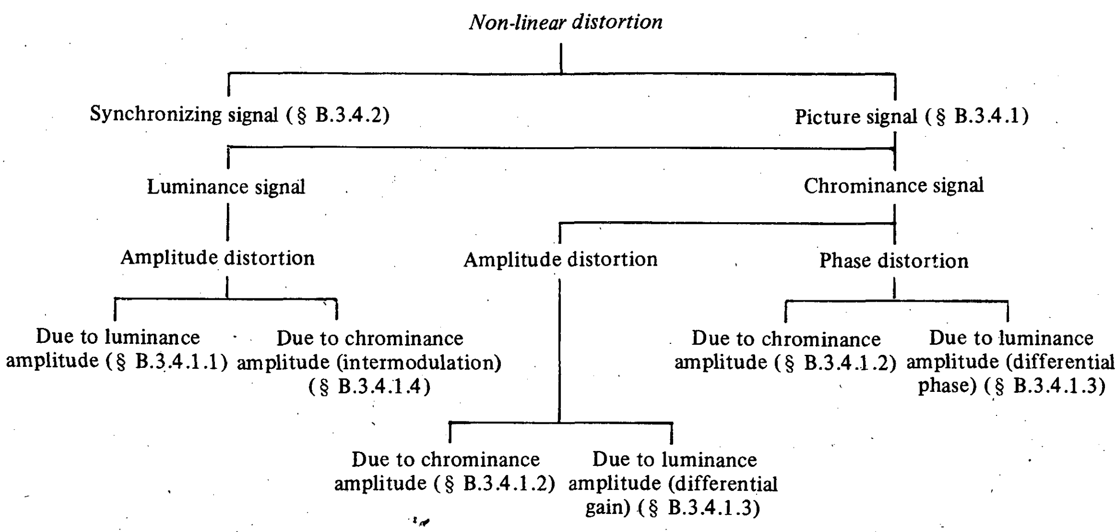
In a television circuit the transmission characteristic may not be completely linear. The extent of the non-linear distortion which is produced will depend primarily on:

- the average picture level, as defined in § B.1;
- the instantaneous value of the luminance signal voltage (D in Fig. 3);
- the amplitude of the chrominance signal (G in Fig. 3).

There would, in general, be little purpose in defining completely the non-linear characteristics of a transmission circuit. It is necessary, therefore, to limit the number of measured quantities by restricting them to those which are recognized as being directly correlated with picture quality. Additionally, the test conditions should be restricted by introducing a systematic classification in the definition of the quantities to be measured. Examples of distortions not included in this classification are given in the documentation [CCIR, 1970-74a and b].

The nature of the video signal is such that, in terms of picture quality, the impairment due to the effect of circuit non-linearity on the synchronizing signal is different from the effect of circuit non-linearity on the picture signal.

Furthermore, the non-linearity may affect the luminance and chrominance signals individually or cause interaction between them. This leads to the following system of classification of non-linear distortions:



The above classification applies for steady-state conditions during a time span which is long in relation to the field period. In this case, the concept of average picture level has a precise significance. If these conditions are not fulfilled, for example, if a sudden change in the APL is introduced, additional non-linear effects may be produced, the extent of which will depend on the long-time transient response of the circuit. This aspect requires further study (see Study Programme 13B/CMTT, Report 636 and [CCIR, 1970-74c, d, and e]).

Additional non-linearity may also occur if a sudden change in signal amplitude occurs [CCIR, 1970-74a].

B.3.4.1 *Picture signal*

B.3.4.1.1 *Luminance signal*

For a particular value of average picture level, the non-linear distortion of the luminance signal is defined as the departure from proportionality between the amplitude of a small step function at the input to the circuit and the corresponding amplitude at the output, as the initial level of the step is shifted from blanking level to white level.

B.3.4.1.2 *Chrominance signal*

Gain

For fixed values of luminance signal amplitude and average picture level, the non-linear gain distortion of the chrominance signal is defined as the departure from proportionality between the amplitude of the chrominance sub-carrier at the input to the circuit and the corresponding amplitude at the output, as the amplitude of the sub-carrier is varied from a specified minimum to a maximum value.

Phase

For fixed values of luminance signal amplitude and average picture level, the non-linear phase distortion of the chrominance signal is defined as the variation in the phase of the chrominance sub-carrier at the output, as the amplitude of the sub-carrier is varied from a specified minimum to a maximum value.

B.3.4.1.3 *Intermodulation from the luminance signal into the chrominance signal*

Differential gain

If a constant small amplitude of chrominance sub-carrier, superimposed on a luminance signal, is applied to the input of the circuit, the differential gain is defined as the change in the amplitude of the sub-carrier at the output as the luminance varies from blanking level to white level, the average picture level being maintained at a particular value.

Differential phase

If a constant small amplitude of chrominance sub-carrier without phase modulation, superimposed on a luminance signal, is applied to the input of the circuit, the differential phase is defined as the change in the phase of the sub-carrier at the output as the luminance varies from blanking level to white level, the average picture level being maintained at a particular value.

B.3.4.1.4 *Intermodulation from the chrominance signal into the luminance signal*

If a luminance signal of constant amplitude is applied to the input of a circuit, the intermodulation is defined as the variation of the amplitude of the luminance signal at the output resulting from the superimposition on the input signal of a chrominance signal of specified amplitude, the average picture level being maintained at a particular value.

B.3.4.2 *Synchronizing signal*

B.3.4.2.1 *Steady-state distortion*

If a video signal of specified average picture level and containing synchronizing pulses of nominal amplitude (*S* in Fig. 3) is applied to the input of the circuit, the steady state non-linear distortion is defined as the departure from nominal of the mid-point amplitude of the synchronizing pulses at the output.

B.3.4.2.2 *Transient distortion*

If the average picture level of the video signal is stepped from a low value to a high value, or from a high value to a low value, the transient non-linear distortion is defined as the maximum instantaneous departure from the nominal value of the mid-point amplitude of the synchronizing pulses at the output.

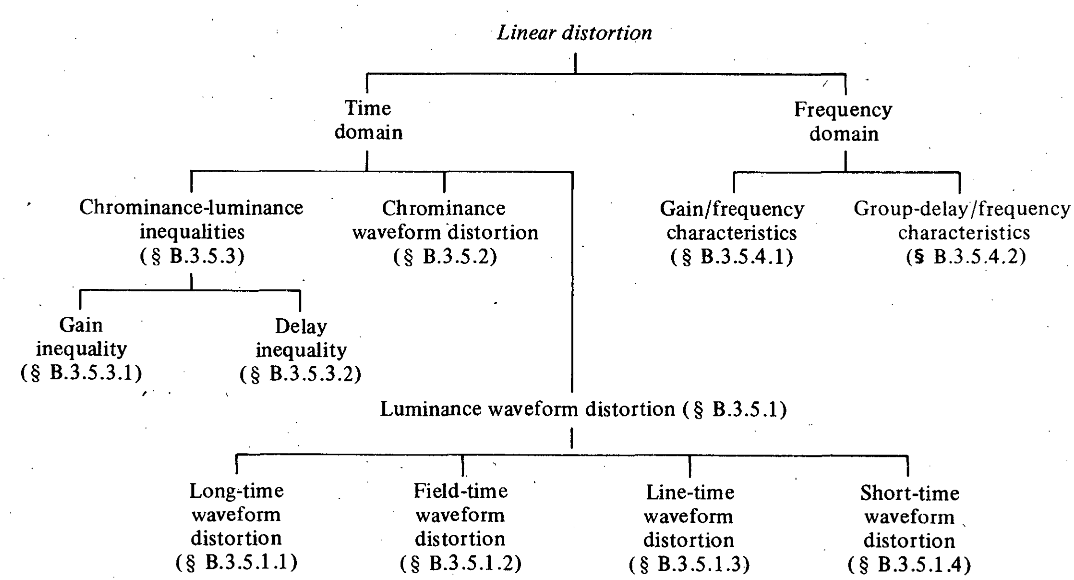
B.3.5 *Linear distortion*

Linear distortions are those which can be caused by linear networks. Such distortions do not depend on the average picture level, or the amplitude, or the position of the test signals.

In the case of networks which are affected by a small amount of non-linearity, measurements can still be carried out. However, as the results can be somewhat affected by the average picture level and the amplitude and position of the test signals, it is good practice, when presenting the results, to specify the measurement conditions.

Linear distortions can be measured either in the time domain or in the frequency domain.

The quantities which can be measured in the two domains may be classified as shown below.



B.3.5.1 *Waveform distortion of the luminance signal*

The distortion of the video waveform due to a television circuit will in general be represented by a continuous function in the time domain.

In practice, however, the form of the video signal and the effects on a displayed picture are such that the resulting impairments may be classified by considering four different time scales which are comparable to the durations of many fields (long-time waveform distortion), one field (field-time waveform distortion), one line (line-time waveform distortion), and one picture element (short-time waveform distortion).

In considering each of these time scales, therefore, impairments appropriate to the other three are excluded by the measurement method.

B.3.5.1.1 *Long-time waveform distortion*

If a video test signal, simulating a sudden change from a low average picture level to a high one or a high average picture level to a low one, is applied to the input of a circuit, long-time waveform distortion is present if the blanking level of the output signal does not accurately follow that of the input. This failure may be either in exponential form or, more frequently, in the form of damped very low-frequency oscillations.

B.3.5.1.2 *Field-time waveform distortion*

If a square-wave signal with a period of the same order as one field and of nominal luminance amplitude is applied to the input of the circuit, the field-time waveform distortion is defined as the change in shape of the square wave at the output. A period at the beginning and end of the square wave, equivalent to the duration of a few lines, is excluded from the measurement.

B.3.5.1.3 *Line-time waveform distortion*

If a square-wave signal with a period of the same order as one line and of nominal luminance amplitude is applied to the input of the circuit, the line-time waveform distortion is defined as the change in shape of the square wave at the output. A period at the beginning and end of the square wave, equivalent to a few picture elements, is excluded from the measurement.

B.3.5.1.4 *Short-time waveform distortion*

If a short pulse (or a rapid step-function) of nominal luminance amplitude and defined shape is applied to the input of the circuit, the short-time waveform distortion is defined as the departure of the output pulse (or step) from its original shape. The choice of the half-amplitude duration of the pulse (or the rise-time of the step) will be determined by the nominal cut-off frequency, f_c, of the television system. (See Report 624.)

B.3.5.2 *Chrominance waveform distortion*

If a test signal in the form of an amplitude-modulated sub-carrier is applied to the input of a circuit, chrominance waveform distortion is defined as the change in the shape of the envelope and phase of the modulated sub-carrier of the output test signal.

B.3.5.3 *Chrominance-luminance inequalities*

B.3.5.3.1 *Gain inequality*

If a test signal having defined luminance and chrominance components is applied to the input of the circuit, the gain inequality is defined as the change in amplitude of the chrominance component relative to the luminance component between the input and output of the circuit.

B.3.5.3.2 *Delay inequality*

If a composite signal, consisting of a defined luminance test signal in fixed amplitude and time relationship with a chrominance sub-carrier modulated by the same luminance test signal, is applied to the input of the circuit and if the luminance signal at the output is compared with the modulation envelope of the chrominance signal, then the delay inequality of the circuit is defined as the change in relative timing of corresponding parts of the two waveforms between input and output.

B.3.5.4 *Steady state characteristics*

B.3.5.4.1 The gain/frequency characteristic of the circuit is defined as the variation in gain between the input and output of the circuit over the frequency band extending from the field repetition frequency to the nominal cut-off frequency of the system, relative to the gain at a suitable reference frequency.

B.3.5.4.2 The group delay/frequency characteristic of the circuit is defined as the variation in group delay between the input and the output of the circuit, over the frequency band extending from the field repetition frequency to the nominal cut-off frequency of the system, relative to the delay at a suitable reference frequency. It is for practical reasons, an approximation to the slope (derivative) of the phase/frequency characteristic of the circuit.

REFERENCES

CCIR Documents

[1970-74]: **a.** CMTT/188 (Germany (Federal Republic of)); **b.** CMTT/189 (Germany (Federal Republic of)); **c.** CMTT/5 (United Kingdom); **d.** CMTT/21 (USA); **e.** CMTT/40 (Germany (Federal Republic of)).

PART C – MEASUREMENT METHODS AND TEST SIGNALS

C.1 Introduction

Section numbering in this part is related to the section numbering of Part B.

The test signal elements contained in Annex I may be combined in any suitable way to form test signals. Unless otherwise specified, the average picture level of test signals so obtained should be 50%. It should be noted that some practical circuits require the presence of synchronizing signals for correct functioning.

Test signals can be used either as repetitive signals or, with certain exceptions, as insertion test signals in connection with active lines chosen to give the required average picture level. During programme periods however, due consideration must be given to the effects of the variations of the average picture level upon measurements made with insertion test signals.

For full-field tests, some administrations may wish to use full-field sequences containing the same signals as specified for use as insertion test signals (Recommendation 473). In this case, measurement methods should be those specified in Annex III to Part C of this Recommendation.

The measurements described in § C.3.2 to C.3.5 are valid provided that the insertion gain of the circuit is within the stated requirements.

Circuit non-linearity may introduce into the received signals spectral components which are not present in the original test signals and which are not related to picture impairment. In such cases, it is suggested that a phase-corrected low-pass filter be inserted before the measuring equipment to eliminate spurious out-of-band components. An example of a filter suitable for 625-line measurements is described elsewhere [CCIR, 1970-74a].

C.2 Measurements of equipment and signal characteristics at points of video interconnection

C.2.1 *Nominal impedance*

At points of video interconnection, the input and output impedance will be specified. The actual impedances will be measured, in terms of departure from the nominal value, by the return loss.

C.2.2 *Return loss*

Return loss may be measured in the time domain or in the frequency domain. If the return loss to be measured is independent of frequency both methods will yield the same numerical result.

To measure return loss in the time domain, test signal elements A, $B1$, $B2$ or $B3$, and F shall be used. The return loss is the ratio of the incident to the reflected test signal element, both being measured in peak-to-peak terms. The return loss for each of the above four test signal elements shall be equal to, or greater than, the value specified in Part D.

To measure return loss in the frequency domain, any of several well-established methods may be used. The return loss at all frequencies within the nominal bandwidth of the television system shall be equal to, or greater than, the value specified in Part D.

Note – Care should be taken to ensure that any spectral components produced by the test signal source above the nominal cut-off frequency, f_c, of the television system are attenuated by at least 40 dB relative to components below f_c.

C.2.3 *Non-useful d.c. component*

A signal consisting of synchronizing pulses and blanking-level is used. The potential of the blanking-level with respect to earth is measured with a d.c.-coupled instrument.

C.2.4 *Nominal signal amplitude*

The nominal signal amplitude at points of interconnection is specified in Part D. Conformity with this specification should be assessed by measurement of a composite video signal which contains element $B2$ or $B3$.

C.3 Measurements of transmission performance

C.3.1 *Insertion gain*

The signal element used is $B3$ for 625-line systems and $B2$ or $B3$ for 525-line systems. The amplitude L is measured between the centre of the bar (point b_2 in Fig. 4) and blanking level (point b_1 in Fig. 4). The resulting value of the received signal must remain inside the limits specified in Part D.

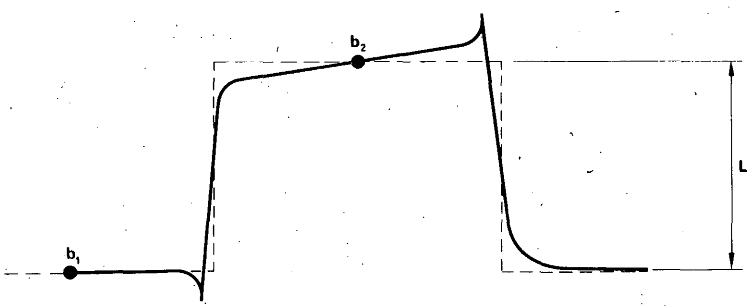


FIGURE 4 – *Insertion gain measurement*

C.3.2 *Noise*

C.3.2.1 *Continuous random noise*

Measuring equipment

In general, measurements should be made with r.m.s.-reading instruments. Depending on the type of instrument to be used, the circuit will carry either no signal or a specified repetitive signal. The latter case may be used if clamping devices have to be activated. For power measurement, the measuring instrument should have, an effective time constant or integrating time of approximately 1 s.

In some cases it may be desirable to precede the noise measuring equipment by a sub-carrier notch filter, so as to eliminate any sub-carrier periodic noise component from the random noise measurement. Consideration must however be given to the effect of such a filter upon the accuracy of measurement.

When the measurements are made by assessing the quasi peak-to-peak amplitude of the noise, administrations are asked to determine the peak factor appropriate for their measuring methods and to express the results in terms of r.m.s. noise amplitude.

Band limiting

The measuring instrument should be preceded by band-limiting filters (see § 1 and 2 of Annex II to Part C of this Recommendation). The lower band limit is such that power-supply hum and microphonic noise are excluded. The upper limit is so selected as to eliminate noise which occurs outside the wanted band of the video signal.

If the circuit carries a signal, band limiting may be necessary, using a 200 kHz high-pass filter, as described in Annex III to Part C of this Recommendation.

Weighting

The measuring instrument should also be preceded by a unified weighting network (see § 3 of Annex II to Part C of this Recommendation).

C.3.2.2 *Low-frequency noise*

Low-frequency noise voltages are usually measured by means of an oscilloscope. The measuring instrument should be preceded by a band-pass filter. The low-pass section of this filter can be as described in § 2 of Annex II to Part C of this Recommendation. In cases where line-frequency synchronizing pulses are required on the circuit under test and where field frequency synchronizing pulses can be omitted, the sharp cut low-pass filter described in [CCIR, 1970-74b] may be preferred. The high-pass section of the filter requires further study.

C.3.2.3 *Periodic noise*

Conventional measuring methods may be used. Measurements of power supply hum including lower-order harmonics should be made through the low-pass filter described in § 2 of Annex II to Part C of this Recommendation. In cases where line-frequency synchronizing pulses are required on the circuit under test and where field-frequency synchronizing pulses can be omitted, the sharp cut low-pass filter described in [CCIR, 1970-74b], may be preferred.

When the frequency of the periodic noise is higher, selective measurement may be necessary for the separation of random noise from periodic noise.

C.3.2.4 *Impulsive noise*

Impulsive noise voltages are measured by means of an oscilloscope.

C.3.3 *Cross-talk from another television channel*

The mechanism which produces cross-talk noise may be dependent upon a signal being transmitted on the disturbed circuit. Accordingly, measurements should be made both with and without a signal on the disturbed circuit.

Suitable combinations of elements $B1$, $B2$, $B3$ and F may be used.

Different values are specified, depending on whether the cross-talk appears more or less uniformly throughout the frequency range of the interfering signal or selectively (differentially), affecting mainly the higher frequencies in the range.

C.3.4 *Non-linear distortion*

C.3.4.1 *Picture signal*

C.3.4.1.1 *Luminance signal*

Luminance non-linearity is measured using the 5-riser staircase test signal element ($D1$) shown in Figs. 11 and 12. At the receiving end, the test signal is passed through a differentiating and shaping network whose effect is to transform the staircase into a train of 5 pulses (by way of example, Annex II (§ 4) to this Part of the Recommendation shows a possible filter, the response of which approximates the sine-squared shape).

Comparing the amplitudes of the pulses, the numerical value of the distortion is found by expressing the difference between the largest and the smallest amplitude as a percentage of the largest.

Note – Some administrations may, on an interim basis, use CCIR Test Signal No. 3 (Recommendation 421-3, Geneva, 1974) instead of the 5-riser test signal.

C.3.4.1.2 *Chrominance signal*

Chrominance non-linearity is measured with the 3-level chrominance signal shown in Figs. 15 ($G2$) and 16.

Gain

Gain non-linearity is defined as the larger of the two values in % obtained by substituting $i = 1$ or $i = 3$ in the expression:

$$100 \times \left| \frac{A_i - k_i A_2}{k_i A_2} \right|$$

where,

A : amplitude of received sub-carrier,

i : position of burst on signal G or $G2$ (1 being the smallest, 3 the largest),

$k_i = \dfrac{2i - 1}{3}$ for 625-line signal $G2$

$k_i = 2^{i-2}$ for 525-line signal G.

It is desirable that the chrominance-luminance gain inequality of the circuit should be within the stated requirements when this measurement is made.

Signal amplitudes should be measured peak-to-peak. A sub-carrier bandpass filter is of assistance in carrying out the measurement.

Phase

Phase non-linearity is defined as the largest difference (in degrees) obtained by comparing the phase of the three bursts in the received signal G or $G2$.

If a vector display is used, it is convenient to normalize the phase of the smallest burst.

C.3.4.1.3 *Intermodulation from the luminance signal into the chrominance signal (Differential gain, differential phase)*

Intermodulation is measured with the test signal element $D2$ shown in Figs. 11 and 12, consisting of a 5-riser staircase with superimposed sub-carrier. At the receiving end, the sub-carrier is filtered from the rest of the test signal and its six sections are compared in amplitude and phase.

Note – Some administrations may, on an interim basis, use a modified version of CCIR Test Signal No. 3 (Recommendation 421-3, Geneva, 1974) with superimposed colour sub-carrier.

Differential gain

Differential gain is expressed by two values, $+x\%$ and $-y\%$, which represent the maximum (peak) differences in amplitude between the sub-carrier on the treads of the received test signal and the sub-carrier on its blanking level, expressed as a percentage of the latter. In the case of a monotonic characteristic either x or y will be zero.

Differential gain in % referred to blanking level, can be found from the expressions below:

$$x = 100\left|\frac{A_{max}}{A_0} - 1\right| \qquad y = 100\left|\frac{A_{min}}{A_0} - 1\right|$$

Peak-to-peak differential gain can be found from the expression:

$$x + y = 100\left|\frac{A_{max} - A_{min}}{A_0}\right|$$

where,

A_0 : amplitude of the received sub-carrier at blanking level;

A : amplitude of the sub-carrier on any relevant tread of the staircase between 0 (blanking level tread) and 5 (top tread) inclusive.

Note – Some administrations use methods in which the denominator in the above expressions for x and y is A_{max} rather than A_0. Results obtained by this method will differ only slightly from those defined above if the magnitude of distortion is not excessive.

Differential phase

Differential phase is expressed by two values, $+x$ and $-y$, in degrees, which represent the maximum (peak) differences in phase between the sub-carrier on the treads of the received test signal, and the sub-carrier on its blanking level expressed in degrees difference from the latter. In the case of a monotonic characteristic either x or y will be zero.

Differential phase in degrees referred to blanking level can be found from the expression below:

$$x = |\Phi_{max} - \Phi_0| \qquad y = |\Phi_{min} - \Phi_0|$$

Peak-to-peak differential phase can be found from the expression:

$$x + y = |\Phi_{max} - \Phi_{min}|$$

where,

Φ_0 : phase of the received sub-carrier at blanking level;

Φ : phase of sub-carrier on any relevant tread of the staircase between 0 (blanking level tread) and 5 (top tread) inclusive.

C.3.4.1.4 *Intermodulation from the chrominance signal into the luminance signal*

Chrominance-luminance intermodulation is measured on element *G*, *G*1 or *G*2 after suppressing the incoming colour sub-carrier. It is defined as the difference between the luminance amplitude in element *G*1, or in the last section of element *G* or *G*2 (b_5 in Figs. 15 and 16) and the amplitude of the succeeding section (b_6 in Figs. 15 and 16) in which the test signal has no sub-carrier, expressed as percentage of the luminance bar amplitude.

C.3.4.2 *Synchronizing signal*

C.3.4.2.1 *Steady-state distortion*

Synchronizing signal steady-state non-linear distortion may be measured using any test signal which will allow the requisite values of average picture level to be obtained.

The distortion is expressed as the difference between sync. amplitude and its normalized value (i.e. 3/7 luminance bar amplitude for 625-line systems, 4/10 luminance bar amplitude for 525-line systems), expressed as a percentage of the normalized value. Measurement is made between the mid-point amplitude of the synchronizing pulse and the mean blanking level.

C.3.4.2.2 *Transient distortion*

Measurement method and test signal are still under study.

C.3.5 *Linear distortion*

C.3.5.1 *Waveform distortion of the luminance signal*

Practical circuits sometimes exhibit amplitude-dependent distortions which show up as linear distortions and which are not detected by the normal non-linear distortion measurement methods [CCIR, 1970-74c and d].

C.3.5.1.1 *Long-time waveform distortion*

Long-time waveform distortion usually deserves consideration only when it assumes the form of a damped, very low-frequency oscillation. It can be measured using any test signal which will allow an adequate change of average picture level to be obtained.

Three parameters can be measured:

- the peak amplitude of the overshoot of the signal (expressed as a percentage of nominal luminance amplitude);
- the time taken for the oscillation to decay to a specified value;
- the slope at the beginning of the phenomenon, expressed in %/s.

C.3.5.1.2 *Field-time waveform distortion*

Field-time waveform distortion is measured with the field-frequency square wave (signal *A*) shown in Figs. 5 and 6a. The magnitude of the distortion is obtained from the maximum departure in level of the bar top from the level at the centre of the bar expressed as a percentage of the bar amplitude as its centre. The first and last 250 μs (approximately 4 lines) are neglected in this measurement.

Alternatively, field-time waveform distortion for 525-line systems is measured with the field bar of the window signal shown in Fig. 6b. The use of the window signal must be noted in the measurement results.

Note – In Canada and the USA, field-time waveform distortion is normally measured as the peak-to-peak level variation over the whole of the bar top, excluding the first and last 250 μs.

C.3.5.1.3 *Line-time waveform distortion*

Line-time waveform distortion is measured with element *B*3 (Fig. 7) for 625-line systems and *B*3 or *B*2 (Fig. 8) for 525-line systems. The magnitude of the top distortion is obtained from the maximum departure in the level of the bar top from the level at the centre of the bar, expressed as a percentage of the bar amplitude as its centre. The first and last 1 μs are neglected in this measurement.

Note – In Canada and the USA, line-time waveform distortion is normally measured as the peak-to-peak level variation over the whole of the bar top, excluding the first and last 1 μs.

The magnitude of bottom distortion (base-line distortion) is obtained from the difference between the level at the point:

– 400 ns for 625-line systems
– 500 ns for 525-line systems

after the half amplitude point of the trailing edge of the bar and the level at a point which follows the bar by an interval equal to half the duration of the bar, and is expressed as a percentage of the bar amplitude. The distortion is to be measured after the bandwidth of the signal has been limited. Limitation may be achieved by the use of a network, the design of which is based on "Solution 3" in [Thomson, 1952], having its first zero at 3.3 MHz, or by an equivalent technique.

Note – Line time waveform distortion (measured at the top of the bar) and base line distortion are likely to be different, both in shape and magnitude.

C.3.5.1.4 *Short-time waveform distortion*

Short-time waveform distortion is measured with $B3$ for 625-line systems and $B3$ or $B2$ for 525-line systems and the sine-squared pulse test signal element $B1$ shown in Figs. 7 and 8. Two measurements of distortion can be made with these signals. The first consists of expressing the amplitude of the pulse as a percentage of the amplitude of the line time bar (element $B2$ or $B3$ in Figs. 7 and 8, as appropriate). The second consists of expressing the amplitude of the lobes, lagging or leading the pulse or bar as a time-weighted percentage of the amplitude of the received pulse or bar respectively.

The results of the foregoing measurements using the sine-squared pulse can be expressed in a compact form in terms of the *K*-rating method which is briefly described in Annex IV to Part C. In this method, equal *K*-values for the different parameters approximately correspond to equal degrees of subjective impairment. Measurements of the bar-edge response of 525-line systems can be expressed in terms of the *S*-rating which is a more recent method, based on broadly similar principles.

C.3.5.2 *Chrominance waveform distortion*

Experience suggests that this form of distortion need not be measured because circuits meeting the requirements for the other parameters in Part D have negligible chrominance waveform distortion.

C.3.5.3 *Chrominance-luminance inequalities*

C.3.5.3.1 *Gain inequality*

Chrominance-luminance gain inequality can be measured with the luminance bar $B2$ and the elements G, $G1$ and $G2$. Alternatively, the chrominance component of the composite pulse F may be used. The magnitude of the distortion is obtained from the departure in peak-to-peak amplitude of the modulated sub-carrier in $G1$, in F, in the last step of G or $G2$, from the amplitude of the luminance bar $B2$, expressed as a percentage of the latter. Account must be taken of the relative amplitudes of $B2$ and G in the original signal for the 525-line case.

A further alternative is to compare the chrominance component of signal F with its luminance component.

C.3.5.3.2 *Delay inequality*

Chrominance-luminance delay inequality is measured on the composite pulse element F. It is expressed in ns, the value being positive when the chrominance element lags the luminance.

C.3.5.4 *Steady state characteristics*

C.3.5.4.1 *Gain*

The gain/frequency characteristic is measured by means of a sweep-frequency method or with the multi-burst test signal C shown in Figs. 9 and 10.

C.3.5.4.2 *Delay*

The delay/frequency characteristic is measured by means of a group-delay measuring set.

REFERENCES

THOMSON, W. E. [1952] The synthesis of a network to have a sine-squared impulse response. *Proc. IEE*, Part III, **99**, 373.

CCIR Documents

[1970-74]: **a.** CMTT/207 (Italy); **b.** CMTT/210 (Italy); **c.** CMTT/188 (Germany (Federal Republic of)); **d.** CMTT/189 (Germany (Federal Republic of)).

ANNEX I TO PART C

TEST SIGNAL ELEMENTS

An indication of the signal elements required to carry out the tests mentioned in this Recommendation is given below in the form of figures. Preferred assemblies for insertion test signals are given in Recommendation 473. The preferred assemblies of elements for full-field measurements is the subject for further study. The reference designations used to describe these elements (e.g. signal $B1$) are the same as the reference designations in Recommendation 473. This Recommendation also contains full specifications of the test signal elements, with the exception of signals A, $B3$ and the window (Fig. 6b).

Note 1 – In the case of PAL and NTSC transmissions, the chrominance sub-carrier of test signal elements should be locked at the phase listed in Table I, where each phase angle is described with reference to the positive $(B-Y)$ axis.

Note 2 – For measurements requiring a change in average picture level (APL), test signals repeating a pattern composed of one line with assemblies of test signal elements followed by three or four consecutive flat lines (e.g. full white, half white, black) should be used. The signal sequence in each field should start at line 24 and 337 in the 625-line system, line 22 and 285 in the NTSC system and line 19 and 282 in the M/PAL system.

TABLE I

Element \ System	PAL	M/PAL (1)	NTSC
*D*2	60 ± 5°	180 ± 1°	180 ± 1°
F	60 ± 5°	180 ± 1°	Not defined
G	60 ± 5°	180 ± 1°	90 ± 1°

(1) Refer to Report 624 for system characteristics.

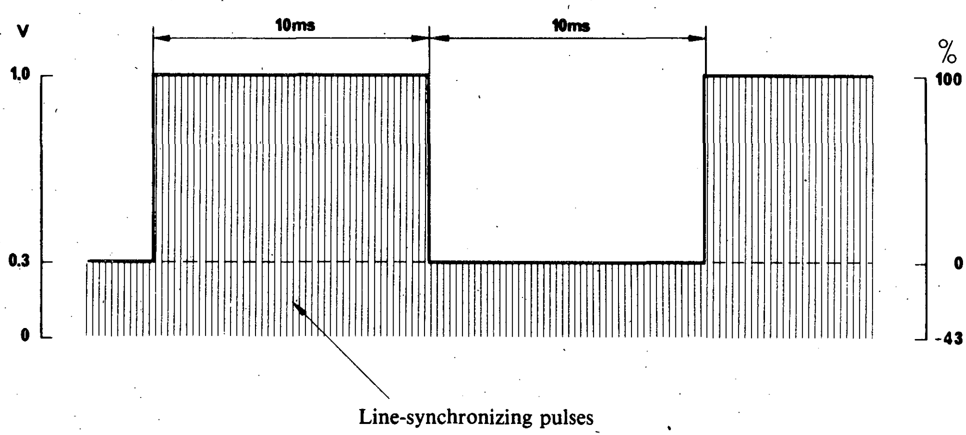


FIGURE 5 – *Signal A for 625-line systems*

Note – This signal may contain field-synchronizing pulses.

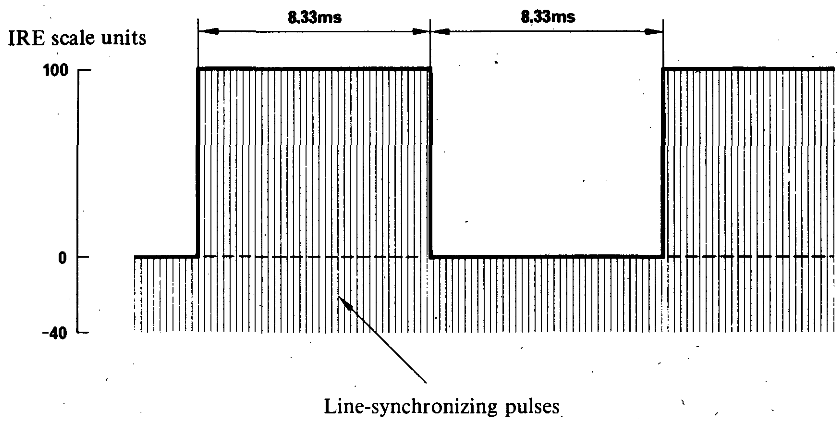


FIGURE 6a – *Signal A for 525-line systems*

Note – This signal may contain field-synchronizing pulses.

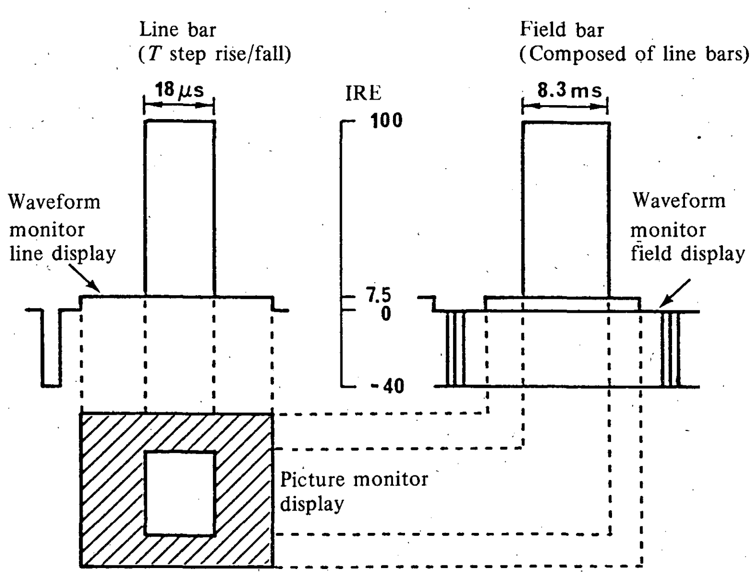


FIGURE 6b – *The window signal for 525-line systems*

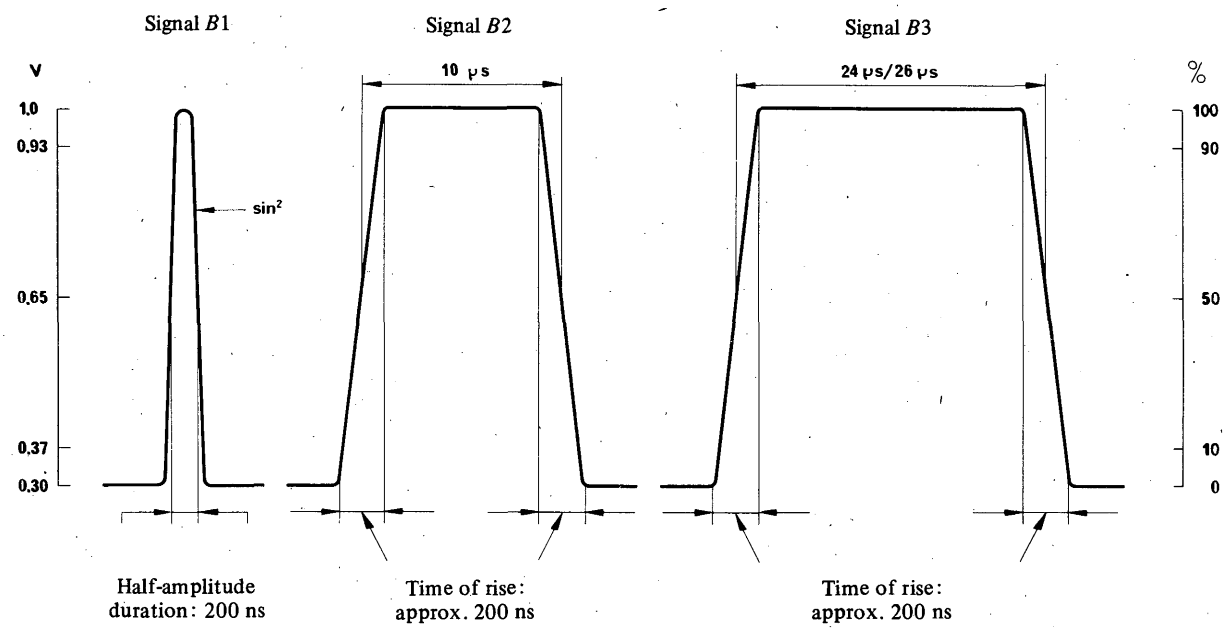


FIGURE 7 – *Signal B for 625-line systems*

Note 1 – In some OIRT countries, a half-amplitude duration of 160 ns is used for *B*1 and a time of rise of 80 ns for *B*2.

Note 2 – In France, the normal time of rise of *B*2 and *B*3 is approximately 110 ns.

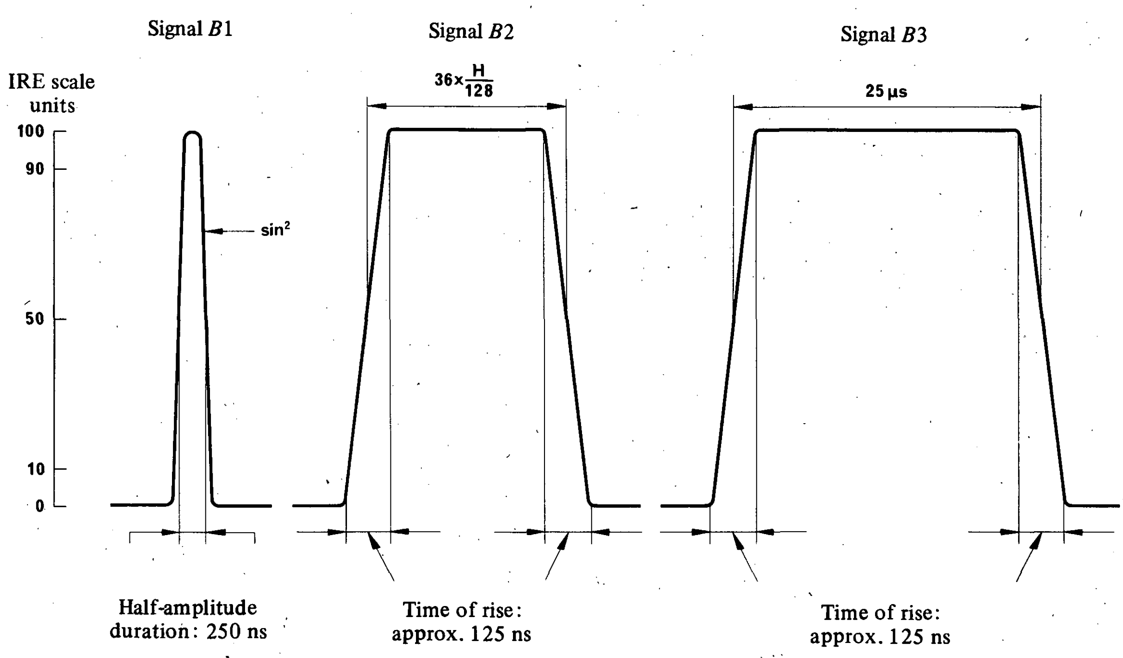


FIGURE 8 – *Signal B for 525-line systems*

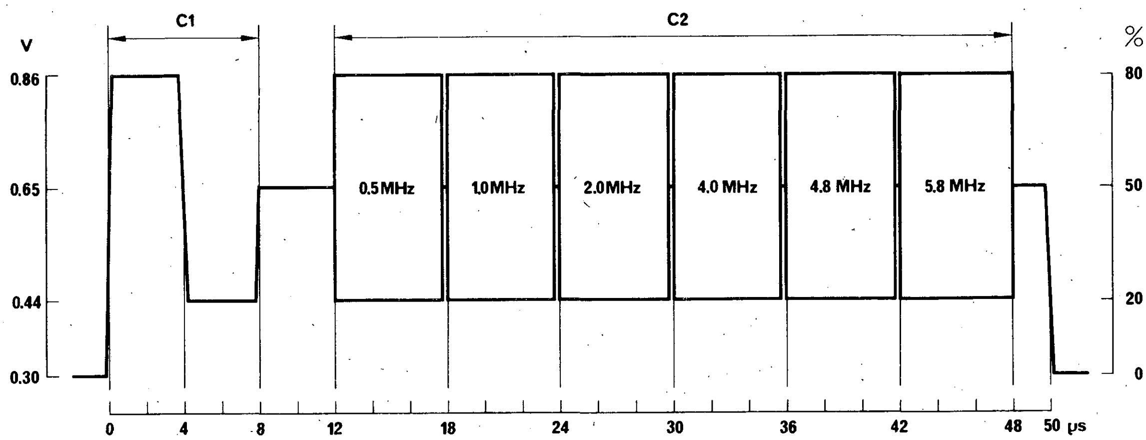


FIGURE 9 – *Signal C for 625-line systems*

Note – Some OIRT countries use 1.5 MHz and 2.8 MHz for the 2nd and 3rd bursts.

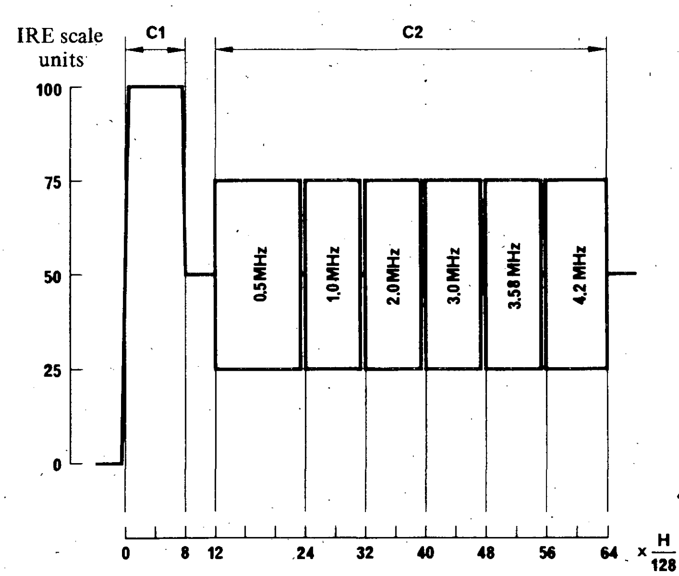


FIGURE 10 – *Signal C for 525-line systems*

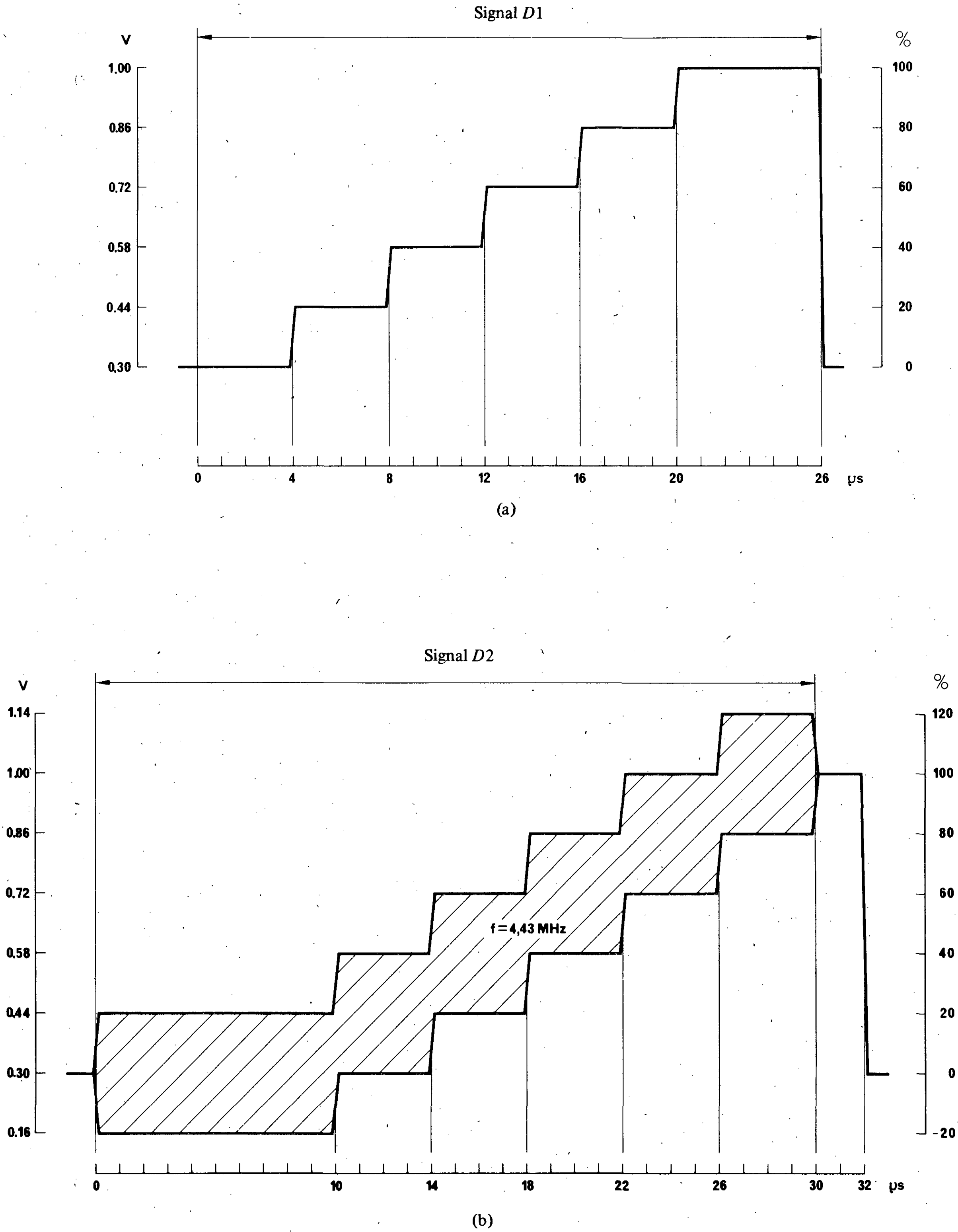


FIGURE 11 – *Signal D for 625-line systems*

Note – In full-field test signals, each tread of the staircase may have a duration of 8.66 μs.

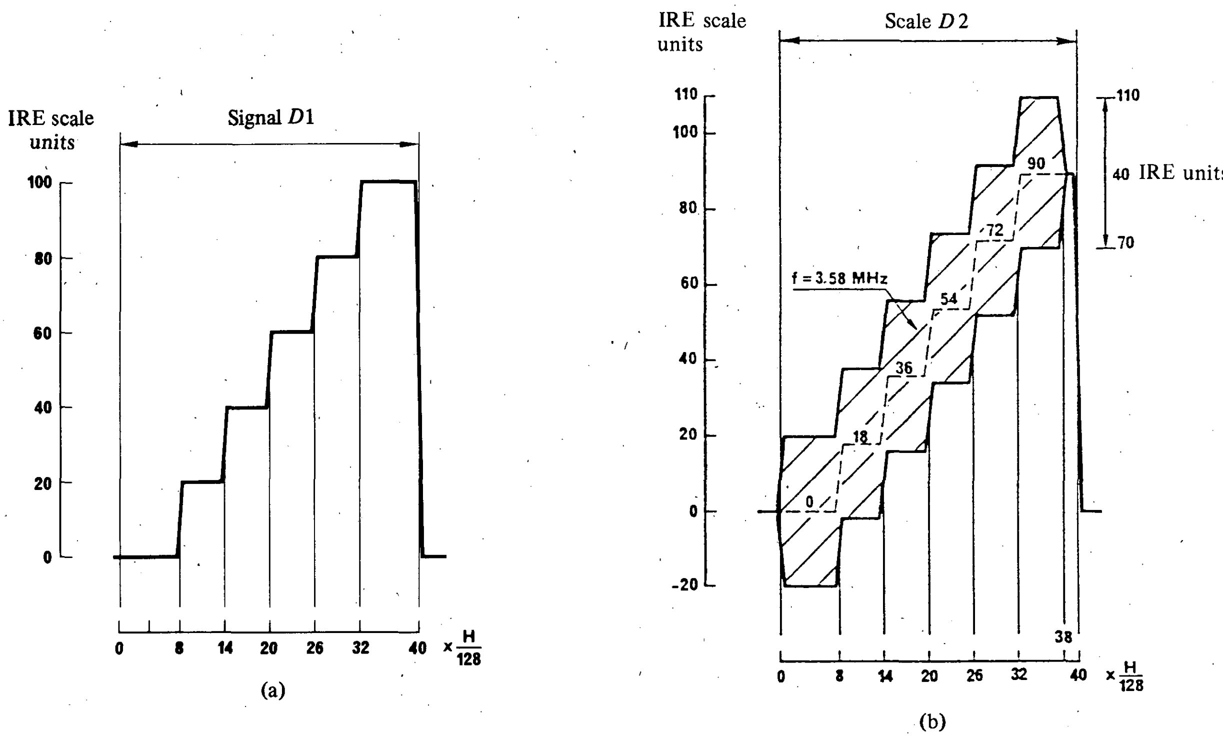


FIGURE 12 – *Signal D for 525-line systems*

Note 1 – Vertical scales give signal amplitudes. In Fig. 12b, the tread levels (in IRE units) are indicated on the dashed line.

Note 2 – Sub-carrier amplitude is ± 20 IRE units.

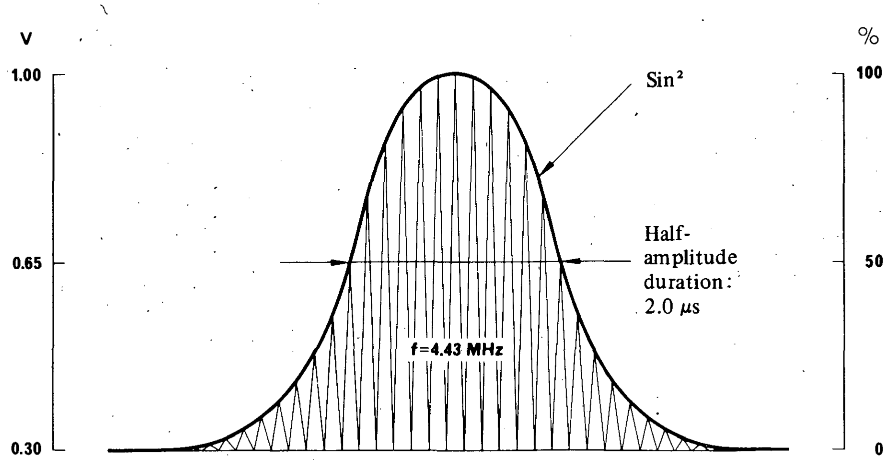


FIGURE 13 – *Signal F for 625-line systems*

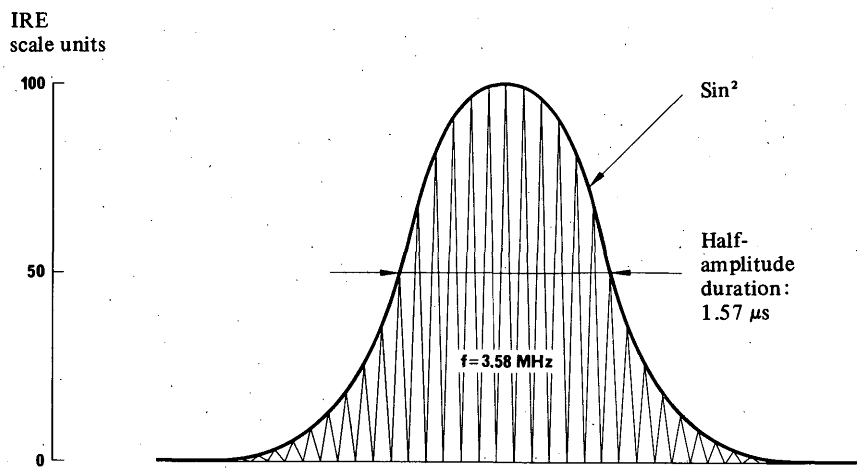


FIGURE 14 – *Signal F for 525-line systems*

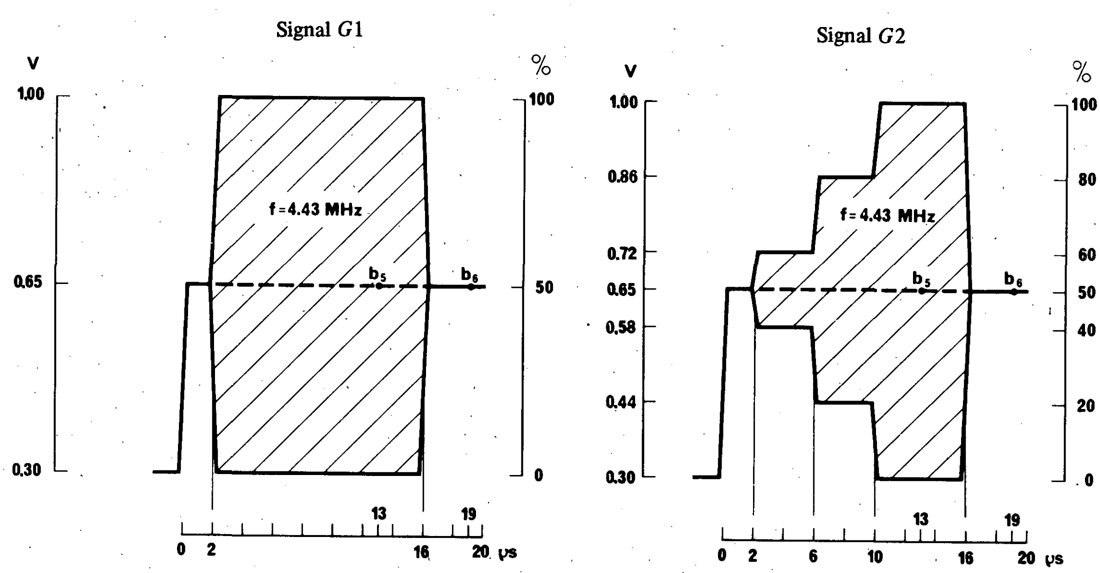


FIGURE 15 – *Signal G for 625-line systems*

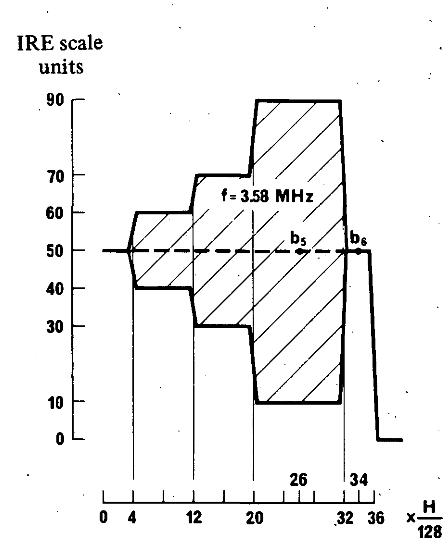


FIGURE 16 – *Signal G for 525-line systems*

ANNEX II TO PART C

DESIGN OF FILTERS USED FOR MEASUREMENTS

1. Low-pass filter for use in noise measurements

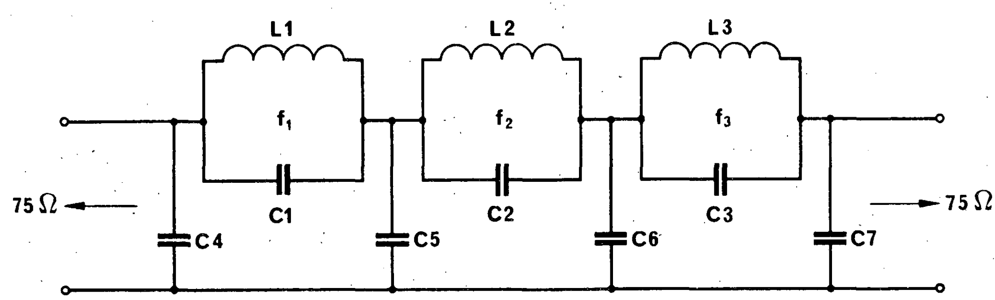


FIGURE 17 – *Low-pass filter diagram*

TABLE OF VALUES

Component	Multistandard value (f_c = 5 MHz)	Tolerance
C1	100	Note 2
C2	545	
C3	390	
C4	428	
C5	563	
C6	463	
C7	259	
L1	2·88	Note 3
L2	1·54	
L3	1·72	
f_1	9·408	
f_2	5·506	
f_3	6·145	

Note 1 – Inductances are given in μH, capacitances in pF, frequencies in MHz.

Note 2 – Each capacitance quoted is the total value, including all relevant stray capacitances, and should be correct to ±2%.

Note 3 – Each inductor should be adjusted to make the insertion loss a maximum at the appropriate indicated frequency.

Note 4 – The *Q*-factor of each inductor measured at 5 MHz should be between 80 and 125.

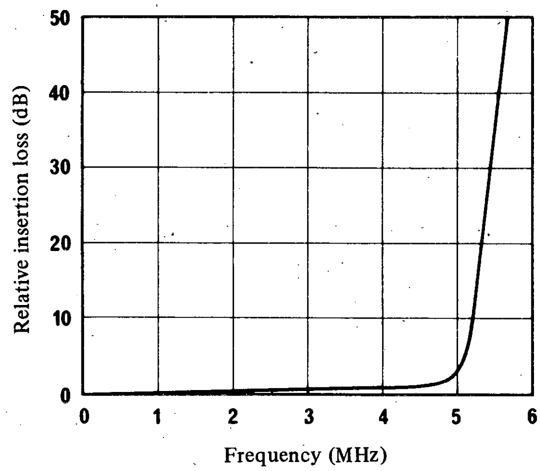


FIGURE 18 – *Low-pass filter characteristic*

2. Combined high-pass, low-pass filter (f_c = 10 kHz)

The high-pass section is used in series with the low-pass filter described in § 1 for measuring continuous random noise.

The low-pass section is used to measure power-supply hum.

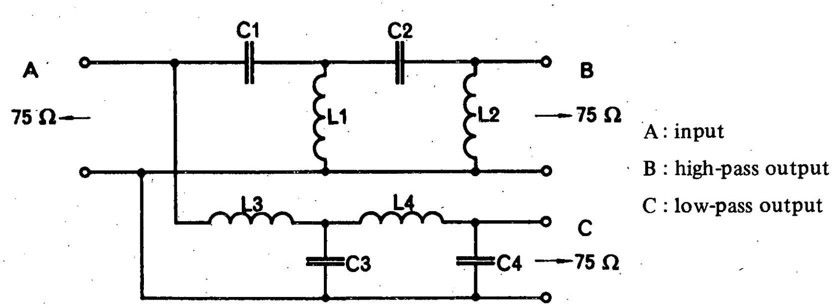


FIGURE 19 – *Combined filter design diagram*

TABLE OF VALUES

Component	Value	Tolerance
C1	139 000	± 5%
C2	196 000	
C3	335 000	
C4	81 200	
L1	0·757	± 2%
L2	3·12	
L3	1·83	
L4	1·29	

Note 1 – Inductances are given in mH, capacitances in pF.

Note 2 – The *Q*-factor of each inductor should be equal to, or greater than, 100 at 10 kHz.

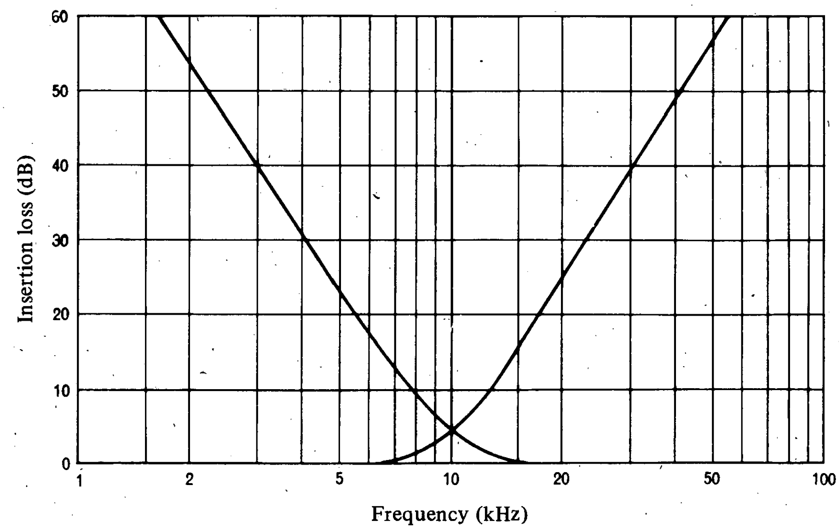


FIGURE 20 – *Combined filter characteristic*

3. Unified weighting network for random noise

3.1 *Network configuration*

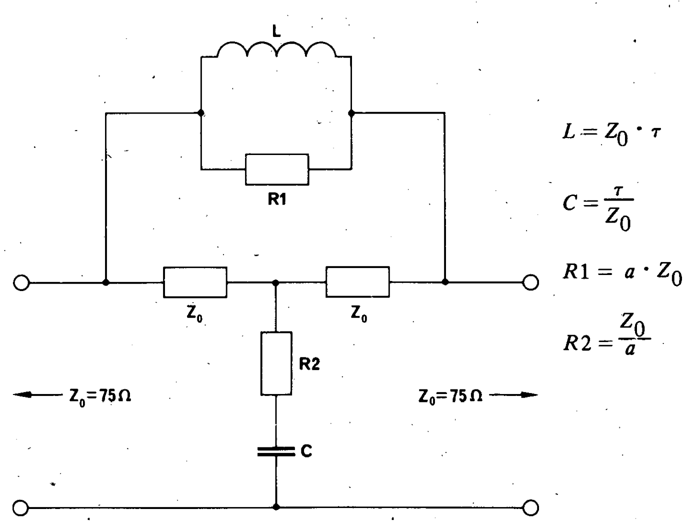


FIGURE 21 – *Network diagram*

3.2 *Insertion loss A*

$$A = 10 \log \frac{1 + \left[\left(1 + \frac{1}{a}\right) \omega\tau\right]^2}{1 + \left[\frac{1}{a}\, \omega\tau\right]^2} \qquad \text{dB}$$

at high frequencies: $A_\infty \rightarrow 20 \log (1 + a)$

where:

$\tau = 245$ ns; $a = 4.5$ $\qquad (A_\infty \rightarrow 14.8$ dB$)$

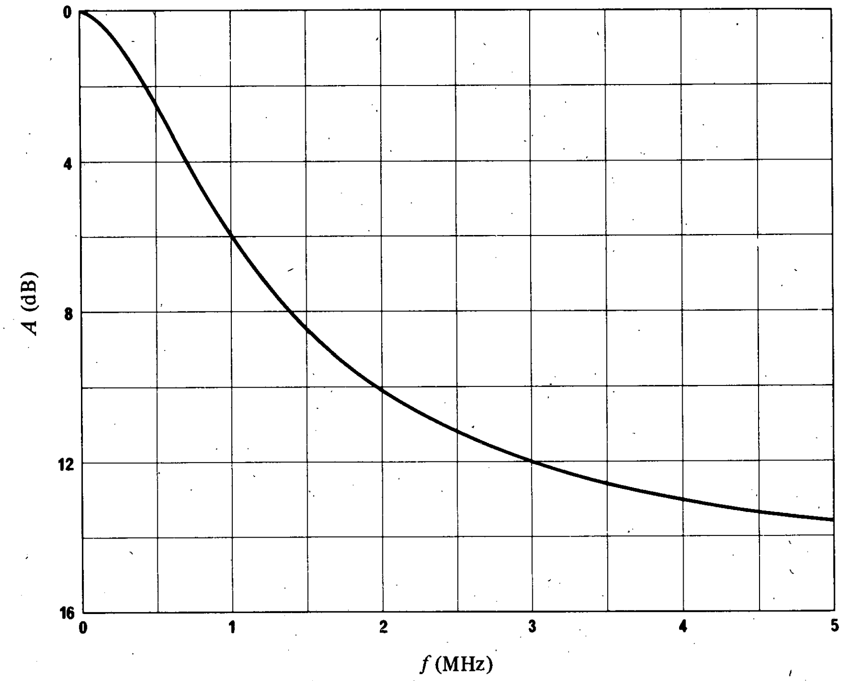


FIGURE 22 – *Unified weighting characteristic*

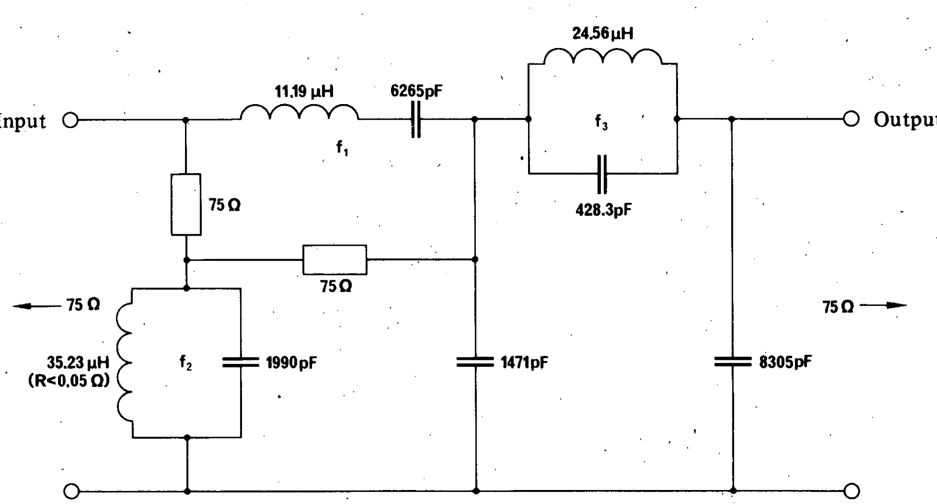
3.3 *Noise weighting factors in a 5 MHz band*

Flat noise: 7.4 dB Triangular noise: 12.2 dB

4. Examples of differentiating and shaping network for luminance non-linearity measurement

Note that the networks shown below have equivalent transfer characteristics.

4.1 *Non-constant resistance form*



f_1 = 601.1 kHz f_2 = 601.1 kHz f_3 = 1551.8 kHz

FIGURE 23 – *Non-constant resistance network diagram*

Note 1 – Capacitor and resistor tolerances ±1%.

Note 2 – Each inductor should be adjusted to resonate at the appropriate indicated frequency.

Note 3 – This network requires to be operated between 75 Ω terminations for correct performance.

4.2 *Constant resistance form*

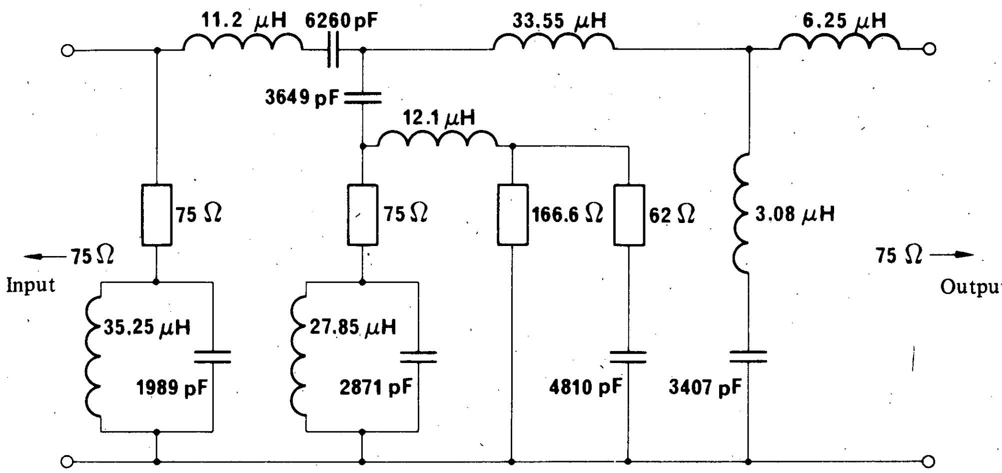


FIGURE 24 – *Constant resistance network diagram*

Note – Capacitor and inductor tolerances ±2%, resistor tolerance ±1%. The Q-factor of each inductor should be equal to, or greater than, 80 at 1 MHz.

4.3 *Step response of staircase differentiating network*

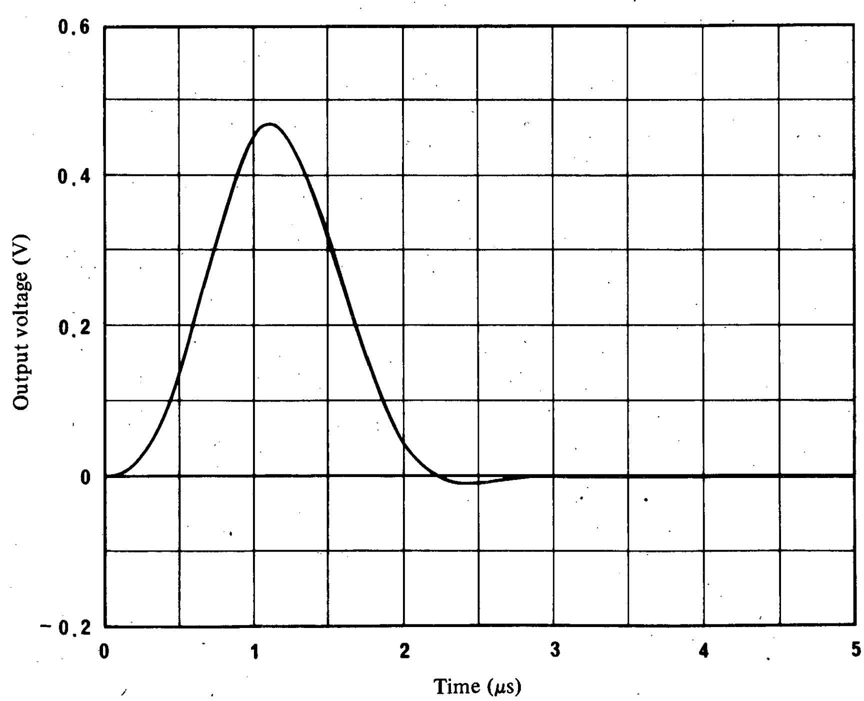


FIGURE 25 – *Transient response of the network*

5. **Thomson filter for use in measurement of line-time waveform distortion**

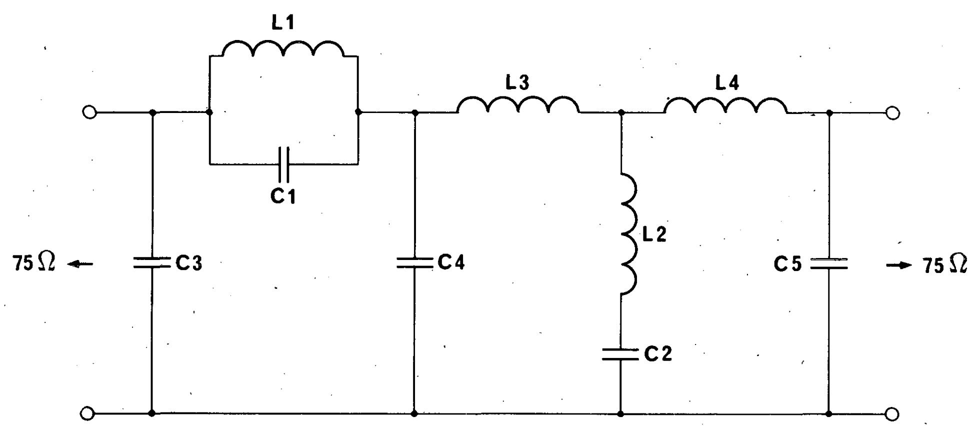


FIGURE 26 – *Thomson filter diagram*

TABLE OF VALUES

Component	Value (f_∞ = 3.3 MHz)
C1	147.7
C2	4044
C3	141.6
C4	1057
C5	310.5
L1	2.948
L2	0.5752
L3	5.767
L4	5.664

Note 1 – f_∞ is the frequency of the first zero of the output/input transfer function.

Note 2 – Inductances are given in μH, capacitances in pF.

Note 3 – For further details see MacDiarmid and Phillips, *Proc. IEE,* Vol. 105B, 440.

ANNEX III TO PART C

METHODS OF MEASUREMENT USING INSERTION TEST SIGNALS

1. **Introduction**

The international insertion test signals (ITS) are described in Recommendation 473. They may be used for out-of-service checks to give results that are closely related to the methods given in the body of this Recommendation or for in-service checks.

Such measurements may give results which differ from those obtained with full-field test signals because:

- the test signal elements may not be identical with those used for full-field tests or their arrangement may be different;
- the measurement result may depend upon the content of the preceding line(s);
- the average picture level (APL) depends upon the nature of the programme signal;
- measurements made on the basis of a single test-line per field may not be fully representative of the performance of any circuit in which half-field-rate dispersal is employed (e.g. in satellite circuits).

To reduce changes in the measurement results due to errors affecting one or several lines it is desirable for the line preceding the test-line(s) to carry a signal having a mean-level of approximately 50%. Such a signal may be a line-bar at 50% of white level or a data signal waveform having a mean-level of approximately 50%.

When measurements with insertion test signals are effected outside programme transmission periods they should be associated with the normal low, mean and high values of average picture level.

Because the methods of measurement specified in Recommendation 569 may in some cases differ from those of this Annex, measurements made by automatic measuring equipment may give results which differ from those obtained normally in accordance with this Annex.

2. Measurement methods

The references given in brackets below refer to the section numbering of Part C.

2.1 *Measurements which are different with insertion test signals*

Insertion gain (§ C.3.1) – Luminance bar amplitude

The signal element used is the luminance bar ($B2$) in line 17.

The amplitude L of the luminance bar is defined as the difference in level between the mid-point of the luminance bar (b_2 in Figs. 27 and 28) and a specified reference point (b_1 in Figs. 27 and 28).

Continuous random noise (§ C.3.2.1)

Measurements are made using either a specified line, which is effaced at the insertion point (lines 22 and 335 for 625-line signals) or on the top of the luminance bar.

Band limiting and noise weighting are specified in § C.3.2.1. In some cases, the measuring instrument may have to be preceded by alternative band limiting filters, for example, when sampling techniques are used and low frequency energy is transferred into the band to be measured. Another problem is described in [CCIR, 1974-78] which also requires different band limiting filters. In such cases upper limiting shall be done by the filter according to § 1 of Annex II to Part C of this Recommendation. Lower limiting shall be done by a first order 200 kHz high-pass filter with a slope of 20 dB per decade. The lower band limit is such that power supply hum, microphonic noise and pulses of line frequency are excluded. The upper limit is so selected as to eliminate noise which occurs outside the wanted band of the video signal.

Note. – Table I of Recommendation 569 shows the effects of using a 200 kHz high-pass filter on the measurement of continuous random noise.

Luminance non-linearity (§ C.3.4.1.1)

§ 3.4.1.1 of Part C applies, except that the test signal element used for 525-line colour transmissions may be $D2$.

Line-time waveform distortion (§ C.3.5.1.3)

The signal element used is the luminance bar ($B2$) in line 17.

The magnitude of the bar-top distortion is the maximum departure in the level in the interval between b_4 and b_3 (Figs. 27 and 28) expressed as a percentage of the bar amplitude.

As shown in Figs. 27 and 28 the first and the last 1 μs are neglected in this measurement.

The magnitude of the base-line distortion is obtained from the difference between the level at the point:

- 400 ns for 625-line systems,
- 500 ns for 525-line systems,

after the half-amplitude point of the trailing edge of the luminance bar and the level at the reference point (b_1 in Figs. 27 and 28) expressed as a percentage of the luminance bar amplitude.

The distortion is to be measured after bandwidth limitation as in § C.3.5.1.3.

2.2 *Measurements which are, in principle, the same with insertion test signals*

The following measurements take the luminance-bar amplitude, as defined in § 2.1 above, as the reference level:

- chrominance-luminance intermodulation (§ C.3.4.1.4);
- short-time waveform distortion (The reference point for the pulse amplitude is also b_1 in Figs. 27 and 28) (§ C.3.5.1.4);
- chrominance-luminance gain inequality (§ C.3.5.3.1).

2.3 *Measurements which are identical with insertion test signals*

- return loss (§ C.2.2);
- cross-talk from another television channel (§ C.3.3);
- chrominance non-linearity (§ C.3.4.1.2);
- differential gain and differential phase (§ C.3.4.1.3);
- synchronizing signal amplitude (§ C.3.4.2);
- chrominance waveform distortion (§ C.3.5.2);
- chrominance-luminance delay inequality (§ C.3.5.3.2);
- steady-state gain/frequency characteristic (§ C.3.5.4.1).

2.4 *Measurements which are impracticable with insertion test signals*

- non-useful d.c. component (§ C.2.3);
- periodic noise (§ C.3.2.2);
- impulsive noise (§ C.3.2.3);
- long-time waveform distortion (§ C.3.5.1.1);
- field-time waveform distortion (§ C.3.5.1.2);
- steady state delay/frequency characteristic (§ C.3.5.4.2).

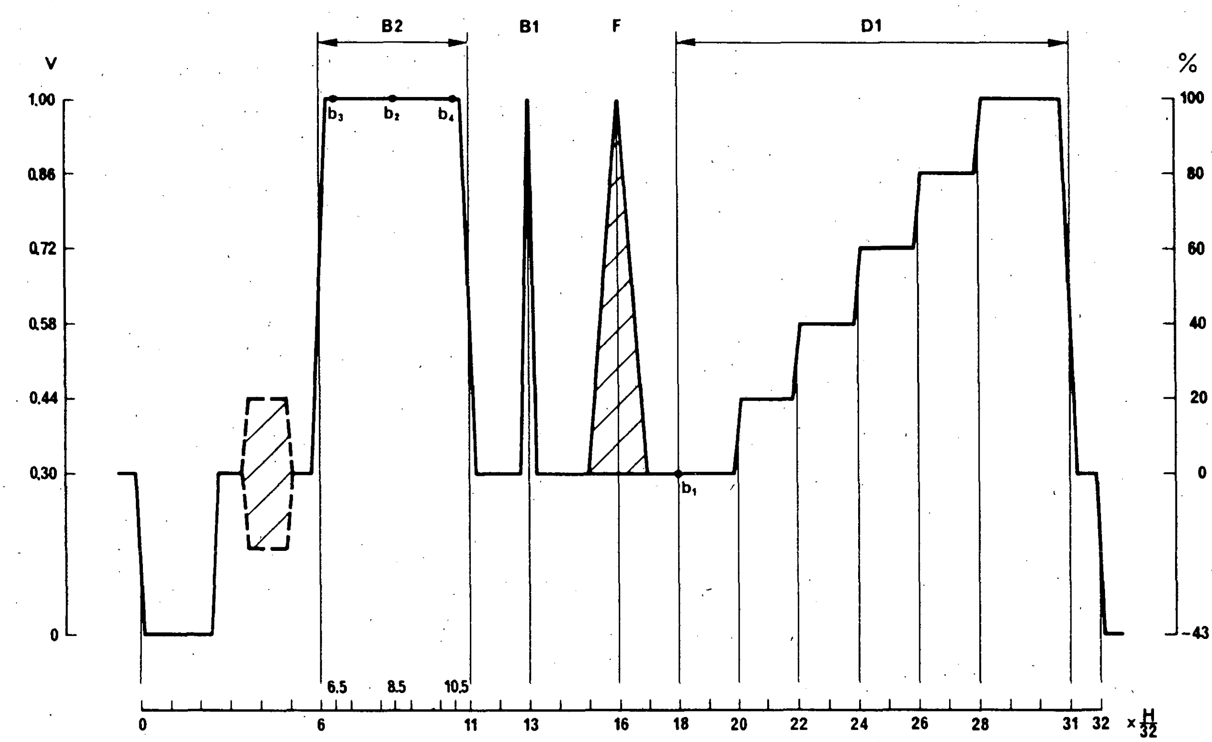


FIGURE 27 – *Line 17 for 625-line systems*

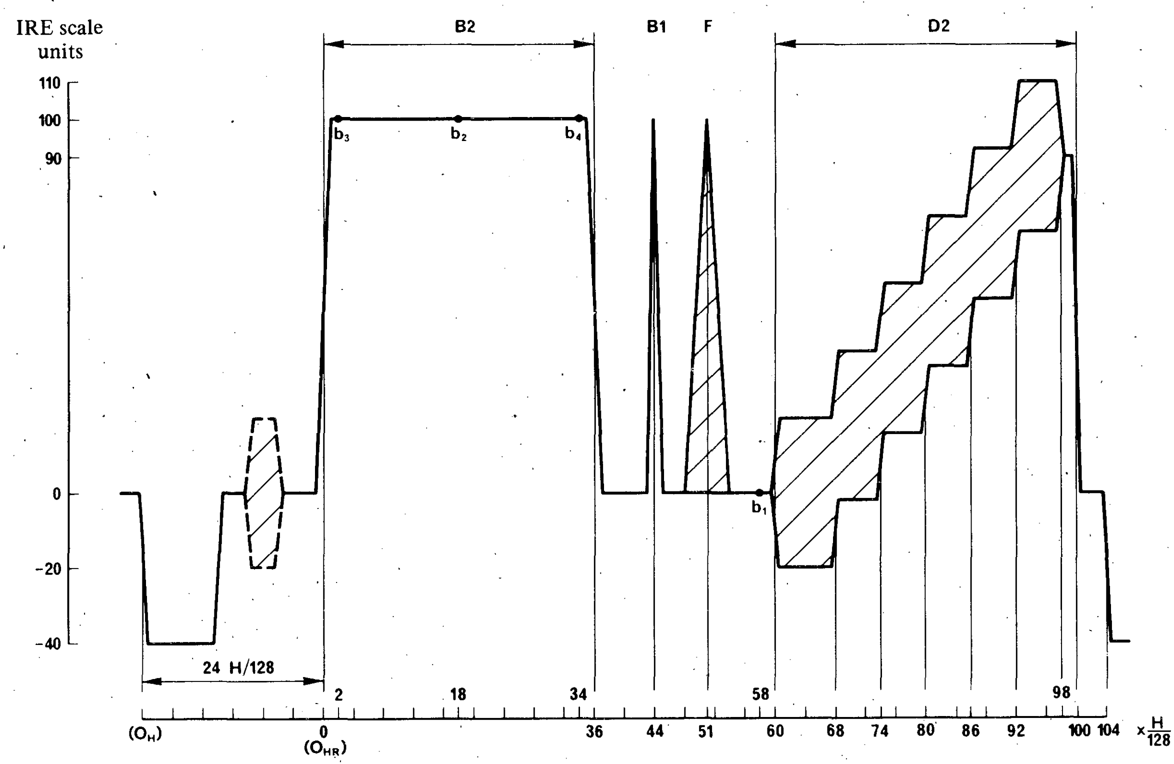


FIGURE 28 – *Line 17/field 1 for 525-line systems*

REFERENCES

CCIR Documents

[1974-78]: CMTT/246 (Germany (Federal Republic of)).

BIBLIOGRAPHY

CCIR Documents

[1974-78]: CMTT/36 (United Kingdom); CMTT/57 (European Broadcasting Union); CMTT/59 (European Broadcasting Union); CMTT/76 (Germany (Federal Republic of)); CMTT/77 (Germany (Federal Republic of)).

ANNEX IV TO PART C

SHORT-TIME WAVEFORM DISTORTION – THE *K*-RATING METHOD OF ASSESSMENT

1. Introduction

This Annex briefly describes the *K*-rating method of assessment of short-time waveform distortion which provides a compact method of expressing the results of the measurements outlined in § C.3.5.1.4. It is based on a deleted Recommendation (Annex II to Recommendation 451, Geneva, 1974) which in turn was based on published papers by [Lewis, 1954] and [Macdiarmid, 1959]. A more recent method, called *S*-rating, for assessing measurements of the bar-edge response on 525-line systems in a broadly similar manner, has been described [Siocos and Chouinard, 1979].

The *K*-rating method, as originally described, was in fact two methods ideally giving the same results:

– the routine-test method, and
– the acceptance-test method.

The routine-test method is based on parameters which can easily be measured on an oscilloscope to give results quickly. The acceptance-test method, based on the response to a T sine-squared pulse, is more rigorous and well suited to the analysis of systems and networks in addition to acceptance tests on hardware. The rating method has been devised so that equal K-values obtained for the various parameters correspond approximately to equal subjective impairments on pictures.

Section 2 shows how the performance objectives and tolerances for short-time waveform distortion can be expressed using the routine-test K-rating method. Section 3, for completeness, outlines how the acceptance-test method could be used.

2. Routine-test method

For the first two parameters, the response to the $2T$ sine-squared pulse ($B1$) and one of the bar elements ($B2$ or $B3$) are used. The third parameter is not normally measured on circuits and equipments for the transmission of composite colour signals. It is included here for possible future use on circuits for colour signals in analogue component form. The test-signal element required is a T sine-squared pulse, where $T = 1/2F_c$ (F_c is the nominal bandwidth of the channel-under-test).

2.1 *2T pulse response*

For a particular value of $K_{(2T)}$, a mask of the type shown in Figs. 29a or 29b is required. The tolerances on the response at the time intervals shown in Fig. 29a correspond to $\pm\, 4K$ at $\pm$ 200 ns, $\pm\, 2K$ at $\pm$ 400 ns and $\pm\, K$ at $\pm$ 800 ns and beyond, with the same values at the greater times in Fig. 29b.

For the masks illustrated in Figs. 29a and 29b:

$$K_{(2T)} = 3\%$$

2.2 *2T pulse/bar ratio*

The $2T$ pulse/bar ratio (P/B) is related to $K_{(P/B)}$ by:

$$K_{(P/B)} = \frac{1}{4}\left|\frac{B}{P} - 1\right| \times 100\%$$

2.3 *T pulse response*

This measurement is not necessary when the circuit has to meet the close tolerances on chrominance-luminance gain and delay inequalities required for composite colour signals. In other cases, the tests using only the $2T$ pulse leave distortions in the upper half of the transmission band virtually untested, so that a test using the T pulse becomes necessary.

Limits to the response to the T pulse cannot be specified rigidly because the spectrum of the T pulse extends far beyond the nominal upper frequency limit of the circuit, and the response must therefore contain irrelevant information. A partial solution has been found by the insertion of a phase-corrected low-pass filter with a sharp cut-off at the edge of the nominal channel band, between the channel under test and the oscilloscope. The filter is first measured using a local test signal. The pulse-to-bar ratio is then, say y (y will be in the region of 0.82). The channel under test is then connected to the filter and the pulse-to-bar ratio measured. From this, the T-pulse rating is, approximately:

$$K_{(T)} = \frac{1}{4}\left|y \cdot \frac{B}{P} - 1\right|$$

Delay errors near the edge of the channel pass-band can also affect the T-pulse K-rating. An estimate of the effects of such errors can be obtained from the change, caused by the channel, in the first pre- and post-overshoots measured at the filter output. The change in overshoot (normalized to the pulse amplitude) is, approximately, $3K_{(T)}$.

3. Acceptance-test method

From the measured T-pulse response and the measured or assumed response of the measuring equipment itself, the "filtered impulse response" is derived and expressed in the form of a normalized time series [Lewis, 1954]. The "main" term of this series represents the ideal or non-distorting part, and the "echo" terms represent the distorting parts. The amplitudes of the echo terms should meet the following four sets of limits giving four values of K.

Let the time series representing the filtered impulse response be:

$$B(rT) = \ldots B_{-r}, \ldots B_{-1}, B_0, B_{+1}, \ldots B_{+r}, \ldots$$

and assume that this has already been normalized so that $B_0 = 1$; let the serial product of $B(rT)$ and the series [½, 1, ½] be

$$C(rT) = \ldots C_{-r}, \ldots C_{-1}, C_0, C_{+1}, \ldots C_{+r}, \ldots$$

where:

$$C_r = \frac{1}{2} B_{r-1} + B_r + \frac{1}{2} B_{r+1}$$

then:

$$K1 \geqslant \frac{1}{8} \left| r \cdot \frac{C_r}{C_0} \right| \qquad \text{for } -8 \leqslant r \leqslant -2 \quad \text{and} \quad +2 \leqslant r \leqslant +8$$

$$K1 \geqslant \left| \frac{C_r}{C_0} \right| \qquad \text{for } r \leqslant -8 \quad \text{and} \quad r \geqslant +8$$

and:

$$K2 = \frac{1}{4} \left| \left(\frac{1}{C_0} \sum_{-8}^{+8} B_r \right) - 1 \right|$$

$$K3 = \frac{1}{6} \left| \left(\sum_{-8}^{+8} B_r \right) - 1 \right|$$

$$K4 = \frac{1}{20} \left\{ \left(\sum_{-8}^{+8} |B_r| \right) - 1 \right\}$$

The series $C(rT)$ represents fairly closely the response to a $2T$ pulse. $K1$ is thus approximately equivalent to $K_{(2T)}$ in the routine test method. $K2$ places limits on the $2T$ pulse/bar ratio and is approximately equivalent to $K_{(P/B)}$ in the routine-test method. $K3$ places limits on the pulse/bar ratio of the response to a hypothetical pulse-and-bar test signal in which the pulse is an ideal filtered impulse and is approximately equivalent to $K_{(T)}$ in the routine-test method. $K4$ places an upper limit on the average amplitude, ignoring signs, of the 16 central echo terms, to protect against rarely-met distortions such as a long train of echoes whose magnitudes are not great enough individually to reach one of the other limits. It has no routine-test equivalent.

REFERENCES

LEWIS, N. W. [1954] Waveform responses of television links. *Proc. IEE*, Vol. 101, Part III, 258-270.

MACDIARMID, I. F. [1959] Waveform distortion in television links. *POOEJ*, Vol. 52, 108-114 and 188-195.

SIOCOS, C. A. and CHOUINARD, G. [June, 1979] Subjective impairment units in relation with oscilloscope graticules for evaluating short-time linear waveform distortion of the luminance signal in 525-line television. *IEEE Trans. Broadcasting*, Vol. BC-25, **2**, 63-71.

PART D – DESIGN OBJECTIVES AND TOLERANCES APPLICABLE TO HYPOTHETICAL REFERENCE CIRCUITS

D.1 Introduction

The purpose of this Part is to provide design objectives and tolerances for the transmission performance characteristics described in § 2 and 3 of Part B. The design objectives and tolerances shown are applicable to circuits which are required to transmit 525- and/or 625-line television signals, these being either monochrome signals or colour signals coded in accordance with such systems as NTSC, PAL or SECAM, as described in Report 624. International circuits using equipment designed by the time this Recommendation is adopted may have characteristics different from those in this part.

Section numbering in this part is related to section numbers of Part B. Sections which are not necessarily relevant to monochrome transmission are D.3.4.1.2, D.3.4.1.3, D.3.4.1.4, D.3.5.2 and D.3.5.3.

This Recommendation does not contain any explicit definition of the bandwidth required for the transmission of colour television signals having at various times any of the standards defined in Report 624. As some of these standards require a bandwidth of 6 MHz, this is the only figure that can be considered to be fully satisfactory. However, because this bandwidth would cause considerable difficulty in countries using standards requiring appreciably smaller bandwidths, it is proposed that, for international circuits which may be required to transmit signals on any standard, it will be sufficient to define the performance of these circuits up to a frequency of 5.5 MHz, unless otherwise specified in the manner indicated in § D.3.5.4. It is noted, however, that countries using 6 MHz bandwidth may have to be protected from interference lying in the unspecified frequency band (from 5.5 to 6 MHz), which for them is in-band interference by means of a phase-corrected low-pass filter. An example of a suitable filter is given elsewhere [CCIR, 1970-74a].

D.2 Objectives and tolerances at points of video interconnection

D.2.1 *Nominal impedance*

At video interconnection points, the input and output impedance (Z_0) of each section should be either unbalanced to earth with a nominal value of 75 Ω resistive, or balanced to earth with a nominal value of 124 Ω resistive.

D.2.2 *Return loss*

At video interconnection points, the return loss, relative to Z_0, of a measured impedance Z should not be less than 30 dB.

D.2.3 *Non-useful d.c. component*

At video interconnection points, any non-useful d.c. component should not exceed 2.75 V, when terminated in the nominal impedance, or 5.5 V in an open circuit.

D.2.4 *Nominal signal amplitude*

The nominal peak-to-peak amplitude of the monochrome video signal (M in Fig. 3) is 1.0 V.

The nominal peak-to-peak amplitude of a composite colour video signal (H in Fig. 3) will depend upon the characteristics of the particular colour television system employed (see the equations given in Item 2.9 of Table II of Report 624), but for circuits required at various times to transmit all systems covered by that Report, a maximum value of 1.25 V should be assumed.

D.3 Transmission performance objectives and tolerances

Tolerances proposed here are expected to apply for most of the time but may be exceeded for part of the time. Further study is required [CCIR, 1970-74b].

It is assumed that any earth stations in the circuit will be operating with a G/T ratio of no less than 40.7 dB and will be transmitting sound on a separate carrier. The tolerances do not necessarily apply to stations operating under different conditions.

D.3.1 *Insertion gain*

The insertion gain, after initial or routine adjustment, should be 0 ± 0.5 dB.

D.3.1.1 *Variations in the insertion gain*

Any variation of insertion gain with time should not exceed the following limits:

– short period variations (e.g. 1 s): ± 0.3 dB;
– medium period variations (e.g. 1 h): ± 0.5 dB.

D.3.2 *Noise*

D.3.2.1 *Continuous random noise*

When the noise is band limited and weighted in accordance with Part C of this Recommendation, the signal-to-weighted-noise ratio should not fall below 53 dB for more than 1% of any month, nor below 45 dB for more than 0.1% of any month.

Note 1 – Where colour television is concerned, noise measurements using the unified network can only be considered as giving a valid indication of the subjective impairment due to the noise in cases where the noise power per unit bandwidth at 5 MHz does not exceed that at 1 MHz by more than about 11 dB. This condition will be met in the majority of cases with existing transmission systems and only the recommended weighting network will be used for operational purposes. For new systems that would not meet this condition, designers should verify by other means that signal-to-weighted-noise ratio is satisfactory and that the recommended weighting network gives satisfactory results (see Recommendation 568).

Note 2 – There may be administrations which need, for national purposes, values of signal-to-noise ratio which differ from 53 dB.

Note 3 – At present satellite circuits may not be able to meet the performance objective for continuous random noise. Currently achievable values of signal-to-noise ratios are given in Report 965.

D.3.2.2 *Low-frequency noise*

It is not possible to give objectives for low-frequency noise at the present time. A signal-to-noise ratio of 43 dB has been suggested by one administration for System M. Contributions on this parameter are invited from other administrations.

D.3.2.3 *Periodic noise*

For power supply hum including lower-order harmonics, the signal-to-noise ratio should not be less than 35 dB. For single-frequency noise between 1 kHz and 5.5 MHz, the signal-to-noise ratio should not be less than 55 dB.

Note – Circuits which are only required to transmit 525-line signals need only be tested up to 4.2 MHz.

D.3.2.4 *Impulsive noise*

For impulsive noise of a sporadic or infrequently-occurring nature, the signal-to-noise ratio should not be less than 25 dB.

D.3.3 *Cross-talk from another television channel*

If the cross-talk is substantially undistorted, the signal-to-cross-talk ratio should not be less than 58 dB. If the cross-talk is substantially "differentiated" (i.e. cross-talk voltage proportional to frequency), the signal-to-cross-talk ratio should not be less than 50 dB.

D.3.4 *Non-linear distortion*

In this section the terms "low average picture level" and "high average picture level" are used. For low average picture level, either 10% or 12.5% is acceptable. For high average picture level, either 87.5% or 90% is acceptable.

The figures shown for a sending level of +3 dB are included only as a guide for designers and require further study. The question as to whether +3 dB is the optimum test signal level for use in specifying circuit overload performance also requires further study.

D.3.4.1 *Picture signal*

D.3.4.1.1 *Luminance signal*

On circuits designed for colour television transmission, the distortion should not exceed 5% at high or low average picture level. The figure applicable to a test signal sent at 3 dB above normal level is 10% at the same average picture levels. On circuits designed for monochrome television transmission only, these objectives should be 12% and 24% respectively.

D.3.4.1.2 *Chrominance signal*

For System M, the following figures apply:

Chrominance gain non-linearity. The distortion should not exceed 4% at high or low average picture level. In addition, when the signal is sent at 3 dB above the normal amplitude, the distortion should not exceed 8% at the same average picture levels.

Chrominance phase non-linearity. The distortion should not exceed 4° at high or low average picture level. In addition, when the signal is sent at 3 dB above the normal amplitude, the distortion should not exceed 8° at the same average picture levels.

Figures for other systems require further study.

D.3.4.1.3 *Intermodulation from the luminance signal into the chrominance signal*

Differential gain

Differential gain should not exceed the following figures at high or low average picture level:

For 3.58 MHz: x or y or $x + y$: 10%

For 4.43 MHz: x or y : 10%; $x + y$: 12%.

The figures applicable to a signal sent at +3 dB are double those shown above.

Differential phase

Differential phase should not exceed the following figures at high or low average picture level:

For 3.58 MHz: x or y or $x + y$: 5°

For 4.43 MHz: x or y : 5°; $x + y$: 6°.

The figures applicable to a signal sent at +3 dB are double those shown above.

D.3.4.1.4 *Intermodulation from the chrominance signal into the luminance signal*

The distortion should not exceed ± 3% at low or high average picture levels. The figure applicable to a test signal sent at 3 dB above normal amplitude is ± 6% at the same average picture levels.

D.3.4.2 *Synchronizing signal*

D.3.4.2.1 *Steady-state distortion*

The distortion should not exceed ± 10% at high or low average picture level. The figure applicable to a test signal sent at 3 dB above normal amplitude is ± 20% at the same average picture levels.

D.3.4.2.2 *Transient distortion*

It is not possible to give limits for transient distortion at this time (see Report 636).

D.3.5 *Linear distortion*

D.3.5.1 *Waveform distortion of the luminance signal*

D.3.5.1.1 *Long-time waveform distortion*

It is not possible to give limits for long-time waveform distortion at this time (see Report 636).

D.3.5.1.2 *Field-time waveform distortion*

Field-time waveform distortion should not exceed ± 6%.

Note – This objective applies to circuits which do not contain waveform clamping devices.

D.3.5.1.3 *Line-time waveform distortion*

Line-time waveform distortion should not exceed ± 3%. This figure applies to a measurement on the top of the bar. A limit for base-line distortion requires further study.

D.3.5.1.4 *Short-time waveform distortion*

The sine-squared pulse-to-bar ratio should lie within the limits 100 ± 12%, corresponding to $K_{(P/B)} = 3\%$.

The pulse lobes should lie within the limits shown in Fig. 29a for 625-line systems and in Fig. 29b for 525-line systems, corresponding to $K_{(2T)} = 3\%$.

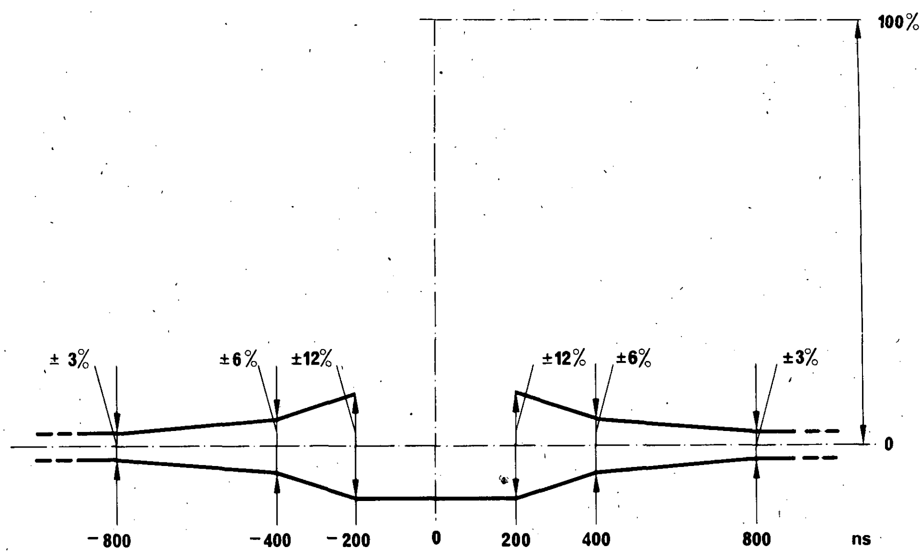


FIGURE 29a – *Mask for response to test signal B1 (625-lines)*
(Half-amplitude duration 200 ns)

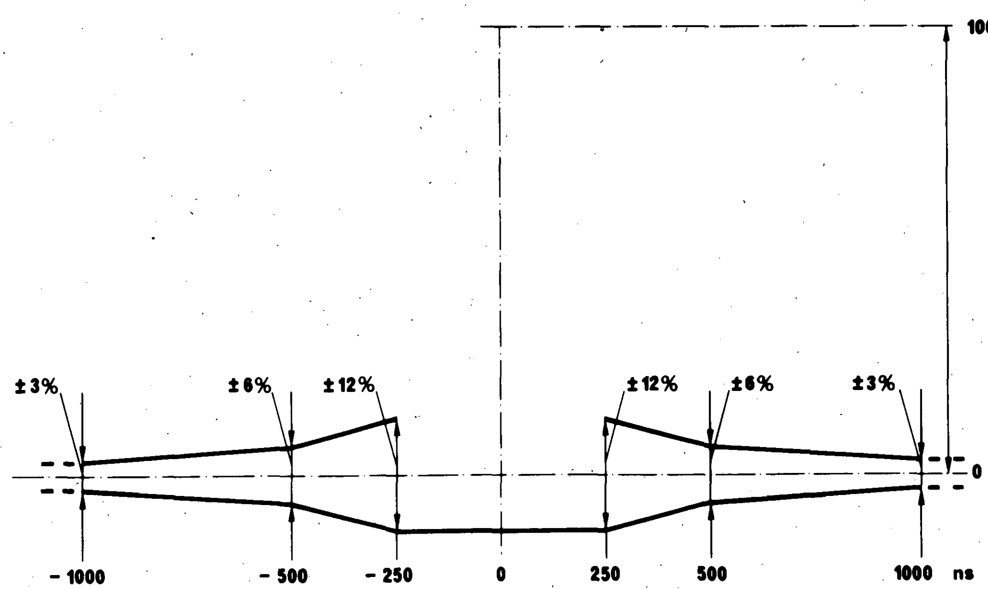
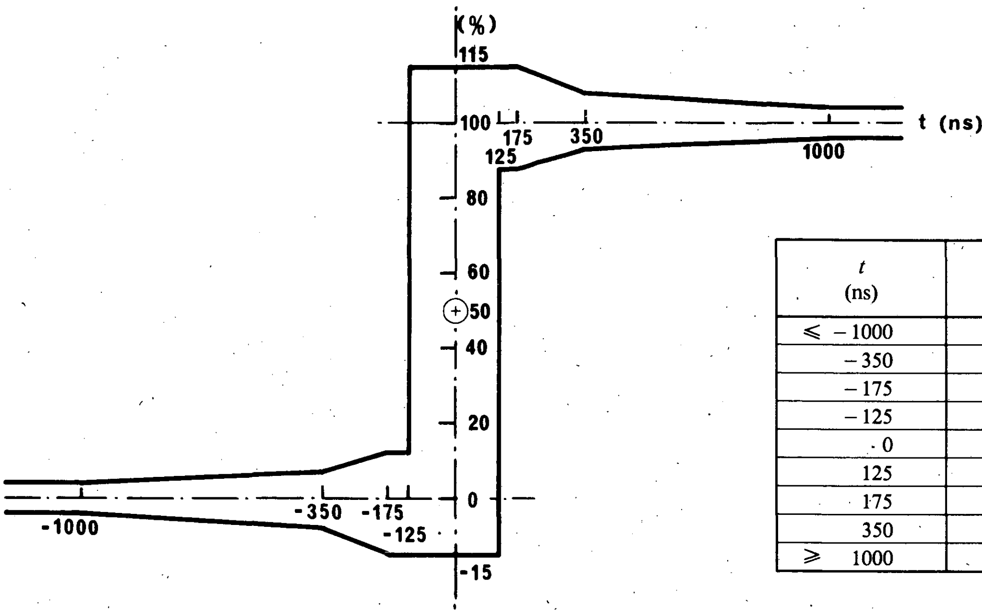


FIGURE 29b – *Mask for response to test signal B1 (525-lines)*
(Half-amplitude duration: 250 ns)

The response to $B2$ or $B3$ test signals for 525-line systems in Japan and Canada only should lie within the limits shown in Fig. 29c.



t (ns)	Outer dimension (%)	Inner dimension (%)
⩽ −1000	−4	4
−350	−8	7
−175	−15	12
−125	−15	12
0	115/−15	–
125	115	88
175	115	88
350	108	93
⩾ 1000	104	96

FIGURE 29c – *Mask for response to B2 or B3 test signals (525-lines for Japan and Canada only)*
(Time of rise: approx. 125 ns)

Note – The mask depicted in Fig. 29c is based on a reference distortion (S) of 3% [CCIR, 1978-82a, b; Siocos and Chouinard, 1979]. For other values of reference distortion (S), the graticule dimensions at the critical points of inflection (at ±175, ±350 and ±1000 ns) are given by the following:

$$\text{Outer dimension: } (100+) \text{ or } (0-) \; \frac{100 \cdot S \cdot A}{100 - S \cdot A} \;\%$$

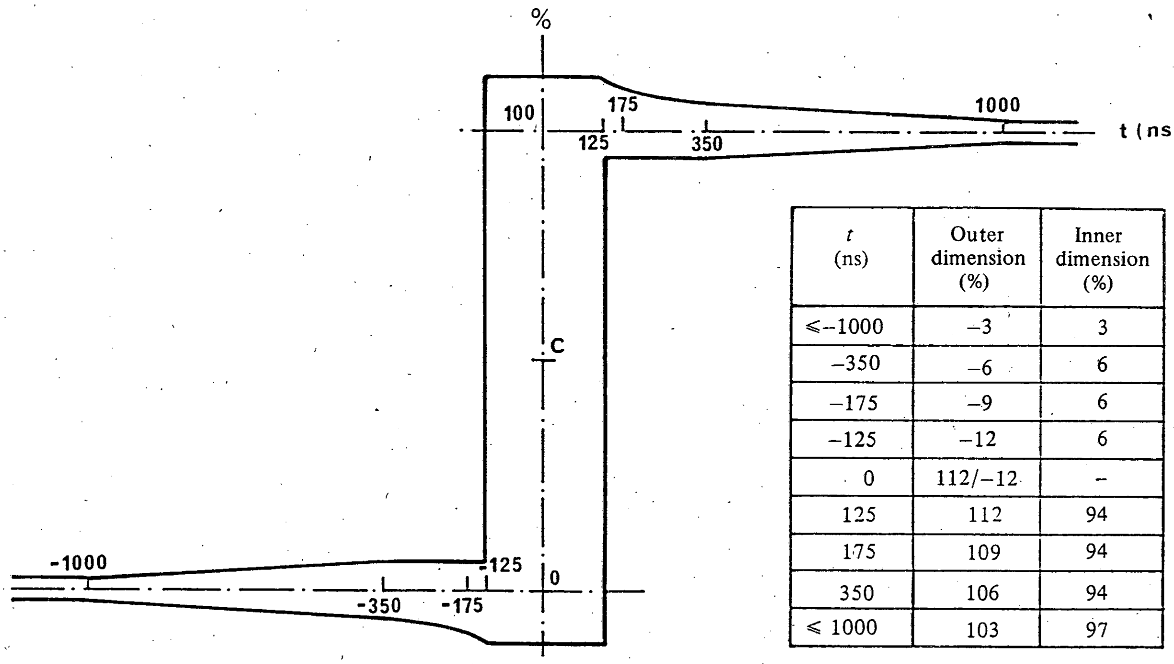
$$\text{Inner dimension: } (100-) \text{ or } (0+) \; \frac{100 \cdot S \cdot A}{100 + S \cdot A} \;\%$$

where S is the defined reference distortion (%) and A is the weighting constant at the critical points of inflection given for t(ns) relative to the reference time at the graticule centre as follows:

TABLE II

t (ns)	A
±175	4.455
±350	2.4128
±1000	1.3414

The response to $B2$ or $B3$ test signals for 525-line systems in the United States of America only should lie within the limits shown in Fig. 29d.



t (ns)	Outer dimension (%)	Inner dimension (%)
⩽−1000	−3	3
−350	−6	6
−175	−9	6
−125	−12	6
0	112/−12	–
125	112	94
175	109	94
350	106	94
⩽ 1000	103	97

FIGURE 29d – *Mask for response to B2 or B3 test signals (525-lines for the United States of America only)*
(Time of rise: approximately 125 ns)

Note – The mask depicted in Fig. 29d is based on a reference distortion (S) of 3% [IEEE, 1979].

D.3.5.2 *Chrominance waveform distortion*

Refer to § C.3.5.2.

D.3.5.3 *Chrominance-luminance inequalities*

D.3.5.3.1 *Gain inequality*

Gain inequality should not exceed ± 10%.

D.3.5.3.2 *Delay inequality*

Delay inequality should not exceed ± 100 ns.

D.3.5.4 *Steady-state characteristics*

The following limits may be found useful by designers but, because the relationships between time domain and frequency domain characteristics are so complex, their use may sometimes give rise to results which conflict with those obtained with test waveforms. If this occurs the waveform results should be considered to be definitive.

D.3.5.4.1 *Gain*

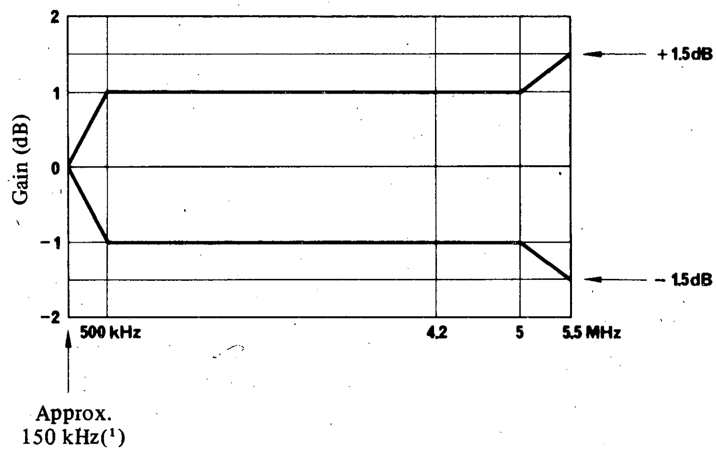


FIGURE 30 – *Limits for gain/frequency characteristic*

(1) May be referred to element $C1$ of test signal C.

Note – For 525-line transmission the limits need be met up to a frequency of only 4.2 MHz.

D.3.5.4.2 *Delay*

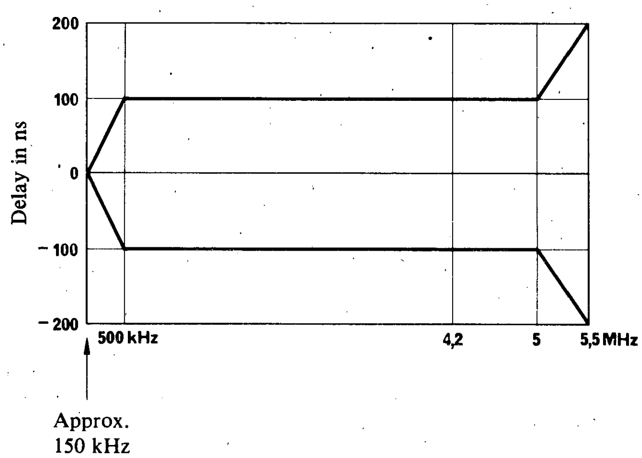


FIGURE 31 – *Limits for delay/frequency response characteristic*

Note – For 525-line transmission the limits need be met up to a frequency of only 4.2 MHz.

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SIOCOS, C. A. and CHOUINARD, G. [June, 1979] Subjective impairment units in relation with oscilloscope graticules for evaluating short-time linear waveform distortion of the luminance signal in 525-line television. *IEEE Trans. Broadcasting*, Vol. BC-25, **2**, 63-71.

CCIR Documents

[1970-74]: **a.** CMTT/207 (Italy); **b.** CMTT/65 (Italy).

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PART E – PERFORMANCE OF CIRCUITS SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT

E.1 Introduction

The purpose of Part E is to give some indication of the performance of circuits that have fewer or more video-to-video sections than the three sections of the hypothetical reference circuit as defined in § A.1.2 of this Recommendation. The effect of the length and the configuration of the circuit relative to the hypothetical reference circuit is also considered. The laws of addition in such circuits may be accurately established only if the statistical behaviour and the composition of the instantaneous values of the parameters is known [Lari *et al.*, 1974].

The values calculated from Tables III and IV provide only indications of the probable performance but for differential phase and gain, and luminance-chrominance gain inequality, such values may be considered accurate enough for practical applications. These values should be used with caution when studying the design of equipment because the laws of addition are not precisely known for every type of impairment.

E.2 Laws of addition

E.2.1 *Comments on the use of the laws of addition*

The definition of a circuit in terms of a single multiple of the hypothetical reference circuit is impossible if the number of video-to-video sections and the length of the circuit differ from those of the hypothetical reference circuit by different ratios, i.e. if $n/3 \neq L/l$ where

n: number of video-to-video sections,

L: length of the circuit,

l: 2500 km.

In such cases two definitions of the circuit in terms of the hypothetical reference circuit should be used, one for those parameters primarily proportional to the circuit configuration, and a second for those parameters (e.g. continuous random noise) primarily proportional to the length of the circuit.

TABLE III

n	$\left(\frac{n}{3}\right)^{1/h}$		
	$h = 1$	$h = 3/2$	$h = 2$
1	0.33	0.48	0.58
2	0.67	0.76	0.82
3	1.00	1.00	1.00
4	1.33	1.21	1.15
5	1.67	1.41	1.29
6	2.00	1.59	1.41
7	2.33	1.76	1.53
8	2.67	1.92	1.63
9	3.00	2.08	1.73
10	3.33	2.23	1.83
11	3.67	2.38	1.91
12	4.00	2.52	2.00
13	4.33	2.66	2.08
14	4.67	2.79	2.16
15	5.00	2.92	2.24

TABLE IV

§ of Part B	Characteristic	D_3 expresed in	h (1)	Notes
3.1	*Insertion gain (error)*	dB	2	
	Insertion gain variations	dB	2	
3.2.1	*Continuous random noise*			1, 8
3.2.2	*Low-frequency noise*	dB	No law	–
3.2.3	*Periodic noise*			
	Power-supply hum	{ Noise	2	2, 7
	Single frequency	voltage	2	3
3.2.4	*Impulsive noise*	Noise voltage		4
3.3	*Crosstalk*	Crosstalk voltage	3/2	
3.4.1	*Non-linear distortion of the picture signal*			
	Luminance	%	3/2	–
	Chrominance gain	%	3/2	–
	Chrominance phase	Degrees	3/2	–
	Chrominance-to-luminance intermodulation	%	2	–
	Differential gain	%	3/2 or 2	9
	Differential phase	Degrees	3/2 or 2	9
3.4.2	*Non-linear distortion of the synchronizing signal*			
	Steady state distortion	%	3/2	
3.5.1	*Linear waveform distortion*			
	Long-time waveform distortion			11
	Field-time waveform distortion	%	1	
	Line-time waveform distortion	%	2	
	Short-time waveform distortion	$K_{(P/B)}$, %	2	10
		$K_{(2T)}$, %	3/2	10
		or S, %	3/2	
3.5.3	*Chrominance-luminance inequalities*			
	Gain inequality	%	2	5
	Delay inequality	ns	2	5
3.5.4	*Steady state characteristics*			
	Gain/frequency	dB	3/2	6
	Delay frequency	µs	3/2	6

(1) Further information on laws of addition is to be found in [CCIR, 1966-69a], [CCIR, 1970-74b and c] and Report 636.

Note 1 – For circuits on coaxial cables, quadratic addition ($h = 2$) applies to random noise expressed in terms of r.m.s. voltage. For circuits on radio-relay links, see Recommendation 555.

Note 2 – Considering the probability of arithmetic addition of power-supply hum in circuits of few sections, it may be advisable to put $h = 1$ when $n \leqslant 3$.

Note 3 – Considering the probability of arithmetic addition when periodic noise consists of a few components that are very close in frequency, it may be advisable to put $h = 1$ when the number of such components is small.

Note 4 – When each of a number of sources of impulsive noise is operative for a small percentage of the time (e.g. $< 0.1\%$), arithmetic addition of the percentages will apply.

Note 5 – Quadratic addition ($h = 2$) for gain and delay inequalities is based on the assumption that positive and negative values are made equally likely by the use of correcting networks or equivalent means.

Note 6 – In Canada and in the United States of America, the practice is to use $h = 2$.

Note 7 – Further information is given in [CCIR, 1966-69b].

Note 8 – Further information is given in [CCIR, 1970-74d].

Note 9 – Use $h = 2$ if the link is equalized with respect to the mean values of differential gain and phase, otherwise $h = 3/2$.

Note 10 – See Part C, Annex IV.

Note 11 – No values yet assigned.

E.2.2 *Law pertaining to circuit configuration*

For the first definition of the circuit in terms of the hypothetical reference circuit use the following equation for all parameters in Table IV except for "continuous random noise".

If D_3 : design objective as expressed in this Recommendation, or the parameter derived therefrom and indicated in Table IV, that is permitted in the hypothetical reference circuit, and

D_n : performance, or the parameter mentioned above, permitted in n sections, then

$$D_n = D_3 \left(\frac{n}{3} \right)^{1/h}$$

where h has the value 1, 3/2 or 2 in accordance with Table IV: $h = 1$ gives linear or arithmetic law of addition, $h = 3/2$ gives the "three-halves power" law of addition and $h = 2$ gives the quadratic (r.s.s.) law of addition.

Calculated values of $(n/3)^{1/h}$ are given in Table III.

E.2.3 *Law pertaining to circuit length*

For the second definition of the circuit in terms of the hypothetical reference circuit use the following equations for "continuous random noise voltage" only. When distance is considered the law of addition becomes:

$$D_n = D_3 \left(\frac{L}{l} \right)^{1/h}$$

where D_n, D_3, L and l are as defined in § E.2.1 and E.2.2. If 20 km $\leqslant L \leqslant$ 280 km, set $L =$ 280 km in the expression for D_n.

Note 1 – Further information on this additional law is to be found in [CCIR, 1970-74a; 1982-86].

Note 2 – A national network may include many circuits less than 20 km in length. The signal/noise ratio required for such circuits depends on the number which may be used in a chain. The value chosen is considered to be a national matter, but, desirably it should be compatible with the requirements for the international hypothetical reference circuit.

E.2.4 *Tables and formulas for the addition of noise voltages and distortions*

For practical applications the following information is useful for the addition of distortions with differing magnitudes.

– Addition of two noise voltages:

If S is the difference between the higher signal-to-noise ratio r_2 and the lower signal-to-noise ratio r_1 (in dB), the resulting signal-to-noise ratio r_{res} after addition will be:

$$r_{res} = r_1 - X(S)$$

where $X(S)$ (dB) is obtained from Table V for the difference value S.

– Addition of two distortions:

If T is the numerical ratio between the distortion D_2 with the greater magnitude and the distortion D_1 with the smaller magnitude, i.e. $T = D_2 / D_1$, the resulting distortion D_{res} will be:

$$D_{res} = D_2 \cdot Y(T, h)$$

where $Y(T, h)$ is obtained from Table VI for the value $T = D_2 / D_1$.

D_{res} can also be calculated from the following:

$$D_{res} = [D_1{}^h + D_2{}^h]^{1/h} = D_2 [1 + T^{-h}]^{1/h}$$

TABLE V

S (dB)	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
X (dB)	3.0	2.5	2.1	1.8	1.5	1.2	1.0	0.8	0.6	0.5	0.4	0.3	0.3	0.2	0.2	0.1	0.1	0.1	0.1	0.05	0.0

TABLE VI

T		1	1.5	2	2.5	3	3.5	4	5	6	7	8	9	10
$Y(T, h)$	$h = 1$	2.00	1.67	1.50	1.40	1.33	1.29	1.25	1.20	1.17	1.14	1.13	1.11	1.10
	$h = 3/2$	1.59	1.34	1.22	1.16	1.13	1.10	1.08	1.06	1.04	1.04	1.03	1.03	1.02
	$h = 2$	1.41	1.20	1.12	1.08	1.05	1.04	1.03	1.02	1.01	1.01	1.01	1.01	1.01

REFERENCES

LARI, M., MORGANTI, G. and SANTORO, G. [February, 1974] The statistical addition of distortion in transmission systems. *EBU Rev. Tech.*, 143.

CCIR Documents

[1966-69]: **a.** CMTT/170 (OIRT); **b.** CMTT/49 (OIRT).

[1970-74]: **a.** CMTT/42 (Germany (Federal Republic of)); **b.** CMTT/149 (Canada); **c.** CMTT/57 (Italy); **d.** CMTT/56 (Italy).

[1982-86]: CMTT/17 (Germany (Federal Republic of)).

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RECOMMENDATION 722

UNIFORM TECHNICAL STANDARDS AND UNIFORM OPERATIONAL PROCEDURES FOR SATELLITE NEWS GATHERING (SNG)*

(Question 13/CMTT and Study Programme 13H/CMTT)

(1990)

The CCIR,

CONSIDERING

(a) that Satellite News Gathering (SNG) using portable transmitting earth stations is essential for broadcast operations and provides a valuable method of transmission for the rapid acquisition and broadcasting of news events;

(b) that to facilitate the international coverage of news and to optimize the design of equipment, it would be desirable to adopt uniform technical standards and uniform operational procedures for SNG, taking into account the possibility for interference to other satellites and systems;

(c) that SNG requirements include various communication and transmission support systems and that it is necessary to provide, preferably on the same satellite transponder, auxiliary signals for the operation of SNG earth stations;

(d) that SNG is temporary and occasional and its activation often cannot be determined long in advance;

(e) that SNG earth stations operate mainly in the fixed-satellite service, and should comply with the relevant provisions of the Radio Regulations, and any relevant domestic regulation requirements;

(f) that the ITU Constitution states in its Preamble: " ... fully recognizing the sovereign right of each State to regulate its telecommunication ... ";

(g) that for the successful operation of SNG, it is essential that there be expeditious authorization for the activation of SNG earth stations, for transmissions to a telecommunication satellite, in conformance with the administrative procedures of the host country and operational and technical criteria established for those systems;

(h) that the means to simplify the procedure required to obtain, as expeditiously as possible, temporary authorization to operate SNG facilities are under investigation by the CCIR in Report 1237;

(j) that SNG would be facilitated through the availability of an SNG user's guide from satellite operators (space segment providers) and host countries,

UNANIMOUSLY RECOMMENDS

1. that SNG earth station transmissions should comply with the uniform technical standards as described in Annex I. In particular, SNG earth stations should provide two-way communication circuits, which must be available prior to the SNG transmission in addition to the vision and associated sound or sound programme channel;

2. that the operation of SNG should comply with the uniform operating procedures as described in Annex II;

3. that to facilitate the temporary authorization of SNG operations, administrations and relevant organizations are encouraged to consider harmonization of expeditious and simplified procedures (e.g. earth station approval, satellite reservation and frequency co-ordination, etc.) (see § 2 of Report 1237);

4. that each administration should establish a point of contact for the exchange of information and guidance on frequency coordination and administrative procedures of the host country;

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* This Recommendation should be brought to the attention of CCIR Study Groups 4, 10 and 11.

5. that in order to simplify operations and minimize delays, satellite space segment providers should develop user guides for SNG operational procedures of their individual systems and take steps to harmonize these procedures between their systems;

6. that host countries are encouraged to develop SNG user guides or other documents which may be in the form of national regulations to facilitate operations;

7. that satellite organizations should, on request, provide an easily identifiable carrier to facilitate the operation of SNG earth stations;

8. that SNG transmissions include an appropriate identification signal, notified to the host country to assist in interference abatement.

ANNEX I

MINIMUM TECHNICAL PARAMETERS APPLICABLE TO SNG EARTH STATIONS

1. General performance

An SNG terminal must be able to be rapidly deployed, to transmit (with a minimum of impairments) video and associated audio signals or sound broadcasting signals, to provide limited receiving capability to assist in the pointing of the antenna and to monitor (where possible) the transmitted signals, and to provide two-way communications for operation and supervision.

2. Transmission performance requirements

The baseband signal shall be transmitted with a minimum of impairment.

2.1 *SNG for television broadcasting*

Video signal: refer to Recommendation 567. (Requirements for random noise may be relaxed by the user.)

Audio signal: refer to Recommendation 505. (Requirements for random noise may be relaxed by the user.)

2.2 *SNG for sound broadcasting*

Baseband: as per Recommendation 504-2 (Geneva, 1982). (Requirements for random noise may be relaxed by the user.)

3. RF performance requirements

3.1 *Off-axis e.i.r.p. density*

Shall comply with Recommendation 524 or the satellite operator's requirements, whichever is more stringent.

3.2 *Polarization discrimination*

Should be better than 35 dB within the −1 dB points of the main axis of the beam and 30 dB elsewhere. However, SNG earth stations generally operate with low e.i.r.p., so a corresponding reduction in polarization discrimination performance may be accepted by the satellite system operator.

3.3 *E.i.r.p.**

SNG for television broadcasting	SNG for sound broadcasting
70 dBW (nominal)	55 dBW (nominal)

3.4 *RF bandwidth**

SNG for television broadcasting	SNG for sound broadcasting
17.5-36 MHz	200 kHz

* Actual value shall be determined considering characteristics of the satellite which is used.

4. Modulation characteristics

4.1 *SNG for television transmission*

Modulation: FM.

Deviation: 10-28 MHz p-p for 1 V p-p baseband signal input.

Sense of modulation: positive (positive-going voltage for positive-going frequency).

Energy dispersal: 0-4 MHz p-p.

Associated audio: Subcarrier or sound-in-sync (SIS) techniques may be used.

4.2 *SNG for sound transmission*

Modulation: FM.

Deviation: 150 kHz p-p for modulating signal of 1 kHz at +9 dBm0s.

Pre-emphasis: 75 μs.

Energy dispersal: fixed/adaptive, depending upon requirement to meet Recommendation 524.

5. Auxiliary signal

Modulation: analogue or digital.

6. Identification signal

Transmitted by suitable means (method to be studied and recommended by CCIR Study Groups 4, 10 and 11 and CMTT).

ANNEX II

UNIFORM OPERATING PROCEDURES FOR SNG*

The harmonization of Satellite System Operators' Procedures for SNG should include as a minimum the following items:

- obtaining the relevant operational parameters;
- SNG earth station line-up, including pre-transmission testing to insure no interference is caused;
- establishment of auxiliary circuits;
- activation of programme transmission;
- SNG earth station close down;
- setting up the next SNG user.

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* It is requested that the Director, CCIR, forward Annex II to all organizations concerned with operating telecommunication satellites requesting them to address the unique requirements for SNG by harmonizing their operating procedures.

RECOMMENDATION 568

SINGLE VALUE OF THE SIGNAL-TO-NOISE RATIO FOR ALL TELEVISION SYSTEMS

(Question 13/CMTT and Study Programme 13A/CMTT)

(1978)

The CCIR,

CONSIDERING

(a) that results of studies made by various administrations have shown the need for weighting the measured noise so that, in principle, a single value of signal-to-weighted noise ratio would result in the same subjective opinion value irrespective of the spectral distribution of the noise;

(b) that such studies have also shown that the various television standards have different sensitivities to random noise, and consequently various noise weighting characteristics have been produced by many administrations;

(c) that the acceptance of a single weighting characteristic for application to the hypothetical reference circuit for all television systems is necessarily based on a substantial compromise among the various noise weightings used by various administrations;

(d) that it is furthermore necessary to measure random noise in a unified bandwidth in order to specify a single value of signal-to-weighted noise ratio as an objective for all television systems, so as to avoid any possible confusion when comparing results obtained at different points along an international television circuit;

(e) that a general agreement has been obtained on a unified noise weighting network and on a single value of the signal-to-weighted noise ratio objective for application to the hypothetical reference circuit for all television systems,

UNANIMOUSLY RECOMMENDS

1. that the weighting characteristic and the weighting network shown in Annex II to part C of Recommendation 567 should be adopted for all international television circuits;

2. that the signal-to-weighted noise ratio should always refer to noise as measured in the band 0.01 to 5 MHz;

3. that a single value of signal-to-weighted noise ratio of 53 dB should be the objective for the 2500 km hypothetical reference circuit for 99% of any month.

Note 1 – Measurements for colour television systems using the unified weighting network can be considered as giving a valid indication of the subjective impairment due to the noise only in cases where the noise power per unit bandwidth at 5 MHz does not exceed that at 1 MHz by more than about 11 dB. This condition will be met in the majority of cases with existing transmission systems and only the recommended weighting network will be used for operational purposes. For new systems that would not meet this condition, designers should verify by other means that signal-to-weighted noise ratio is satisfactory and that the recommended network gives suitable results (see Annex I).

Note 2 – There may be administrations which need for national purposes values of signal-to-noise ratio which differ from 53 dB.

ANNEX I

TREATMENT OF SPECIAL CASES

In cases of new transmission systems where the noise spectral distribution might not meet the requirement given in Note 1 (of the Recommendation) it should be verified during the design stage that the proposed noise distribution will give satisfactory results for all colour television systems. Some guidance on how to do this is given below.

– For system I, the signal-to-weighted noise ratio may be determined by the Hypothetical Composite Weighting Network [Allnatt and Prosser, 1966] and in this case it should not be worse than 49 dB;
– In Italy, for PAL colour systems B and G, the measurement may be carried out with the weighting characteristic indicated in [CCIR, 1966-69];
– For systems D and K the measurement may be carried out with the weighting characteristic indicated in [CCIR, 1970-74].

REFERENCES

ALLNATT, J. W. and PROSSER, R. D. [1966] Subjective quality of colour television pictures impaired by random noise. *Proc. IEE*, **113**, 551-557, Appendix 7.3.

CCIR Documents:

[1966-69]: CMTT/160 (Italy).

[1970-74]: CMTT/72 (USSR).

RECOMMENDATION 603

HYPOTHETICAL REFERENCE CHAIN FOR TELEVISION TRANSMISSIONS OVER VERY LONG DISTANCES

(Question 13/CMTT and Study Programme 13D/CMTT)

(1982)

The CCIR,

CONSIDERING

(a) that, in some parts of the world, television transmission chains much longer than 2500 km are either existing or are under consideration;

(b) that it is desirable to define one or more hypothetical reference chains corresponding to various typical connections;

(c) that the concept of hypothetical reference chains of known composition would be very useful in progressing the studies under Study Programmes 13C/CMTT and 14A/11;

(d) that Recommendation 567 recommends that the design objectives and tolerances for a terrestrial hypothetical reference circuit should be the same as those of a hypothetical reference circuit in the fixed-satellite service;

(e) that, for television, inter-continental and very long distance transmission is normally by satellite,

UNANIMOUSLY RECOMMENDS

1. that it is only necessary to define one hypothetical reference chain which represents very long distance connections enabling any two points on the Earth's surface to be interconnected;

2. that the hypothetical reference chain should be defined as equivalent to five hypothetical reference circuits in cascade;

3. that the performance of the hypothetical reference chain be determined from the performance of the hypothetical reference circuit as defined in Recommendation 567 by the use of the methods recommended in Part E of that Recommendation.

Note 1 – For the purpose of defining the hypothetical reference chain and using it to investigate the adequacy of the performance of very long distance connections, it is not essential to define what the five parts represent. However, by way of illustration they could be taken to represent two national networks, two satellite links and 2500 km of terrestrial network interconnecting the intermediate pair of earth stations and/or the terminal earth stations and the national networks; also, standards converters are likely to be present in very long distance connections. From this it can be seen that a chain comprising five hypothetical reference circuits is fully adequate to represent a very long television connection in all but the most extreme cases.

Note 2 – In real circuits of similar complexity to the hypothetical reference chain, clamps may be required to control the accumulation of power-supply hum, field-time waveform distortion and long-time waveform distortion.

Note 3 – In real circuits of similar complexity to the hypothetical reference chain, automatic correction of differential phase and/or linear waveform distortion may be desirable.

Note 4 – The view has been expressed that when a television signal is transmitted over a very long distance connection, as represented by the hypothetical reference chain, the picture quality may not be adequate. It would therefore be desirable to study further the performance of very long distance connections, as expressed in subjective terms.

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RECOMMENDATION 604-2 *

DIGITAL TELEVISION TRANSMISSION OVER LONG DISTANCES – GENERAL PRINCIPLES

(Question 14/CMTT and Study Programmes 14A/CMTT, 14B/CMTT and 14C/CMTT)

(1982-1986-1990)

The CCIR,

CONSIDERING

(a) that a Recommendation 601 on digital coding in television studios based on a family of component codes has been adopted for 525-line and 625-line standards;

(b) that it is desirable to transmit digital component signals through digital circuits;

(c) that there will be different applications for long distance digital television transmission, for example:

– *contribution:* to carry signals to production centres where post-production processing may take place;

– *distribution:* to carry television programmes when no further post-production processing is expected (see Note 1);

(d) that it is highly desirable to facilitate the international exchange of television programmes by making digital television transmission circuits throughout the world transparent to television signals to the same or a higher degree than is possible with analogue circuits;

(e) that when establishing a digital transmission circuit on a global basis, account should be taken of the different digital transmission hierarchies and the H-channel rates and interfaces for ISDN recommended by the CCITT;

(f) that bit-rate reduction techniques can be effective for reducing transmission costs,

UNANIMOUSLY RECOMMENDS

1. that, when carrying a television signal originated in digital form, entirely digital transmission circuits should be preferred. For these circuits the principles given in this Recommendation should be followed;

2. that the signals from digital television studios as described in Recommendation 601 should be preserved in their component coded form and should be the basis for signals transmitted over digital links;

Note 1 – In the present Recommendation and in the texts of CMTT concerning Question 14/CMTT, the term "distribution" refers to primary distribution (for example the transmission to the input of broadcast transmitters) and not to secondary distribution (delivery to the television consumer). It should be noted that the quality objectives in the two cases may be different.

Note 2 – Depending on various applications, for which examples are given in CONSIDERING *(c)*, the coding techniques (which are under study according to Study Programme 14A/CMTT) have to take into account the corresponding post-production processing requirements and quality requirements defined by Study Group 11.

3. that for each application, for which examples are given in CONSIDERING *(c)*, a single transmission coding technique (see Note 2) should be used for each of the 525-line and 625-line television signals;

4. that for each application, for which examples are given in CONSIDERING *(c)*, a single transmission multiplexing structure for video, sound and ancillary signals should be used for each of the 525-line and 625-line standards, and, if possible, for both standards;

5. that the bit-rate of each multiplex structure should be compatible with an appropriate level of the digital transmission hierarchy or of the H-channel rates for ISDN recommended by the CCITT.

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* This Recommendation should be brought to the attention of CCIR Study Groups 4, 9, 10 and 11 and CCITT Study Group XVIII.

RECOMMENDATION 658-1 *

MIXED ANALOGUE-AND-DIGITAL TRANSMISSION OF ANALOGUE COMPOSITE TELEVISION SIGNALS OVER LONG DISTANCES

(Question 14/CMTT and Study Programmes 14A/CMTT, 14B/CMTT, 14C/CMTT and 14D/CMTT)

(1986-1990)

The CCIR,

CONSIDERING

(a) that, although television studios may progressively adopt operation based on separate component coding (e.g. according to Recommendation 601 for digital systems), existing analogue operation using composite signals will continue for a considerable time (see Note 1);

(b) that a transitional period will occur during which analogue composite television signals will be transmitted through a circuit comprising analogue and digital sections in tandem;

(c) that to facilitate the international transmission of television programmes during the transitional period, the methods of coding should preserve the quality of the colour television signals (e.g. NTSC, SECAM, PAL) and of the ancillary and additional signals (e.g. insertion test signals, teletext, etc.);

(d) that, when establishing digital circuit sections, account should be taken of the different digital transmission hierarchies and the H-channel rates and interfaces for ISDN recommended by the CCITT;

(e) that while the use of bit-rate reduction techniques could be economically desirable, further studies are required before coding methods to comply with (*c*) above can be recommended,

UNANIMOUSLY RECOMMENDS

1. that when carrying a television signal presented in analogue composite form, preference should be given to all-analogue paths. However, in cases where mixed analogue-and-digital paths are unavoidable, the principles given in this Recommendation should be followed;

2. that the number of digital sections in a real circuit should be kept to the minimum;

3. that the hypothetical reference circuit for mixed analogue-and-digital transmission shall be equivalent to that defined in § A.1.2 of Recommendation 567, which applies to the case where the three sections all use analogue transmission. When one or more sections use digital transmission, the same structure should apply but modifications may be required for sections employing digital transmission (see Note 2). The signals at the input and output, and at the intermediate interconnection points of the hypothetical reference circuit, are in analogue form (see Note 3);

4. that the design objectives and tolerances specified for a hypothetical reference circuit in Recommendation 567 should also apply for mixed analogue-and-digital transmission (see Note 4);

5. that the overall quality of the hypothetical reference circuit, when considered against the objective and subjective criteria defined by Study Group 11, shall be no worse than the equivalent analogue system. In practice this may be achieved by using the parameter values defined in Annex I (see Note 5).

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* This Recommendation should be brought to the attention of CCIR Study Group 11 and CCITT Study Groups XV and XVIII.

Note 1 – This Recommendation does not apply either to component analogue signals or to encrypted signals. These areas require further study which should lead to new Recommendations.

Note 2 – The detailed requirements for the digital sections of the hypothetical reference circuit require further study on such issues as their lengths, the number of digital sections permitted, etc. Section 6 of Report 646 gives details of the progress achieved in defining the digital sections and of the areas where further studies are required.

Note 3 – In real circuits, when two digital sections are interconnected, it is not necessary, apart from adjustment periods, to introduce an analogue interface.

Note 4 – Some additions to the methods of test specified in Recommendation 567 may need to be added to this new Recommendation for application where digital sections are included (see Report 819). Moreover, additional tests and objectives may be required to deal with new types of impairment caused by digital coding. These matters are under study (see Report 646).

Note 5 – The parameter values in Annex I have been shown to meet this criterion. Other parameter sets may meet this criterion, but their conformance would need to be demonstrated. Administrations are reminded that they have the right to make bilateral agreements on coding parameters for a single colour television system, should this be required. If, however, such a circuit is to form part of an international television connection, it should meet the requirements specified in this Recommendation.

ANNEX I

A SPECIFICATION FOR DIGITAL SECTIONS OF MIXED LINKS

This specification applies to one of the three equal sections of the hypothetical reference circuit specified in Recommendation 567. Since the distortion introduced by a digital section (apart from transmission errors) is due entirely to the analogue-to-digital converter (ADC) and digital-to-analogue converter (DAC), it follows that any number of digitally-connected links of any length may be considered as making up one such digital section.

(a) Composite television signals must not be decoded into components.

(b) The lowpass filters in ADCs and DACs must be such that six filters in tandem would satisfy the requirements for short-time waveform distortion given in § D.3.5.1.4 of Recommendation 567. In practice the luminance filter specified in Annex III to Recommendation 601 would meet this requirement.

(c) Sampling frequency must be 13.0 MHz or higher.

(d) Uniform quantization coding must be used.

(e) ADC and DAC must be monotonic.

(f) The conversion range of the ADC must be 1.75 V ± 10 mV (see Note 1, below).

(g) The signal-to-quantizing-noise ratio of the ADC/DAC combination must be better than 58 dB. The signal-to-quantizing-noise ratio should be measured in the presence of a line sawtooth signal or a 0.7 V p-p low frequency sinewave both with and without a superimposed 0.35 V p-p sinewave at a high frequency (> 4 MHz); with the superimposed high frequency sinewave an extra implementation margin of up to 6 dB may be allowed. (Unified weighted r.m.s. noise in 5.0 MHz bandwidth relative to 0.7 V, see Note 2 below.)

(h) Signals containing synchronising pulses and blanking intervals must be clamped at the input to the ADC so that black level is at one-quarter of the conversion range (level 128 in a 9-bit system). The coder must clamp both 625/50 signals and 525/60 signals correctly.

(i) The clamp time constant must be at least 2 ms (see Note 3 below).

(j) Any waveform not containing synchronising pulses and blanking intervals must not be clamped but must be coupled to the ADC input in such a way that its mean level is close to the centre of the conversion range.

(k) All parts of the signal must be transmitted without modification.

(l) No bit rate reduction coding may be used.

(m) If no error correction is used, the long-term mean bit error ratio (BER) must be better than 10^{-8}. Bursts of errors with BER worse than 10^{-6} should not be longer than 5 s and there should not be more than one such burst in any hour (see Note 4 below).

(n) If error correction is used, residual BER after correction in the two most significant bits must be no worse than that specified in *(m)* (see Note 4 below).

(o) Jitter on the regenerated samples at the DAC must be less than 0.3 ns r.m.s. (see Note 5 below).

Note 1 – This conversion range allows the transmission of 100% colour bars at 3 dB overload without crushing. However, modern equipment might allow the specification of a lower overload margin; further work is required. The use of a lower conversion range will improve the signal-to-quantizing noise ratio.

Note 2 – This requirement ensures that the quantizing noise contributed by a digital section is no more than one third of the total allowance for the three-section HRC specified in Recommendation 567. It is shown in [CCIR, 1986-90a] that this requirement may be met by the use of nine bits per sample at a sampling frequency of 13 MHz. The frequency of the high-frequency sinewave may depend on the television standard and measuring equipment in use. Further work is needed to standardize this parameter.

If the frequency used lies within the passband of the 5 MHz noise measuring filter, it will be necessary to remove the high-frequency sinewave with a supplementary filter and to correct the result for the noise bandwidth of the supplementary filter as described in [Devereux, 1982].

Note 3 – A shorter time constant gives clamp streaking from noise. A time constant of 2 ms gives 6 dB suppression of mains hum; if mains hum is not a problem, the time constant could be longer. Devereux [1982] gives more details.

Note 4 – Experiments reported in [Ratliff, 1974] showed that random errors were "imperceptible" at a BER better than 10^{-8} and "definitely perceptible but not disturbing at a BER of 10^{-6}" and that, for linear PCM, protecting the two MSB's is as effective as protecting all bits of the video sample word.

Note 5 – Devereux [1971]; Devereux and Wilkinson [1973] show that 0.3 ns r.m.s. is the threshold of perception for timing jitter on PAL signals.

It is recognized that due to jitter produced by practical multiplexers and demultiplexers – for example, multiplexing from 140 Mbit/s to 565 Mbit/s and back to 140 Mbit/s – (waiting time jitter) this specification may be difficult to meet in practice. Further work is needed.

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DEVEREUX, V. G. [1971] Pulse code modulation for video signals: subjective tests on acceptable limits for timing jitter in the decoded analogue samples. BBC Research Department Report 1971/42.

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DEVEREUX, V. G. [1982] Tests on eight video p.c.m. codecs in tandem handling composite PAL and monochrome video signals. BBC Research Department Report 1982/19. Information in this Reference is also contained in EBU Rev. *(Tech.)* 199, June 1983, 114-131.

RATLIFF, P. A. [1974] Digital video: Subjective assessment of an experimental Wyner – Ash error corrector. BBC Research Department Report 1974/41.

CCIR Documents

[1986-90]: **a.** CMTT/176 (United Kingdom).

RECOMMENDATION 723*

TRANSMISSION OF COMPONENT-CODED DIGITAL TELEVISION SIGNALS FOR CONTRIBUTION-QUALITY APPLICATIONS AT THE THIRD HIERARCHICAL LEVEL OF CCITT RECOMMENDATION G.702

(Question 14/CMTT and Study Programmes 14A/CMTT, 14D/CMTT)

(1990)

The CCIR,

CONSIDERING

(a) that for contribution-quality application, transmission should be based on component-coded digital video signals conforming with CCIR Recommendation 601;

(b) that such transmission would satisfy user requirements for contribution-quality codecs at 30-45 Mbit/s, as specified by Study Group 11 in Report 1211;

(c) that according to these user requirements such transmission should preserve the picture quality inherent to the 4:2:2 encoding process based on Recommendation 601 to the maximum extent possible, taking into account the bit rate available to the user;

(d) that such transmission should similarly preserve the downstream processing capabilities by maintaining the spatial and temporal resolution of the 4:2:2 signals as given by Recommendation 601;

(e) that additional transmission capacity should be provided for stereo sound-channels, ancillary signals (e.g. teletext, test signals) and accompanying error-protection data;

(f) that through the use of appropriate bit rate reduction techniques it is likely that these objectives can be reached at an acceptable level of complexity and cost for transmission at bit rates of the order of 30-45 Mbit/s,

RECOMMENDS

that for the transmission of component-coded digital video signals according to Recommendation 601 at the CCITT Recommendation G.702 third hierarchical level bit rates, the bit rate reduction codec should be characterized as shown in Table I.

* Equipment built to the specifications given in this Recommendation has not yet been shown to meet the users requirements defined in Report 1211. The Administrations of Australia, Canada and the United States of America therefore withhold their approval of this Recommendation until confirmation that the required performance can be achieved. The Administrations of Italy and Spain withhold their approval of this Recommendation until the evaluations suggested in Annex IV of Report 1235 are undertaken.

TABLE I – *Specification of CMTT codecs for 32-45 Mbit/s*

Video input/output	Standard	525-line or 625-ligne digital video in component form. Manual or automatic selection of the video standard is at the manufacturer's discretion.
	Coding	4:2:2 level of Recommendation 601.
	Interface	Bit-parallel or bit-serial in accordance with Recommendation 656.
Signal preprocessing	Horizontal	Full digital active line of 720 samples for luminance (Y) and 360 samples for each colour difference (C_R, C_B).
	Vertical	525 lines: 248 lines per field (Note 1) Field 1: lines 16 to 263 Field 2: lines 278 to 525 625 lines: 288 lines per field Field 1: lines 23 to 310 Field 2: lines 336 to 623
	Range of preprocessing	In Recommendation 601, the range of values of Y, C_R and C_B is from 0 to 255. For processing purposes an offset of -128 is added and the values are expressed as 2's complement integers of 8 bits (including sign).
Coding	Modes	Three modes (intra-field, inter-field and motion compensated inter-frame) are used. The three following processings are applied either on 8 × 8 intra-field blocks (intra-field mode) or on differential blocks obtained by difference between the current 8 × 8 intra-field block and a reference block taken in the previous field (inter-field mode) or in the field with same parity in the previous frame (inter-frame mode) (see Annex I).
	DCT	Discrete cosine transform applied on rectangular blocks of 8 lines of 8 samples for the three components Y, C_R and C_B (see Annex II).
	Prediction of the block	For each block processed according to interfield mode, the reference block is determined with pixels of the previous field without motion compensation. For each block processed according to interframe mode, the reference block is taken in the previous frame. Its position is derived from the position of the current block by application of a displacement vector (see Annex III).
	Motion compensation	Motion compensation is applied to "macro-blocks". Each macro-block (two adjacent 8 × 8 blocks for Y and the two co-positioned C_R and C_B blocks) is assigned a single displacement vector with half-pixel accuracy (see Annex IV).
	Quantization	A different quantization characteristic is used for each coefficient. Its parameters are adapted to the buffer occupancy, the criticality of the block and the type of the block (luminance/chrominance). The shape of the characteristic is nearly uniform (see Annex V).
	Variable length coding	VLCs are used to encode the quantized DCT coefficients and motion information (see Annex VI) (Note 2).
Buffer memory capacity		1 572 864 bits
Video framing		(see Annex VII) (Note 2)
Video data error protection		Reed Solomon (255,239) interleaving factor 2 (see Annex VIII).
Sound and data		The following capacity is available: – 2048 kbit/s for 34.368 Mbit/s channel; – 1544 kbit/s for 32.064 Mbit/s and 44.736 Mbit/s channels.
Channel framing		Frame structures for 32.064, 34.368 and 44.736 Mbit/s channels are under study (see Annex IX).

Note 1 – Only 244 lines per field are significant; lines 16, 17, 18, 19 and 278, 279, 280, 281 are encoded but not displayed.

Note 2 – Three systems have been proposed and are under study in TG CMTT/2.

ANNEX I

INTRA-FIELD, INTER-FIELD AND INTER-FRAME MODES

Two processing modes are used:

1. Intra-field mode (Fig. 1)

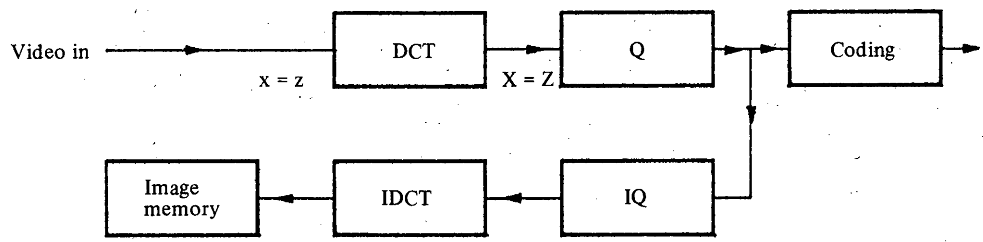


FIGURE 1 – *Intra-field mode*

2. Inter-field and inter-frame mode (Fig. 2)

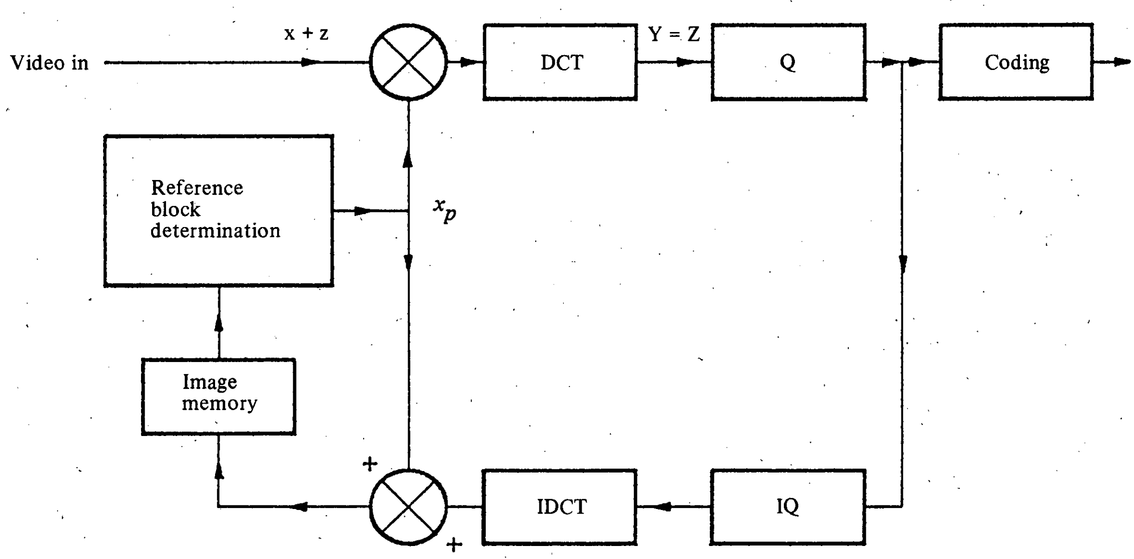


FIGURE 2 – *Inter-field and intra-frame mode*

3. Definition of the different modules

DCT: discrete cosine transformation (for 8 × 8 blocks).

IDCT: inverse discrete cosine transformation (for 8 × 8 blocks).

Q: quantification (see Annex V).

Coding: see Annex VI.

IQ: the IQ module builds a DCT-coefficients block from the corresponding transmitted information, by assigning to the coefficients the reconstruction values corresponding to the transmitted quantization levels (see Annex V).

Image memory: the image memory provides storage for:

- the decoded current field. This field is used as reference for coding the next image;
- the two last previously decoded fields, which are used to determine the current reference block.

+ *For the inter-field mode:* the reference block is computed with pixels of the previous field according to the interpolation process described in Annex III.

+ *For the inter-frame mode:* The reference block is taken in the field of the previous frame with the same parity as the current field. Its position is obtained from the position of the current block by a translation given by a motion vector. The specification of the motion vector is given in Annex IV, the exact computation of the reference block for the inter-frame mode is presented in Annex III.

4. Notations

$x(i, j)$: 8 × 8 pixels block

$x_p(i, j)$: 8 × 8 reference block

$z(i, j)$: $= x(i, j)$ for the intra-field mode
$= x(i, j) - x_p(i, j)$ for the inter-frame or inter-field mode

$X(k, l)$: is the 8 × 8 DCT coefficients block in intra-field mode

$Y(k, l)$: is the 8 × 8 DCT coefficients block in inter-frame or in inter-field mode

$Z(k, l)$: $= X(k, l)$ for intra-field mode
$Y(k, l)$ for inter-frame or inter-field mode

(i, j): coordinates in the image domain
i: line index (range: 0 to 7 from left to right)
j: column index (range: 0 to 7 from top to bottom)

(k, l): coordinates in the transform domain
k: line index (range: 0 to 7)
l: column index (range: 0 to 7)

5. Mode choice

The chosen mode (intra-field, inter-field or inter-frame) is coded and transmitted for each processed macro-block (see Annex VII), so no specification is required for the mode choice since it concerns only the coder side.

The inter-field and the inter-frame scheme presented in Fig. 2 allows the use of *a priori* choice (decision done before coding steps) or *a posteriori* choice (decision done after having coded the blocks according to both modes).

In the inter modes, the $z(i, j)$' elements must be in the range (−128, 127); the mode decision is forced, when necessary, in order to satisfy this constraint.

To avoid the temporal propagation of transmission errors effects, it is recommended to use an intra-field refreshing processing. This processing only concerns the coder and need not be specified.

ANNEX II

DISCRETE COSINE TRANSFORM

For each component (Y, C_R, C_B), the discrete cosine transformation is applied to blocks composed of eight lines of eight samples. The data to be processed are, for each block, the samples of the present field or the differences between the present field samples and those obtained from a reference block (see Annex III). The direct transformation is computed according to the following formula:

$$Z(k, l) = \frac{1}{4} C_k C_l \sum_{i=0}^{7} \sum_{j=0}^{7} z(i, j) \cos \frac{\pi(2i + 1)k}{16} \cos \frac{\pi(2j + 1)l}{16}$$

and the inverse transformation is given by:

$$z(i, j) = \frac{1}{4} \sum_{k=0}^{7} \sum_{l=0}^{7} C_k C_l Z(k, l) \cos \frac{\pi(2i + 1)k}{16} \cos \frac{\pi(2j + 1)l}{16}$$

with the conventions of Annex I

$$C_k = \frac{1}{\sqrt{2}} \quad \text{for } k = 0 \qquad\qquad C_l = \frac{1}{\sqrt{2}} \quad \text{for } l = 0$$
$$= 1 \quad \text{elsewhere} \qquad\qquad = 1 \quad \text{elsewhere}$$

$Z(0,0)$ is called the DC coefficient: the other coefficients are AC coefficients.

The input to the DCT is expressed as 2's complement integers of 8 bits (including sign). The output of the DCT is expressed as 12 bit 2's complement numbers, of which the integer part is 11 bits (including sign).

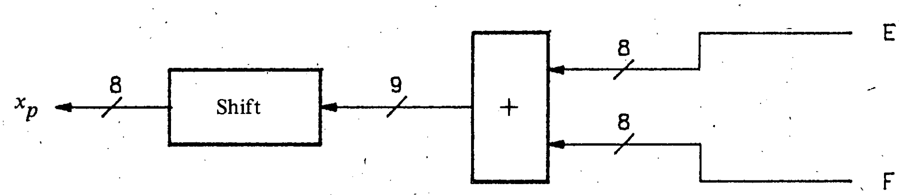
The accuracy of the performance of the inverse DCT computation is in accordance with that specified in CCITT draft Recommendation H.261.

ANNEX III

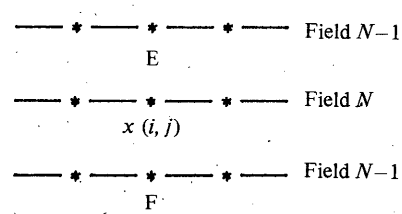
PREDICTION OF THE BLOCK

1. Inter-field mode

The reference block x_p for the current block x in field N is computed with pixels of field $N - 1$ with the following interpolation scheme:



E and F are defined below:



2. Inter-frame mode

The position of the reference block is obtained from the position of the currently processed block by a translation.

For motion compensation, the translation vector *(x, y)* is as described in Annex IV.

Interpolation scheme for non-integer vectors

There is no ambiguity in the definition of the reference block when the coordinates *(x, y)* are integers. If one of the coordinates has a non-zero fractional part, an interpolation scheme has to be used to build the reference block.

This scheme is described below for 1/2 pixel accuracy:

A+	.P	+B
Q.	.R	.S
C+	.T	+D

A, B, C, D: reconstituted pixels of the previous frame (in the field of same parity). Integer coordinates.

P, Q, R, S, T: interpolated pixels of the previous frame (in the field of same parity).

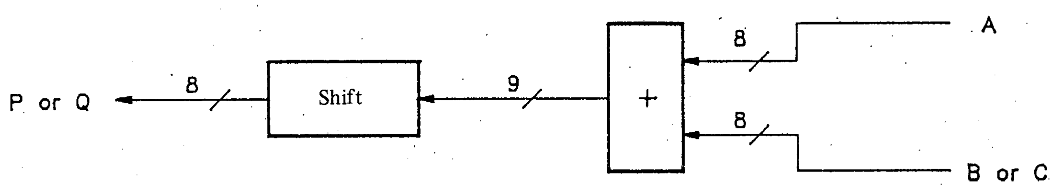
The values assigned to interpolated pixels are:

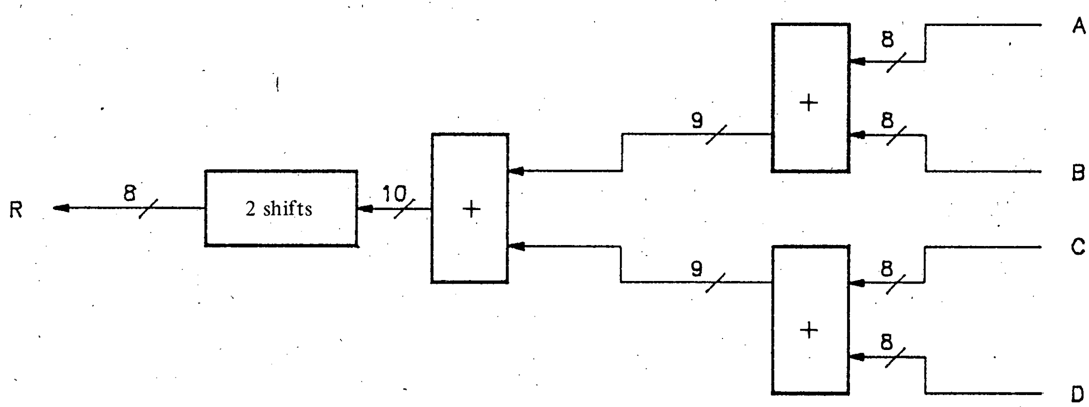
P = [(A + B)/2]

Q = [(A + C)/2]

R = [(A + B + C + D)/4]

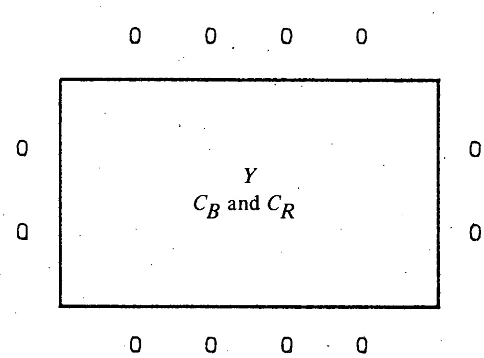
as illustrated below:





3. Video level outside active picture

In the definition of the reference block given in the two previous paragraphs, the pixels outside the active picture must be set to zero, expressed in 2's complement (8 bits):



ANNEX IV

MOTION COMPENSATION

Only one motion vector is used for the blocks belonging to a macro-block.

The parameters of the motion compensation are given below:

Search areas	± 15.5 pixels and ± 7.5 lines
Resolution	½ pixel and ½ line
Number of possible vectors	1953 (all vectors within the search area are permitted)

The method of estimation need not be specified since it only concerns the coder side.

The motion vector points to the pixel in the previous frame that is used in te interframe prediction.

If the vector components are defined as:

– x increasing from left to right, from -15.5 to $+15.5$;

– y increasing from top to bottom, de -7.5 to $+7.5$.

The x component of the vector is expressed as a six-bit two's complement number, the integer part of which is 5 bits (including sign). The y component is expressed as a five-bit two's complement number, the integer part of chich is 4 bits (including sign). It is coded by differential variable-length coding as described in Annex VI.

The motion vector to apply to the C_R, C_B blocks, is derived from the macro-block luminance motion vector in the following way:

– the vertical coordinate is identical to that of the luminance vector;

– the horizontal coordinate is equal to half that of the luminance vector.

Chrominance samples at quarter-pixel points are obtained by interpolation according to Annex III on a grid of samples coincident with luminance samples. The necessary intermediate chrominance samples are obtained using the same method but are based on the original chrominance grid. The resulting process is equivalent to bilinear interpolation to quarter-pixel accuracy.

ANNEX V

QUANTIZATION OF DCT COEFFICIENTS*

1. AC coefficient quantization

A different quantization characteristic is used for each coefficient. The quantization is achieved in two steps.

1.1 *Computation of relative coefficients*

$C(k, l) = 2Z(k, l) / (S(k, l, m, f))$

where:

$S(k, l, m, f)$: transmission threshold for (k, l) coefficients and where $S(k, l, m, f)$ is of the form:

$S(k, l, m, f) = 2^{n(k, l, m, f)/16}$ where $n(k, l, m, f)$ is an integer;

$m = 0, 1, 2, 3$ according to the block criticality;

f: transmission factor, related to buffer occupancy.

---

* Quantization does not strictly need to be specified since it concerns only the coder side. Sections 1 and 2 of this Annex give information on the calculation of parameters necessary for the operation of the inverse quantizer, which is specified in § 3.

1.2 *Quantization of relative coefficients*

The quantization law is close to linear and is described in Table II.

1.2.1 *Determination of transmission threshold matrix*

The S matrix for each component depends on the relative visibility matrix V defined in Figs. 3a and 3b for both components, and buffer factor f which is sent before each stripe of the DCT blocks and the criticality factor m which is sent for each macro-block.

The value of f is computed according to the buffer occupancy in order to provide a mean rate not greater than the bit rate available for video in the transmission multiplex.

The value of m is coded with two bits per macro-block.

The modules which realize the computation of f and the choice of the value for m are only in the coder and the corresponding information is sent to the decoder.

Referring to Figs. 4a, 4b and 4c, the scaler control parameter $n\,(k, l, m, f)$ for each component is obtained in the following way:

$$p\,(k, l, m) = \text{Min}\,[p_0\,(k, l) + Tr\,(m),\ Th\,(m)]$$

where $p_0\,(k, l)$ is defined by:

$$V(k, l) = 2^{p_0(k, l)/8}$$

and where p is an integer between 0 and 52.

$Tr\,(m)$ and $Th\,(m)$ are parameters depending on criticality (m) and are defined as follows:

Criticality (m)	Translation $Tr(m)$ [Y or C_R C_B coefficients]	Limit for Y $Th(m)$	Limit for C_R C_B $Th(m)$
0	+8	no (i.e., 44 + 8)	no (i.e., 26 + 8)
1	+2	no (i.e., 44 + 2)	no (i.e., 26 + 2)
2	0	34	16
3	0	24	9

Then $n\,(k, l, m, f)$ is given by:

$$n\,(k, l, m, f) = \text{Min}\,[n'(k, l, m, f),\ 175]$$

where:

$$\left.\begin{aligned} q\,(k, l, m, f) &= \text{Min}\,[2p\,(k, l, m) - 48, f] + f \\ n'(k, l, m, f) &= \text{Max}\,[q\,(k, l, m, f), 0] \end{aligned}\right\}$$

1.2.2 *Data accuracy*

Data	*Total* (including sign bit)
AC – DCT coefficients $Z\,(k, l)$	12 bits
Relative coefficients $C\,(k, l)$	12 bits
Quantized coefficients	11 bits

1.2.3 *Ranges of quantization parameters*

Information	*Range* (power of $2^{1/16}$)
Transmission threshold $n\,(k, l, m, f)$	0 to 175
Transmission factor f	0 to 175
	(power of $2^{1/8}$)
Relative visibility $p_0\,(k, l)$	0 to 44

The transmission factors are transmitted for each stripe of blocks and are each coded with 8 bits.

1.2.4 *Quantization characteristic*

Table II defines the quantization levels for the nearly-linear law for luminance and chrominance information.

TABLE II – *Quantizer, nearly-linear characteristic*

The quantization law is symmetric, so the characteristic is presented only for positive input values

Input values $C(k, l)$ or intervals	Quantizer levels	Quantized values $C'(k, l)$ (1)
0	0	0
1	1	1
2	2	2
:	:	:
255	255	255
256:257	256	256
258:259	257	:
:	:	:
510:511	383	510
512:515	384	513
516:519	385	:
:	:	:
1020:1023	511	1021
1024:1031	512	1027
1032:1039	513	:
:	:	:
2040:2047	639	2043

(1) Outputs of the inverse quantizer.

$p_0(k, l)$	0	1	2	3	4	5	6	7
0	0	0	2	8	12	18	22	28
1	0	6	6	10	16	18	22	34
2	0	6	10	14	18	20	24	38
3	2	6	12	16	18	20	26	40
4	6	12	14	16	20	22	28	42
5	10	14	14	18	22	24	30	42
6	14	16	16	18	22	24	34	44
7	14	18	18	20	24	30	38	44

FIGURE 3a – *Relative visibility matrix for luminance*

$$V(k, l) = 2^{p_0(k, l)/8}$$

$p_0(k, l)$	0	1	2	3	4	5	6	7
0	0	0	3	4	6	8	8	11
1	0	1	2	3	6	8	9	13
2	2	2	3	4	7	9	10	16
3	3	4	5	5	8	10	12	16
4	5	6	6	7	9	11	13	17
5	8	7	9	9	11	14	16	21
6	10	11	11	11	14	16	19	24
7	12	12	12	12	17	18	20	26

FIGURE 3b – *Relative visibility matrix for chrominance*

$$V(k, l) = 2^{p_0(k, l)/8}$$

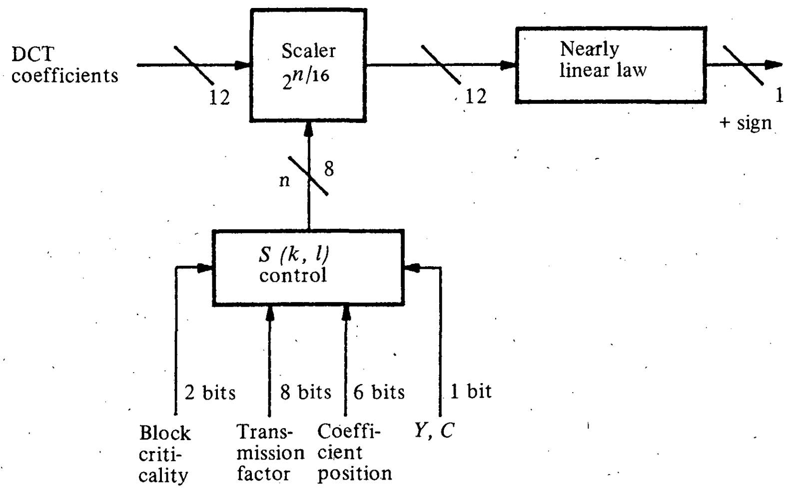


FIGURE 4a

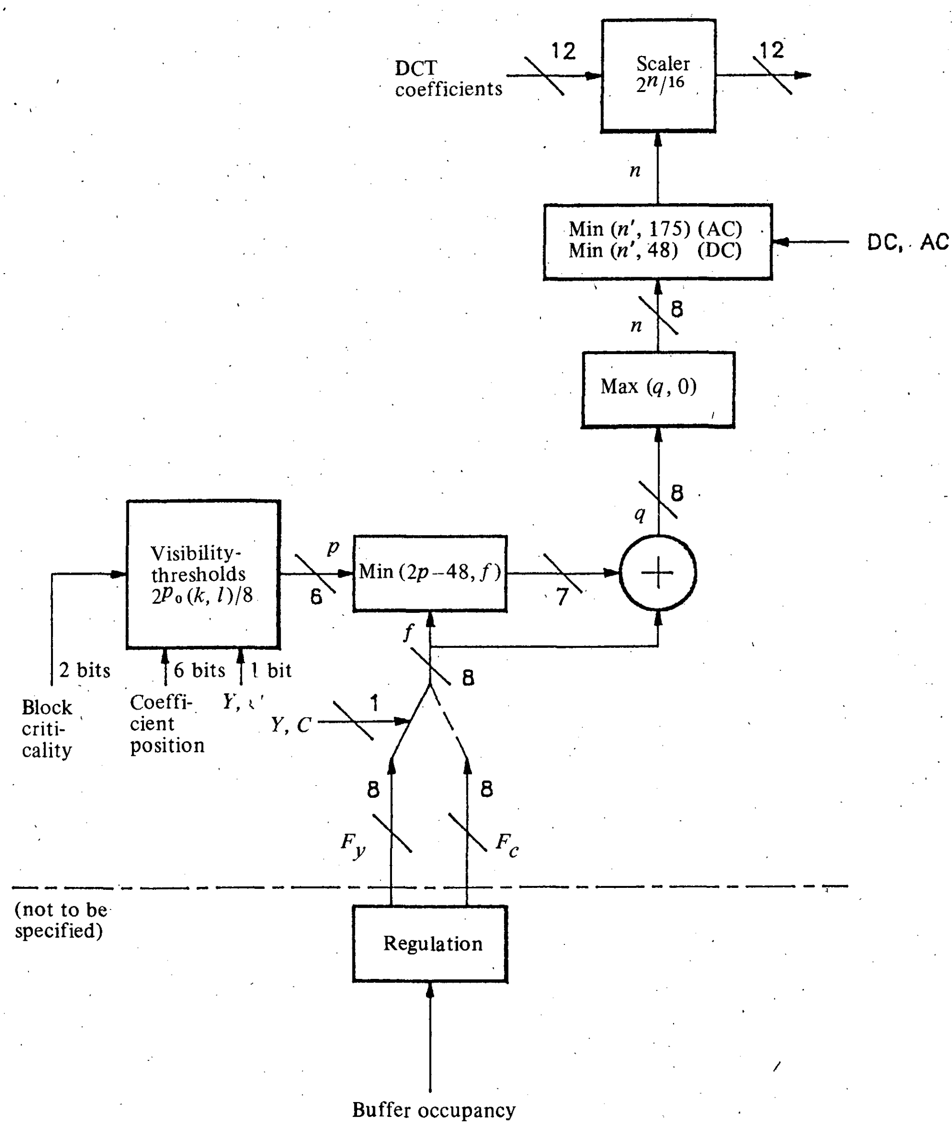


FIGURE 4b – *S (k, l) control*

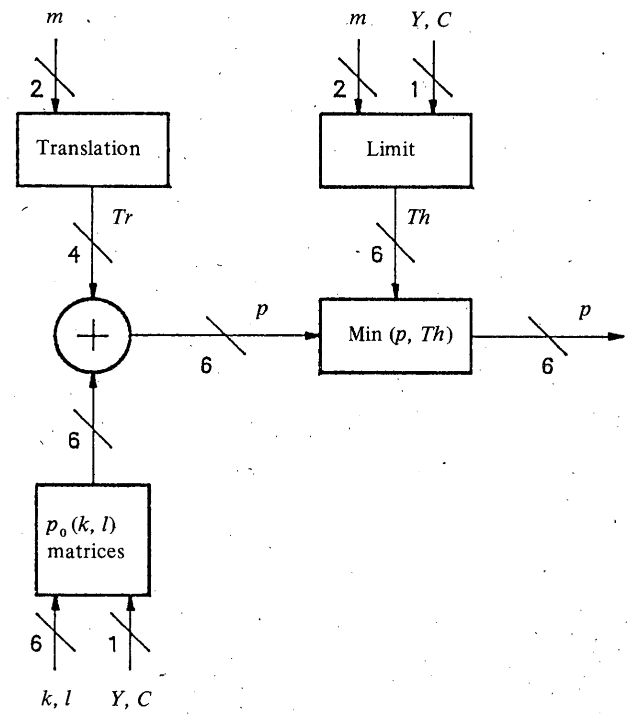


FIGURE 4c – *Computation of p (k, l, m)*

2. DC-coefficient quantization

The DC coefficient $Z\,(0,\,0)$ is quantized using the same process as the AC coefficients, but the scaling factor $n\,(0,\,0,\,m,\,f)$ of the DC coefficient is limited to the range (0, 48).

3. Inverse quantization

The reconstructed DCT coefficients are given by the following formula:

$$Z'(k, l) = C'(k, l) * S(k, l, m, f) * 1/2$$

where:

$S(k, l, m, f) = 2^{n(k, l, m, f)/16}$, as defined in § 1 above

$C'(k, l)$: is the quantized value corresponding to the transmitted quantizer level

$n(k, l, m, f)$ may be expressed as:

$n(k, l, m, f) = 16\,q + r$

q, r integers, $0 \leqslant r < 16$

so that:

$$Z'(k, l) = (C'(k, l) * 2^q) * 2^{r/16}$$

or:

$$Z'(k, l) = D(k, l) * 2^{r/16}$$

with:

$$D(k, l) = C'(k, l) * 2^q$$

The 12 bits values for $2^{r/16}$ are given in Table III. Note that the same set of values may be used in the quantizer.

$D(k, l) = C'(k, l) * 2^q$ is obtained by a binary left shift of q bits performed on the 12 bit value $C'(k, l)$. Only the rightmost 12 bits of the result are significant and are used in the following multiplication.

$Z'(k, l)$ is the result of the multiplication of $D(k, l)$ by $2^{r/16}$, truncated to 12 bits.

TABLE III – *Values of $2^{r/16}$*

r	$2^{r/16}$	$2048 * 2^{r/16}$
0	1.00000000000	2048
1	1.00001011011	2139
2	1.00010111001	2233
3	1.00100011100	2332
4	1.00110000011	2435
5	1.00111101111	2543
6	1.01001100000	2656
7	1.01011010110	2774
8	1.01101010000	2896
9	1.01111010001	3025
10	1.10001010110	3158
11	1.10011100010	3298
12	1.10101110100	3444
13	1.11000001101	3597
14	1.11010101100	3756
15	1.11101010010	3922

ANNEX VI

Under study in TG CMTT/2.

ANNEX VII

Under study in TG CMTT/2.

ANNEX VIII

FORWARD ERROR PROTECTION

The transmission signal is protected from transmission errors by an RS (255,239) code which is used to correct 8 byte errors and has 2 byte interleaving.

The generator polynomial of the RS code is given by

$$\prod_{i=0}^{15} (x + \alpha^i)$$

where α is a root of the binary primitive polynomial $x^8 + x^4 + x^3 + x^2 + 1$. A data byte $(d_7, d_6, \ldots, d_1, d_0)$ is identified with the element $d_7\alpha^7 + d_6\alpha^6 + \ldots + d_1\alpha + d_0$ in GF (256), the finite field with 256 elements.

The redundancy of the forward error correction (FEC) coding in 6.69%.

The data stream at the output of the video encoder is arranged in a matrix of 16 rows of 239 columns.

The RS (255,239) code is computed on each of the 2 rows of bytes and the 16 byte error control group is added to the corresponding row. The byte read-in and transmission (read-out) are performed from the first column with the sequence shown in Fig. 5.

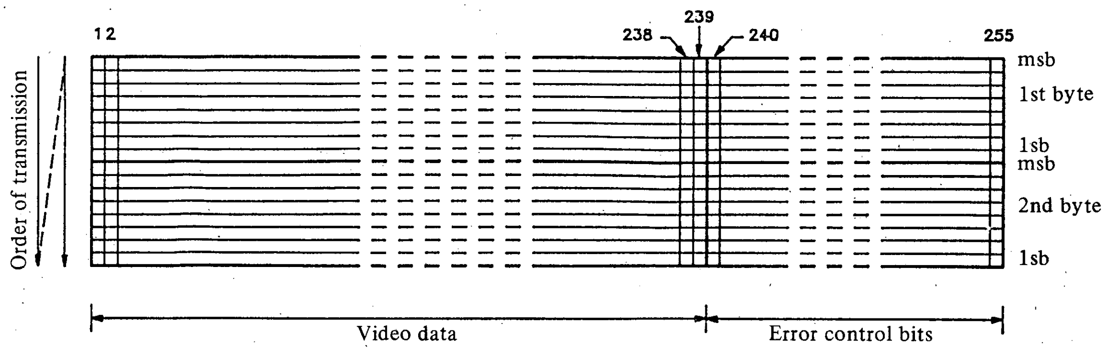


FIGURE 5

At the FEC decoder, error detection and correction is performed on a matrix obtained by organizing the received bit stream as in Fig. 5.

ANNEX IX

CHANNEL FRAMING

Framing is provided to synchronize the error correcting blocks of video information (see Annex VIII), to multiplex the sound and data channel together with the error corrected video and appropriate information for the link management, video and sound and data channel justification, if required, and synchronization of the video clock at the decoder side.

– Channel framing for the 32.064 Mbit/s channel to be studied.
– Channel framing for the 34.368 Mbit/s channel to be studied.
– Channel framing for the 44.736 Mbit/s channel to be studied.

RECOMMENDATION 721*

TRANSMISSION OF COMPONENT-CODED DIGITAL TELEVISION SIGNALS FOR CONTRIBUTION-QUALITY APPLICATIONS AT BIT RATES NEAR 140 Mbit/s**

(1990)

The CCIR,

CONSIDERING

(a) that for contribution-quality applications transmission should be based on component-coded digital video signals conforming with Recommendation 601;

(b) that such transmission should satisfy the user requirements for contribution-quality codecs at 60/70 to 140 Mbit/s as specified by Study Group 11 in Report 1211;

(c) that according to these user requirements such transmission should preserve the picture quality inherent in the 4:2:2 encoding process based on Recommendation 601 to the maximum extent possible;

(d) that such transmission should similarly preserve the downstream processing capabilities by maintaining the spatial and temporal resolution of the 4:2:2 signals as given by Recommendation 601;

(e) that besides component-coded digital video signals conforming with Recommendation 601, TV signals resulting from other source coding methods, in particular, component-coded TV signals for distribution, multiplexed analogue component (MAC) signals in digital form, or composite signals encoded according to Recommendation 658 should also be allowed for transmission;

(f) that additional transmission capacity should be provided for two pairs of stereo-sound channels, ancillary signals (e.g. teletext, test signals) and error-protection data;

(g) that for transmission the complete TV signal could be fitted into the fourth hierarchical level given by CCITT Recommendation G.702 as well as into the STM-1 level given by CCITT Recommendation G.707;

(h) that such transmission could be realized by hardware implementation at a moderate level of complexity and cost,

RECOMMENDS

that for transmission at data rates near 140 Mbit/s of component-coded digital video signals according to Recommendation 601, the bit rate reduction codec should be characterized as follows:

1. *Video input/output*

A standard 4:2:2 video signal conforming with Recommendation 601 is used. Bit-parallel or bit-serial interfacing in accordance with Recommendation 656 can be applied.

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* This Recommendation should be drawn to the attention of CCIR Study Group 11 and CCITT Study Groups XV and XVIII.

** Equipment built to the specifications given in this Recommendation has not yet been shown to meet the user requirements defined in Report 1211. The Administration of the United States of America therefore witholds its approval of this Recommendation, until confirmation that the required performance can be achieved.

2. *Signal pre-processing*

The horizontal and vertical blanking intervals are removed. Ancillary data such as teletext or test signals which are normally transmitted in the vertical blanking interval of the video signal are assigned to separate slots in the video multiplex.

No sub-sampling is used in order to satisfy the down-stream processing requirements of inter-studio links.

3. *Coding scheme*

Fixed two-dimensional predictors are applied both for the luminance and colour-difference components. Non-adaptive hybrid DPCM known as the Van Buul technique [Van Buul, 1978] in combination with folded quantizers as proposed by Bostelmann [Bostelmann, 1974] form the main elements of the coding scheme which reduces significantly the picture quality degradations due to overload effects as shown by simple DPCM systems. In addition, sensitivity to transmission errors comparable to systems with PCM representation is obtained.

Details of the quantizer characteristics used for the DPCM and PCM quantization of the luminance and colour-difference components are given in Table I.

No variable-length coding will be used; no post-processing is provided for the output signal.

4. *Video bit rate*

6 bit/sample for each of the luminance and colour-difference components results in a video bit rate of 124 416 kbit/s.

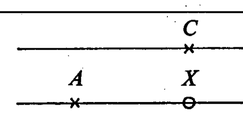
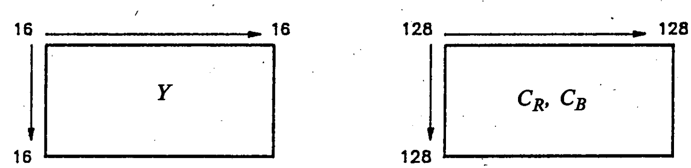
The main characteristics of this Recommendation are summarized in Table II.

TABLE I – *Quantizer characteristic*

Level No.	From	Value	To
0	0	0	0
1	1	1	1
2	2	2	2
3	3	3	3
4	4	4	4
5	5	5	5
6	6	6	6
7	7	7	7
8	8	8	8
9	9	9	9
10	10	11	12
11	13	14	15
12	16	17	18
13	19	20	21
14	22	23	24
15	25	26	27
16	28	30	32
17	33	35	37
18	38	40	42
19	43	45	47
20	48	50	52
21	53	55	57
22	58	61	64
23	65	68	71
24	72	75	78
25	79	82	85
26	86	89	92
27	93	96	99
28	100	103	106
29	107	110	113
30	114	117	120
31	121	124	127

The same folded quantizer characteristic is used for the DPCM and FCM quantization of the luminance and colour-difference components.

TABLE II – *Summary of the main characteristics recommended for the transmission of component-coded digital television signals for contribution-quality applications at bit rates near 140 Mbit/s*

Video input/output	Standard	525- or 625-line digital video in component form
	Coding	4:2:2 signals according to Recommendation 601
	Interface	Bit-parallel or bit-serial in accordance with Recommendation 656
Pre-processing	Blanking	Removal of horizontal and vertical blanking intervals
	Sub-sampling	None
	Pre-filtering	None
Coding	Predictor	Two-dimensional intra-field for luminance and colour-difference components
	Calculation of the prediction value X	$X = \frac{A + C}{2}$  The prediction value X is calculated with 8-bit accuracy
	Predictor preset	Video levels outside the active picture are set to 16 for Y and 128 for C_R, C_B to preset the initial value of the predictor in both the coder and decoder 
	Adaptive control of predictors	None
	Motion compensation	None
	Quantizer characteristics	Folded quantizer combined with Van Buul technique
	Bits/sample	6, for each of the luminance Y, and colour-difference components C_R, C_B
	Variable-length code	None
Post-processing		None
Video data rate		124 416 kbit/s

REFERENCES

BOSTELMANN, G. [1974] A simple high quality DPCM codec. *Nachrichtentechn.* 7., Vol. 27, **3**, 115-117.

VAN BUUL, M.C.W. [1978] Hybrid D-PCM, a combination of PCM and DPCM. *IEEE Trans. Comm.*, Vol. COM-26, **3**, 362-368.

ANNEX I

MULTIPLEXING FOR TRANSMISSION

1. Introduction

Two different approaches have been made to build up an appropriate multiplex for the transmission of component-coded video signals for contribution-quality applications.

1.1 *Multiplexing scheme (A)*

1.1.1 *Main characteristics*

This scheme, described in [CCIR, 1986-90a] makes use of a so-called TV container with a data rate of 138 240 kbit/s that is able to convey different video source signals with data rates of 135 000 kbit/s and audio/data signals with data rates of 2048 kbit/s. Within the video frame rate of 135 Mbit/s two 2048 kbit/s channels are available for data transmission which can be used for the transmission of two AES/EBU stereo sound signals.

Although the main interest of this TV container is directed to Y, C_R, C_B contribution signals and MAC/packet family signals, other component or composite signals are applicable. By means of stuffing or mapping techniques the TV container could be adapted to the channel framing of different transport media; in particular, the TV container will fit into a channel framing according to CCITT Recommendation G.751 as well as those of CCITT Recommendations G.707-G.709.

Table III summarizes the main characteristics of the TV container.

1.1.2 *Multiplexing scheme*

Video bit rate

6 bit/sample either for luminance and colour-difference components results in a video bit rate of 124 416 kbit/s.

Sample arrangement of Y, C_R, C_B within the multiplex scheme

For multiplexing, the 6-bit structure of the video data words is changed to an 8-bit structure whereby the transmission order according to Recommendation 656 C_B, Y, C_R, Y, C_B, Y, C_R etc. will remain unchanged. So the YUV encoder provides a data output according to the format given in Fig. 1.

Sound/data

For sound/data, two 2048 kbit/s channels associated with positive justification are included in the video multiplex. Each of these channels can carry a stereo sound signal according to the AES/EBU standard.

Ancillary data

A data rate of 4608 kbit/s is available inside the video multiplex for ancillary data (e.g. teletext, test signals) which are normally transmitted in the vertical blanking interval of the video signal.

Video multiplex

The sound/data and ancillary data are interleaved within the video data stream as shown in Fig. 2. In addition, the video multiplex includes 4 byte synchronization words (consisting of F1, F2 respectively $\overline{\text{F1}}$, $\overline{\text{F2}}$) which indicate the beginning of each line and each field. The bit structure of the two synchronization bytes is given by the following sequences:

F1: 0011 0001
F2: 1000 0011

The video multiplex including video, sound/data, ancillary data and sync words results in a data rate of 129 600 kbit/s.

TABLE III – *Main characteristics of TV container*

Video data rate	124 416 kbit/s
Sound/data 1, 2	2 × 2048 kbit/s (2 stereo AES audio signals)
Ancillary data (e.g. teletext)	460.8 kbit/s
Video framing	*YUV mode:* 2-octet synchronization word per video line (= subframe, 1125 octets) *MAC mode:* 4-octet synchronization word per video frame (5184 octets)
Video frame data rate	129 600 kbit/s
Video and sound/data error protection	Forward error correction using a two-dimensional product code: [110,108,3] RS code for rows, [102,100,3] RS code for columns, (symbol length: 1 octet each)
FEC redundancy	3.89%
Video multiplexing data rate ("TV container" input data rate)	135 000 kbit/s [(1)]
Sound/data 3	2048 kbit/s (2 stereo DS-1 audio signals)
Service channels (S1 ... S8)	8 × 64 kbit/s
"TV container" framing	256 kbit/s
"TV container" output data rate	138 240 kbit/s
– Channel framing according to CCITT Recommendation G.751 Channel rate	G.751 compatible framing using a frame length of 2928 bit (6 groups of 488 bits) and a 12 bit synchronization word 139 264 kbit/s
– Channel framing according to CCITT Recommendation G.709 Channel rate	G.709 compatible framing for direct synchronous or asynchronous mapping into virtual container VC-4 149 760 kbit/s

(1) In the case of ATM transmission this video multiplexing data rate could be fully conveyed by ATM cells offering a net data rate of 149 720 kbit/s × 48/53 = 135 631.7 kbit/s at the STM-1 level.

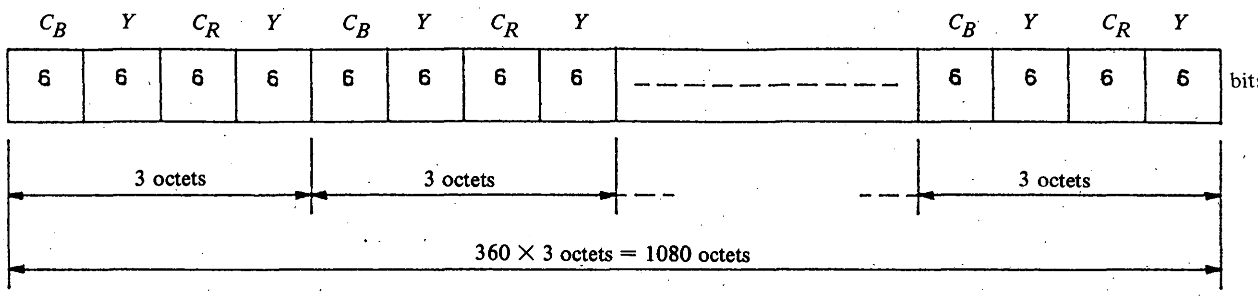


FIGURE 1 – *Structure of a TV line*

Frame length: (1 TV line)	1080 octets (1 octet = 8 bits)
Repetition rate: (576 active lines)	14 400 kHz
Bit rate:	124 416 kbit/s

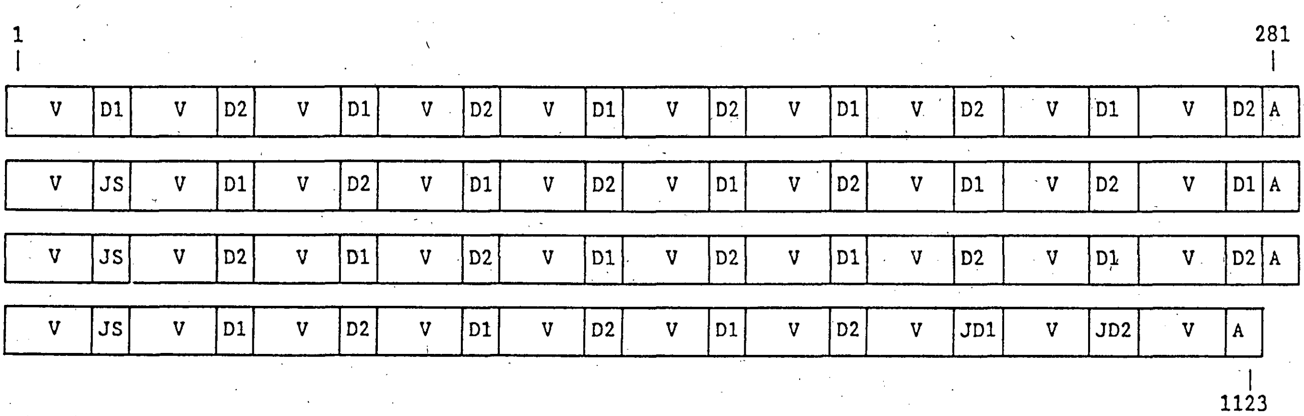


FIGURE 2 – *One row of the video multiplexing frame for YUV contribution signal*

			No. of octets
Slots:	V	Video information	40 × 27 = 1080
	D1	Data channel 1	17 × 1 = 17
	D2	Data channel 2	17 × 1 = 17
	A	Ancillary data	4 × 1 = 4
	JS	Justification service	3 × 1 = 3
	JD1	Justification data channel 1	1 × 1 = 1
	JD2	Justification data channel 2	1 × 1 = 1
			Total: 1123

Octets:	JS	J1 J1 J1 J1 J2 J2 J2 J2 binary
	JD1	D1 D1 D1 D1 D1 D1 S1 F binary
	JD2	D2 D2 D2 D2 D2 D2 S2 F binary

Bits:	J1	1 = justifying data channel 1, 0 = not justifying
	J2	1 = justifying data channel 2, 0 = not justifying
	D1, D2	Bits from data channel 1, 2
	S1, S2	Justification bit data channel 1, 2
	F	Free

Error protection

The complete video multiplex is protected from transmission errors by a two-dimensional FEC product code using a 1 octet correcting RS coder in each direction.

Code for rows:
(110, 108, 3) RS code defined over GF(2^8)

Code for columns:
(102, 100, 3) RS code defined over GF(2^8)

The field generator polynomial is:
$f(\alpha) = \alpha^8 + \alpha^4 + \alpha^3 + \alpha^2 + 1$

The code generator polynomial is:
$g(x) = (x + \alpha)(x + 1) = x^2 + \alpha^{25} x + \alpha$,
with α = 0000 0010

The redundancy is 3.89%.

FEC framing

The data stream at the FEC coder is arranged in a matrix of 102 rows of 110 columns of octets (see Fig. 3). 9.6 TV lines including ancillary data and video sync words will be transmitted within each FEC frame.

The transmission of the FEC frames results in a data rate of 134 640 kbit/s; the repetition rate of the FEC frame is 1500 Hz.

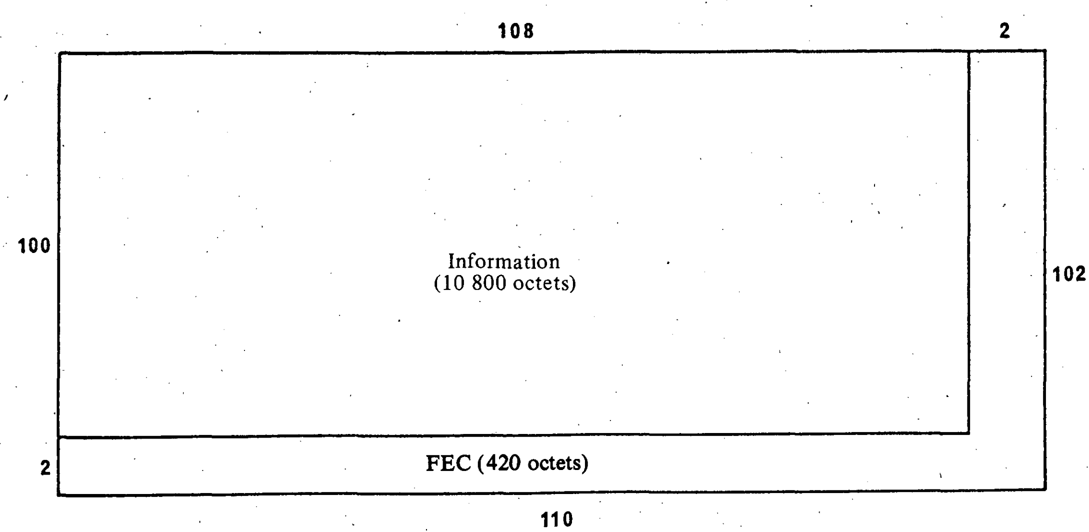


FIGURE 3 – *FEC framing*

Repetition rate:	1500 Hz	
Video, data, sync:	10 800 octets →	129 600 kbit/s
FEC:	420 octets →	5 040 kbit/s
Total:	11 220 octets →	5 040 kbit/s

Channel framing

The structure of the channel framing will provide access to the video channel of the TV container which offers a total bit rate of 135 000 kbit/s for the complete TV input signal. Each output frame contains 9.6 complete TV lines whereby each row of the frame starts with a 2-octet synchronization word (see Fig. 4). The bit structure of the four synchronization octets is given by the following sequences:

C1: 0011 1100
C2: 1101 0101
$\overline{\text{C1}}$: 1100 0011
$\overline{\text{C2}}$: 0010 1010

This framing arrangement leads exactly to the bit rate offered by the video channel of the TV container.

Bit rate budget

The bit rate budget chosen for the video multiplexing, FEC framing and TV container is given in Table IV.

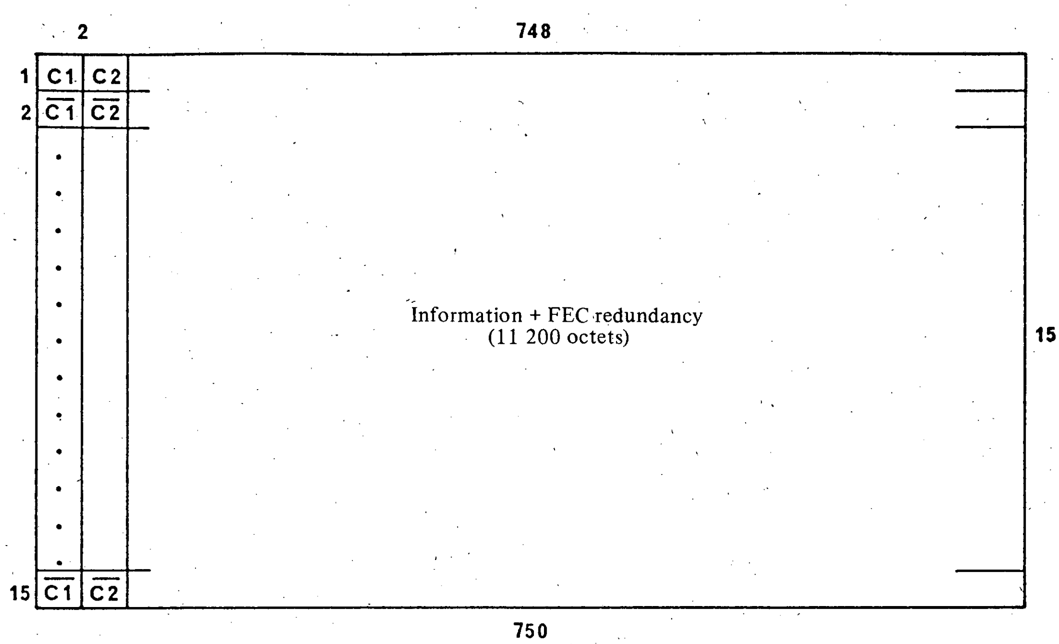


FIGURE 4 – *Channel framing*

Repetition rate:	1500 Hz	C1: 0011 1100 C2: 1101 0101
Video, data, FEC:	11 220 octets → 134 640 kbit/s	$\overline{\text{C1}}$: 1100 0011
Framing:	30 octets → 360 kbit/s	$\overline{\text{C2}}$: 0010 1010
Total:	11 250 octets → 135 000 kbit/s	

TABLE IV – *Bit rate budget*

(a) Video multiplexing frame: (Repetition rate: 25 Hz)	YUV contribution signal	
Channel	Number of bits	Data rate (kbit/s)
Video information	576 × 8 640	124 416.000
Data channel 1	576 × 142	2 044.800
Justification (1)	576 × 1	14.400
Justification service	576 × 12	172.800
Data channel 2	576 × 142	2 044.800
Justification (1)	576 × 1	14.400
Justification service	576 × 12	172.800
Ancillary data	576 × 32	460.800
Video framing	576 × 16	230.400
Free	576 × 2	28.800
Total sum	576 × 9 000	129 600.000
(b) FEC frame: (Repetition rate: 1500 Hz) Channel	Number of bits	Data rate (kbit/s)
Information	86 400	129 600.000
FEC redundancy	3 360	5 040.000
FEC framing	240	360.000
Total sum	90 000	135 000.000
(c) TV container:	Number of 64 kbit/s slots	Bit rate (kbit/s)
Video channel	2 109	134 976
Video justification (2)	1	64
Video justification control (2)	3	192
Video channel data rate: 135 000 kbit/s ± (0/59.26/177.78) ppm		
Audio/data channel	31	1 984
Audio/data justification (2)	1	64
Audio/data justification control (2)	3	192
Audio/data channel data rate: 2048 kbit/s ± (0/3906) ppm		
Service channels (S1 ... S8)	8	512
Channel framing	4	256
Total sum	2 160	138 240

(1) Positive justification; channel data rate: 2048 kbit/s (+5468/−1562) ppm.

(2) Positive-zero-negative justification.

1.2 *Multiplexing scheme (B)*

1.2.1 *Main characteristics*

This scheme, described in [CCIR, 1986-90b] starts from the need expressed by some broadcasters for the transmission of two stereo pairs of high-quality sound in addition to the video. Two 1920 kbit/s channels are then provided for sound transmission. The general structure of the frame is in accordance with CCITT Recommendation G.751 with respect to frame length, alignment and organization. Video and audio justification is provided such that the jitter of the recovered clock is maintained within the limits specified by CCIR Recommendation 656.

The total rate of 139 264 kbit/s can be mapped into the channels defined by CCITT Recommendations G.707-G.709. The proposed transmission frame at 140 Mbit/s easily allows the transmission of all component coded, multiplexed analogue components or composite signals. Whatever the kind of signal to be transmitted, the rate assigned to the video frame with error protection ("FEC frame" at 133 824 kbit/s) is identical.

One important application of this multiplex is the connection of digital video recorders (one 4:2:2 video signal and two stereo pairs of sound encoded according to the AES/EBU standard) for which experiments have already been completed. The total system, including channel framing and FEC, has been tested by IWP 11/7.

1.2.2 *Multiplexing frame*

Video bit rate

6 bits/sample
124 416 kbit/s.

Video framing

The multiplex arrangement for the video data and test signals includes 12 bit synchronization words which indicate the beginning of each line and each field. The ancillary data lines are shared among all the lines of the video field.

Video data error protection

Forward error correction and interleaving. The transmitted video data (and ancillary data) are protected from transmission errors by six parallel working FEC coders each using a BCH (324,306) code which is able to correct 2 bits in each path.

The field generator polynomial is:

$f(a) = a^9 + a^4 + 1$

The code generator polynomial is:

$g(X) = X^{18} + X^{15} + X^{12} + X^{10} + X^8 + X^7 + X^6 + X^3 + 1$

The redundancy is 5.9%.

To improve the correction capabilities for burst errors, a 4-symbol interleaving in each path is adopted. The protection is applied on the whole video frame (video synchronization, ancillary data and video data). A "protection frame" is built with 6×43 symbols. Each frame begins with a 6×8 bit synchronization word and contains nine complete video data lines (Fig. 5). This synchronization word includes the interleaving initialization slot. In this way error bursts of length up to 48 bits can be corrected.

Channel framing

CCITT Recommendation G.751 structure.

Recommendation G.751 proposes a channel frame for transmission at 139 264 kbit/s, which uses a 12 bit synchronization word and is 2928 bits long shared into six groups of 488 bits (Fig. 6).

Bit rate budget

The total available bit rate has to contain not only the video information, but also data associated with the picture, sound, exploitation channels, error protection data and framing information for the multiplex.

For video including FEC and framing, a bit rate of 133 824 kbit/s is assigned.

For sound, two 1920 kbit/s channels, associated with a positive justification, are available. Each sound interface is in accordance with the EBU/AES digital stereo interface (20-bit samples – 48 kHz sampling frequency).

The complete bit rate budget for the system is given in Table V.

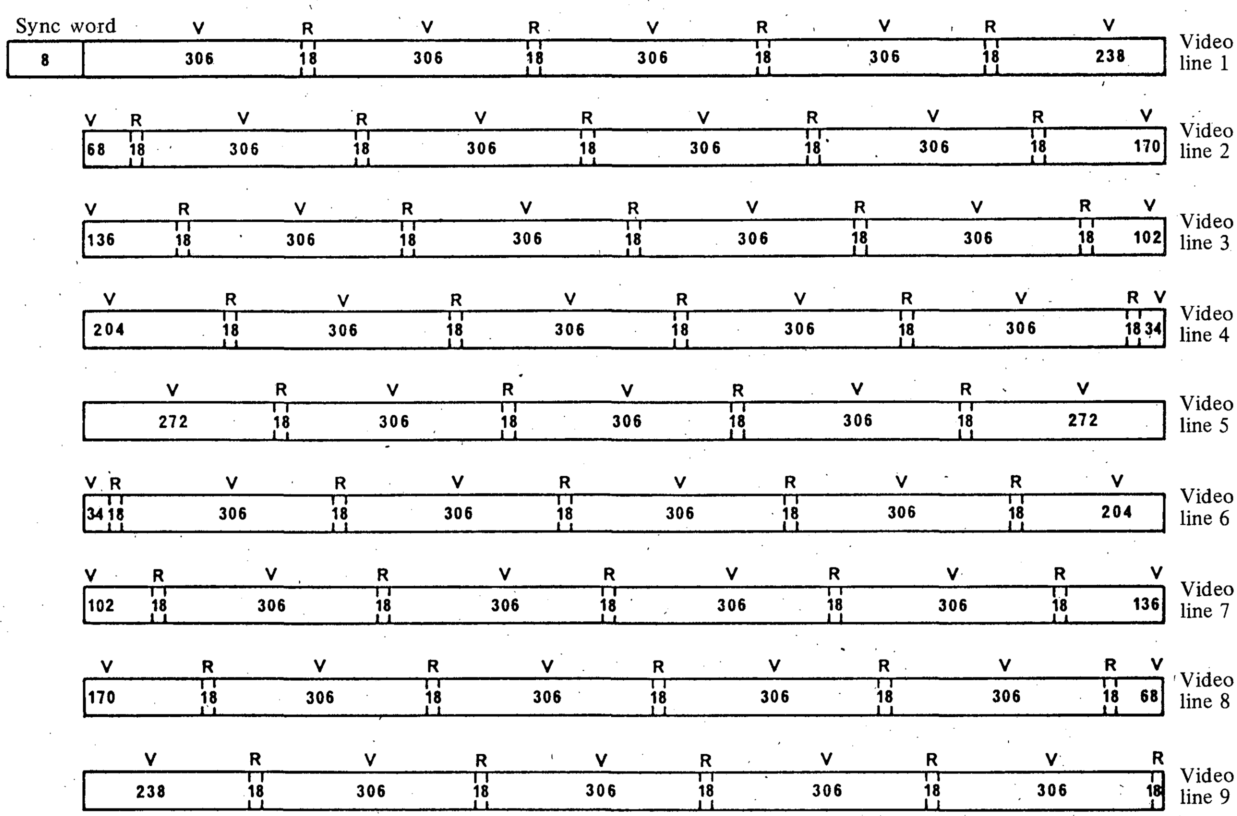


FIGURE 5 – *Protection frame*

V: video
R: redundancy

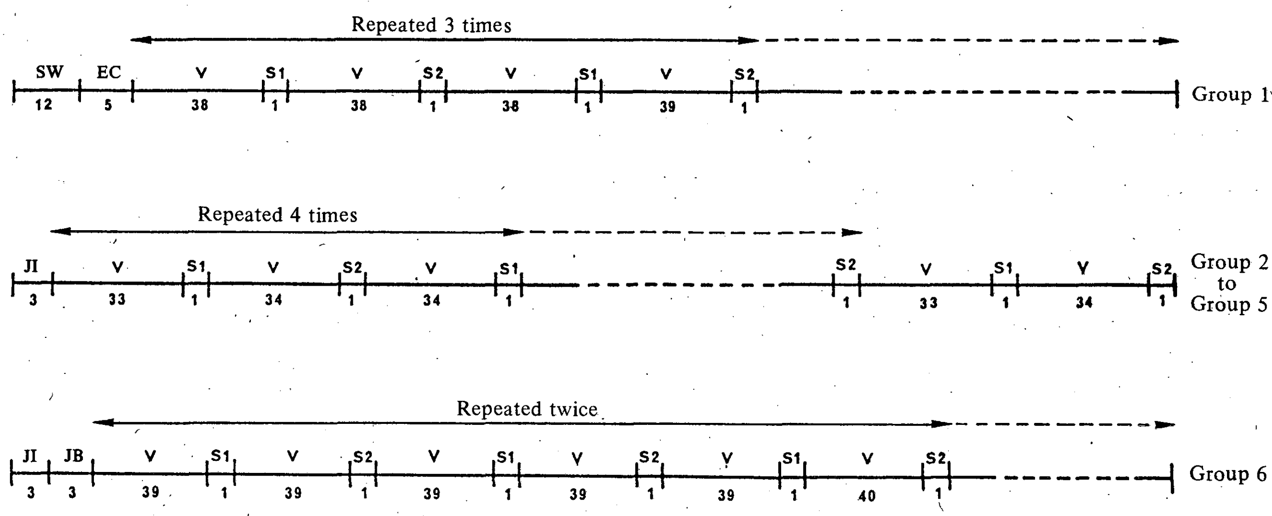


FIGURE 6 – *Channel framing*

SW: synchronization word
EC: exploitation channels
V: video data
S1, S2: sound channels 1 and 2
JI: justification control
JB: justification time slots

TABLE V – *Bit rate budget (kbit/s)*

Video frame Picture FEC Ancillary data Framing	 124 416 7 507 1 728 173
Video justification Average margin Control	 18 238
Audio Global bit rate	 3 840
Audio justification Average margin Control	 60 475
Other data Channel framing Exploitation channels	 571 238
Total	139 264
Frame length (bits)	2 928

REFERENCES

CCIR Documents
[1986-90]: **a.** IWP CMTT/2-130 (Germany (Federal Republic of)) **b.** IWP CMTT/2-100 (France).

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SECTION CMTT B: METHODS OF OPERATION AND ASSESSMENT OF PERFORMANCE OF TELEVISION TRANSMISSIONS

RECOMMENDATION 473-5*

INSERTION OF TEST SIGNALS IN THE FIELD-BLANKING INTERVAL OF MONOCHROME AND COLOUR TELEVISION SIGNALS

(Study Programmes 15A/CMTT and 12A/11)

(1970-1974-1978-1982-1986-1990)

The CCIR,

CONSIDERING

(a) that it is already current practice in a number of countries to use insertion test signals in the field-blanking interval of monochrome and colour television signals;

(b) that such signals can be used for the measurement of performance and the monitoring, control and correction of characteristics of international transmission circuits;

(c) that Report 314 proposes that certain specific lines in each field be allocated for the insertion of special test signals for international transmissions;

(d) that traffic demands may make it necessary to perform all operational measurements by means of insertion test signals, to an accuracy approaching that of conventional out-of-service measuring methods,

UNANIMOUSLY RECOMMENDS

1. that for the international transmission of television signals, insertion test signals in accordance with Annex I (625-line systems) and Annex II (525-line systems) may be inserted at the origin of the circuit;

Note. – As an interim measure some administrations may decide to omit some of the waveforms described. Where waveforms are omitted:

- waveforms other than those described should not be inserted;
- care must be taken to ensure that the line-time luminance components of equivalent lines in each field (e.g. 17 and 330 for 625-line systems) are similar.

2. that these signals should neither be removed nor replaced on the international circuit, except possibly at a point of conversion of either standard or colour system.

ANNEX I

625-LINE SYSTEMS

1. Introduction

For the international transmission of 625-line television signals, Recommendation 472 and Report 314 propose the use of lines 17 (330) and 18 (331) for insertion test signals.

This Annex describes a comprehensive arrangement of insertion test signals (see Note) to which the following general considerations apply:

- it is assumed that the line duration H is divided into 32 equal time periods. This division defines the characteristic instants;
- the time periods shall not differ from each other by more than ± 40 ns;
- the characteristic instants are referred to the mid-amplitude point of the leading edge of the synchronizing pulse. The half-amplitude points of the luminance and chrominance transitions and the peaks of the pulses occur at the characteristic instants;

* CCITT Recommendation N.67 has been amended to bring it into agreement with this Recommendation.

- the actual characteristic instants of any luminance waveform shall not differ by more than 250 ns from their nominal positions;
- except in the case of the $20T$ composite pulse, the actual characteristic instants of any chrominance waveform shall not differ by more than 500 ns from their nominal positions;
- the colour burst is present in the line-blanking period only in colour transmission;
- in the case of PAL transmissions, the chrominance sub-carrier of the insertion signals is locked at 60 ± 5° from the positive $(B-Y)$ axis;
- harmonic distortion components of the sub-carrier shall be at least 40 dB below the level of the fundamental;
- the frequency of the sub-carrier is 4.433 618 75 MHz ± 10 Hz.

Note. – These are intended for use with colour television signals. The basic insertion test signal for monochrome transmissions is identical, with the following exceptions:

Line 17: element F is omitted;

Line 18: the luminance pedestal and elements C_1 and C_2 are omitted;

Line 330: element D_2 is replaced by D_1;

Line 331: the luminance pedestal and elements G and E are omitted.

The following additions to the basic monochrome insertion test signal may be found useful:

(a) the luminance pedestal on lines 18 and 331 and elements C_1 and C_2 on line 18

and/or

(b) the element F on line 17.

Alternatively, the whole of the signals may be utilized. The above modifications to the basic monochrome insertion test signal should, however, only be made with the agreement of the administrations concerned.

2. Particulars of signals inserted in line 17 (Fig. 1)

2.1 *Luminance bar* (reference white level) (B_2)

- position of transitions: $6H/32$ and $11H/32$, duration of bar $5H/32$;
- bar amplitude: 0.700 ± 0.007 V;
- rise and fall times of transitions: derived from the shaping network of the sine-squared pulse (element B_1);
- overshoot and undershoot ⩽ 0.5%;
- tilt ⩽ 0.5%.

2.2 *2T sine-squared pulse* (B_1)

- peak position: $13H/32$;
- amplitude: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- half-amplitude duration: 200 ± 10 ns (see Note).

Note. – In some countries, members of the OIRT, the half-amplitude duration of the $2T$ sine-squared pulse may be 160 ns.

2.3 *Composite 20T pulse* (F)

- position of peak: $16H/32$;
- position of base: $15H/32$ - $17H/32$;
- amplitude: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- half-amplitude duration: 2 ± 0.06 µs;
- perturbations of the pulse base-line, due to inherent chrominance-luminance amplitude and delay inequalities and different shape of the luminance and chrominance components: ⩽ 0.5% peak amplitude.

2.4 *Five-riser luminance staircase* (D_1) (see Note)

- position of successive transitions: $20H/32$, $22H/32$, $24H/32$, $26H/32$, $28H/32$ and $31H/32$ (fall);
- peak-to-peak amplitude of the staircase: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);

- nominal amplitude of risers: 1/5 of amplitude of the luminance bar (B_2) (nominal value: 0.140 V). The difference in amplitude between the largest and smallest risers must be less than 0.5% of the largest amplitude;
- rise and fall times of transitions: shaped by a Thomson filter (or similar network) with a transfer function modulus having its first zero at 4.43 MHz to restrict the amplitude of components of the luminance signal in the vicinity of the colour sub-carrier.

Note. – Some administrations may wish to superimpose a chrominance sub-carrier signal on this staircase. In this case, the position and duration of the sub-carrier are determined by instants $18H/32$ and $31H/32$. The other characteristics of this signal are identical to those described in § 4.3.2.

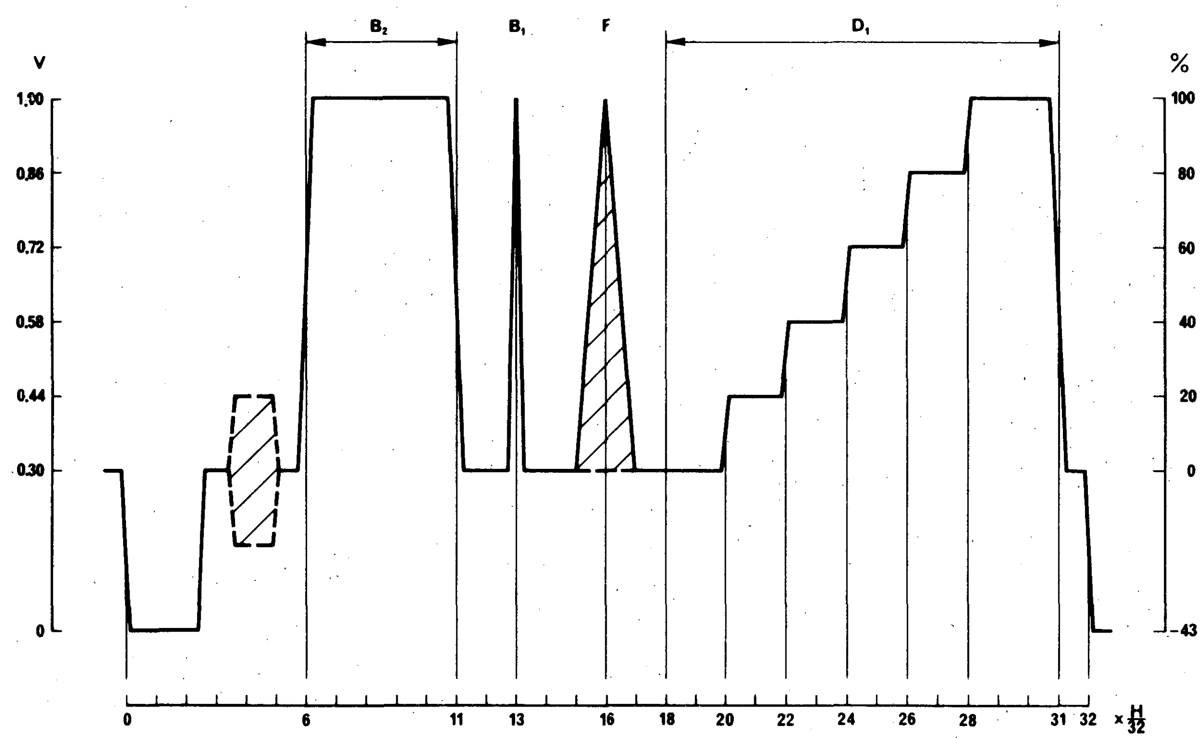

FIGURE 1 – *Line 17*

3. **Particulars of signals inserted in line 18** (Fig. 2)

3.1 *Luminance pedestal*

- position of transitions: $6H/32$, $31H/32$;
- amplitude measured from blanking level: within ± 1% of one-half of the amplitude of luminance bar (B_2) (nominal value: 0.350 V);

3.2 *Reference bar signal* (C_1)

- position of transitions: $6H/32$, $8H/32$, $10H/32$;
- amplitudes measured from blanking level:
  1st section: within ± 1% of four-fifths of the amplitude of the luminance bar (B_2) (nominal value: 0.560 V);
  2nd section: within ± 1% of one-fifth of the amplitude of the luminance bar (B_2) (nominal value: 0.140 V);
- rise and fall times of transitions: derived from the shaping network of the sine-squared pulse (element B_1).

3.3 *Sine-wave signals superimposed on the pedestal* (C_2)

– starting positions and frequencies of the bursts (see Table I);

TABLE I

Burst No.	Precise starting position(1) (2)	Frequency MHz(3)
1	12*H*/32	0·5
2	15*H*/32	1·0(4)
3	18*H*/32	2·0(4)
4	21*H*/32	4·0
5	24*H*/32	4·8
6	27*H*/32	5·8

(1) The starting point of each burst shall be at zero phase of the sine-wave, and each burst shall consist of the maximum number of complete cycles. The gaps between successive bursts shall not be shorter than 0.4 μs nor longer than 2.0 μs in duration.

(2) Some administrations may prefer to use burst durations different from those shown above and in Fig. 2.

(3) Spectral components of the bursts may cause interference to sub-carriers or noise detection circuits and the out of band energy should be limited by suitable design techniques. Other frequencies near to the above mentioned may be used, subject to an agreement between the administrations concerned.

(4) In some countries, members of the OIRT, the frequencies of bursts Nos. 2 and 3 may be 1.5 MHz and 2.8 MHz respectively.

– the peak-to-peak amplitude of bursts shall be within ± 1% of the peak-to-peak amplitude of the reference bar signal (C_1) (nominal value: 0.420 V);
– the d.c. component of each burst shall not exceed 0.5% of the amplitude of the reference bar signal (C_1);
– the harmonic distortion components of each burst are to be at least 40 dB (see Note) below the fundamental.

Note. – This value is subject to further study.

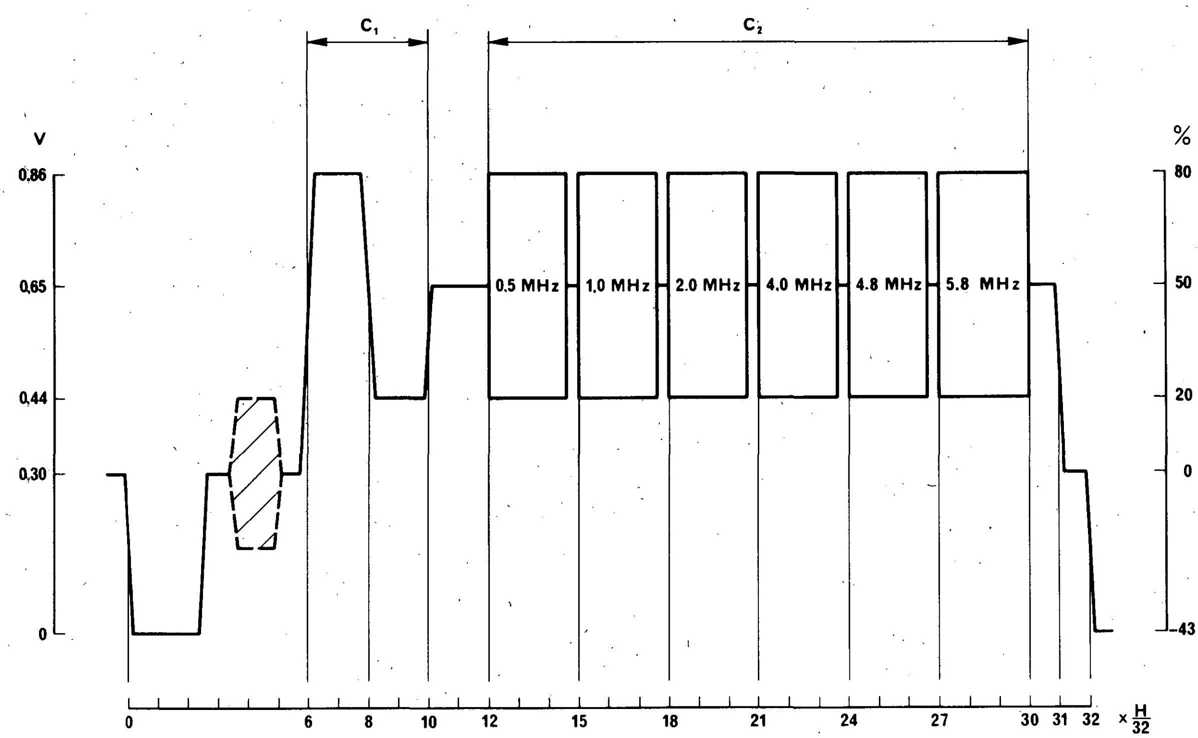

FIGURE 2 – *Line 18*

4. Particulars of signals inserted in line 330 (Fig. 3)

4.1 *Luminance bar* (reference white level) (B_2)

- position of transitions: $6H/32$ and $11H/32$, duration of bar $5H/32$;
- bar amplitude: 0.700 ± 0.007 V;
- rise and fall times of transitions: derived from the shaping network of the sine-squared pulse (element B_1);
- overshoot and undershoot ⩽ 0.5%;
- tilt ⩽ 0.5%.

4.2 *2T sine-squared pulse* (B_1)

- peak position: $13H/32$;
- amplitude: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- half-amplitude duration: 200 ± 10 ns. (In some countries, members of the OIRT, the half-amplitude duration of the $2T$ sine-squared pulse may be 160 ns.)

4.3 *Five-riser luminance staircase* (D_1) *and superimposed five-riser staircase* (D_2)

4.3.1 The five-riser luminance staircase has the following characteristics:

- position of successive transitions: $20H/32$, $22H/32$, $24H/32$, $26H/32$, $28H/32$ and $31H/32$ (fall);
- peak-to-peak amplitude of the staircase: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- nominal amplitude of risers: 1/5 of amplitude of the luminance bar (B_2) (nominal value: 0.140 V). The difference in amplitude between the largest and smallest risers must be less than 0.5% of the largest amplitude;
- rise and fall times of transitions: shaped by a Thomson filter (or similar network) with a transfer function modulus having its first zero at 4.43 MHz to restrict the amplitude of components of the luminance signal in the vicinity of the colour sub-carrier.

4.3.2 The chrominance signal superimposed on the five-riser luminance staircase (D_1) has the following characteristics:

- position and duration: $15H/32$ to $30H/32$. The superimposed sub-carrier may be limited to $28H/32$;
- peak-to-peak amplitude: 0.280 V ± 2%;
- inherent differential-gain distortion: ⩽ 0.5%;
- inherent differential-phase distortion: ⩽ 0.2°;
- rise and fall times of the envelope of the chominance transition: 1 μs approximately.

5. Particulars of signals inserted in line 331 (Fig. 4)

5.1 *Luminance pedestal*

- position of transitions: $6H/32$, $31H/32$;
- amplitude measured from blanking level: within ± 1% of one-half of the amplitude of the luminance bar (B_2) (nominal value: 0.350 V);
- rise and fall times of transitions: derived from the shaping network of the sine-squared pulse (element B_1).

5.2 *Superimposed chrominance bar signal* (G_1)

- position of transitions: $7H/32$, $14H/32$;
- peak-to-peak amplitude: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- rise and fall times of the envelope of the chrominance signal transactions: 1 μs approximately;
- inherent chrominance-luminance cross-talk: ⩽ 0.5% of luminance pedestal amplitude;
- phase difference between the sub-carrier superimposed on the staircase in line 330 and the sub-carrier superimposed on line 331: ⩽ 2°.

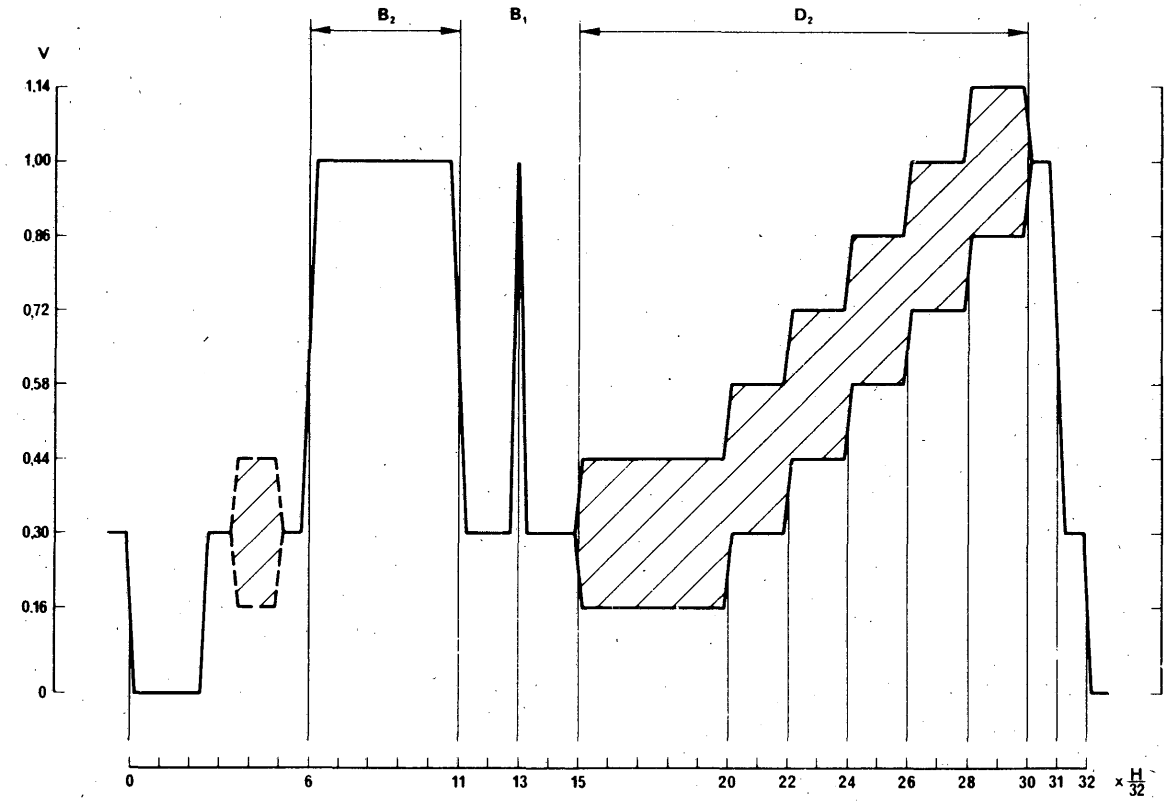

FIGURE 3 – *Line 330*

5.3 *Superimposed three-level chrominance signal* (G_2)

This signal may be used as an alternative to the superimposed chrominance bar signal defined above:

- position of transitions: $7H/32$, $9H/32$, $11H/32$ and $14H/32$;
- peak-to-peak amplitudes:
  1st section: within ± 1% of one-fifth of the amplitude of the luminance bar (B_2) (nominal value: 0.140 V);
  2nd section: within ± 1% of three-fifths of the amplitude of the luminance bar (B_2) (nominal value: 0.420 V);
  3rd section: within ± 1% of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- rise and fall times of the envelope of the chrominance signal transitions: 1 µs approximately;
- inherent chrominance-luminance cross-talk: ⩽ 0.5% of luminance pedestal amplitude;
- inherent phase/amplitude distortion: ⩽ 0.5°;
- phase difference between the sub-carrier superimposed on the staircase in line 330 and the sub-carrier superimposed on line 331: ⩽ 2°.

5.4 *Superimposed reference sub-carrier* (E)

This auxiliary signal may be used as a reference sub-carrier for the measurement of differential phase;

- position of transitions: $17H/32$, $30H/32$;
- peak-to-peak amplitude: within ± 1% of three-fifths of the amplitude of the luminance bar (B_2) (nominal value: 0.420 V);
- rise and fall times of the envelope of the chrominance signal transitions: 1 µs approximately;
- phase difference between the sub-carrier superimposed on the staircase in line 330 and the sub-carrier superimposed on the line 331: ⩽ 2°.

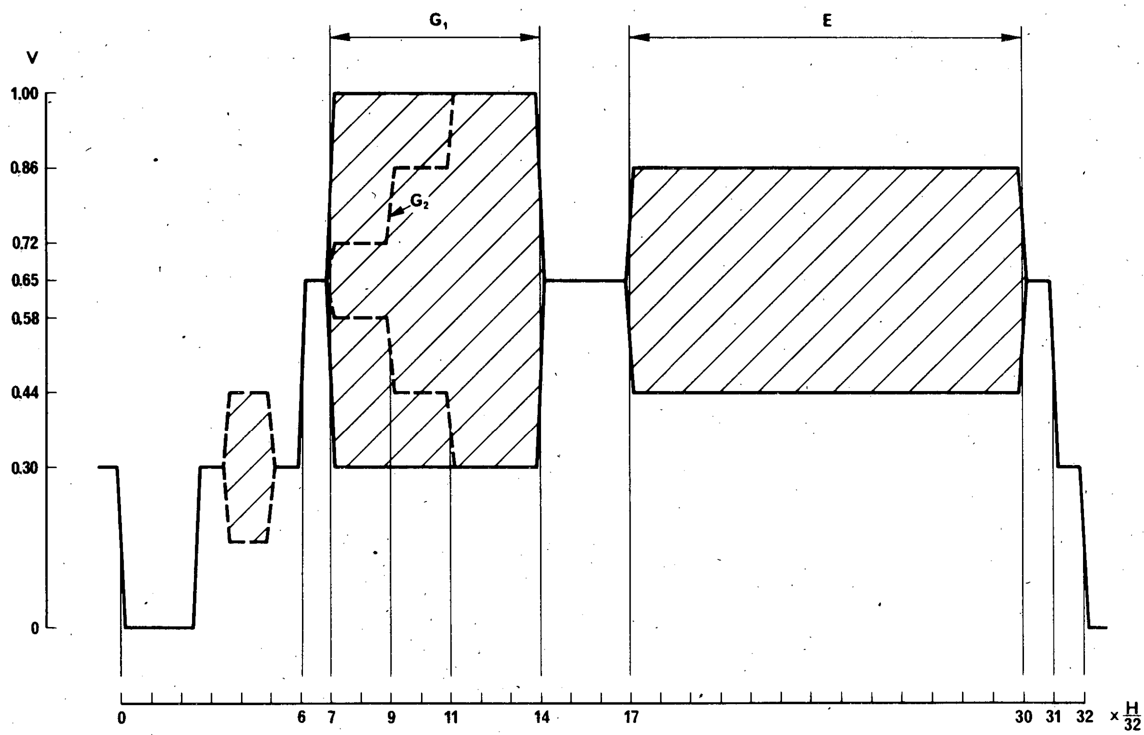

FIGURE 4 – *Line 331*

6. List of measurements which can be made with the defined insertion test signals

TABLE II

Characteristics measured	Waveform used	Line number
Linear distortions		
Insertion gain	B_2	17 and 330
Amplitude/frequency response	C_2 and C_1	18
Line-time waveform distortion	B_2	17 and 330
Short-time waveform distortion:		
— step response	B_2	17 and 330
— pulse response	B_1	17 and 330
Chrominance-luminance gain inequality	B_2 and G_1 or G_2	17 and 330, 331
	B_2 and F	17
Chrominance-luminance delay inequality	F	17
Non-linear distortions		
Luminance line-time non-linearity	D_1	17
Chrominance non-linearity	G_2	331
Luminance-chrominance intermodulation:		
— differential gain	D_2	330
— differential phase	D_2 and E	330, 331
Chrominance-luminance intermodulation	B_2 and G_1 or G_2	17, 331

ANNEX II

525-LINE SYSTEMS

1. Introduction

This Annex describes waveforms and the corresponding specification of insertion test signals to which the following general considerations apply:

- for international transmission of a 525-line television signal, line 17 of both fields (lines 17 and 280 if numbered consecutively), are reserved for international insertion test signals (see Note 1);
- the signals defined in this Annex apply to both monochrome and colour television transmission, as shown in Figs. 5 and 6. For monochrome transmission, some simplifications of the test signal by the omission of one or more of its components may be desirable. Such simplified signals are shown in Figs. 7 and 8;
- the line duration H is divided into 128 equal parts, and the position and duration of test signals are determined in $H/128$. This division defines the characteristic instants;
- the characteristic instants are referred to the half-amplitude points of the leading edge of the luminance bar signal (B_2) in Figs. 5 or 7 and the reference bar signal (C_1) in Figs. 6 or 8, which are inserted in fields 1 and 2 respectively, (O_{HR}). The half-amplitude point of the luminance and chrominance transitions and peak of the pulses occur at the characteristic instants;
- positioning of the reference point (O_{HR}) shall not exceed $24H/128 \pm 125$ ns relative to the mid-amplitude point of the leading edge of the horizontal synchronizing pulse (O_H);
- the systematic offset in the defined characteristic instants of any luminance and chrominance waveforms shall not differ by more than ± 150 ns (see Note 2) and ± 300 ns (see Note 2) respectively, from the nominal points;
- the random error in the defined characteristic instants for both luminance and chrominance waveforms shall not exceed ± 25 ns from a fixed position which lies within the above systematic offset;
- the colour burst is present in the line-blanking period only in colour transmission;
- the frequency of the colour sub-carrier is 3.579 545 MHz for system M/NTSC, and 3.575 611 49 MHz for system M/PAL, ± 10 Hz.

Note 1. – The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

Note 2. – Reduction in these tolerances is a matter for further study.

2. Particulars of signals inserted in line 17 of field 1 (Figs. 5 or 7) (see Note 1 of § 1)

2.1 *Luminance bar* (reference white level) (B_2)

- position of transitions: $0H/128$ (O_{HR}) and $36H/128$, duration of bar $36H/128$;
- bar amplitude: 100 ± 0.5 IRE units (see Note);
- rise and fall times of transitions (integrated sine-squared shape): 125 ± 5 ns;
- overshoot and undershoot: ⩽ 1%;
- tilt: ⩽ 0.5%.

Note. – For 525-line systems, the signal amplitude is expressed in Institute of Radio Engineers (IRE) units. By convention, 100 IRE units correspond to the amplitude comprised between the blanking level and the white level (see Figs. 5 to 8).

2.2 *2T sine-squared pulse* (B_1)

- position of peak: $44H/128$;
- amplitude: within ± 0.5 IRE unit of the amplitude of the luminance bar (B_2) (nominal value: 100 IRE units);
- half-amplitude duration: 250 ± 10 ns.

2.3 *Modulated 12.5T sine-squared pulse* (F) (see Note)

- position of peak: $51H/128$;
- amplitude: within ± 0.5 IRE unit of the amplitude of the luminance bar (B_2) (nominal value: 100 IRE units);
- half-amplitude duration: 1.57 ± 0.05 µs;
- inherent chrominance-luminance amplitude inequality: ⩽ 0.5%;

- inherent chrominance-luminance delay inequality: ⩽ 5 ns;
- other perturbations in the pulse base line: ⩽ 0.5 IRE unit;
- harmonic distortion component of the chrominance sub-carrier: at least 40 dB below the fundamental;
- chrominance sub-carrier is to be phase-locked to colour burst when this is present.

Note. – For monochrome transmissions, this signal is optional.

2.4 *Five-riser luminance staircase* (D_1) (see Note) *and superimposed five-riser staircase* (D_2)

Note. – Monochrome transmission only.

2.4.1 The five-riser luminance staircase (D_1) has the following characteristics:

- position of successive transitions: $68H/128$, $74H/128$, $80H/128$, $86H/128$, $92H/128$ and $100H/128$ (fall);
- peak-to-peak amplitude of the staircase: 100 ± 1 IRE units for the signal D_1 and 90 ± 1 IRE units for the signal D_2;
- nominal amplitude of risers: within ± 1% of 1/5 amplitude of peak-to-peak amplitude of the staircase (nominal value: 20 IRE units for the signal D_1 and 18 IRE units for the signal D_2);
- rise and fall times of transitions: shaped by a $2T$ sine-squared filter to restrict the amplitude of components of the luminance signal in the vicinity of the colour sub-carrier (nominal value: 250 ns).

2.4.2 The chrominance signal when superimposed on the staircase has the following characteristics:

- position of transitions: $60H/128$ and $98H/128$, duration of the chrominance signal $38H/128$;
- peak-to-peak amplitude of the envelope of the chrominance signal: 40 ± 0.4 IRE units;
- inherent differential-gain distortion: ⩽ 0.25% (average picture luminance (APL): 10 to 90%);
- inherent differential-gain distortion: ⩽ 0.2° (APL: 10 to 90%);
- rise and fall times of the envelope of the chrominance signal transitions: 400 ± 25 ns;
- phase difference between the chrominance signal and the mean phase (see Note) of the programme colour burst signal: 0 ± 1° (APL: 10 to 90%).

Note. – The term "mean phase" is particularly significant in the case of M/PAL.

3. Particulars of signals inserted in line 17 of field 2 (Figs. 6 or 8) (see Note 1 of § 1)

3.1 *Reference bar signal* (C_1)

- position of transitions: $0_H/128$ (O_{HR}) and $8H/128$; duration of bar: $8H/128$;
- bar amplitude: within ± 0.5 IRE unit of the amplitude of the luminance bar signal (B_2) (nominal value: 100 IRE units);
- rise and fall times of transitions: (integrated sine-squared shape): 125 ± 5 ns;
- overshoot and undershoot: ⩽ 1%;
- tilt: ⩽ 0.5%.

3.2 *Luminance pedestal*

- position of transitions: $8H/128$ and $100H/128$;
- amplitude: within ± 1% of one half of the amplitude of luminance bar (B_2) (nominal value: 50 IRE units).

3.3 *Multi-burst signal superimposed on the pedestal* (C_2)

- starting positions and frequencies of the bursts (see Table III);
- peak-to-peak amplitude of burst: 50 ± 0.5 IRE units;
- d.c. component of each burst: not to exceed 0.25 IRE unit;
- harmonics shall be at least 40 dB below the fundamental.

3.4 *Superimposed 3-level chrominance signal* (G) (see Note)

- position of transitions: $68H/128$, $76H/128$, $84H/128$ and $96H/128$;
- peak-to-peak amplitudes:
  1st section: 20 ± 0.2 IRE units,
  2nd section: 40 ± 0.4 IRE units,
  3rd section: 80 ± 0.4 IRE units;
- rise and fall times of the envelope of the chrominance signal transitions: 400 ± 25 ns;
- inherent phase/amplitude distortion: ⩽ 0.5°;
- inherent chrominance-luminance intermodulation: ⩽ 0.25 IRE unit;

- chrominance component:
  - in system M/PAL is to be phase-locked to programme colour burst, if present;
  - in system M/NTSC, is to lag the programme colour burst (if present) by 90° ± 1°.

Note. – For monochrome transmission, this signal is optional.

TABLE III

Burst No.	Precise starting position (1)	Frequency (MHz) (2)
1	$12H/128$	0·5
2	$24H/128$	1·0
3	$32H/128$	2·0
4	$40H/128$	3·0
5	$48H/128$	3·58
6	$56H/128$	4·2

(1) The starting point of each burst shall be at zero phase of the sine-wave, and each burst shall consist of the maximum number of complete cycles. The gaps between successive bursts shall not be shorter than 0.4 μs, nor longer than 2.0 μs in duration.

(2) Spectral components of the bursts may cause interference to sound sub-carriers or noise detection circuits and the out-of-band energy should be limited by suitable design techniques. For example, the envelopes of the bursts should have a rise time greater than 300 ns and the envelope should be approximately integrated sine-squared shape.

If harmonics of the burst cause interference, other frequencies near to the above-mentioned may be used, subject to agreement between the administrations concerned.

4. List of measurements which can be made with the defined insertion test signals (see Note 1 of § 1)

TABLE IV

Characteristics measured	Waveform used	Line number
Linear distortion		
Insertion gain	B_2	17/field 1
Amplitude/frequency response	B_2 (1) and C_2	17/field 1 and 2
Line-time waveform distortion	B_2	17/field 1
Short-time waveform distortion:		
— step response	B_2	17/field 1
— pulse response	B_1	17/field 1
Chrominance-luminance gain inequality	B_2 and F	17/field 1
Chrominance-luminance delay inequality	F	17/field 1
Non-linear distortion		
Line-time luminance non-linearity	D_1 (2)	17/field 1
Chrominance non-linearity	G	17/field 2
Luminance-chrominance intermodulation:		
— differential gain	D_2	17/field 1
— differential phase	D_2	17/field 1
Chrominance-luminance intermodulation	G	17/field 2

(1) C_1 (line 17/field 2) may be used in place of B_2, when line-time distortion is suitably small.

(2) D_2 may be used when the chrominance-luminance intermodulation is suitably small.

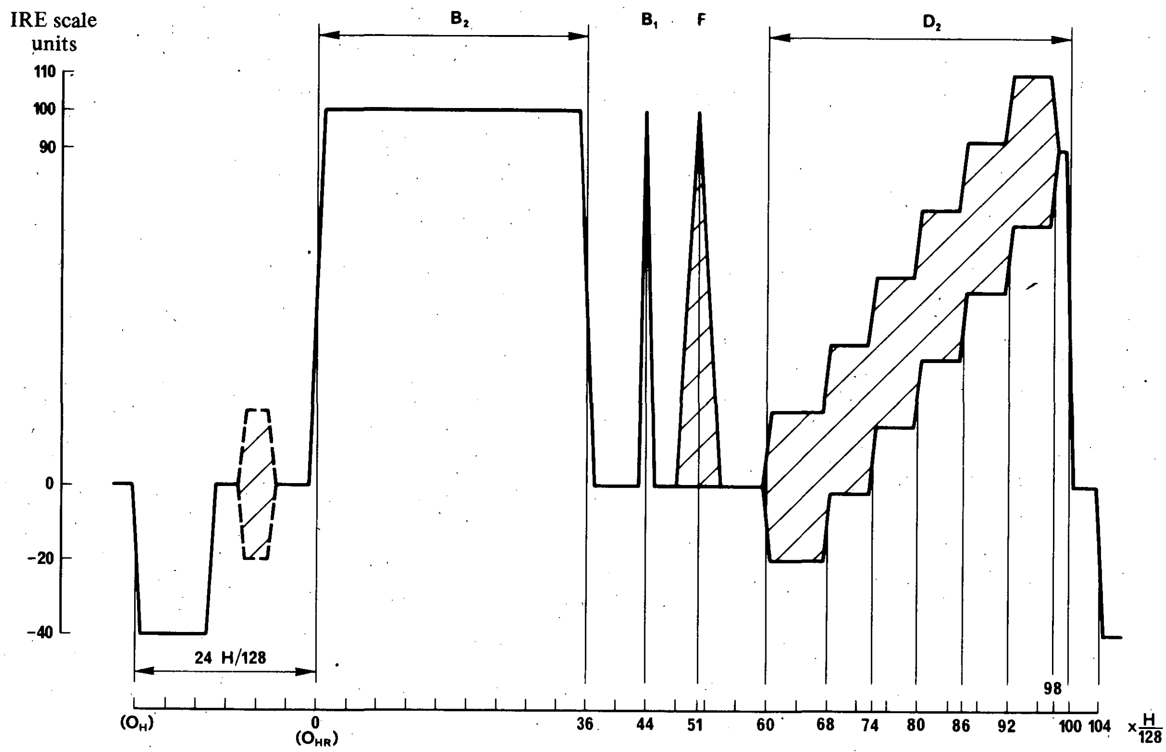

FIGURE 5 – *Line 17 of field 1*

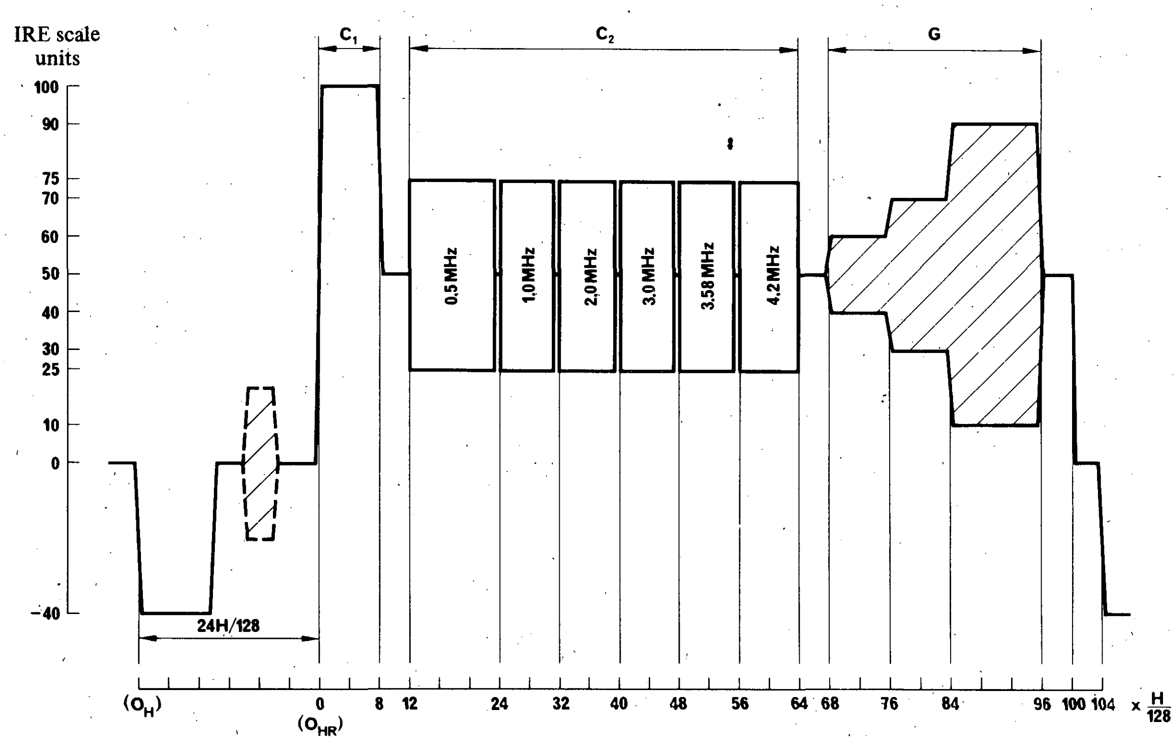

FIGURE 6 – *Line 17 of field 2*

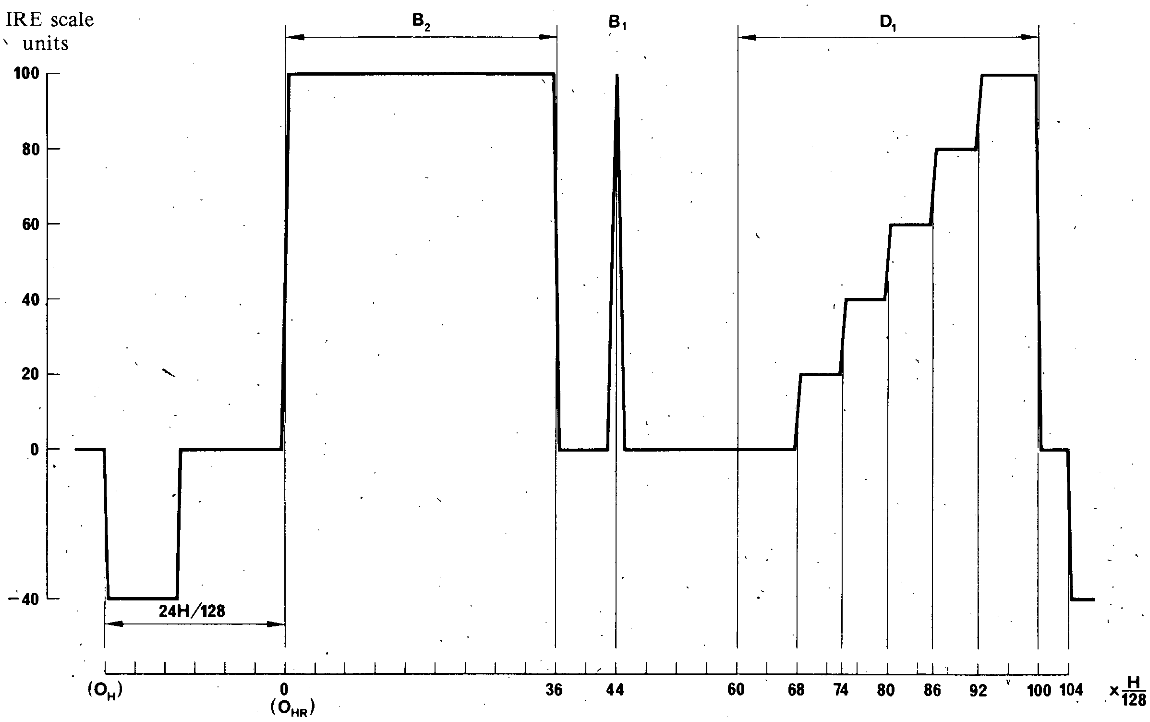

FIGURE 7 – *Line 17 of field 1*

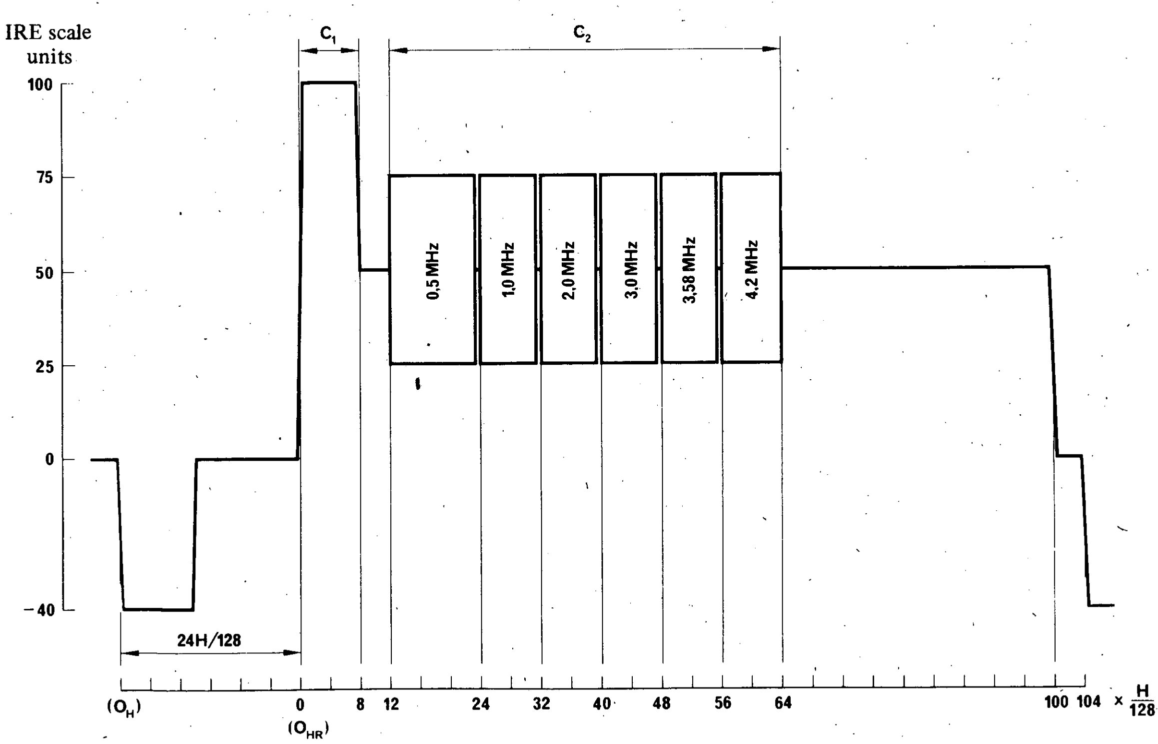

FIGURE 8 – *Line 17 of field 2*

Note. – Figs. 7 and 8 are examples of insertion test signals for monochrome transmission.

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[1982-86]: CMTT/6 (United States of America).

RECOMMENDATION 569-2

DEFINITIONS OF PARAMETERS FOR SIMPLIFIED AUTOMATIC MEASUREMENT OF TELEVISION INSERTION TEST SIGNALS

(Study Programme 15D/CMTT and Question 15/11)

(1978-1982-1986)

The CCIR,

CONSIDERING

(a) that Recommendation 567 is the basic reference which defines the parameters which are to be measured and the test signal elements and measuring methods which are to be used, in order to determine the performance of a television transmission circuit;

(b) that Reports 628 and 411 describe various techniques for automatic measurement and monitoring of the performance of television chains which make use of insertion test signals;

(c) that operational measurements are commonly carried out using insertion test signals which are defined in Recommendation 473;

(d) that Report 314 describes the allocation of lines in the field-blanking interval for special purposes;

(e) that, although automatic measuring equipment exists that can make measurements in accordance with Recommendation 567, simplified automatic measuring equipment also exists that requires modifications of the measurement methods and definitions;

(f) that such simplified automatic measurement of insertion test signals suits the requirements of operational staffs and makes the analysis of measurement results easier,

UNANIMOUSLY RECOMMENDS,

that, when simplified automatic measuring equipment is used to make measurements of an insertion test signal, and when a normalized form of presentation of the results is desired, the definitions used in quantifying the parameters of that signal should be those which are given in Annex I.

ANNEX I

1. Introduction

The need for each of the measurements described in this Recommendation (and possibly other measurements) will depend upon the type of plant in use and the policy of the administrations.

The test signals specified are those shown in Recommendation 473.

The definitions assume that the performance of the measuring equipment employed is such that any harmonic components of the incoming signal, occuring above the nominal video band, will not give rise to measurement errors which exceed the specified accuracy of that equipment. The definitions also assume instrumentation that will substantially eliminate the effects of any noise present on the incoming signal from the measurement of any test signal parameter.

The magnitude of distortions exhibited by signals which have passed through a non-linear transmission circuit tend to vary with average picture level. Therefore it may be desirable also, to measure automatically the value of Average Picture Level (APL) associated with any particular magnitude of distortion or error.

2. Definitions

2.1 *Luminance bar amplitude*

The luminance bar amplitude is defined as the difference between the level corresponding to the mid-point of the bar (element B_2) and the level corresponding to a point immediately following the composite pulse (element F). These points are shown as b_2 and b_1 respectively in Figs. 1 and 2. It is to be expressed as a percentage of the nominal bar amplitude (0.7 V for 625-line signals, 0.714 V for 525-line signals).

2.2 *Luminance bar amplitude error*

The luminance bar amplitude error is defined as the difference between the actual luminance bar amplitude and the nominal value expressed as a percentage of the nominal value (0.7 V for 625-line signals, 0.714 V for 525-line signals).

2.3 *Bar tilt*

The luminance bar tilt is defined as the difference between the level of the luminance bar one microsecond after the half amplitude point of its leading edge (point b_3 in Figs. 1 and 2), and the level one microsecond before the half amplitude point of its trailing edge, (point b_4 in Figs. 1 and 2) expressed as a percentage of the luminance bar amplitude. The sign of the difference is positive if b_4 is higher than b_3.

Note. – The parameter bar tilt as defined above is a unique measurement by automatic devices of a specific form of line time waveform distortion, i.e. the difference in the level of the line bar at two specific reference points. This measurement is different to the measurements of line time waveform distortion described in Recommendation 567 (§ C.3.5.1.3 and Annex III to Part C, § 2.1) where the maximum difference in level at any point between defined reference points is measured.

2.4 *Base-line distortion*

The base-line distortion is defined as the difference between the levels of the signal at point b_7, which is located after the mid-amplitude point of the trailing edge of the bar (element B_2) at a distance of 400 ns for 625-line systems and 500 ns for 525-line systems (see Figs. 1 and 2), and at a reference point b_1 located before the beginning of the staircase in line 17 (see also Figs. 1 and 2).

The base-line distortion is expressed as a percentage of the luminance bar amplitude. It is to be measured after the bandwidth of the signal has been limited (see Note). The sign of the difference is positive if the signal level at point b_7 is higher than the level of reference point b_1.

Note. – Limitation may be achieved by the use of a network, the design of which is based on "Solution 3" [Thomson, 1952], having its first zero at 3.3 MHz, or by an equivalent technique.

2.5 *2T pulse/bar ratio error*

The $2T$ sine-squared pulse/bar ratio error is defined as the difference between the amplitudes of the $2T$ pulse (element B_1) and the luminance bar (element B_2), expressed as a percentage of the luminance bar amplitude. The peak amplitude of the $2T$ pulse is referred to a reference point b_1 (see Note) (Figs. 1 and 2) before the first riser of the staircase. The sign of the difference is positive if the $2T$ pulse amplitude is greater than the luminance bar amplitude.

Note. – To avoid error due to line tilt, it may be preferable to use a reference point exclusively for the measurement of $2T$ pulse/bar ratio error, which is defined to be the linear mean level of the insertion test signal during the periods: 2 to 1 μs before, and 1 to 2 μs after the $2T$ pulse.

2.6 *2T pulse shape distortion*

This definition requires further study.

2.7 *Chrominance-luminance gain inequality*

The chrominance-luminance gain inequality is defined as the difference between the peak-to-peak amplitude of the chrominance component of the element G, G_1, G_2 and the amplitude of the luminance bar (element B_2) expressed as a percentage of the luminance bar amplitude. The sign of the difference is positive if the amplitude of the chrominance component is greater than that of the luminance bar. Note that in the 525-line case the nominal amplitude of element G is 80 IRE units. This factor must be taken into account when normalizing results.

If for any reason signal elements G, G_1 or G_2 are not available, the measurement can be made with the chrominance component of element F.

2.8 *Chrominance-luminance delay inequality*

The chrominance-luminance delay inequality is defined as the time difference (expressed in ns) between the luminance and the chrominance component of the composite pulse (element F). This difference is positive, if the symmetry axis of the demodulated chrominance component lags behind the symmetry axis of the luminance component.

2.9 *Luminance non-linearity*

The luminance non-linearity is to be measured with the staircase signal in line 17 (element D_1 for 625-lines, D_2 for 525-lines). It is defined as the difference between the largest and the smallest step amplitudes, expressed as a percentage of the amplitude of the largest step. As the sign of the difference is not significant it is taken to be positive.

2.10 *Differential gain*

Differential gain is determined by evaluating the amplitude modulation of the colour sub-carrier superimposed on the staircase in element D_2.

Recommendation 567 defines differential gain in terms of two parameters $+x$ and $-y$ which represent the maximum (peak) differences in amplitude between the sub-carrier on the treads of the received test signal and the sub-carrier on its blanking level, expressed as a percentage of the latter. In the case of a monotonic characteristic, either x or y will be zero.

x and y can be found from the expressions below:

$$x = 100\left|\frac{A_{max}}{A_0} - 1\right| \qquad y = 100\left|\frac{A_{min}}{A_0} - 1\right|$$

where:

A_0 : amplitude of the received sub-carrier on the blanking level tread of element D_2.

A_{max} : highest value of sub-carrier on any tread.

A_{min} : lowest value of sub-carrier on any tread.

Two alternative methods of expressing the results are acceptable for automatic measurement. These are:

(a) "peak differential gain", which is defined by either $+x$ or $-y$, depending upon which of these parameters has the larger magnitude.

(b) "peak-to-peak differential gain", which is defined as $x + y$.

Note. – For the measurement of peak-to-peak differential gain some administrations use A_{max} rather than A_0. The formula used then is:

$$x + y = 100\left|\frac{A_{max} - A_{min}}{A_{max}}\right|$$

Results obtained by this method will differ only slightly from those defined above if the magnitude of the distortion is not excessive.

2.11 *Differential phase*

Differential phase is determined by evaluating the phase modulation of the colour sub-carrier superimposed on the staircase in element D_2: (Fig. 5: 625-lines; Fig. 2: 525-lines).

Recommendation 567 defines differential phase in terms of two parameters $+x$ and $-y$ which represent the maximum (peak) differences in phase between the sub-carrier on the treads of the received test signal and the sub-carrier on its blanking level, expressed in degrees difference from the latter. In the case of a monotonic characteristic either x or y will be zero.

x and y can be found from the expressions below:

$$x = |\Phi_{max} - \Phi_0| \qquad y = |\Phi_{min} - \Phi_0|$$

where:

Φ_0 : phase of sub-carrier on the blanking level tread of element D_2.

Φ_{max} : highest value of sub-carrier phase on any tread.

Φ_{min} : lowest value of sub-carrier phase on any tread.

Two alternative methods of expressing the results are acceptable for automatic measurement. These are:

(a) "peak differential phase", which is defined by either $+x$ or $-y$ depending upon which of these parameters has the larger magnitude.

(b) "peak-to-peak differential phase", which is defined as $x + y$.

2.12 *Chrominance-luminance intermodulation*

The chrominance-luminance intermodulation is measured on element G, G_1 or G_2, after suppressing the incoming colour sub-carrier. It is defined as the difference between the luminance amplitude in element G_1, or in the last section of element G or G_2 (b_5 in Figs. 3 and 4) and the amplitude of the succeeding section (b_6 in Figs. 3 and 4) in which the test signal has no sub-carrier, expressed as a percentage of the amplitude of the luminance bar (element B_2). The sign of the difference is positive if the luminance amplitude b_5 is greater than the luminance amplitude of the succeeding section b_6.

Note. – Some administrations use element F instead of G, G_1 or G_2 for measurement of this parameter. In this case measurement of the amplitude of the luminance component of the composite pulse (element F) is made after suppressing the incoming colour sub-carrier. The result will be given by the difference between the composite pulse luminance amplitude and half the luminance bar amplitude, expressed as a percentage of the luminance bar amplitude. The sign of the difference is positive if the amplitude of the composite pulse component is greater than half the luminance bar amplitude. In some cases the result may differ from that given by the preferred method, since the signal element F is not so well suited as element G to the measurement of this distortion.

2.13 *Two-level chrominance amplitude non-linearity*

This parameter is to be measured with element G or G_2. Its value, expressed in per cent, and with a sign, is defined by:

$$\frac{(V_3 - 5V_1)}{V_3} \times 100 \text{ for 625-line signals}$$

$$\frac{(V_3 - 4V_1)}{V_3} \times 100 \text{ for 525-line signals}$$

where V_1 and V_3 are respectively the peak-to-peak amplitudes of the first and last sections of element G or G_2.

2.14 *Two-level chrominance phase non-linearity*

This parameter is to be measured with element G or G_2. Its value, expressed in degrees, and with a sign, is defined by:

$$\Phi_3 - \Phi_1$$

where Φ_3 and Φ_1 are respectively the phases of the last and first sections of element G or G_2.

2.15 *Signal-to-random noise ratio*

2.15.1 *Signal-to-unweighted random noise ratio*

The signal-to-unweighted random noise ratio is defined as the ratio of the amplitude of the luminance bar (element B_2) to the r.m.s. value of the noise measured on a specified line, or part of this line, (line 22, or optionally both lines 22 and 335, in the case of 625-line signals). It is to be given in dB. The noise bandwidth is assumed to be limited by the low pass filter defined in Recommendation 567 Annex II to Part C. Lower frequency limiting shall be done by a 200 kHz high pass filter with a slope of 20 dB per decade (see Note).

To suppress any periodic noise at sub-carrier frequency, a notch filter should be used (see Note).

For 625-line signals, the amplitude/frequency response of the filter should be as in Fig. 8 and a possible implementation of the filter as a constant impedance network is given in [CCIR, 1978-82a].

Note. – The upper limit of the noise bandwidth is selected so as to eliminate noise which occurs outside of the wanted band of the video signal. The high pass filter and the notch filter are used to minimize the effects of periodic noise at low frequencies and at the sub-carrier frequency, respectively. The high pass filter has also been specified to minimize the measurement errors caused by residual wave-form distortion in the measurement period.

Attention is directed to the fact that the high pass filter and the notch filter modify the spectral composition of the random noise and therefore alter its r.m.s. or quasi peak-to-peak value. The conversion factors in dB established for noise with a spectrum ideally limited to 5 MHz are given in Table I (see also [CCIR, 1978-82b]).

2.15.2 *Signal-to-weighted random noise ratio*

The signal-to-weighted random noise ratio is defined as in § 2.15.1 above, save for the addition of the unified weighting network specified by the CCIR in Recommendation 567.

2.15.3 *Signal-to-quasi peak-to-peak noise ratio*

The signal-to-quasi peak-to-peak noise ratio is defined as the ratio of the amplitude of the luminance bar (element B_2) to the value exceeded by the noise voltage deviation for a specified measurement time percentage (see Notes 1 and 2). It may be measured both under weighted or unweighted conditions. The comparison between these parameters and that defined in § 2.15.1 and 2.15.2 is intended to confirm the Gaussian nature of the noise. They are to be given in dB.

Note 1. – The upper limit of the noise bandwidth is selected so as to eliminate noise which occurs outside of the wanted band of the video signal. The high pass filter and the notch filter are used to minimize the effects of periodic noise at low frequencies and at the sub-carrier frequency, respectively. The high pass filter has also been specified to minimize the measurement errors caused by residual wave-form distortion in the measurement period.

Attention is directed to the fact that the high pass filter and the notch filter modify the spectral composition of the random noise and therefore alter its r.m.s. or quasi peak-to-peak value. The conversion factors in dB established for noise with a spectrum ideally limited to 5 MHz are given in Table I (see also [CCIR, 1978-82b]).

Note 2. – Further study is required to specify this percentage.

2.16 *Signal-to-chrominance periodic noise ratio*

This parameter is to be measured on the part of the signal used in § 2.15 above. It is defined as the ratio of the amplitude of the luminance bar (element B_2) to the peak-to-peak amplitude of spurious signals in a total 3 dB bandwidth of 0.2 MHz centred on the appropriate colour sub-carrier frequency as in § 2.15 above. The result of the measurement is to be given in dB.

2.17 *Low-frequency errors*

This parameter is defined as the peak-to-peak amplitude of the fluctuations of the blanking level, measured in a frequency band from 10 Hz to 2 kHz, and expressed as a percentage of the amplitude of the luminance bar (element B_2). Further information is given in [CCIR, 1974-78].

2.18 *Sync. amplitude error*

Sync. amplitude error is defined as the difference between sync. amplitude and its normalized value (i.e. 3/7 luminance bar amplitude for 625-lines, 4/10 luminance bar amplitude for 525-lines) (see Note 1) expressed as a percentage of the normalized value. The sign of the difference is positive if sync. pulses are larger than the normalized value.

To provide a measurement result in the presence of sound-in-syncs signals, sync. amplitude must be measured at the mid-point of the last broad pulse of each field (point b_8 in Fig. 6) (see Note 2).

Note 1. – The luminance bar amplitude is defined in § 2.1.

Note 2. – To avoid error due to field tilt it may be preferable to use a reference point exclusively for the measurement of sync. amplitude error which is placed at point b_9 in Fig. 6 of each field.

2.19 *Chrominance reference amplitude error*

This parameter relates to the variation in amplitude of the colour sub-carrier occuring in the region of blanking level. It is defined as the difference between the peak-to-peak amplitude of the colour sub-carrier on the blanking level tread of element D_2 and its normalized value, (i.e. 4/10 luminance bar amplitude) (see § 2.18, Note 1), expressed as a percentage. The sign of the difference is positive if the amplitude of the colour sub-carrier on the blanking level tread is larger than the normalized value.

2.20 *Gain/frequency characteristic*

2.20.1 *Peak ripple of the multi-burst signal*

This quantity is defined on the basis of two numbers x and y, which represent the maximum (peak) differences between the amplitudes of the bursts of the test signal C (see Note 1) and a reference quantity A_0, the two numbers x and y being expressed as a percentage of A_0.

For 625-line signals, A_0 is the peak-to-peak amplitude of element C_1 (see Fig. 7).

For 525-line signals, A_0 is equal to half the amplitude of the luminance bar, as defined in § 2.1 above (see Note 2).

x and y can be found from the following expressions:

$$x = 100 \left| \frac{A_{max}}{A_0} - 1 \right| \qquad y = 100 \left| \frac{A_{min}}{A_0} - 1 \right|$$

where A_{max} and A_{min} are respectively the highest and the lowest value of the peak-to-peak amplitude of the relevant bursts (see Note 3) measured at their half duration point.

The peak ripple of the multi-burst signal is defined by either $+x$ or $-y$ depending upon which of these parameters has the larger magnitude.

Note 1. – For 625-line signals, the last burst (having a frequency of 5.8 MHz), is not taken into account in this measurement.

Note 2. – Further study is required to check if, as an alternative, A_0 may also be derived from test element C_1.

Note 3. – For 625-line signals, the last burst (having a frequency of 5.8 MHz), is not taken into account in this measurement.

2.20.2 *Amplitude error of the burst at n MHz* (see Note)

This quantity is defined as the difference in terms of magnitude and sign between the peak-to-peak amplitude of the burst at n MHz and the reference quantity A_0 (defined as above), expressed as a percentage of A_0.

Note. – n is the designation of the frequency of the burst taken into account. Note 1 of § 2.20.1 also applies here.

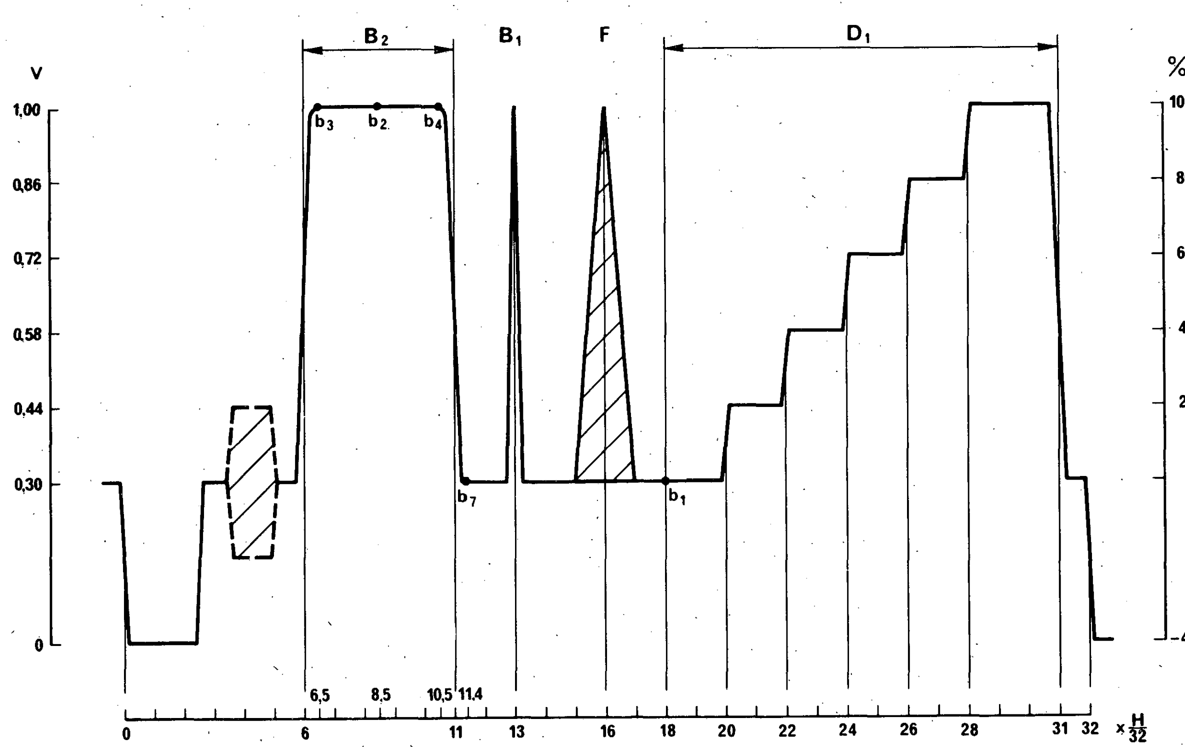

FIGURE 1 – *Line 17 for 625-line systems*

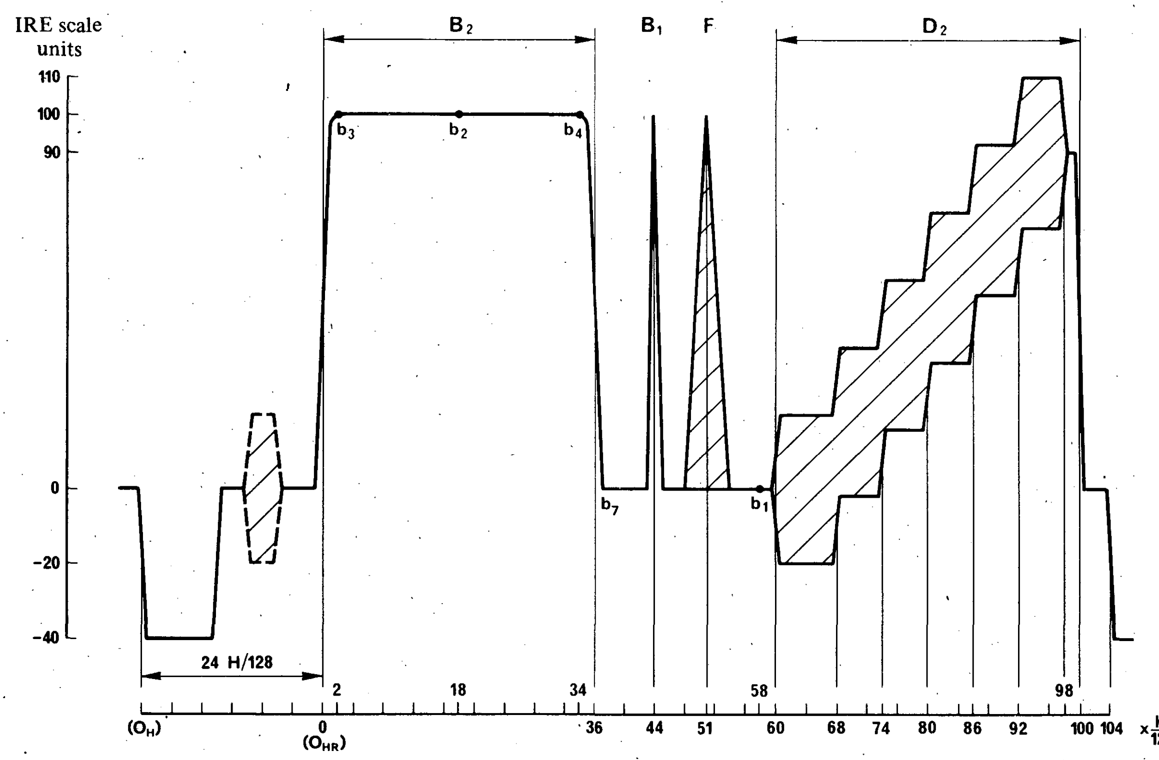

FIGURE 2 – *Line 17 of field 1 for 525-line systems*

Note. – The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

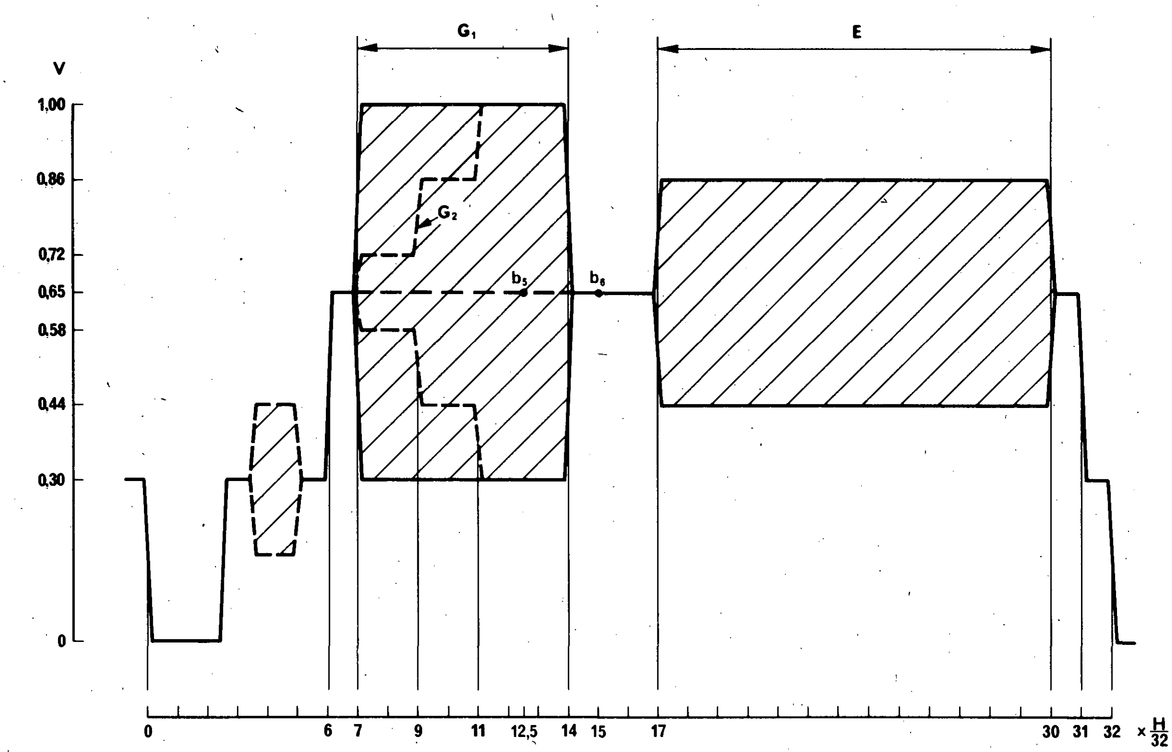

FIGURE 3 – *Line 331 for 625-line systems*

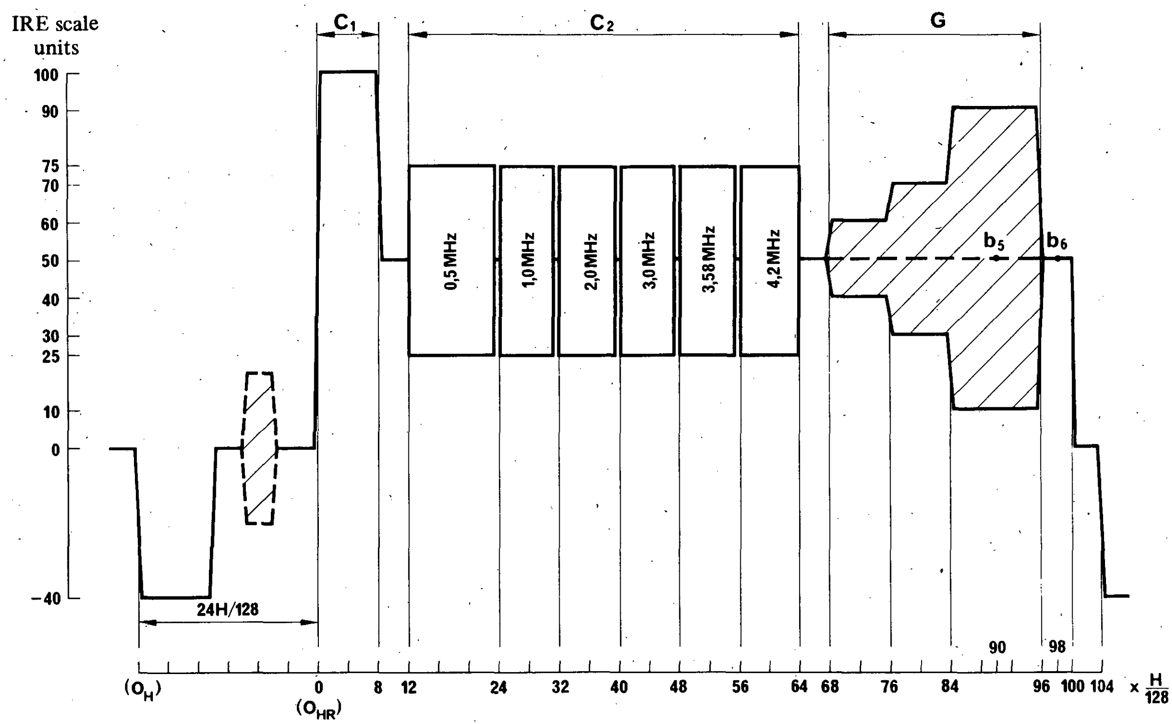

FIGURE 4 – *Line 17 of field 2 for 525-line systems*

Note. – The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

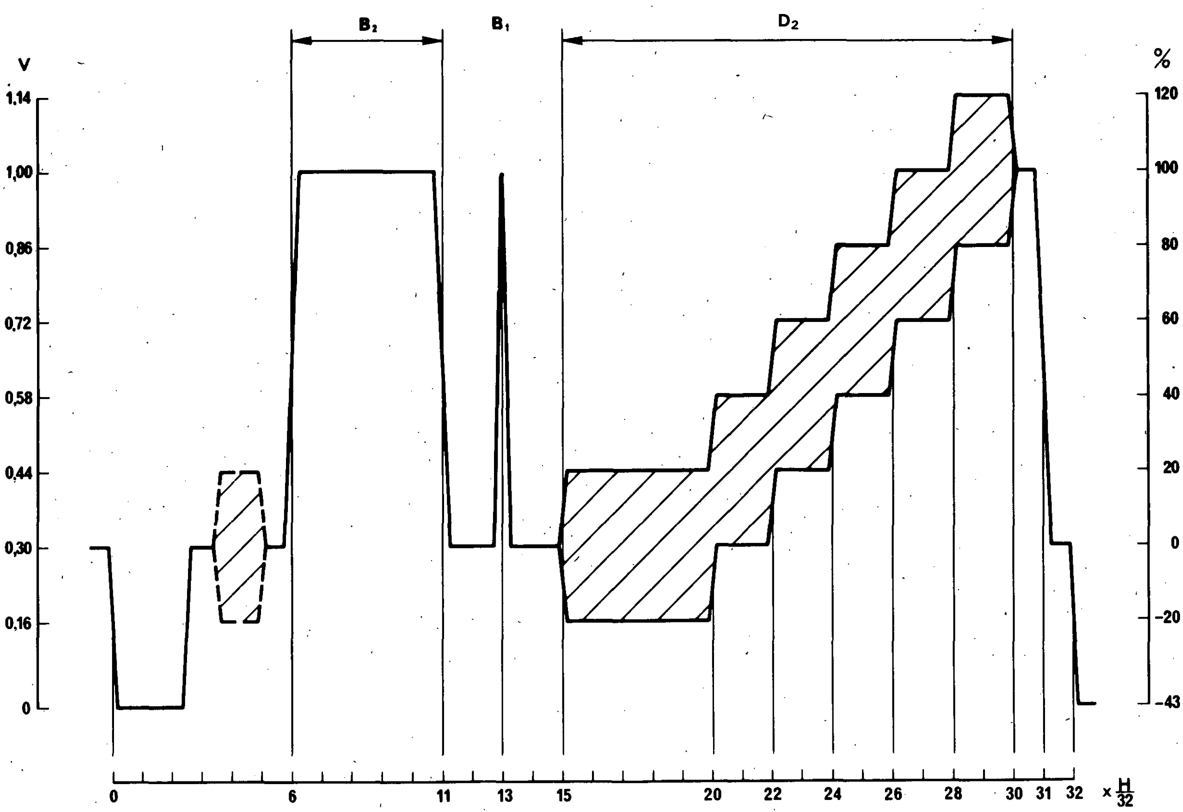

FIGURE 5 – *Line 330 for 625-line systems*

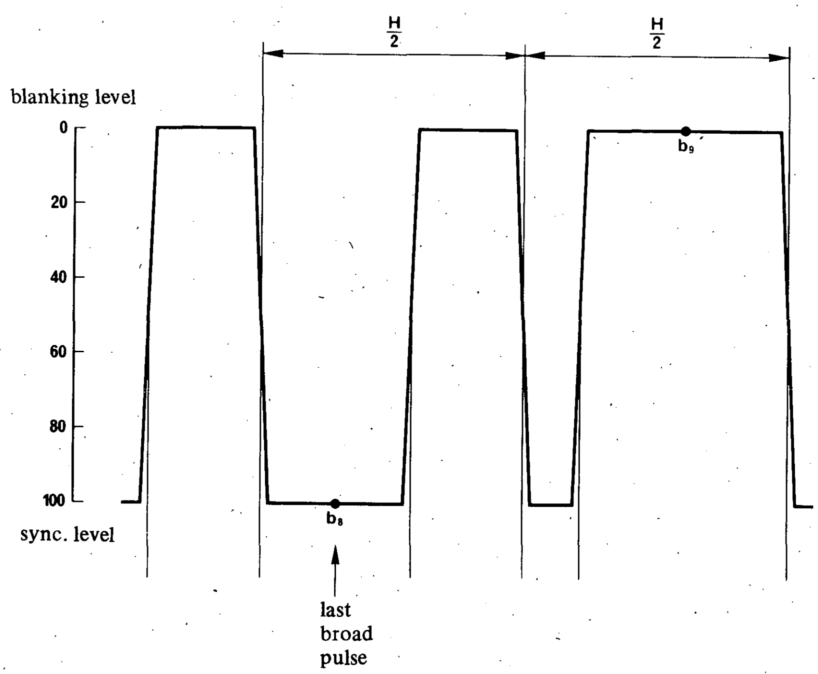

FIGURE 6 – *Measurement points for sync. amplitude error during equalizing pulse period*

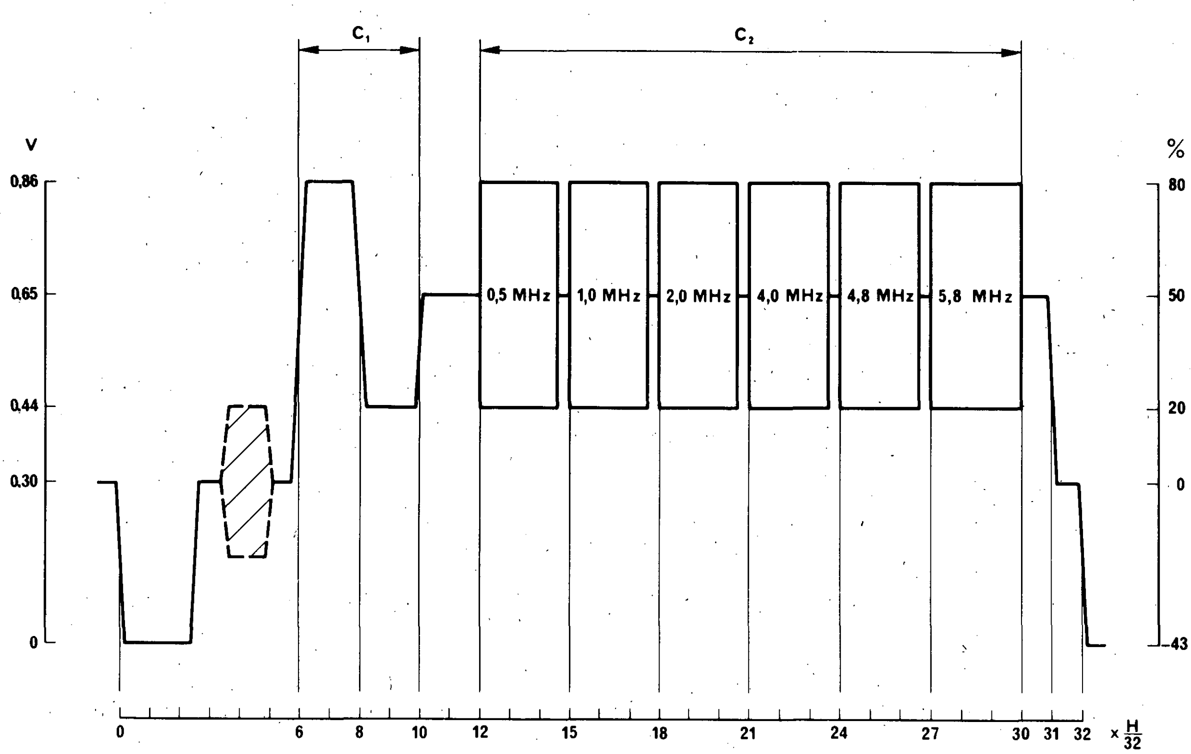

FIGURE 7 – *Line 18 for 625-line systems*

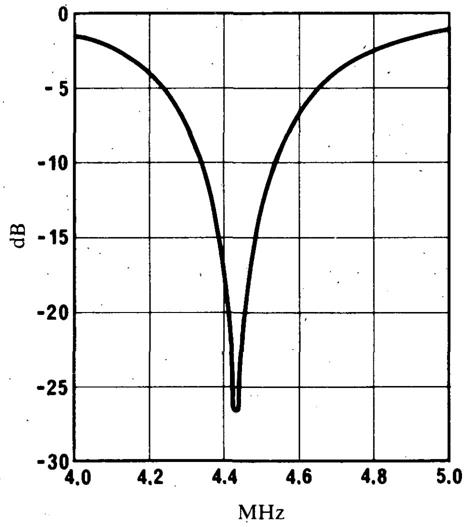

FIGURE 8 – *Response of sub-carrier notch filter for 625-line systems*

3 dB bandwidth: 600 kHz
Attenuation at 4.43 MHz ⩾26 dB

TABLE I – *Theoretical values of conversion factors in dB*
(rounded off to a tenth of dB)

		200 kHz high pass filter 20 dB/decade	Sub-carrier notch filter 3 dB bandwidth = 600 kHz
White noise	unweighted	0.3	0.7
	weighted	1.3	0.2
Triangular noise	unweighted	0.0	1.8
	weighted	0.0	1.3
De-emphasized triangular noise	unweighted	0.0	1.6
	weighted	0.0	0.9

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RECOMMENDATION 570*

STANDARD TEST SIGNAL FOR CONVENTIONAL LOADING OF A TELEVISION CHANNEL

(Study Programme 15E/CMTT)

(1978)

The CCIR,

CONSIDERING

(a) that a common transmission path may be used by one or more television signals and one or more sound-programme channels or telephony channels;

(b) that, due to distortion in the common path, the signals applied to one or more television channels may result in unwanted signals appearing in other channels;

(c) that Report 375 draws attention to the need, where a sound-programme circuit is carried on a radio-relay system for television, for measurements or calculations to be made with the television channel loaded with a standard test signal;

(d) that a similar problem exists for cable systems such as that described in CCITT Recommendation J.73 or that being studied in CCITT Question 20/XV (1976-80);

(e) that a special test signal is required for use as a conventional load for a television channel when measuring, calculating or specifying the noise in other circuits which are carried in a common path with the television channel or channels;

(f) that such a standard test signal for conventional loading of a television channel should preferably be one which is already in general use or which can readily be made available to users and manufacturers of the transmission systems concerned;

(g) that a standard test signal for conventional loading of a television channel should preferably be typical of a wide range of video signals likely to be used in practice, and should include chrominance information and a field-frequency component,

UNANIMOUSLY RECOMMENDS

1. that a composite colour-bar test signal, appropriate to the colour television standard to be carried, should be used as the standard test signal for conventional loading of an analogue colour television channel;

2. that, using the nomenclature of Recommendation 471, the luminance and chrominance amplitudes of the colour bar test signal when used for conventional loading should be 100/0/75/0 for 625/50 standards;

3. that countries which operate on 525/60 standards should employ a colour bar signal of the split field type for conventional loading. The colour bars occupy the larger portion of the active lines and have the characteristic 75/7.5/75/7.5.

Accompanying the colour bars, on a split-field basis is a peak-white bar on all active lines not carrying the colour bars. For example, Electronic Industries Association (USA) RS-189-A colour bars;

4. that the colour bars should be arranged in order of descending luminance as commonly used.

Note 1. – The choice of standard test signals for conventional loading of analogue television channels which are used only for monochrome transmission, and digital channels for monochrome and colour television transmission, should be the subject of further study.

Note 2. – The colour bar test signal at an appropriate saturation is representative of the circuit load caused by the studio outputs. It is not typical, however, of some test signals, electronically-generated captions, or some data and other special signals, of which the disturbing effect on other channels on the same bearer path may be greater than for camera-generated signals. The disturbing effect due to the spectral components of these special signals should be tested by means of the signals themselves.

Note 3. – Administrations making measurements of the disturbing effect of a television signal on other signals which share the same bearer path, should take into account the relative stability of the carrier frequencies of the disturbing and disturbed circuits on the common bearer channel.

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* This Recommendation should be brought to the attention of Study Group XV/CCITT, as well as Study Groups 4 and 9 of the CCIR.

RECOMMENDATION 720*

MEASUREMENT METHODS AND TEST PROCEDURES FOR TELETEXT SIGNALS

(Question 21/CMTT and Study Programme 21C/CMTT)

(1990)

The CCIR,

CONSIDERING

(a) that Report 956, Appendix I to Part 1, provides conceptual definitions to proposed data signal parameters;

(b) that Report 969 is intended to identify measurement methods and test procedures for checking the impairment of special signals resulting from transmission over television circuits;

(c) that operational measurements on teletext signals need no special test signals, because they can be made on the normal teletext lines;

(d) that automatic measurement of teletext signals suits the requirements of operational staff and makes the analysis of results easier;

(e) that the definitions given in this Recommendation may be applicable to other data services,

UNANIMOUSLY RECOMMENDS,

that, when measuring equipment is used to make measurements on teletext signals, the definitions used in quantifying the parameters should be those given in Annex I.

ANNEX I

1. Introduction

The need for each of the measurements described in this Recommendation (and possibly other measurements) will depend on the type of plant in use and the policy of the administrations.

No special test signals are necessary to meet the requirements of this Recommendation.

The definitions of parameters were specifically designed to meet the requirements of automatic measuring equipment. They are also suitable for manual measurements.

To reduce the influence of non-linear distortions the signal shall be bandlimited prior to measurement to a frequency between the upper limit of the television system and the teletext clock frequency.

Due to the random nature of the teletext signals, the results will exhibit some fluctuation between successive measurements.

2. Definition of terms

This section defines terms which are used in § 3 to define the measurement parameters.

2.1 *Mean value of clock run-in*

The mean value of clock run-in is defined as the mean level of the clock run-in waveform excluding the first two bits.

2.2 *All-zeros level*

The all-zeros level is the level resulting from a continuous stream of "zero" pulses. For measuring purposes the all-zeros level is defined as the mean level of the back porch within the nominal duration of the colour burst.

* This Recommendation should be brought to the attention of the IEC and Study Group 11.

2.3 *All-ones level*

The all-ones level is the level resulting from a continuous stream of "one" pulses. For measuring purposes the all-ones level is defined as twice the mean value of clock run-in minus the all-zeros level.

2.4 *Basic amplitude*

The basic amplitude is the difference between the all-ones level and the all-zeros level.

2.5 *Nominal teletext signal amplitude*

The nominal teletext signal amplitude is defined as a fixed percentage of the luminance bar amplitude and represents the ideal binary "1" amplitude in any teletext system (Fig. 1). If no luminance bar signal is present, the nominal value of the luminance bar signal is used.

Note – The luminance bar amplitude is defined in Recommendation 569. The relationship of the nominal teletext signal amplitude to the luminance bar amplitude is defined in Recommendation 653.

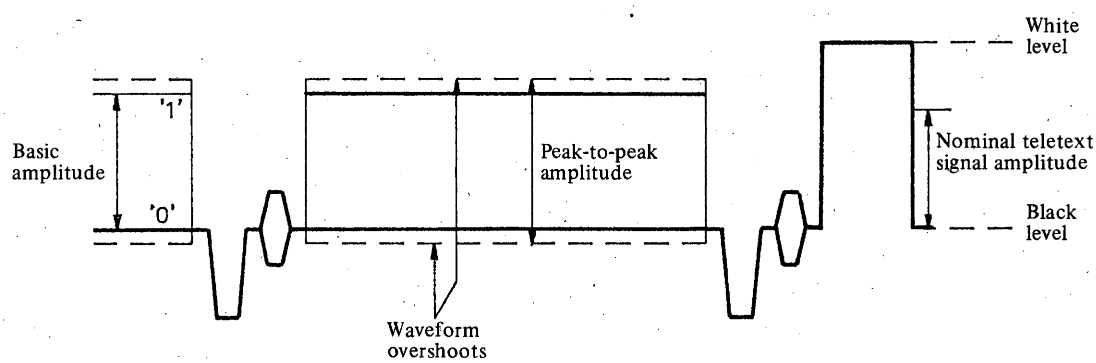

FIGURE 1 – *Teletext parameters*

2.6 *Timing reference*

The timing reference for each line is a uniform sequence of timing instants whose timing is derived only from the clock run-in of that line excluding the first two bits.

The timing of these instants is such that they coincide with the average timing of the points where the clock run-in crosses the mean value defined in § 2.5.

2.7 *Sampling instants for decoding margin*

The sampling instants for decoding margin are half way between the timing instants defined in § 2.6.

3. Definition of parameters

3.1 *Basic amplitude error*

This parameter is defined as the difference between the basic amplitude and the nominal teletext signal amplitude expressed as a percentage of the latter. In terms of abbreviations in Fig. 1 the basic amplitude error is:

$$\frac{E - D}{D} \times 100\%$$

3.2 *Peak-to-peak amplitude*

The peak-to-peak amplitude is defined as the sum of basic amplitude zero overshoots and ones overshoots. It is expressed as a percentage of the basic amplitude (see Fig. 1).

3.3 *Decoding margin*

The decoding margin is defined as the difference between the highest "0" bit level and the lowest "1" bit level measured at the sampling instants for a bit error ratio of 10^{-3}. The difference is expressed as percentage of the basic amplitude.

3.4 *Number of run-in bits*

This parameter counts the number of the "1" and "0" run-in bits present at the start of the teletext waveform prior to the framing code. The result will be always an even number because a "0" bit follows every "1" run-in bit. The counting starts with the first bit with amplitude exceeding the mean value of the clock run-in.

3.5 *Data timing*

In teletext System B the data timing is defined as the time difference between the peak of the penultimate "1" run-in bit and the line time datum (see Report 624). In teletex System A the data timing is defined as the time difference between the leading edge of the data signal and the line time datum (see Fig. 2).

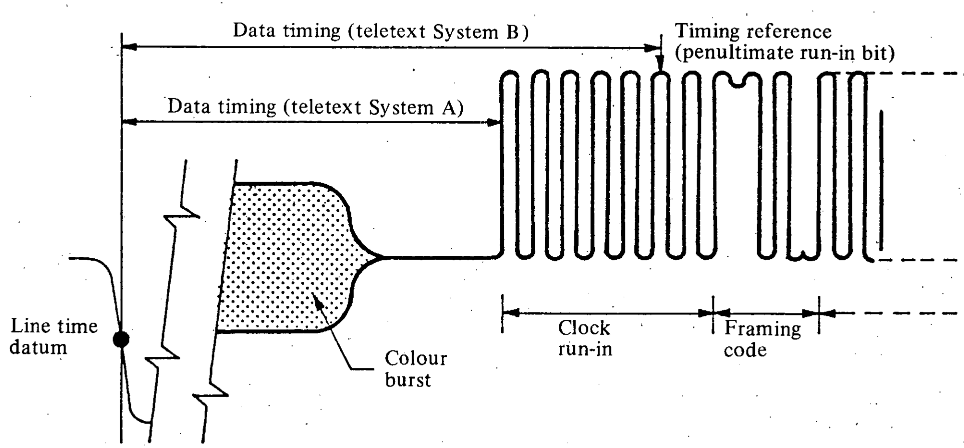

FIGURE 2 – *Data timing*

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[1986-90]: CMTT/34, CMTT/207 (Germany (Federal Republic of)).

SECTION CMTT C: TRANSMISSION STANDARDS AND PERFORMANCE OBJECTIVES FOR SOUND-PROGRAMME CHANNELS

RECOMMENDATION 502-2 *

HYPOTHETICAL REFERENCE CIRCUITS FOR SOUND-PROGRAMME TRANSMISSIONS **

Terrestrial systems and systems in the fixed-satellite service

(Question 17/CMTT)

(1974–1978–1982)

The CCIR,

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 Fig. 1), 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 Fig. 2) 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.

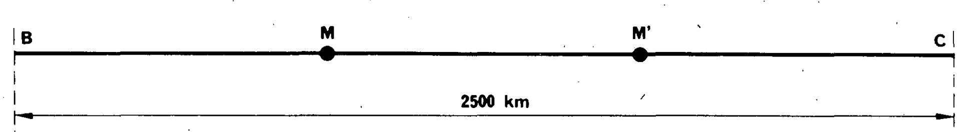

FIGURE 1 – *The hypothetical reference circuit for sound programme transmissions over a terrestrial system*

* This Recommendation should be brought to the attention of CCITT Study Group XVIII.

** – The hypothetical reference circuits defined in this Recommendation should apply for both analogue and digital systems.
– For maintenance purposes there may be a need to define other circuits of which an illustration is shown in Annex I of this Recommendation.

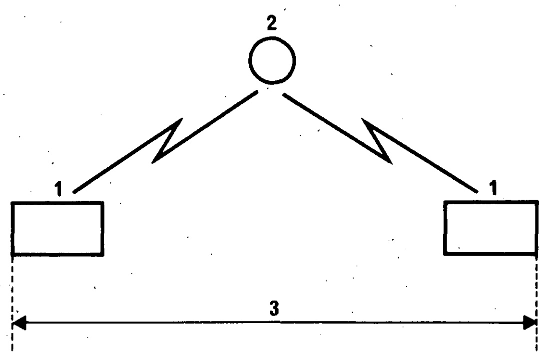

FIGURE 2 – *Hypothetical reference circuit for sound-programme transmissions over a system in the fixed-satellite service*

1: Earth station

2: Space station

3: Hypothetical reference circuit

ANNEX I

ILLUSTRATION OF AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

Figure 3 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.

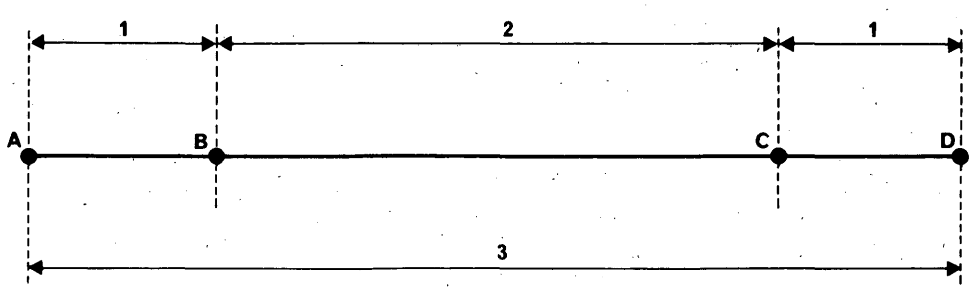

FIGURE 3 – *An international sound-programme connection*

1: Local sound-programme circuits

2: International sound-programme circuit

3: International sound-programme connection

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 Volume IV, Fascicle 3 of the CCITT Red Book. 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.

RECOMMENDATION 503-4*

PERFORMANCE CHARACTERISTICS OF 7 kHz-TYPE (NARROW-BANDWIDTH) SOUND-PROGRAMME CIRCUITS**

Circuits for medium quality monophonic transmission

(Question 17/CMTT and Study Programme 17G/CMTT)

(1974-1978-1982-1986-1990)

The CCIR,

CONSIDERING

(a) that it is necessary to set transmission standards for sound-programme circuits;

(b) that quality requirements for the hypothethical 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,

UNANIMOUSLY RECOMMENDS

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out below.

1. Application

This Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation 502.

For estimation of the performance of circuits shorter or longer than the HRC, see Recommendation 605.

Note 1 – For all-digital circuits, a separate Recommendation might be envisaged after further study.

Note 2 – For further information, 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***
System output impedance, provisionally	Low, balanced

* This Recommendation is also proposed by the CMTT as a revision of CCITT Recommendation J.23 and should be brought to the attention of CCITT Study Group XV.

** a) For the definition of absolute power, relative power and noise levels, see Recommendation 574.

b) For old 5 kHz, 6.4 kHz and 10-kHz-type systems still in use, see Recommendation 503-1 (Kyoto, 1978) and Recommendation 504-2 (Geneva, 1982) respectively.

*** The tolerance, permitted reactance and degree of unbalance need further study.

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.

2.3 *Levels*

Input maximum programme level	+ 9 dBm0s
Insertion gain (1 kHz at − 12 dBm0s)	0 dB
Adjustment error, within	± 0.5 dB
Variation over 24 hours not to exceed	± 0.5 dB
Relative level (see CCITT Recommendation J.14)	+ 6 dBrs

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization 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

TABLE I

Frequency (kHz)	Response (dB)
$0.05 \leq f < 0.1$	+1 to −3
$0.1 \leq f \leq 6.4$	+1 to −1
$6.4 < f \leq 7$	+1 to −3

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

3.1.2 *Group delay variation*

Differences between the value of group delay at the following frequencies and the minimum value are:

kHz	$\Delta\tau$ (ms)
0.05	80
0.1	20
6.4	5
7	10

Between the points defined above, the tolerance limit varies linearly on a linear-delay/logarithmic-frequency diagram.

3.1.3 *Noise*

The measurement to be made with an instrument conforming to Recommendation 468.

For radio-relay systems, the requirements 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 II

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
Idle channel noise, maximum (dBq0ps)	−44	−49
Programme-modulated noise, maximum (dBq0ps)	−32	−37

Programme-modulated noise usually occurs on sound programme circuits which are equipped with compandors (e.g. types of circuits corresponding to CCITT 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.

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.

Note – For further information on digital systems see Report 647.

3.1.4 *Single tone interference*

Level of any individual tone:

$$\leqslant -(73 + \psi) \qquad \text{dBm0s}$$

where ψ is the weighting filter response factor (positive or negative) as given in 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, it is recommended that the stop filters should have a 3 dB bandwidth of less than 3%, referred to the mid-frequency. The use of stop filters which affect 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 harmonic distortion 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 CCITT Recommendations N.21 and N.23.

The total harmonic distortion (THD) when measured with a true r.m.s. meter shall not be greater than the following values:

TABLE III

Input frequency (kHz)	THD
$0.05 \leqslant f < 0.1$	2% (−25 dBm0s)
$0.1 \leqslant f \leqslant 2.0$	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, k, meets the requirement:

$$k = \sqrt{k_2^2 + k_3^2}$$

where:

k_2 : second harmonic coefficient and

k_3 : third harmonic coefficient.

3.1.6.2 *Intermodulation*

With input signals of 0.8 kHz and 1.42 kHz, 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 Report 640-1 (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 following values:

TABLE IV

Frequency (kHz)	Cross-talk attenuation
$f < 0.5$	Slope 6 dB/octave
$0.5 \leqslant f \leqslant 3.2$	74 dB
$f > 3.2$	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 or methods for measuring this parameter.

Note 2 – The attention of administrations is drawn to the fact that in some cases it may be 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). This is because under these circumstances they occupy the same position in the line-frequency band (see CCITT 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 CCITT 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 multiple cross-talk (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 505).

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. CCITT Recommendation O.33 recommends 1020 Hz.

3.3.2 *Unbalance of the limitation level*

The difference between the 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 (f_0) is combined with the in-band audio signals (f_i) or out-of-band interfering signals (f_a).

3.3.3.1 *In-band intermodulation*

The following combination rule applies: $f_d = f_0 - nf_i$.

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 following f_i and f_d values is sufficient:

TABLE V

	$n = 2$		$n = 3$	
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: $f_d = nf_0 \pm f_a$.

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 following f_a and f_d values is sufficient:

TABLE VI

	$n = 1$		$n = 2$	
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 Report 647).

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RECOMMENDATION 505-4 *

PERFORMANCE CHARACTERISTICS OF 15 kHz-TYPE SOUND-PROGRAMME CIRCUITS **

Circuits for high quality monophonic and stereophonic transmissions

(Question 17/CMTT and Study Programme 17G/CMTT)

(1974-1978-1982-1986-1990)

The CCIR,

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,

UNANIMOUSLY RECOMMENDS

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out below.

1. Application

This Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation 502.

For estimation of the performance of circuits shorter or longer than the HRC, see Recommendation 605.

Note 1 – For all-digital circuits, a separate Recommendation might be envisaged after further study.

Note 2 – For further information, 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 ***
System output impedance, provisionally	Low, balanced

* This Recommendation is also proposed by the CMTT as a revision of CCITT Recommendation J.21 and should be brought to the attention of CCITT Study Group XV.

** For the definition of absolute power, relative power and noise levels, see Recommendation 574.

*** The tolerance, permitted reactance and degree of unbalance need further study.

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 dBm0s)	0 dB
Adjustment error, within	± 0.5 dB
Variation over 24 hours not to exceed	± 0.5 dB
Relative level (see CCITT Recommendation J.14)	+ 6 dBrs

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization 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

TABLE I

Frequency (kHz)	Response (dB)
$0.04 \leq f < 0.125$	+0.5 to −2
$0.125 \leq f \leq 10$	+0.5 to −0.5
$10 < f \leq 14$	+0.5 to −2
$14 < f \leq 15$	+0.5 to −3

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

3.1.2 *Group delay variation*

Differences between the value of group delay at the following frequencies and the minimum value:

kHz	Δτ (ms)
0.04	55
0.075	24
14	8
15	12

Between the points defined above, the tolerance limit varies linearly on a linear-delay/logarithmic-frequency diagram.

3.1.3 *Noise*

The measurement to be made with an instrument conforming to Recommendation 468.

For radio-relay systems, the requirements 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 II

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
Idle channel noise, maximum (dBq0ps)	−42	−51
Programme-modulated noise, maximum (dBq0ps)	−30	−39

Programme-modulated noise usually only occurs on sound-programme circuits which are equipped with compandors (e.g. types of circuits corresponding to CCITT 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.

Report 493 indicates that if a compandor is used, an improved signal-to-noise ratio is necessary to avoid objectional effects with some programme material.

Note – For further information on digital systems see Report 647.

3.1.4 *Single tone interference*

Level of any individual tone:

$$\leqslant -(73 + \psi) \qquad \text{dBm0s}$$

where ψ is the weighting filter response factor (positive or negative) as given in 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, it is recommended that the stop filters should have a 3 dB bandwidth of less than 3% referred to the mid-frequency. The use of stop filters which affect 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 harmonic distortion shall be measured with the input signal at +9 dBm0s for frequencies up to 2 kHz, and 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 CCITT Recommendations N.21 and N.23.

The total harmonic distortion when measured with a true r.m.s. meter shall not be greater than the following values:

TABLE III

Input frequency (kHz)	Total harmonic distortion (THD)	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)

Note – If a companding system is used, the selective measuring method should be applied in order to avoid any influece of the programme-modulated noise on the measured values.

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 0.5% (−43 dBm0s).

Note – Attention is drawn to the fact that in transmission systems using compandors, a 3rd order difference-frequency 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 following additional requirements apply:

TABLE IV

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 Report 640-1 (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 following values:

TABLE V

Frequency (kHz)	Cross-talk attenuation
$f = 0.04$	50 dB
$0.04 < f < 0.05$	Oblique straight-line segment on linear-decibel and logarithmic-frequency scales
$0.05 \leqslant f \leqslant 5$	74 dB
$5 < f < 15$	Oblique straight-line segment on linear-decibel and logarithmic-frequency scales
$f = 15$	60 dB

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 in some cases it may be 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). This is because under these circumstances they occupy the same position in the line-frequency band (see CCITT 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 CCITT 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 multiple cross-talk (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 following values:

TABLE VI

Frequency (kHz)	Gain difference (dB)
$0.04 \leqslant f < 0.125$	1.5
$0.125 \leqslant f \leqslant 10$	0.8
$10 < f \leqslant 14$	1.5
$14 < f \leqslant 15$	3.0

3.2.2 The phase difference between the A and B channels shall not exceed the following values:

TABLE VII

Frequency (kHz)	Phase difference
$f = 0.04$	30°
$0.04 < f < 0.2$	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
$0.2 \leqslant f \leqslant 4$	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 scales
$f = 15$	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 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 in Table VIII.

TABLE VIII

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 this case, the nominal 1 kHz test signal must be offset. CCITT Recommendation O.33 recommends 1020 Hz.

3.3.2 *Unbalance of the limitation level*

The difference between the 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 (f_0) is combined with the in-band audio signals (f_i) or out-of-band interfering signals (f_a).

3.3.3.1 *In-band intermodulation*

The following combination rule applies: $f_d = f_0 - nf_i$.

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 following f_i and f_d values is sufficient:

TABLE IX

	$n = 2$		$n = 3$	
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: $f_d = nf_0 \pm f_a$.

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 following f_a and f_d values is sufficient:

TABLE X

	$n = 1$		$n = 2$	
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 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.

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RECOMMENDATION 605-1

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT

(Study Programme 17D/CMTT)

(1982–1986)

The CCIR

UNANIMOUSLY RECOMMENDS

that for the estimation of transmission performance of sound-programme circuits shorter or longer than the hypothetical reference circuit the following rules be applied:

Laws of addition

1. Comments on the use of the laws of addition

The definition of a circuit in terms of a single multiple of the hypothetical reference circuit is impossible if the number of audio-to-audio sections and the length of the circuit differ from those of the hypothetical reference circuit by different ratios, i.e. if $n/3 \neq L/l$ where

n: number of audio-to-audio sections,

L: length of the circuit,

l: 2500 km.

In such cases two definitions of the circuit in terms of the hypothetical reference circuit should be used, one for those parameters primarily a function of the circuit configuration, and a second for those parameters (e.g. continuous random noise) primarily a function of the length of the circuit.

2. Law pertaining to circuit configuration

For the first definition of the circuit in terms of the hypothetical reference circuit use the following equation for all parameters in Table II except for "continuous random noise".

Let D_3: design objective or the parameter derived therefrom and indicated in Table II, permitted value for three homogeneous sections of the hypothetical reference circuit,

and D_n: performance, or the parameter mentioned above, to be estimated for n sections.

The following expressions apply:

– for absolute values and logarithmic deviations:

$$D_n = D_3(n/3)^{1/h} \tag{1}$$

– for absolute levels:

$$D_n = D_3 + \frac{20}{h}\log(n/3) \tag{2}$$

– for logarithmic ratios:

$$D_n = D_3 - \frac{20}{h}\log(n/3) \tag{3}$$

h has the value 1, 3/2 or 2 in accordance with Table II: $h = 1$ gives linear or arithmetic law of addition, $h = 3/2$ gives the "three-halves power" law of addition and $h = 2$ gives the quadratic (r.m.s.) law of addition.

Calculated values of $\left(\frac{n}{3}\right)^{1/h}$ and $\frac{20}{h}\log\left(\frac{n}{3}\right)$ are given in Table I.

3. Law pertaining to circuit length

For the second definition of the circuit in terms of the hypothetical reference circuit use the following equation for "continuous random noise levels" only. When distance is considered the law of addition becomes:

$$D_n = D_3 - \frac{20}{h} \log (L/l) \tag{4}$$

where D_n, D_3, L and l are as defined in § 1 and § 2. If $l <$ 280 km, set $l =$ 280 km in (4).

Note – The values calculated as per this Recommendation provide only indications of the probable performance. They should be used with caution when studying the design of equipment, because the laws of addition are not precisely known for every type of impairment. Further studies are necessary, especially to cover the case of mixed analogue-and-digital circuits.

TABLE I

n	Formula (1)			Formulas (2) and (3)		
	$\left(\frac{n}{3}\right)^{1/h}$			$\frac{20}{h} \log \left(\frac{n}{3}\right)$		
	$h = 1$	$h = 3/2$	$h = 2$	$h = 1$	$h = 3/2$	$h = 2$
1	0.33	0.48	0.58	−9.5 dB	−6.4 dB	−4.8 dB
2	0.67	0.76	0.82	−3.5	−2.3	−1.8
3	1.00	1.00	1.00	0.0	0.0	0.0
4	1.33	1.21	1.15	2.5	1.7	1.2
5	1.67	1.41	1.29	4.4	3.0	2.2
6	2.00	1.59	1.41	6.0	4.0	3.0
7	2.33	1.76	1.53	7.4	4.9	3.7
8	2.67	1.92	1.63	8.5	5.7	4.3
9	3.00	2.08	1.73	9.5	6.4	4.8
10	3.33	2.23	1.83	10.5	7.0	5.2
11	3.67	2.38	1.91	11.3	7.5	5.6
12	4.00	2.52	2.00	12.0	8.0	6.0
13	4.33	2.66	2.08	12.7	8.5	6.4
14	4.67	2.79	2.16	13.4	8.9	6.7
15	5.00	2.92	2.24	14.0	9.3	7.0

TABLE II

	Parameter	D_3 expressed in	Applicable formula	h (tentative value)
Parameters relevant to mono and stereo circuits	Insertion gain (1.0 kHz)			
	Adjustment error	dB	1	2
	Variation during 24 h	dB	1	2
	Gain/frequency response	dB	1	3/2
	Group delay/frequency response referred to minimum	ms	1	1
	Maximum weighted noise level	dBq0ps	4	2
	Single tone interference level	dBm0	2	3/2
	Ratio of disturbing modulation by power supply to reference signals	dB	3	3/2
	Non-linear distortion	%	1	3/2
	Error in reconstituted frequency	Hz	1	3/2
	Intelligible crosstalk ratio	dB	3	3/2
	Error in amplitude/amplitude response	dB	1	3/2
Additional parameters relevant to stereo circuits	Difference in gain between A and B channels	dB	1	(1)
	Phase difference between A and B channels	degree	1	(1)
	Intelligible crosstalk ratio between A and B channels	dB	3	3/2
	Non-linear crosstalk ratio between A and B channels	dB	3	3/2

(1) Some administrations have found $h = 3/2$ appropriate. Tests conducted in Japan [CCIR, 1978-82], however, showed better agreement with quadratic law of addition ($h = 2$) than 3/2 law of addition.

REFERENCES

CCIR Documents

[1978-82]: CMTT/233 (Japan).

RECOMMENDATION 474-1

MODULATION OF SIGNALS CARRIED BY SOUND-PROGRAMME CIRCUITS BY INTERFERING SIGNALS FROM POWER SUPPLY SOURCES

(Study Programme 17F/CMTT, Geneva, 1982)

(1970–1982)

The CCIR

UNANIMOUSLY RECOMMENDS

that the ratio of a sine-wave test signal applied to the sound programme circuit to the highest level unwanted side component due to modulation caused by interfering signals from power supply sources should be greater than 45 dB.

Note 1 – This limit is identical to that which is considered tolerable for other types of transmission (FM and AM-VF telegraphy, facsimile transmission, speech, telephone signalling and data transmission).

Note 2 – This limit applies only where the interfering signals are the usual low order mains frequency harmonics. For modulation by much higher frequencies a more stringent limit is likely to apply.

Note 3 – For design purposes this limit should be taken as applying to the hypothetical reference circuit of length 2500 km.

RECOMMENDATION 659*

DIGITAL TRANSMISSION OF SOUND PROGRAMMES – GENERAL PRINCIPLES

(Question 18/CMTT and Study Programmes 18A/CMTT, 18C/CMTT, 18D/CMTT and 18E/CMTT)

(1986)

The CCIR,

CONSIDERING

(a) that it is desirable to use common standards for the transmission of high-quality sound;

(b) that it is desirable to use as few code conversions as possible in the international digital sound exchange procedure;

(c) that to facilitate the exchange of signals, it is desirable to render the transmission interfaces as transparent as possible to the transmitted message content;

(d) that the interface bit rates should take into account the hierarchical levels recommended by the CCITT;

(e) that it may be efficient to apply bit-rate reduction and error-protection methods to lower transmission costs;

(f) that quality and availability should not be restricted by the signal processing equipment or methods employed in the transmission circuit,

UNANIMOUSLY RECOMMENDS

1. that a sound programme originating in digital form should preferably be maintained in digital form for transmission;

2. that it should be possible to transmit digitally coded studio signals at the quality level required for each particular application;

3. that all the components of an individual sound programme (e.g. the left and right channel of a stereophonic programme signal) should be transmitted as one composite digital signal.

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* This Recommendation should be brought to the attention of CCIR Study Group 10 and CCITT Study Groups XV and XVIII.

RECOMMENDATION 724*

TRANSMISSION OF DIGITAL STUDIO QUALITY SOUND SIGNALS OVER H1 CHANNELS**

(Question 18/CMTT, Study Programmes 18A/CMTT, 18B/CMTT, 18D/CMTT, 18E/CMTT, 18F/CMTT and 18J/CMTT)

(1990)

The CCIR,

CONSIDERING

(a) that the source encoding for digital sound signals in broadcasting studios is given in Recommendation 646;

(b) that the 2-channel digital audio interface is specified in Recommendation 647;

(c) that a format for digital studio-quality sound signal connections should be based on these Recommendations;

(d) that the sound quality and the auxiliary information carried over the Recommendation 647 interface should be maintained as far as possible;

(e) that the hierarchical bit rates for digital networks are given in CCITT Recommendation G.702;

(f) that a digital hierarchy for interworking between networks using different transmission hierarchies is given in CCITT Recommandation G.802;

(g) that the access levels for the H1 channels in the ISDN are given in CCITT Recommendation I.412;

(h) that simple interworking between the different hierarchies is necessary;

(j) that account must be taken of network impairments such as signal bit errors, error bursts and controlled slips;

(k) that, for some applications, the introduction of excessive delay can cause operational problems,

RECOMMENDS

that for transmission of digital studio-quality sound signals the coding and multiplexing format given in Annex I should be used.

ANNEX I

CODING AND MULTIPLEXING FORMAT

1. Introduction

The transmission format is based on the 2-channel digital audio interface described in Recommendation 647, which should be read in conjunction with this text.

The 32 bits in each sub-frame of the Recommendation 647 interface are treated in the following way in the contribution format:

– Preamble (bit 0-3):	not transmitted
– Bit 4-7:	not transmitted
– Audio sample word (bit 8-27):	companded
– Validity flag (bit 28):	not transmitted
– User data (bit 29):	transmitted transparently in H12 only
– Channel status (bit 30):	subjected to data compression
– Parity bit (bit 31):	not transmitted. Replaced by a parity bit for the audio sample word.

* This Recommendation should be brought to the attention of Study Group 10 and CCITT Study Groups XV and XVIII.

** The United States of America withholds its approval of this Recommendation.

2. Compatibility between H11 and H12-based systems

The H12 level provides a total of 20 bits per sample, and H11 provides a total of 16 bits per sample. To simplify interworking between H11 and H12 channels, the companding of the audio signal is such that the samples are compressed for transmission in the H11 channel. In the H12 channel, extra bits may be conveyed, to improve resolution of the audio coding and provide a user data channel.

The essential data occupies the entire available capacity of the H11 channel, and the first 24 available octets of each frame of the H12 channel.

3. Coding law

Near instantaneous companding from 20 to 15 bits/sample is applied. A 1 ms companding block is used with 8 coding ranges.

The coding table is shown in Fig. 1a and the transmitted bits in Fig. 1b. Unused bits are set to 1 (one).

The 1 ms companding block introduces a fundamental delay of 2 ms per codec. In practice, the total delay will be slightly longer.

4. Sample error detection

One parity bit is applied to the 7 most significant bits of each transmitted sound sample such that the parity group is odd.

5. Sample interleaving

The companding block contains 96 sound samples (48 from each sound signal). The sound samples within the companding block are organized in 8 successive frames complying with CCITT Recommendation G.704. Each frame contains 6 samples from each sound signal, with the associated parity bits. Adjacent sound samples within the companding block are separated by four Recommendation G.704 frames. This is shown diagrammatically in Fig. 2. The first four frames carry all the odd-numbered samples from both sound signals; the second four frames carry all the even numbered samples.

In the event of an error burst corrupting consecutive bits for a period equivalent to up to four frames, the erroneous samples should be concealed by interpolation between adjacent samples (from part of the block unaffected by the error burst).

6. Signalling in parity [Chambers, 1985]

There are 96 parity bits per 1 ms companding block. Some extra data bits are carried by modifying the parity bits as follows:

6.1 *Scale factor transmission*

Each bit of each 3-bit scale factor word is carried in the parity of 8-sound samples defined in § 6.4, according to the following rule: a scale factor bit which is "0" causes the parity of the 8 samples to be unchanged; a scale factor bit which is "1" modifies the parity. At the decoder, a majority decision process is used to determine the scale factor bits and restore the original parity bit. The samples are then checked for errors in the normal way.

6.2 *Channel status*

Compressed channel status (see § 9) is carried in exactly the same manner as the scale factor bits.

6.3 *Framing signals*

Multiframe alignment signals (MFA) (see § 7) and frame slip detection signals (FSD) (see § 8) are carried by modifying the parity of single samples. These signals do not have the benefit of majority decision decoding, but are inherently predictable and may be decoded reliably.

6.4 *Signalling within the companding block*

The sample-interleaved companding block is represented in Fig. 3 by the rectangle at the bottom of the diagram. Each row within the rectangle represents a Recommendation G.704 frame, and is subdivided into 12 squares representing the 12 samples. The diagram further shows the modification of the parity bits associated with the 12 samples in each frame by the signalling in parity mentioned above.

Sign bit																			LSB	Scale factor S_2	S_1	S_0
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	0
0	0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	1
0	0	0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	1	0
0	0	0	0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	1	1
0	0	0	0	0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	1	0	0
0	0	0	0	0	0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	1	0	1
0	0	0	0	0	0	0	1	X	X	X	X	X	X	X	X	X	X	X	X	1	1	0
0	0	0	0	0	0	0	0	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1
1	1	1	1	1	1	1	1	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1
1	1	1	1	1	1	1	0	X	X	X	X	X	X	X	X	X	X	X	X	1	1	0
1	1	1	1	1	1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	1	0	1
1	1	1	1	1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	1	0	0
1	1	1	1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	1	1
1	1	1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	1	0
1	1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	1
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	0

Bits truncated in H11 channels

Bits truncated in H11 and H12 channels

X = 1 or 0

FIGURE 1a – *Coding table*

Sign bit b_0														b_{14}			LSB b_{17}	Scale factor S_2	S_1	S_0
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	0
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	1
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	1	0
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	1	0	1	1
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	0	0
0	1	X	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	1	0	1
0	1	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	1	1	1	0
0	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	1	1	1	1	1
1	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	1	1	1	1	1
1	0	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	1	1	1	0
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	1	0	1
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	1	1	1	0	0
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	1	0	1	1
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	1	0
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	1
1	0	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	X	0	0	0

← Bits transmitted in H11 channels →

← Bits transmitted in H12 channels →

X = 1 or 0

FIGURE 1b – *Transmitted bits*

0 ms

1A	1B	3A	3B	5A	5B	7A	7B	9A	9B	11A	11B	Odd-numbered samples
13A	13B	15A	15B	17A	17B	19A	19B	21A	21B	23A	23B	
25A	25B	27A	27B	29A	29B	31A	31B	33A	33B	35A	35B	
37A	37B	39A	39B	41A	41B	43A	43B	45A	45B	47A	47B	
2A	2B	4A	4B	6A	6B	8A	8B	10A	10B	12A	12B	Even-numbered samples
14A	14B	16A	16B	18A	18B	20A	20B	22A	22B	24A	24B	
26A	26B	28A	28B	30A	30B	32A	32B	34A	34B	36A	36B	
38A	38B	40A	40B	42A	42B	44A	44B	46A	46B	48A	48B	

Channel B
Channel A

FIGURE 2 – *Sample interleaving within the companding block*

1 ms = 1 companding block

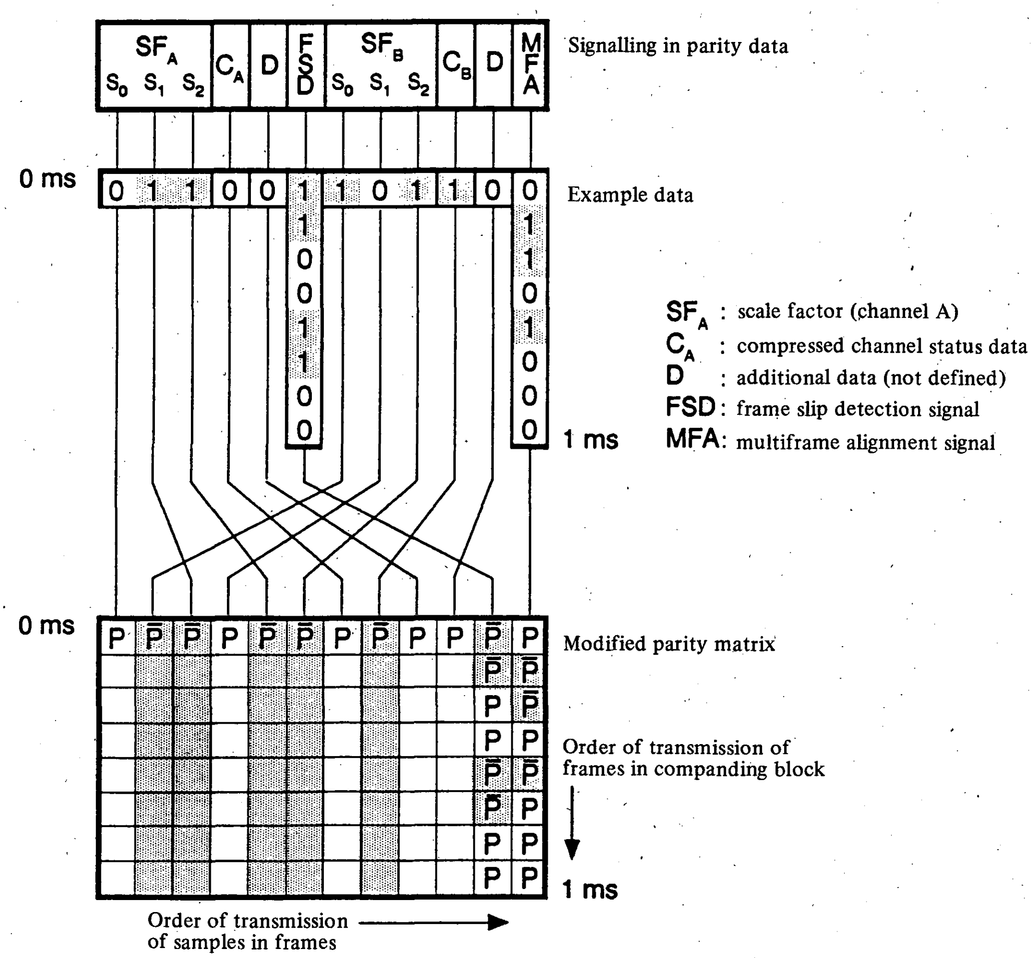


FIGURE 3 – *Signalling in parity matrix and data*

7. Synchronization and frame alignment

The sampling frequency of the sound signal must be synchronous with the bit clock of the transmission system.

Frame alignment of the companding block and multiframe (see § 9) is accomplished by application of a 1536-bit chain code, MFA. The generator is shown in Fig. 4a and the corresponding circuit for locking in Fig. 4b [Chambers, 1985].

The start word of the MFA generator is 10000110100 and it corresponds to the first frame of the multiframe. The generator will produce a sync pulse after 1536 bits (192 ms) and automatically reset to the start word.

The start of the multiframe may be locked to the interface Z-preamble as shown in Fig. 6.

The 1 ms companding block sync is generated by decoding the corresponding register contents.

The multiframe alignment signal (MFA) is signalled in parity (see § 6).

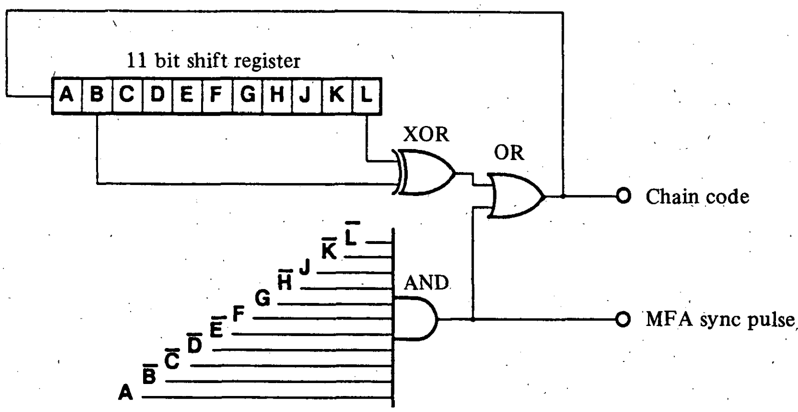


FIGURE 4a – *Chain code generator*

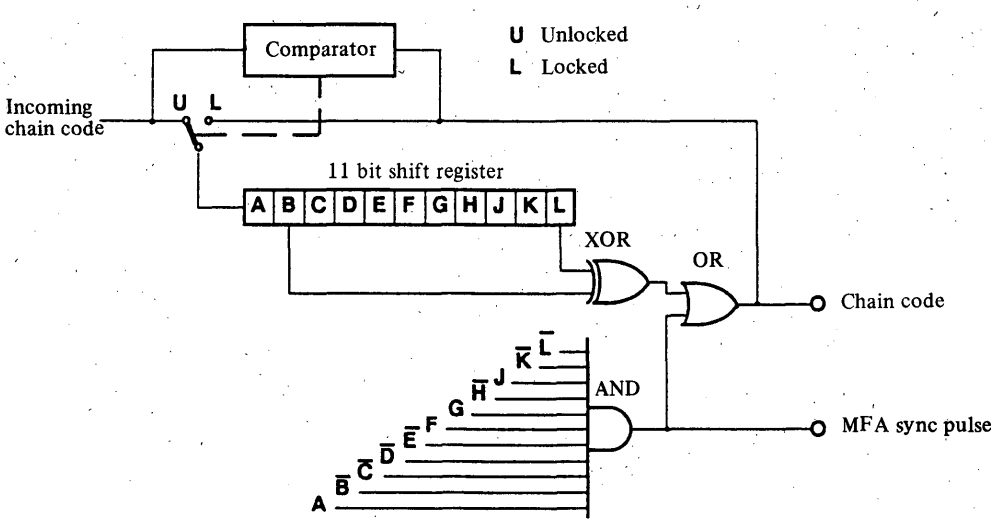


FIGURE 4b – *Chain code synchronization circuit*

8. Frame slip detection and management

A controlled slip is defined as suppression or repetition of a Recommendation G.704 frame.

A fixed of bits, FSD, (... 110011001100...) is signalled in parity to assist decoders to detect controlled slips during transmission. The FSD is transmitted as shown in Fig. 3 (i.e. the FSD is frame-aligned with the companding block).

With a companding block of 1 ms and traditional framing methods, it would normally take a number of such frames (7-8 ms) to detect a slip and re-align the companding frame.

With the FSD, it will only take a few Recommendation G.704 frames to detect that a slip has occurred as the phase of the sequence will shift + or −90° (depending on whether a frame has been suppressed or repeated). Through a modulo-2 operation on the received and the expected FSD it is possible to detect, within two frames or more precisely, in which "pair" (1 and 2, 3 and 4, 5 and 6, or 7 and 8) of frames within the companding block the slip has occurred.

A suggested strategy for interpolation after a detected slip is shown in Fig. 5a and 5b, where the decoder produces a decoded block of the same length as the received block. Only channel A is shown; channel B is handled identically (Fig. 2 shows the transmitted sample sequence).

Positive slip – one frame is repeated

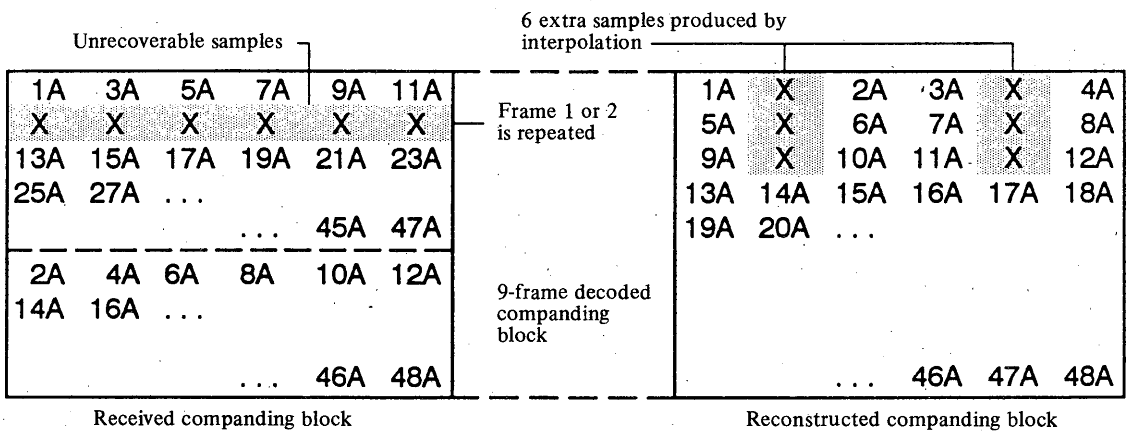


FIGURE 5a – *Strategy for interpolation after a detected positive slip*

By suitable decoder arrangements the scale factor can still be correctly decoded.

As all sound samples are stored for 1 ms (the length of the companding block) in the decoder, it is possible to move the companding frame boundary pointer such that the correct scale factor is applied to the correct number of samples.

Negative slip – one frame is deleted

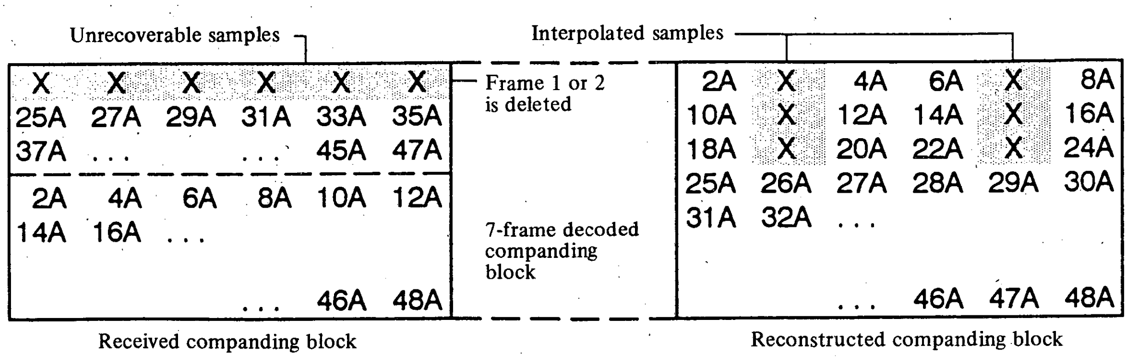


FIGURE 5b – *Strategy for interpolation after a detected negative slip*

Note that it is not possible to tell which frame has been deleted. Consequently, of the 12 transmitted samples, 6 are not received and 6 cannot be identified. In the reconstructed companding block, 6 samples are simply omitted (to adjust the length of the reconstructed block) and 6 are replaced by interpolation.

9. Channel status

Channel status data in the digital audio interface consists of a 192 bit (24 byte) cycle, which is repeated in 4 ms (one block of the interface).

The channel status is signalled in parity according to the description in § 6.2 and 6.4. This method of signalling provides one channel status bit for each audio signal per 1 ms companding block, enabling the system to carry one channel status block every 192 ms. This is shown in Fig. 6.

Because only one data block out of 48 is transmitted, the two counters (local sample and time of day address code) must be incremented in the decoder by the appropriate amount.

The start of the multiframe is signalled by the multiframe alignment signal defined in § 7. The time codes carried in the compressed channel status refer to the timing of the first sample in the multiframe.

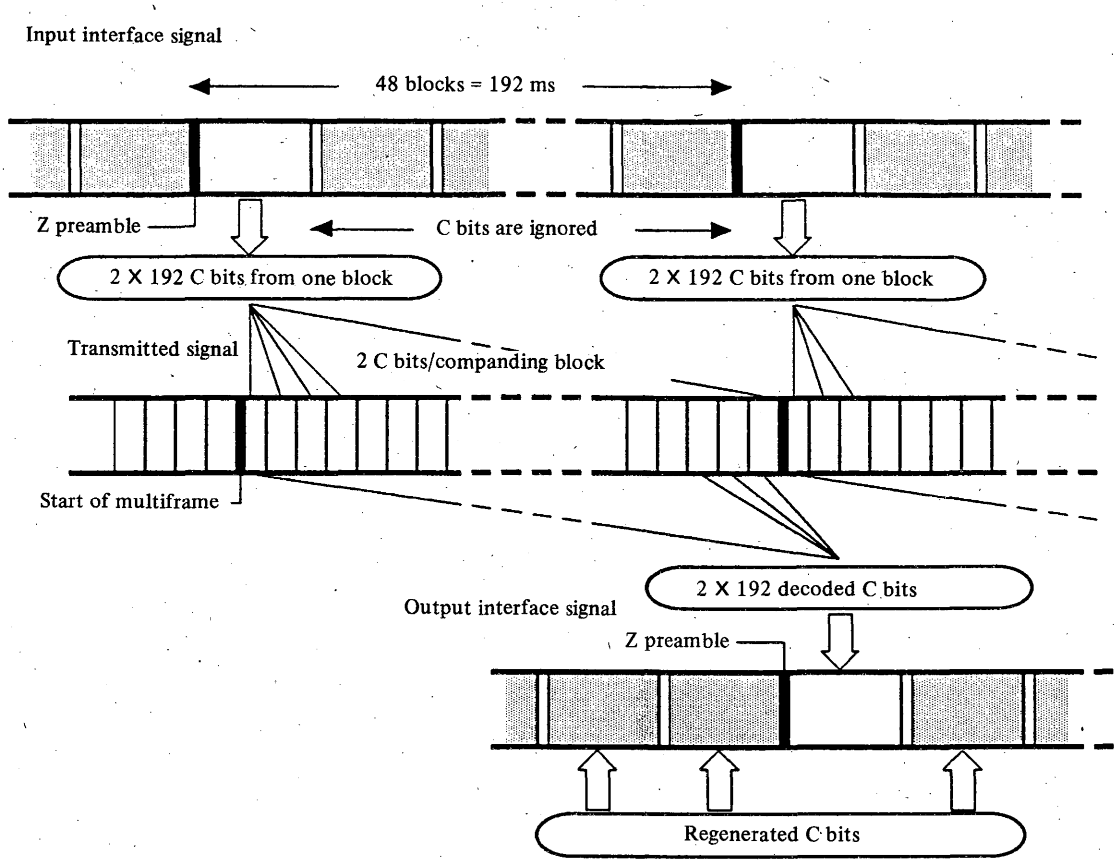


FIGURE 6 – *Transmission of channel status data*

10. User data

The user data bit in each sub-frame of the interface is transmitted transparently in the H12 channel only.

11. Frame structure and bit interleaving

The Recommendation G.704 frame for the H11 channel contains 192 usable bits, and that for the H12 channel contains 240 usable bits. Twelve 15-bit sound samples, each accompanied by its parity bit, may be carried in either type of frame. Additionally, the H12 frame may carry sufficient extra audio bits to increase the length of the compressed audio samples to 18 bits, and one user data bit per sample.

The organization of the 24 octets of data which are common to both H11 and H12 channels are identical in both types of frame, to facilitate remultiplexing at the interface between H11 and H12 channels. The common data occupies the entire available capacity of the H11 frame, and the first 24 available octets in the H12 frame, as shown in Fig. 7. The remaining 6 octets in the H12 frame carry user bits and extra audio bits.

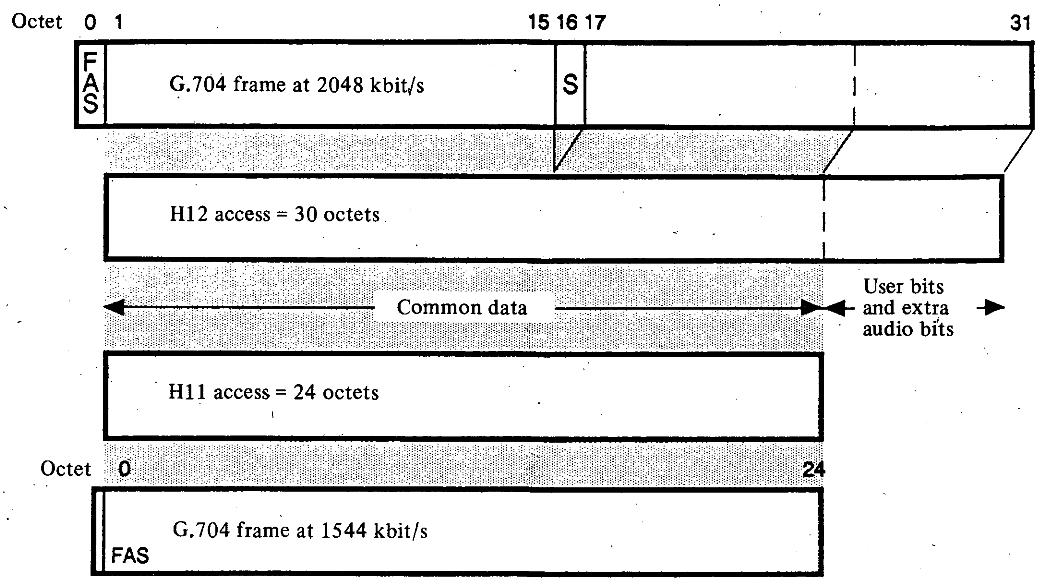


FIGURE 7 – *Organization of H11 and H12 frames*

FAS: frame alignment signal
S: signalling

11.1 *Organization of the 24 octets common to both H11 and H12*

The number of bits available for interleaving is 192. The bit-interleaved sample block can be described by a 8 × 24 matrix (Fig. 8). The numbers designate the samples within any frame of a companding block, interleaved as described in § 5 (see Fig. 2), in transmission order.

Row/column	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
1	1	1	1	2	1	7	1	8	1	3	1	4	1	9	1	10	2	5	2	6	2	11	2	12
2	2	1	2	2	2	7	2	8	7	3	7	4	7	9	7	10	7	5	7	6	7	11	7	12
3	8	1	8	2	8	7	8	8	8	3	8	4	8	9	8	10	3	5	3	6	3	11	3	12
4	3	1	3	2	3	7	3	8	4	3	4	4	4	9	4	10	4	5	4	6	4	11	4	12
5	9	1	9	2	9	7	9	8	9	3	9	4	9	9	9	10	10	5	10	6	10	11	10	12
6	10	1	10	2	10	7	10	8	5	3	5	4	5	9	5	10	5	5	5	6	5	11	5	12
7	6	1	6	2	6	7	6	8	6	3	6	4	6	9	6	10	11	5	11	6	11	11	11	12
8	11	P1	11	P2	11	P7	11	P8	12	P3	12	P4	12	P9	12	P10	12	P5	12	P6	12	P11	12	P12

Protected bits and parity bits
Unprotected bits
Order of transmission

Numbering convention

1 corresponds to bits from samples 1A, 13A, 25A, ... or 38A
2 corresponds to bits from samples 1B, 13B, 25B, ... or 38B
3 corresponds to bits from samples 3A, 15A, 27A, ... or 40A
etc.

See Fig. 2

FIGURE 8 – *Bit-interleaving matrix*

Each even-numbered column represents the 7 protected bits (b_0-b_6) and the parity bit of one sample. These bits are filled into the matrix column by column. The progression of samples is 1-2-7-8-3-4-9-10-5-6-11-12 in order to maximize the distance between any two neighbouring samples of one channel and at the same time keep co-timed samples (in stereo mode) together. The transmission sequence always starts with the LSB and the MSB always precedes the P-bit.

The 8 least significant bits of the 15-bit word (b_7-b_{14}) are filled into the odd columns of the matrix, but in this case row by row. These bits will be kept close after interleaving, in order to minimize the number of impaired samples when long error-bursts occur. The transmission sequence is: b_7-b_9-b_{11}-b_{13}-b_8-b_{10}-b_{12}-b_{14}. Bits read from columns 1, 5, 9, 13, 17 and 21 are inverted prior to transmission.

The first 24 available octets of the Recommandation G.704 frame are filled using the data represented in Fig. 8, by reading row by row from the matrix. The distance between protected sample-bits or parity bits from each sample is 24.

Octets 0 and 16 of the 2048 kbit/s Recommandation G.704 frame are not interleaved.

Note – When the connection is established over 1544 kbit/s circuits which are not bit-sequence independent, the minimum pulse density requirement may be guarantedd by forcing to digital "1" bits in columns 7, 15 or 23, where necessary. This operation is equivalent to the "z-operation" described in CCITT Recommendation G.802, § 2.1, and should be performed at the interface with such circuits.

11.2 *Organization of the last 6 octets in H12*

The last 3 LSB and the user bit from each sample are transmitted in the remaining 6 octets in the H12 frame by reading row by row from the matrix shown in Fig. 9. As before, the numbers designate samples, and the order of transmission is LSB first.

Row/column

	1	2	3	4	5	6	7	8
1	1	1	1	U1	2	2	2	U2
2	7	7	7	U7	8	8	8	U8
3	3	3	3	U3	4	4	4	U4
4	9	9	9	U9	10	10	10	U10
5	5	5	5	U5	6	6	6	U6
6	11	11	11	U11	12	12	12	U12

User bits

Unprotected bits

Order of transmission

Numbering convention: see Fig. 8

FIGURE 9 – *Interleaving of supplementary audio bits and user bit in the H12 channel*

REFERENCES

CHAMBERS, J. P. [1985] Signalling in parity: a brief history. British Broadcasting Corporation, BBC RD 1985/15.

RECOMMENDATION 718*

DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON DISTRIBUTION CIRCUITS USING 480 kbit/s (496 kbit/s) PER AUDIO CHANNEL

(Question 18/CMTT, Study Programmes 18A/CMTT, 18B/CMTT, 18C/CMTT, 18D/CMTT, 18E/CMTT and 18J/CMTT)

(1990)

The CCIR,

CONSIDERING

a) that the distribution of high-quality sound-programme signals from the studio to transmitters and users does not require further audio downstream processing;

b) that in general more than one high-quality sound-programme signal has to be conveyed over distribution circuits as in the case of DBS;

c) that a sampling frequency of 32 kHz is recommended for the digital transmission of high-quality sound-programme signals (Recommendation 606);

d) that some digital broadcasting applications may require system performances that go beyond those offered by equipment complying with CCIR Recommendation 660;

e) that the high-quality sound-programme signals should interface the ISDN at the H1 level as specified in CCITT Recommendation I.412,

UNANIMOUSLY RECOMMENDS

1. that for distribution applications where a sampling frequency of 32 kHz is used and where a dynamic range corresponding to more than 14 bits is required the coding method described in § 1 of Annex I should be used on links providing a BER of less than 10^{-5};

2. that for transmission at H12 level, two stereophonic programmes or four monophonic programmes should be multiplexed according to the format described in § 2 of Annex I;

3. that for cases where a higher ancillary data capacity is required and dedicated 2048 kbit/s links are available, the coding method and multiplexing format described in Annexe II should be used.

Note – Exchange of international digital sound-programme signals on networks with other hierarchical rates (1544 kbit/s in North America) shall be made using Recommendation 660.

* This Recommendation is of interest to CCITT Study Groups XV and XVIII.

ANNEX I

DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON DISTRIBUTION CIRCUITS USING 480 kbit/s AUDIO CHANNEL

1. Coding characteristics

1.1 *Sampling frequency*

The sampling frequency shall be 32 kHz. The sampling frequency tolerance shall be $\pm 5 \times 10^{-5}$, as specified in CCITT Recommendations G.732 and G.733 for primary PCM multiplex equipment. This sampling frequency is consistent with that indicated in CCIR Recommendation 606.

1.2 *Coding method*

16/14-bit floating point companding with a 2 ms coding block length (i.e. 64 consecutive samples per block) and a 3-bit scale factor (transmitted by signalling in parity).

The 16-bit samples of the sound signal are represented in a 2 s complement format. The first bit of each word is the MSB (sign bit, $0 \sim +$), and the last the LSB. Using a floating point system, the 16-bit samples are converted into 14-bit code words for transmission.

A 3-bit scale factor applying to a block of 64 samples indicates how many of the bits (0 ... 7) following the sign bit (y_1) in all sampled words have the same value as the sign bit (Fig. 1a)). The redundancy indicated by the scale factor does not need to be transmitted. Instead, the samples and their relevant information must be shifted towards the sign bits (floating-point system). This allows the 15th and 16th bits of the source code words to be transmitted in the case of low signal amplitudes. The bits marked Z1 to Z5 have not yet been assigned (Fig. 1b)).

At the receiving end the scale factor is used to shift the bits of the samples back to their original value. This yields 16-bit samples and limits the effects of unrecognized bit errors to the amplitude range indicated by the scale factor.

1.3 *Sample error-protection*

After having applied the floating-point technique for reducing the bits per sample from 16 to 14, a parity bit is calculated on the seven most significant bits of each sample, such that the number of "1"s in the group of the seven protected bits and the parity bit is odd. Thus, the best protection against clicks due to bit errors is guaranteed provided concealment in the form of the arithmetic mean value of the samples adjacent to the faulty sample is performed at the receiving end. If the concealment is done after the 14/16 bit reconversion, errors in least significant bits can also be concealed in an optimum way.

1.4 *Ancillary data*

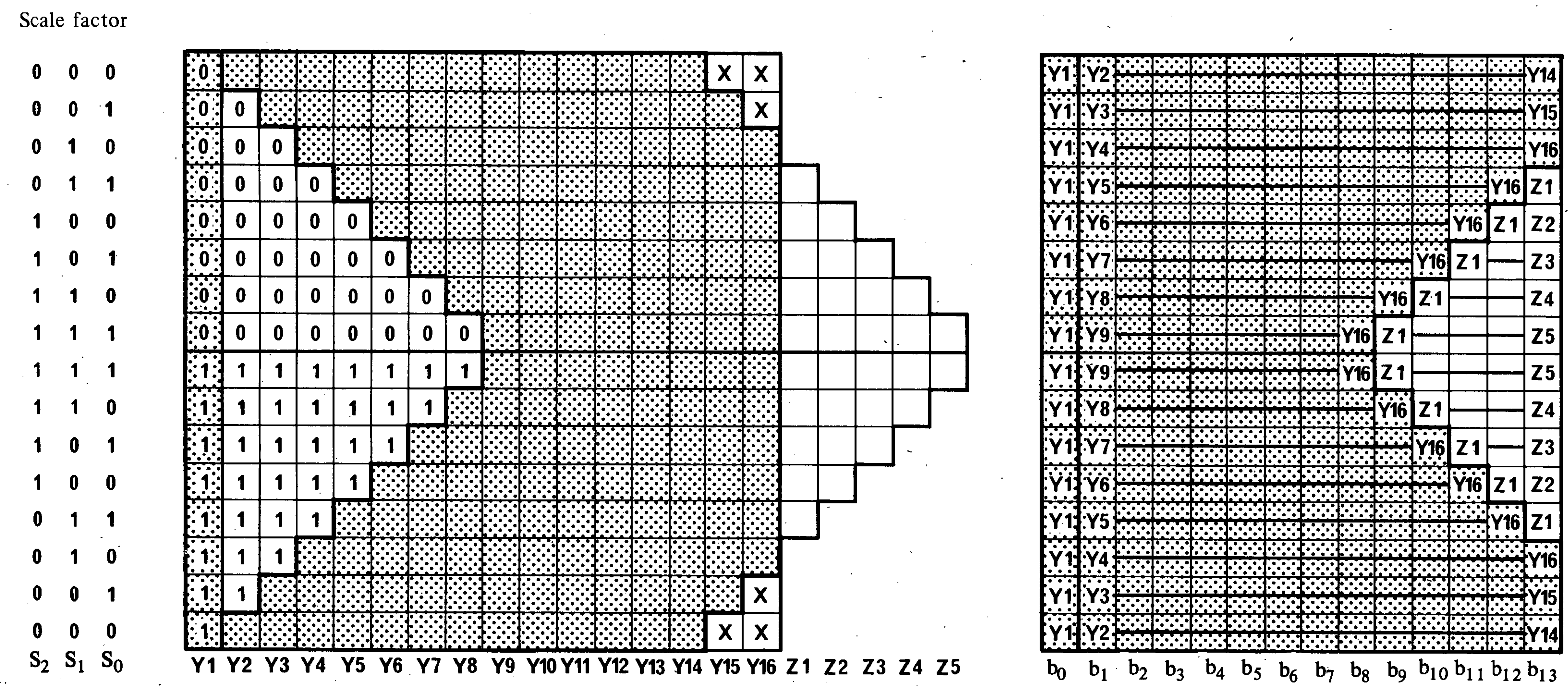
A data capacity of 4 kbit/s per channel is transmitted by signalling in parity.

1.5 *Signalling in parity* [Chambers, 1985]

Signalling in parity is achieved by transmitting the parity bits of an odd number of successive samples without inversion or after inversion, depending on the bit to be signalled. Inversion has to be done, if the bit to be transmitted is 1. The signalling technique used for the scale factor, its related parity bit, and the ancillary data is based on majority-decision logic at the receiver: for each channel, it processes twelve groups of five consecutive samples (three for the scale factor, one for its parity bit and eight for the ancillary data) to recognize simultaneously the odd or even parity of the sample and the data signalled in parity (see Fig. 2). A similar process is used to recognize the synchronization of the 2 ms frames through one group of four consecutive samples.

1.6 *Total bit rate*

With the parameters mentioned above the total bit rate required for one monophonic channel is 480 kbit/s [32 kHz $\times$ (14 +1) bit].



a) Coding scheme

b) Transmission format

FIGURE 1 – *16/14 bit floating-point method*

relevant sound signal code word range of the 16-bit source signal words

X X non-transmittable bits of the 16-bit source signal words

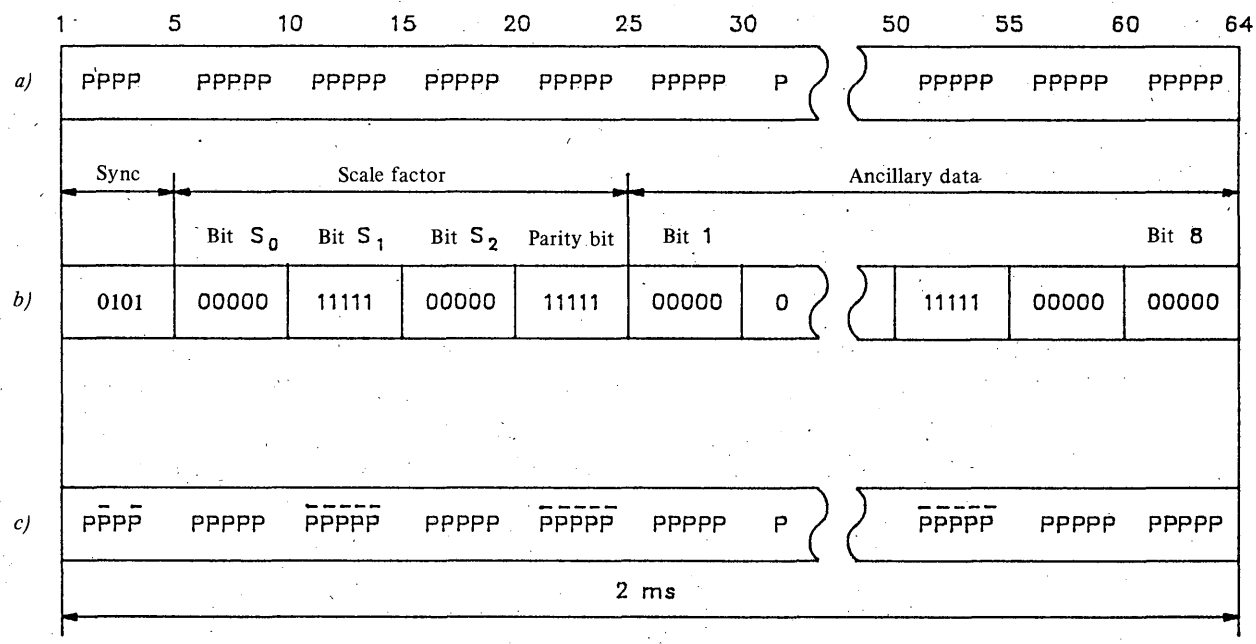


FIGURE 2 – *Signalling in parity coding for one audio channel*

a) Parity bits (P denotes the parity bit associated to each individual audio sample).

b) Sync pattern, scale factor, parity bit for the scale factor and ancillary data bits. The parity bit for the scale factor should be even as shown in the example.

c) Modified parity bits.

2. Frame structure for transmission

Within the ISDN, a usable bit rate of 1920 kbit/s for one multiplex frame is available according to CCITT Recommendation G.704.

In order to ensure the compatibility between the signals transmitted at 480 kbit/s according to this Recommendation with those transmitted at 384 kbit/s according to CCITT Recommendation G.737, the bits of each audio channel should be allocated according to the present proposal depicted in Fig. 3. The bits of each audio channel should be transmitted as a group of 30 bits within a half-frame.

Moreover, the bits of each sample are interleaved such that the most significant bit (MSB) is followed by the least significant bit (LSB), etc. (see Fig. 3). This arrangement of bits has proven to be a good protection against double errors, which otherwise cannot be recognized by a single parity bit and thus cannot be concealed.

On this basis, channels using 480 kbit/s and 384 kbit/s can be combined according to Table I:

TABLE I

	Number of channels	
	480 kbit/s	384 kbit/s
I	4	0
II	3	1
III	2	2
IV	1	3
V	0	5

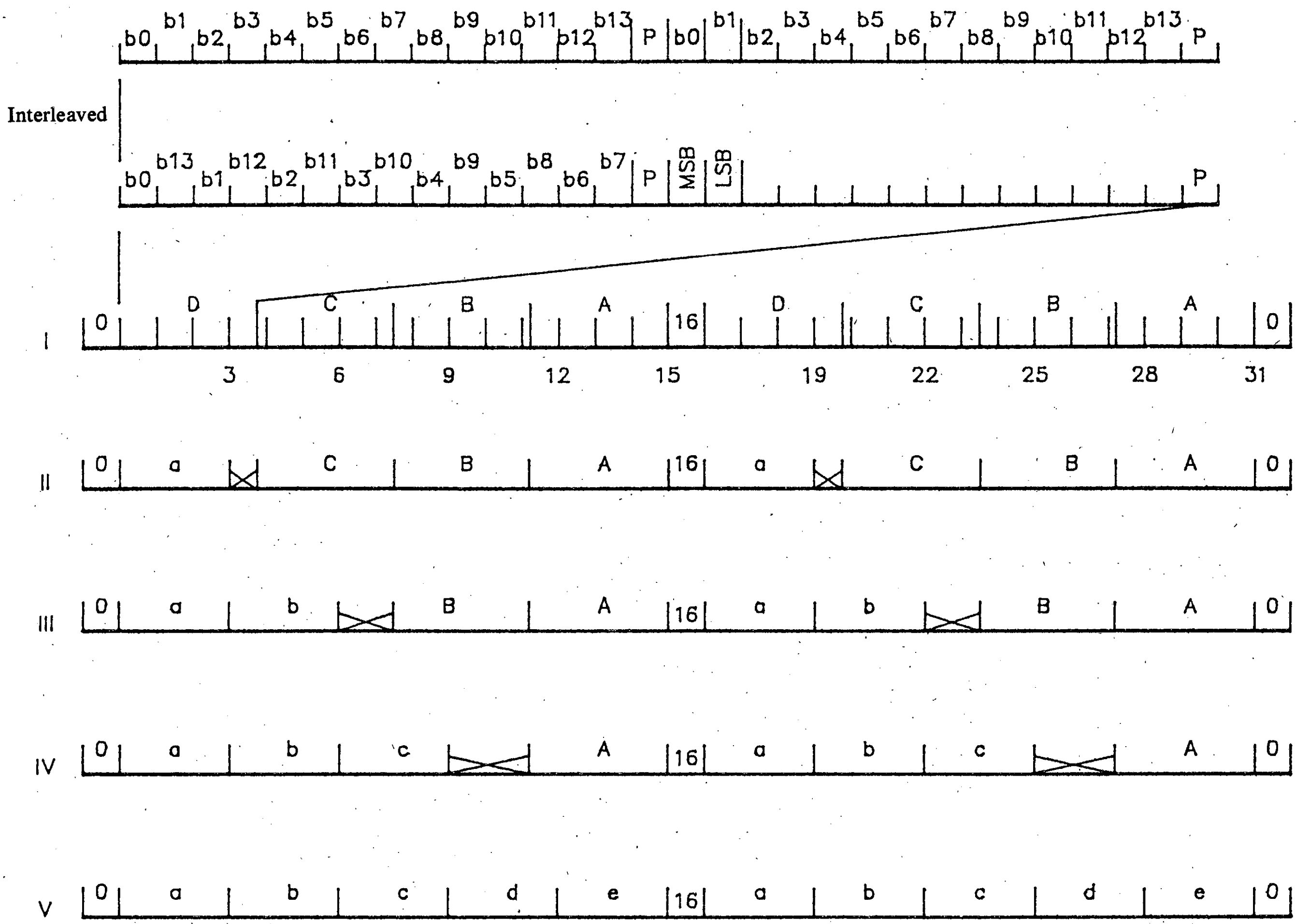


FIGURE 3

A, B, C, D: monophonic channel 480 kbit/s
a, b, c, d, e: monophonic channel 384 kbit/s
MSB: most significant bit
LSB: least significant bit
: not usable for sound transmission

Two monophonic channels of the same coding scheme can be combined to form a stereophonic channel. In some realized systems for 384 kbit/s channels only the combinations a, b and c, d are possible to form stereophonic channels.

REFERENCES

CHAMBERS, J. P. [1985] Signalling in parity: a brief history. British Broadcasting Corporation, BBC RD 1985/15.

ANNEX II

DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNAL ON DISTRIBUTION CIRCUITS USING 496 kbit/s PER AUDIO CHANNEL

1. Coding characteristics

1.1 *Sampling frequency*

See Annex I.

1.2 *Coding method*

See Annex I.

1.3 *Sample error-protection*

See Annex I.

1.4 *Ancillary data*

12 kbit/s per channel are available for transmission of ancillary data.

1.5 *Signalling in parity* [Chambers, 1985]

Signalling in parity is achieved by transmitting the parity bits of an odd number of successive samples without inversion or after inversion, depending on the bit to be signalled. Inversion has to be done, if the bit to be transmitted is "1". The signalling technique used for the scale factor is based on majority-decision logic at the receiver: for each channel, it processes three groups of twenty-one consecutive samples to recognize simultaneously the odd or even parity of the sample and the data signalled in parity (see Fig. 4).

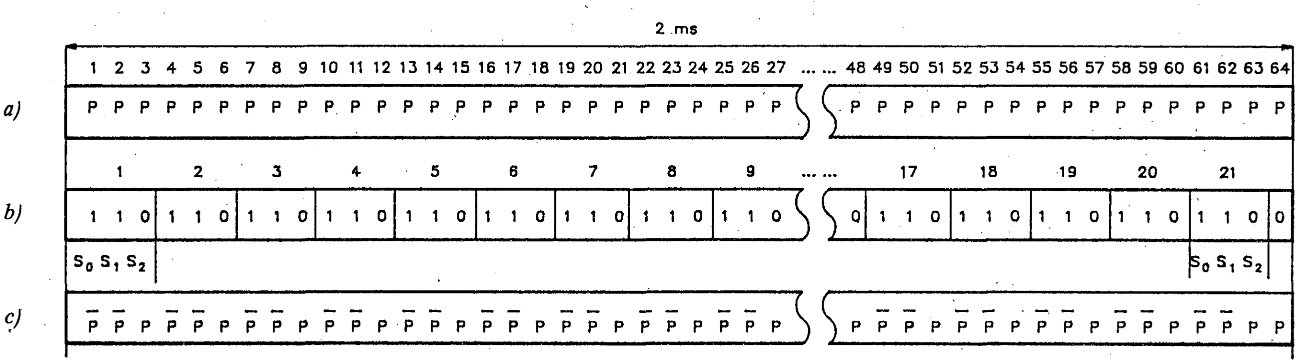


FIGURE 4 – *Signalling in parity coding for one audio channel*

a) Parity bits (P denotes the parity bit associated to each individual audio sample)
b) Scale factor bits
c) Modified parity bits

1.6 *Synchronization*

4 kbit/s are used for synchronization of the floating point companding block.

1.7 *Total bit rate*

With the parameters mentioned above the total bit rate required for one monophonic channel is 496 kbit/s [32 kHz × (14 +1) bits + 12 kbit/s + 4 kbit/s].

2. Frame structure for transmission

The frame structure for transmission is based on an interface operating at 1024 kbit/s.

Two of the 496 kbit/s channels are combined with a frame alignment signal to form a multiplexed signal of 1024 kbit/s, which constitutes the bit rate of the interface. The structure of the frame is almost the same as that specified in CCITT Recommendation G.704 for the primary hierarchical level of 2048 kbit/s. It should be noted that the frame repetition rate is 4 kHz instead of 8 kHz. The frame format is shown in Fig. 5.

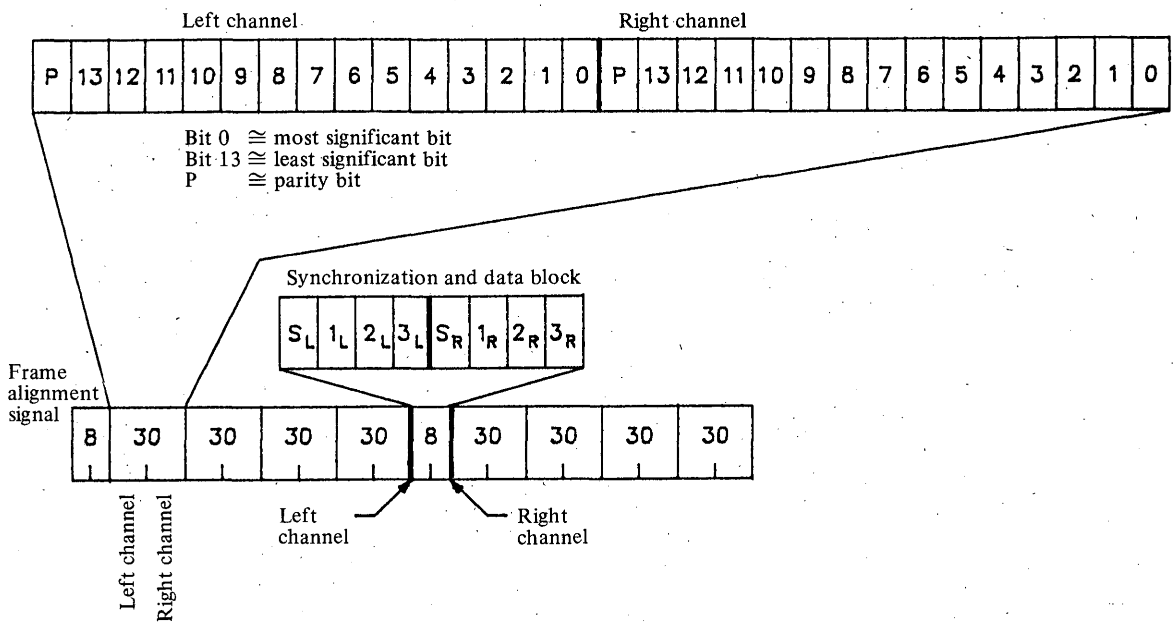


FIGURE 5 – *Frame for the 1024 kbit/s signal with a length of 256 bits and a 4 kHz repetition rate*

S_L, S_R: synchronization signal for floating point companding

1_L, 2_L, 3_L, 1_R, 2_R, 3_R: ancillary data signal

For each channel, the synchronization of the floating point companding block (64 samples carried in 8 frames) is achieved by means of an 8-bit synchronization word.

Two synchronization words are defined:

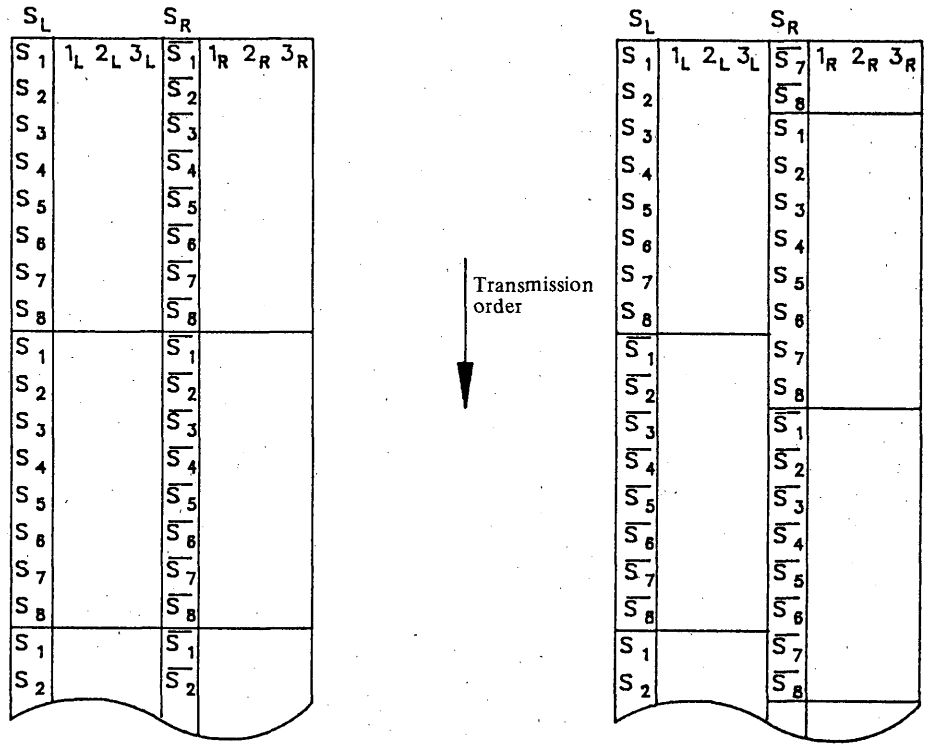
$$S_y = \underset{S_1 \ldots\ldots S_8}{00011011}$$

and the inverted form

$$\overline{S_y} = \underset{\overline{S_1} \ldots\ldots \overline{S_8}}{11100100}$$

For a stereophonic pair the synchronization word used is S_y for the left channel and $\overline{S_y}$ for the right channel.

For monophonic applications, the synchronization word used for each channel is alternatively S_y and $\overline{S_y}$ (see Fig. 6).



a) Stereophonic pair

Note – The relation between S_L and S_R should be as shown.

b) Two monophonic channels

Note – For monophonic application there is no need for a fixed relation between S_L and S_R.

FIGURE 6 – *Transmission of the synchronization words*

By way of synchronous insertion into a 2048 kbit/s frame structure, it is not necessary to transmit the frame alignment signal (FA) which is contained in the frame structure shown in Fig. 5.

Thus, for one channel the net bit rate for transmission is still 496 kbit/s.

In cases where dedicated 2048 kbit/s links are available, up to four 496 kbit/s channels can be combined in one multiplex frame according to CCITT Recommendation G.704.

In order to ensure compatibility between the signals transmitted at 496 kbit/s according to this Annex II with those transmitted at 384 kbit/s according to CCITT Recommendation G.737, the available transmission capacity should be allocated as shown in Fig. 7 [CCIR, 1982-86a].

In Fig. 7 each block L I, R I, L II, R II comprises 60 bits corresponding to a capacity equivalent to 4 samples. Each block results from bit- and sample-interleaving over two consecutive Recommendation G.704 frames.

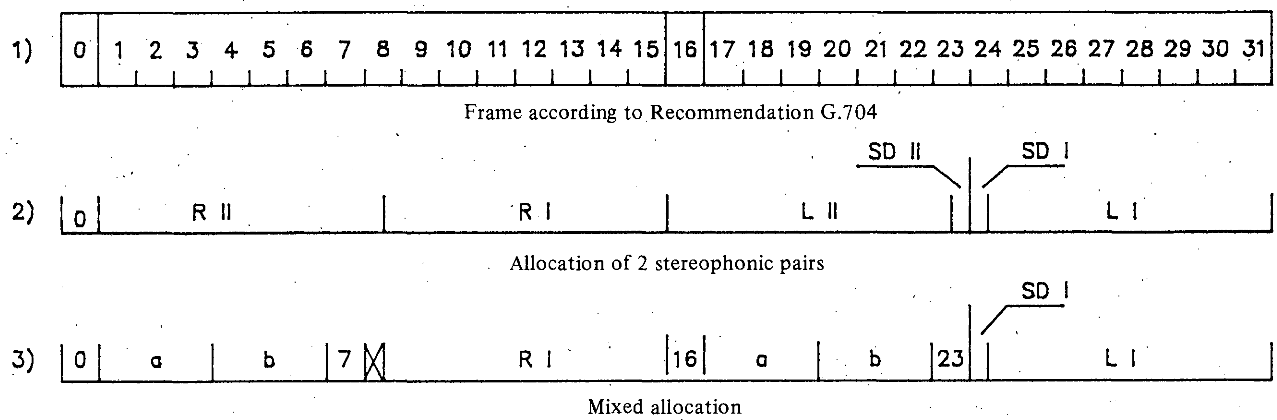


FIGURE 7 — *Allocation of available transmission capacity*

L I: 7.5 × 64 kbit/s = 480 kbit/s for left channel I
R I: 480 kbit/s for right channel I
L II: 480 kbit/s for left channel II
R II: 480 kbit/s for right channel II

SD I: 0.5 × 64 kbit/s = 32 kbit/s, to be divided into 2 × 16 kbit/s.
In the frames containing the frame alignment word the data capacity is allocated to the left channel I, in the other frames to right channel I. In each case 4 kbit/s are used for synchronization of the floating point companding block and 12 kbit/s are used for the transmission of ancillary data

SD II: as SD I but for channels L II and R II

a, b: each 2 × 3 × 64 kbit/s = 384 kbit/s monophonic channel

: capacity not usable (32 kbit/s) in this combination

Note — In line 3) above time slots 7 and 23 are available for telephone; the signalling channel (time slot 16) is not occupied by a sound channel.

REFERENCES

CHAMBERS, J. P. [1985] Signalling in parity: a brief history. British Broadcasting Corporation, BBC RD 1985/15.

CCIR Documents

[1982-86]: **a.** CMTT/214 (Germany (Federal Republic of)).

RECOMMENDATION 719*

TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME ANALOGUE SIGNALS OVER MIXED ANALOGUE/DIGITAL CIRCUITS AT 320 kbit/s

(Question 18/CMTT, Study Programmes 18A/CMTT, 18C/CMTT, 18D/CMTT, 18E/CMTT and 18J/CMTT)

(1990)

The CCIR,

CONSIDERING

a) that the high-quality transmission of sound-programme signals over mixed analogue/digital circuits must meet the requirements of Recommendation 505;

b) that the number of analogue-to-digital and digital-to-analogue conversions and likewise the number of coding methods should be minimized to ensure that, in an international link, the sound-programme signals transmission quality is the same as or better than that obtained in an exclusively analogue link;

c) that the 320 kbit/s rate is sufficient for high-quality sound-programme signal transmission and can, if necessary, interface with the H0 (384 kbit/s) channel recommended by the CCITT;

d) that the digital conversions using the same coding law should not introduce sound-programme signal distortions in digital transits,

UNANIMOUSLY RECOMMENDS

1. that the transmission of sound-programme signals over 320 kbit/s channels with analogue terminals at the input and output of the international link should comply with the criteria for analogue and digital parameters contained in Parts 1 and 2 of this Recommendation;

2. that digital interfaces between administrations which have adopted different systems should operate at 384 kbit/s (H0 channel) and carry signals encoded according to Recommendation 660, § 2.2, method a). Any necessary transcoding will be done by administrations using the system specified in this Recommendation.

PART 1

CRITERIA FOR ANALOGUE PARAMETERS

The transmission of high-quality sound-programme signals over mixed analogue/digital circuits must meet the requirements of Recommendation 505.

1. Codec parameters

The parameters of a codec between its analogue terminals must meet the requirements to be defined as one third of the requirements for the hypothetical reference circuit (Recommendation 505). In calculating the standard parameters of a codec between the analogue terminals, the summation laws given in Recommendation 605 may be applied, with $n = 3$ as the number of low-frequency transit sections.

2. Parameters of codec series connection

The hypothetical reference circuit described in Recommendation 502 is divided into three sections of equal length, each of which may be either analogue or digital. Each digital section must contain not more than one coder and decoder. Hence the number of series-connected codecs in a mixed hypothetical reference circuit comprising both analogue and digital sections is established in accordance with the figures in Table I.

* This Recommendation is of interest to CCITT Study Groups XV and XVIII.

TABLE I – *Number of series-connected codecs in a mixed analogue/digital hypothetical reference circuit*

Number of analogue sections	Number of series connected codecs
0	3
1	2
2	1
3	0

PART 2

CRITERIA FOR DIGITAL PARAMETERS

The digital transits of sound-programme signals converted by the method described in this Recommendation (see § 2, below) must not introduce any analogue signal distortions. In the digital transits of sound-programme signals converted by various methods (e.g., those described in Recommendation 660 and the present Recommendation), distortions must be minimized.

1. Sampling frequency

The sampling frequency shall be 32 kHz. The sampling frequency tolerance shall be $\pm 5 \times 10^{-5}$, as specified in CCITT Recommendations G.732 and G.733 for primary PCM multiplex equipment. This sampling frequency is consistent with that indicated in Recommendation 606.

2. Coding method

The recommended coding characteristic is based on uniform quantization using 14 bits on sampling followed by companding.

Companding is carried out in three stages, as defined below:

a) quasi-instantaneous companding according to a five-range characteristic with compression of 14 to 10 bits, as shown in Fig. 1;

b) division of the samples *x(n)* into two sequences – odd $x(2n + 1)$ and even $x(2n)$ samples and calculation of the difference $\Delta(2n)$ using the formula:

$$\Delta(2n) = x(2n) - \frac{x(2n + 1) + (2n - 1)}{2}$$

c) additional quasi-instantaneous companding of the difference $\Delta(2n)$ according to a 3-range characteristic and compression of 11 to 9 bits, as shown in Fig. 2.

In digital links between administrations using different companding methods, the signals must be coded by method a) of Recommendation 660. The necessary code conversion shall be carried out by an administration using a method different from method a) of Recommendation 660.

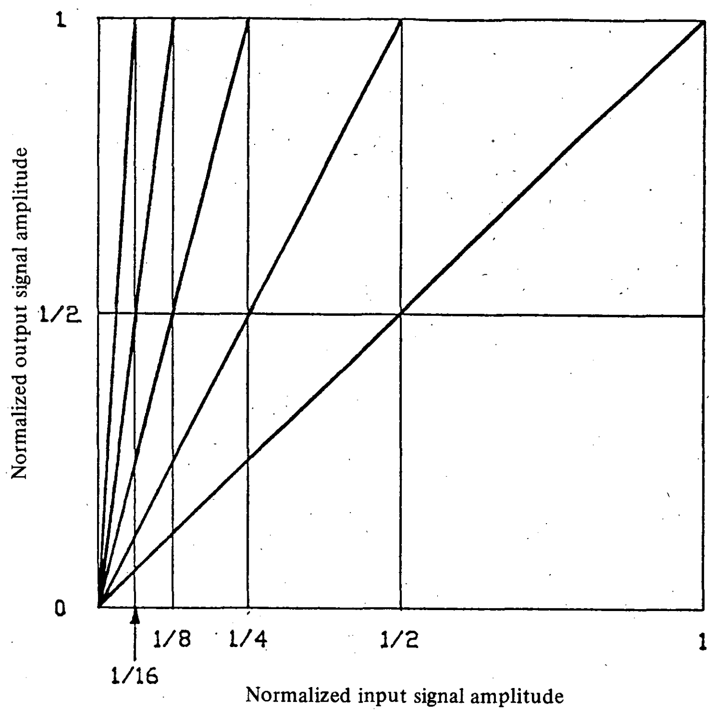


FIGURE 1 – *First stage of companding algorithm*

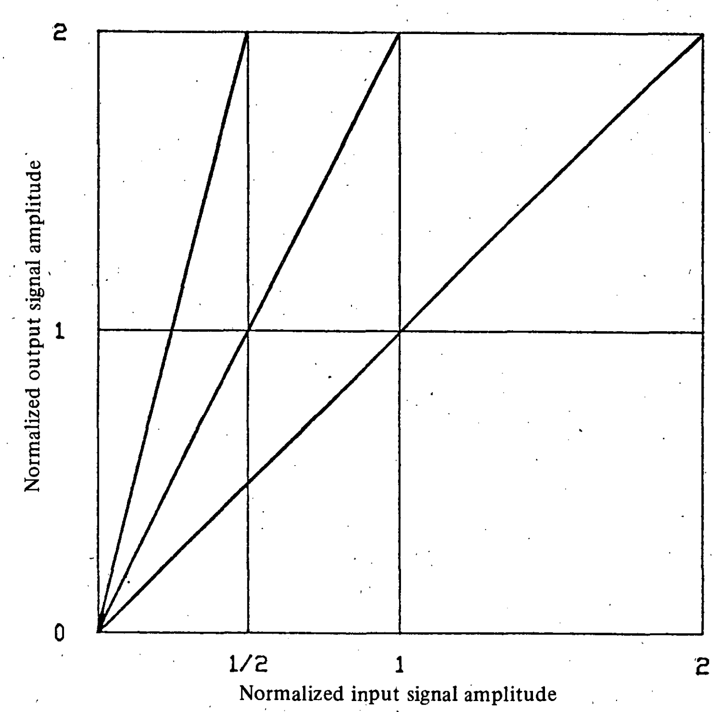


FIGURE 2 – *Third stage of companding algorithm*

3. Coding table

The coding table for quasi-instantaneous 14-10 companding is given in Table II.

The coding table for additional quasi-instantaneous companding of the difference $\Delta(2n)$ 11-9 is given in Table III.

TABLE II – *Coding table for quasi-instantaneous 14-10 companding*

Range	Standard analogue input	Standard analogue output		Compressed digital code		Resolution capacity (bit)
4	+8176 0 −16 −8192	+8192 +16 0 −8176	+8184 +8 −8 −8184	+511 0 −1 −512	(0111111111) (0000000000) (1000000000) (1111111111)	10
3	+4088 0 −8 −4096	+4096 +8 0 −4088	+4092 +4 −4 −4092	+511 0 −1 −512	(0111111111) (0000000000) (1000000000) (1111111111)	11
2	+2044 0 −4 −2048	+2048 +4 0 −2044	+2046 +2 −2 −2046	+511 0 −1 −512	(0111111111) (0000000000) (1000000000) (1111111111)	12
1	+1022 0 −2 −1024	+1024 +2 0 −1022	+1023 +1 −1 −1023	+511 0 −1 −512	(0111111111) (0000000000) (1000000000) (1111111111)	13
0	+511 0 −1 −512	+512 +1 0 −511	+511.5 +0.5 −0.5 −511.5	+511 0 −1 −512	(0111111111) (0000000000) (1000000000) (1111111111)	14

TABLE III – *Coding table for additional quasi-instantaneous 11-9 diversity companding*

Range		Standard analogue input	Standard analogue output		Compressed digital code		Resolution capacity (bit)
	2	+16320	+16384	+16352	+255	(0111111111)	8
		0	+64	+32	0	(0000000000)	
		−64	0	−32	−1	(1000000000)	
		−16384	−16320	−16352	−256	(1111111111)	
4	1	+8160	+8192	+8176	+255	(0111111111)	9
		0	+32	+16	0	(0000000000)	
		−32	0	−16	−1	(1000000000)	
		−8190	−8160	−8176	−256	(1111111111)	
	0	+4080	+4096	+4088	+255	(0111111111)	10
		0	+16	+8	0	(0000000000)	
		−16	0	−8	−1	(1000000000)	
		−4096	−4080	−4088	−256	(1111111111)	
	2	+8160	+8192	+8176	+255	(0111111111)	9
		0	+32	+16	0	(0000000000)	
		−32	0	−16	−1	(1000000000)	
		−8192	−8160	−8176	−256	(1111111111)	
3	1	+4080	+4096	+4088	+255	(0111111111)	10
		0	+16	+8	0	(0000000000)	
		−16	0	−8	−1	(1000000000)	
		−4096	−4080	−4088	−256	(1111111111)	
	0	+2040	+2048	+2044	+255	(0111111111)	11
		0	+8	+4	0	(0000000000)	
		−8	0	−4	−1	(1000000000)	
		−2048	−2040	−2044	−256	(1111111111)	
	2	+4080	+4096	+4088	+255	(0111111111)	10
		0	+16	+8	0	(0000000000)	
		−16	0	−8	−1	(1000000000)	
		−4096	−4080	−4088	−256	(1111111111)	
2	1	+2040	+2048	+2044	+255	(0111111111)	11
		0	+8	+4	0	(0000000000)	
		−8	0	−4	−1	(1000000000)	
		−2048	−2040	−2044	−256	(1111111111)	
	0	+1020	+1024	+1022	−255	(0111111111)	12
		0	+4	+2	0	(0000000000)	
		−4	0	−2	−1	(1000000000)	
		−1024	−1020	−1022	−256	(1111111111)	
	2	+2040	+2048	+2044	+255	(0111111111)	11
		0	+8	+4	0	(0000000000)	
		−8	0	−4	−1	(1000000000)	
		−2048	−2040	−2044	−256	(1111111111)	
1	1	+1020	+1024	+1022	+255	(0111111111)	12
		0	+4	+2	0	(0000000000)	
		−4	0	−2	−1	(1000000000)	
		−1024	−1020	−1022	−256	(1111111111)	
	0	+510	+512	+511	+255	(0111111111)	13
		0	+2	+1	0	(0000000000)	
		−2	0	−1	−1	(1000000000)	
		−512	−510	−511	−256	(1111111111)	
	2	+1020	+1024	+1022	+255	(0111111111)	12
		0	+4	+2	0	(0000000000)	
		−4	0	−2	−1	(1000000000)	
		−1024	−1020	−1022	−256	(1111111111)	
0	1	+510	+512	+511	+255	(0111111111)	13
		0	+2	+1	0	(0000000000)	
		−2	0	−1	−1	(1000000000)	
		−512	−510	−511	−256	(1111111111)	
	0	+2555	+256	+255.5	+255	(0111111111)	14
		0	+1	0.5	0	(0000000000)	
		−1	0	−0.5	−1	(1000000000)	
		−256	−255	−255.0	−256	(1111111111)	

4. Error protection

4.1 *Error detection*

Error detection must be used for the identification of distorted sound-programme signal samples occurring as a result of random digital errors, error bursts, loss of frame alignment or frame slip.

4.2 *Error masking and correction*

Error masking shall be used to maintain the subjective quality of the sound-programme signal at a level of not less than 4.5 on the quality scale shown in Recommendation 562. This quality shall be ensured at the random digital error ratio of 10^{-6} given in CCITT Recommendation G.821.

Further study is required to establish the performance objectives for error bursts.

4.3 *Recovery of frame alignment*

Loss of frame alignment and subsequent realignment in digital systems causes interruptions of sound-programme signal transmission. Further study is required on the permissible limit values of such interruptions.

5. Jitter

Jitter occurring in the digital path in the transmission and asynchronous interfacing of digital signals may cause distortions of the decoded analogue signal. As is stated in Report 648, this point calls for further study.

6. Bit rate

6.1 *Monophonic transmission*

The bit rate for one sound-programme channel is 320 kbit/s, including all necessary additional bits.

6.2 *Stereophonic transmission*

Two 320 kbit/s channels are required to establish a stereophonic sound-programme circuit (stereo-pair). Both channels must be included in one digital transmission path in order to avoid transmission time differences.

7. Connection to the network

Sound-programme signals coded in 320 kbit/s digital channels must be interfaced with with the ISDN at the primary stage at a rate of 2048 kbit/s or 1544 kbit/s.

RECOMMENDATION 660*

TRANSMISSION OF ANALOGUE HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON MIXED ANALOGUE-AND-DIGITAL CIRCUITS USING 384 kbit/s CHANNELS

(Question 18/CMTT, Study Programmes 18A/CMTT, 18C/CMTT, 18D/CMTT and 18E/CMTT)

(1986)

The CCIR,

CONSIDERING

(a) that high-quality sound-programme signals on mixed analogue-and-digital circuits must satisfy the performance requirements of Recommendation 505;

(b) that the number of analogue-to-digital and digital-to-analogue conversions as well as the number of digital coding techniques should be kept to a minimum, in order for the international sound programme connection to provide the same or higher degree of performance than possible with a homogeneous analogue connection;

(c) that the CCITT has recommended a 384 kbit/s H0 bearer channel in Recommendation I.412 which has the capacity to transport sound-programme signals on an integrated services digital network (ISDN);

(d) that digital-to-digital conversions using the same encoding law on tandem sound-programme circuits with different primary rates should not introduce any signal degradation,

UNANIMOUSLY RECOMMENDS

that the transmission of sound-programme signals on 384 kbit/s channels, with an analogue interface at the input and output of the international connection, should satisfy the analogue-and-digital requirements defined in § 1 and 2 respectively of this Recommendation.

Note 1 – Equipment characteristics for the coding of analogue high-quality sound-programme signals using 384 kbit/s channels have been defined in CCITT Recommendation J.41. Characteristics for equipment offering digital access at 384 kbit/s into 2048 kbit/s are given in CCITT Recommendations G.735 and G.737.

Note 2 – Other coding techniques are proposed and are the subject of Question 18/CMTT, for example, coding of analogue sound-programme signals at 316 kbit/s, insertion into a 320 kbit/s channel and multiplexing of six high-quality sound-programme channels within 2048 kbit/s. Techniques which may be used by bilateral agreement of the administrations concerned are listed in Table I of Report 647.

1. Analogue requirements

Transmission of high-quality sound-programme signals on mixed analogue-and-digital circuits shall satisfy the performance requirements of Recommendation 505.

1.1 *Single codec performance*

The analogue-to-analogue transmission performance of a single codec shall be equivalent to one-third the performance characteristics of the hypothetical reference circuit (see Recommendation 505). Where applicable, the laws of addition given in Recommendation 605 can be used to calculate the single codec performance with $n = 3$ analogue-to-analogue sections.

* This Recommendation should be brought to the attention of CCIR Study Group 10 and CCITT Study Groups XV and XVIII.

1.2 *Tandem codec performance*

The hypothetical reference circuit (HRC) given in Recommendation 502 is divided into three sections of equal length which could be either analogue or digital. Each digital section should only contain one encoder and one decoder. Therefore, in a mixed analogue-and-digital HRC, Table I indicates the number of codecs which may be connected in tandem.

TABLE I – *Number of tandem codecs on a mixed analogue-and-digital sound programme circuit*

Analogue sections	Tandem codecs
0	3
1	2
2	1
3	0

2. Digital requirements

Digital-to-digital conversions using the same encoding law as given in Table II or Table III of this Recommendation on tandem sound-programme circuits with different primary rates should not introduce any signal degradation. Digital-to-digital conversions using different encoding laws, as given in Tables II and III, should cause minimum signal degradation.

TABLE II – *Coding table for instantaneous companding of sound-programme signals* *

Normalized analogue input (¹)	Normalized analogue output (¹)	Compressed digital code	Segment No.	Effective resolution (bits)
8160 to 8192 4096 to 4128	8176 4112	895 768	1	9
4080 to 4096 2048 to 2064	4088 2056	767 640	2	10
2040 to 2048 1024 to 1032	2044 1028	639 512	3	11
1020 to 1024 512 to 516	1022 514	511 384	4	12
510 to 512 256 to 258	511 257	383 256	5	13
255 to 256 128 to 129	255.5 128.5	255 128	6	14
127 to 128 0 to 1	127.5 0.5	127 0		

* The positive half of the coding table only is shown.

(¹) The top code word of ± 8192 corresponds to an overload level of +15 dBm0s at the 0 dB insertion loss frequency (2.1 kHz) of the CCITT Recommendation J.17 pre-emphasis circuit with 6.5 dB loss at 800 Hz.

TABLE III – *Coding table for near-instantaneous companding of sound-programme signals*

Range	Normalized analogue input ([1])	Normalized analogue output ([1])	Compressed digital code	Effective resolution (bits)
4	+8176 to +8192 0 to +16 −16 to 0 −8192 to −8176	+8184 +8 −8 −8184	+511 0 −1 −512	10
3	+4088 to +4096 0 to +8 −8 to 0 −4096 to −4088	+4092 +4 −4 −4092	+511 0 −1 −512	11
2	+2044 to +2048 0 to +4 −4 to 0 −2048 to −2044	+2046 +2 −2 −2046	+511 0 −1 −512	12
1	+1022 to +1024 0 to +2 −2 to 0 −1024 to −1022	+1023 +1 −1 −1023	+511 0 −1 −512	13
0	+511 to +512 0 to +1 −1 to 0 −512 to −511	+511.5 +0.5 −0.5 −511.5	+511 0 −1 −512	14

([1]) The top code word of ± 8192 corresponds to an overload level of +12 dBm0s at the 0 dB insertion loss frequency (2.1 kHz) of the CCITT Recommendation J.17 pre-emphasis circuit with 6.5 dB loss at 800 Hz.

2.1 *Sampling frequency*

The sampling frequency shall be 32 kHz. The associated tolerance shall be $\pm 5 \times 10^{-5}$ as specified in CCITT Recommendations G.732 and G.733 for primary PCM multiplex equipment. This sampling frequency conforms with that given in Recommendation 606.

2.2 *Encoding method*

The recommended encoding laws are based on a uniformly quantized 14-bit per sample PCM technique with companding and shall employ either method (a) or method (b) with the appropriate rules of precedence as defined below:

(a) eleven-segment 14-to-11 bit instantaneous A-law companding. The companding characteristic is illustrated in Fig. 1;

(b) five-range 14-to-10 bit near-instantaneous companding. The companding characteristic is illustrated in Fig. 2.

Digital paths between administrations which have adopted different systems should carry signals encoded according to method (a). Where both administrations have adopted the same method, that method should be used on digital paths between them. Any necessary transcoding will be done by administrations using method (b).

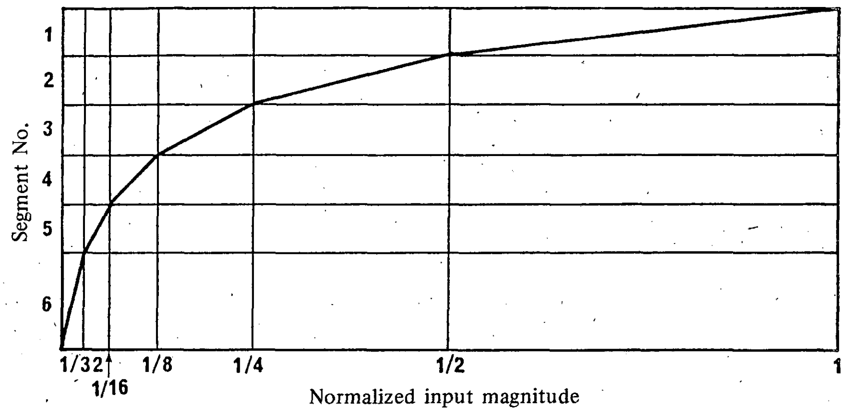


FIGURE 1 – *Instantaneous companding characteristics*

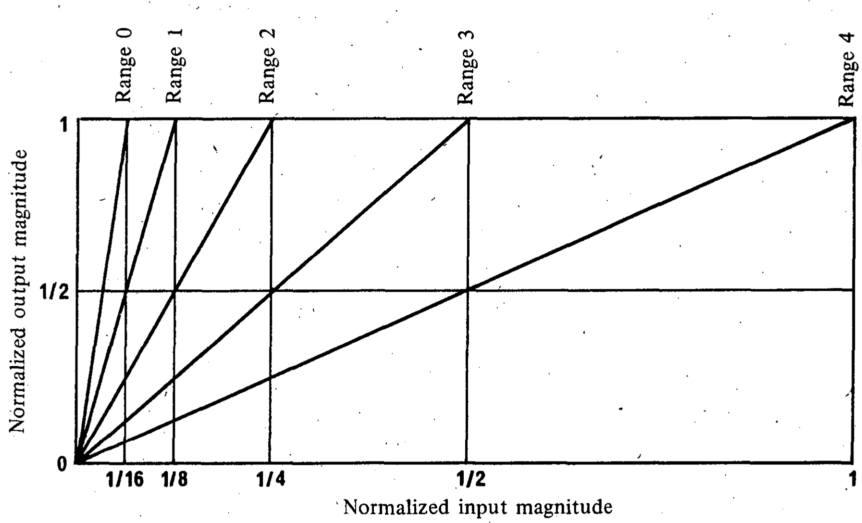


FIGURE 2 – *Near-instantaneous companding characteristics*

2.3 *Coding table*

The coding table for the instantaneous A-law compandor is specified in Table II and the coding table for the near-instantaneous compandor is specified in Table III.

2.4 *Bit error protection*

2.4.1 *Error detection*

Error detection shall be used to detect erroneous sound programme samples which can be caused by random bit errors, burst errors, frame loss or frame slips.

2.4.2 *Error concealment and correction*

Error concealment or correction techniques shall be used to ensure that the subjective quality of the sound-programme signal is not degraded below grade 4.5 as defined in Recommendation 562. This quality shall be maintained for a random bit error ratio of 10^{-6} as defined in CCITT Recommendation G.821. Further study is necessary to define a requirement for burst errors (see Report 648).

2.4.3 *Frame loss/recovery*

Frame loss/recovery on digital channels causes interruptions to sound-programme signals. The tolerable limits for these interruptions require further study (see Reports 642 and 647).

2.5 *Jitter*

Jitter characteristics from the digital interface can produce impairments to the decoded analogue signal after passing through a digital system. This requires further study as outlined in Report 648.

2.6 *Transmission bit rate*

2.6.1 *Monophonic transmission*

The transmission bit rate for one sound-programme channel shall be 384 kbit/s, including all ancillary bits as necessary.

2.6.2 *Stereophonic transmission*

Two separate 384 kbit/s channels shall be used to form a stereophonic (stereo-pair) sound-programme circuit. Both sound-programme channels shall be routed together over the same transmission path to avoid transmission delay differences.

2.7 *Network access*

Sound-programme signals coded into 384 kbit/s digital channels should interface to the ISDN at 1544 kbit/s or 2048 kbit/s as specified in CCITT Recommendation I.412.

RECOMMENDATION 606-1

SAMPLING FREQUENCY TO BE USED FOR THE DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS

(Study Programme 18A/CMTT)

(1982-1990)

The CCIR,

CONSIDERING

(a) the rapid developments in the use of digital systems for the transmission of high-quality sound signals;

(b) that, according to subjective tests, limiting the audio frequency band to 15 kHz does not cause any appreciable degradation in subjective quality, even in critical listening conditions;

(c) that the sampling frequency of 32 kHz is close to the theoretical limit compatible with the nominal passband of 15 kHz;

(d) that the frequency of 32 kHz is already commonly used in a number of equipments;

(e) that this frequency is compatible with the bit rates corresponding to the various hierarchical levels defined by the CCITT;

(f) that the use of a single frequency is calculated to simplify equipment and facilitate exchange,

UNANIMOUSLY RECOMMENDS

that a sampling frequency of 32 kHz shall be used in coding for the digital transmission of high-quality sound channels. The associated tolerance shall be $\pm 5 \times 10^{-5}$ as specified in CCITT Recommendations G.732 and G.733 for primary PCM multiplex equipment, operating at 2048 kbit/s and 1544 kbit/s respectively.

Note – For quality levels of sound programme transmission other than high quality (15 kHz), other sampling frequencies may be preferred. They should always be multiples of 8 kHz.

SECTION CMTT D: METHODS OF OPERATION AND ASSESSMENT OF PERFORMANCE OF SOUND-PROGRAMME TRANSMISSION CHANNELS

RECOMMENDATION 571-2*

A CONVENTIONAL TEST SIGNAL SIMULATING SOUND-PROGRAMME SIGNALS FOR MEASURING INTERFERENCE IN OTHER CHANNELS**

(Study Programme 19D/CMTT)

(1978-1982-1990)

The CCIR,

CONSIDERING

(a) that on FDM systems non-linear cross-talk 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 I and Fig. 1;

2. the conventional test signal can be produced from a Gaussian white noise generator associated with a shaping network conforming with Fig. 2;

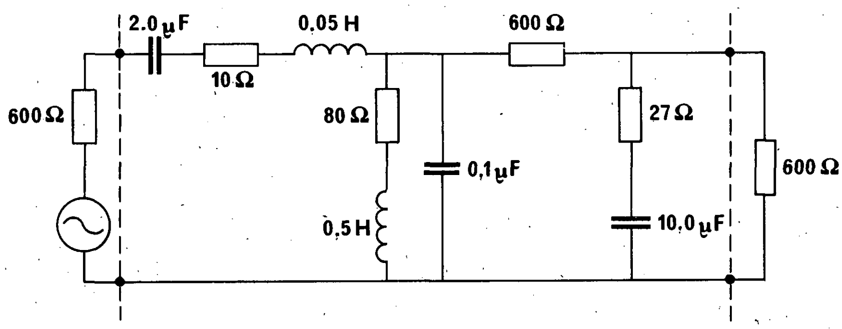
3. the total test signal power applied to a sound-programme circuit under test shall be cyclically changed in level according to Table II.

Note – This Recommendation is derived from studies given in Report 497.

FIGURE 1 – *Insertion-loss/frequency*

* This Recommendation should be brought to the attention of CCITT Study Group XV and CCIR Study Group 10.

** For the definitions of absolute power, relative power and noise levels, see Recommendation 574.



(Non-constant resistance form)

FIGURE 2

TABLE I

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
1000	9.2	0.3
2000	10.6	0.5
3150	13.0	0.5
4000	15.7	0.5
5000	18.8	0.5
6300	22.5	0.5
7100	24.6	0.5
8000	26.6	0.5
9000	28.6	0.5
10000	30.4	1.0
12500	34.3	1.0
14000	36.3	1.0
16000	38.6	1.0
20000	42.5	1.0
31500	50.4	1.0

TABLE II

Step	Level	Time for which signal is applied
1	−4 dBm0s	4 s
2	+ 3 dBm0s	2 s
3	no signal	2 s

RECOMMENDATION 645-1 *

TEST SIGNALS TO BE USED ON INTERNATIONAL SOUND-PROGRAMME CONNECTIONS

(Questions 50/10 and 19/CMTT, Study Programmes 50B/10 and 50E/10)

(1986-1990)

The CCIR,

CONSIDERING

(a) that many impairments in international programme exchange on sound-programme connections are attributed to different national test signal definitions;

(b) that some existing definitions are found in different Recommendations of the CCITT and the CCIR;

(c) that for clarification, a list of those definitions should be available,

UNANIMOUSLY RECOMMENDS

that for an international sound-programme connection only the test signals defined below should be used:

1. Alignment signal (AS)

Sine-wave signal at a frequency of 1 kHz, which is used to align the international sound-programme connection. The signal level corresponds to 0 dBu0s (see Note) (i.e. 0.775 V r.m.s. at a zero relative level point). In accordance with CCITT Recommendation N.13, the period of sending the alignment signal should be kept as short as possible – preferably to less than 30 s.

Note – The notation "dBu0s" is defined in Recommendation 574. Other related texts of the CMTT use the notation "dBm0s" also defined in Recommendation 574.

2. Measurement signal (MS)

Sine-wave signal at a level 12 dB below the alignment signal level which should be used for long-term measurements and measurements at all frequencies (see CCITT Recommendations N.12, N.13, N.21 and N.23).

3. Permitted maximum signal (PMS)

Sine-wave signal at 1 kHz, 9 dB above the alignment signal level, equivalent to the permitted maximum programme-signal level. The sound-programme signal should be controlled by the sending broadcaster so that the amplitudes of the peaks only rarely exceed the peak amplitude of the PMS.

Note – Under these conditions a peak programme meter will indicate levels not exceeding the level of the permitted maximum signal.

A numerical example may serve to clarify this definition. The alignment signal has an r.m.s. voltage of 0.775 V and a peak amplitude of 1.1 V at a zero relative level point. The instantaneous peak amplitude of the sound-programme signal at this point should only rarely exceed 3.1 V.

Although it is intended that the peaks of the sound-programme signal should not exceed the permitted maximum signal level, an overload margin must be provided so that rare excursions of the sound-programme signal above the permitted maximum signal level may be tolerated.

Note – Annex I describes the response of peak programme and vu meters to these test signals.

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* The CMTT and Study Group 10 will coordinate the future development of this Recommendation. This Recommendation should be brought to the attention of Study Group IV of the CCITT.

ANNEX I

ALIGNMENT USING THE RECOMMENDED TEST SIGNALS WITH PEAK PROGRAMME METERS AND VU METERS

1. Broadcasters have evolved, over a period of forty years, procedures for using both types of meter to control programme levels. These procedures are satisfactory to the organizations using them, so that they produce neither over-modulation, leading to distortion, nor under-modulation, leading to impairment from noise.

Although different kinds of programme material deflect the two meters differently, the organizations using them have evolved techniques that produce satisfactory level control and artistic balance within the programme.

2. The sensitivity of peak programme meters (PPM) is such that a sine wave signal at the alignment level, 0 dBu0s, indicates "Test" on an EBU PPM (this corresponds to "4" on the BBC PPM and "-9" on the PPMs of the Federal Republic of Germany and the OIRT (see Fig. 1)).

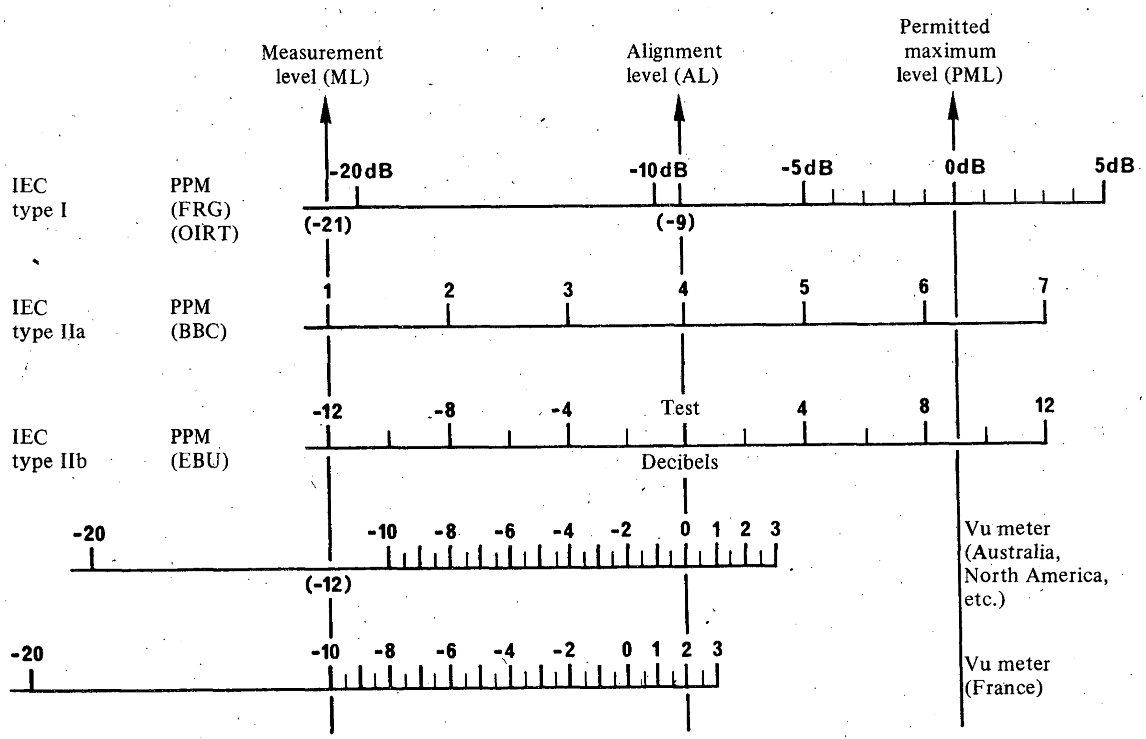

FIGURE 1 – *Indications produced by various types of programme meter with the recommended test signals*

Note. – Meter indications are schematic – not to scale.

3. The sensitivity of the vu meter is such that a sine wave signal at the alignment level, 0 dBu0s, produces nearly full indication, 0 vu in Australia and North America and +2 vu in France (see Fig. 1).

4. The PPM reads "quasi-peak", that is, its peak indication on programme signals reads a little lower than true peaks. Operators are instructed to make the programme peaks give the same indication as a sinusoidal tone at +9 dBu0s (+8 dBu0s in some organizations). The true peaks of the programme are higher than indicated by up to 3 dB. When, additionally, operator errors are taken into account, the true peaks of the programme signal may reach the amplitude of a sinusoidal tone at +15 dBu0s.

5. The vu meter indicates the mean level of the programme, which is generally much lower than the true peak. Operators are instructed to make programmes peak generally to the 0 vu reading. Experience has shown that the true programme peaks are higher than indicated by between +6 dB and +13 dB, depending on the programme material. When, additionally, operator errors are taken into account, the true peaks of the signal may be up to 16 dB higher than indicated, corresponding to the peak amplitude of a sinusoidal tone at +16 dBu0s, or alternatively +14 dBu0s when application of the alignment level signal results in +2 vu indication.

6. Thus, although the dynamic characteristics of the two meters are different, the highest peak levels encountered after control using either meter are very similar.

7. Thus, an international connection between broadcasters will be correctly aligned regardless of the type of meter employed when a sinusoidal signal at alignment level, 0 dBu0s, produces the indication appropriate to that level at both the sending and receiving ends of the circuit.

To avoid any confusion between alignment level and other levels that might be used, it is recommended that the three level tone test signal described in Recommendation 661 be used for the alignment of an international sound-programme connection.

Figure 1 illustrates the indications given by a number of programme level meters when the recommended test signals are applied to them.

RECOMMENDATION 661-1 *

SIGNALS FOR THE ALIGNMENT OF INTERNATIONAL SOUND-PROGRAMME CONNECTIONS

(Question 19/CMTT)

(1986-1990)

The CCIR,

CONSIDERING

(a) that Recommendation 645 defines three test signals to be used on international sound-programme connections;

(b) that no level information can be derived from a single-level test signal concerning its relationship to the levels defined in Recommendation 645;

(c) that many impairments in international programme exchanges can be traced to misinterpretations of the single-level test signals;

(d) that the test signals in Recommendation 645 could be used to align sound-programme commentary connections on the switched telephone network which cannot accomodate high level sinusoidal test signals,

UNANIMOUSLY RECOMMENDS

that international sound-programme connections should be identified and aligned using the definitions in § 1, the test signal format in § 2 and the measurement methods in § 3 of this Recommendation.

1. Definitions

1.1 *Source identification*

An announcement should be used to identify the originating point of the test signals and should be preferably as short as possible. It is suggested that such an announcement contain at least the following information:

- name of originating organization,
- location,
- country.

The sound-programme signal should be controlled by the sending broadcaster so that the amplitudes of the peaks only rarely exceed the peak amplitude of the permitted maximum (sine-wave test) signal.

1.2 *Test signal and level definitions*

1.2.1 *Alignment signal (AS)*

Sine-wave signal at 1 kHz** at a level of 0 dBm0s, which is used to align the international sound-programme connection.

1.2.2 *Measurement signal (MS)*

Sine-wave signal at 1 kHz** at a level 12 dB below the alignment signal level, which should be used for long-term measurements and measurements at all frequencies.

1.2.3 *Permitted maximum signal (PMS)*

Sine-wave signal at 1 kHz**, 9 dB above the alignment signal level, equivalent to the permitted maximum programme-signal level.

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* This Recommendation should be brought to the attention of CCIR Study Group 10 and CCITT Study Group IV.

** This frequency is nominal, and 1020 Hz recommended by CCITT Recommendation O.33 may be used.

2. Test signal format

2.1 A three-level sinusoidal test signal at a reference frequency of 1 kHz* should be used to check the alignment of international sound-programme connections. These three levels should be combined with the source identification and be repeated cyclically as specified in the format shown in Fig. 1 for monophonic and stereophonic connections.

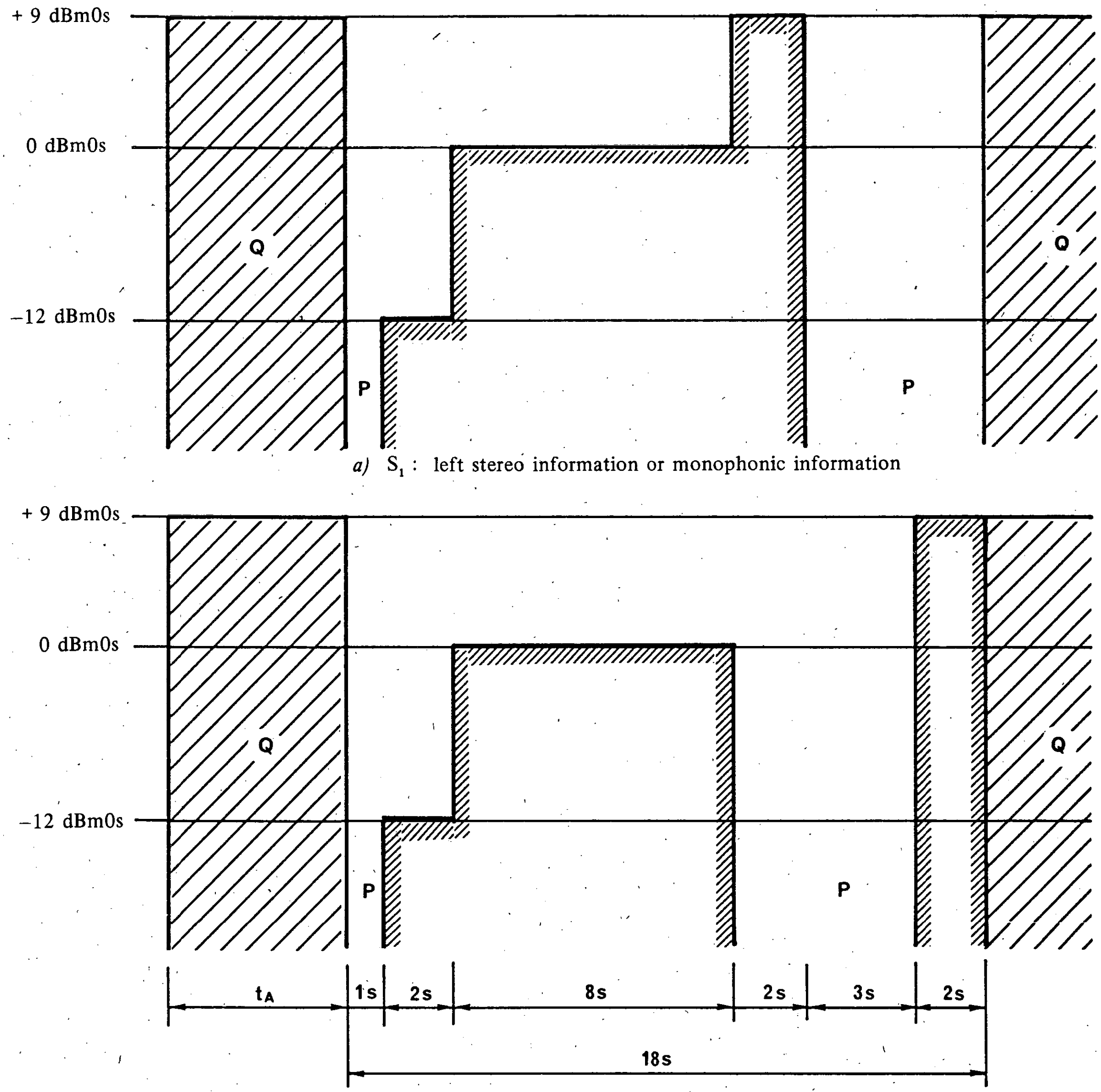

FIGURE 1 – *Format for the three-level test signal for sound programme connections*

Cycle duration = t_A + 18 s

Q: station announcement

t_A: duration of the station announcement. It varies depending on the length of the message

P: signal pauses

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* This frequency is nominal, and 1020 Hz recommended by CCITT Recommendation O.33 may be used.

2.2 These test signals should not be connected directly to the switched telephone network as excessive channel loading or crosstalk into other channels may be produced.

2.3 Some organizations may not have automatically generated test levels as defined in § 2.1. In these cases, the alignment level of 0 dBm0s at 1 kHz (see Note) should be used for the alignment of international sound-programme connections.

Note – This frequency is nominal, and 1020 Hz recommended by CCITT Recommendation O.33 may be used.

3. Measurement methods

The fundamental concept of the test signals defined in this Recommendation is to provide organizations with accurate and well-defined levels [Thiele, 1984]. These levels are intended to provide rapid identification of level errors as well as to allow operational personnel sufficient time to make the necessary level adjustments at the appropriate points in the international sound-programme connection. Alignment of the connection is made by adjusting the alignment signal to the appropriate point on the programme level meter as defined in Annex I to Recommendation 645. Identification of left and right channels is provided as shown in Fig. 1. Organizations should give consideration to Reports 292 and 820 when establishing measurement procedures.

The three-level test signal defined in § 2.1 makes provision for a brief subjective and/or objective noise measurement in the signal pauses (P) as shown in Fig. 1. These measurement pauses are not intended to replace the maintenance practices defined in the CCITT Series N Recommendations but rather to confirm that there are no gross noise or cross-talk impairments on the circuit.

REFERENCES

THIELE, A. N. [September, 1984] Three-level-tone test signal for setting audio levels. AES Australian Convention, Melbourne, Australia.

SECTION CMTT E: TRANSMISSION OF SIGNALS WITH MULTIPLEXING OF VIDEO, SOUND AND DATA, AND SIGNALS OF NEW SYSTEMS

RECOMMENDATION 717*

TOLERANCES FOR TRANSMISSION TIME DIFFERENCES BETWEEN THE VISION AND SOUND COMPONENTS OF A TELEVISION SIGNAL

(Study Programme 21A/CMTT)

(1990)

The CCIR,

CONSIDERING

a) that a perceptible time difference between the sound and vision components of a television signal impairs the viewer's enjoyment of the programme;

b) that Report 1081 gives figures for time differences between sound and vision components that are "detectable" and "subjectively annoying";

c) that the effect is less critical when the sound is delayed with respect to the picture;

d) that transmission links are not the only elements in the broadcast chain that may cause a time difference between sound and vision components;

e) that the time difference between sound and vision components introduced by a transmission link is not necessarily related to the lenght of the link,

UNANIMOUSLY RECOMMENDS

that for any connection used for the international exchange of television signals the time difference between the sound and vision components should not exceed 20 ms if the sound is advanced with respect to the picture or 40 ms if the sound is delayed with respect to the picture.

Note – The application of these values to circuits is a subject for further study.

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* This Recommendation should be brought to the attention of CCIR Study Groups 10 and 11 and CCITT Study Group IV.

RECOMMENDATION 572

TRANSMISSION OF ONE SOUND-PROGRAMME ASSOCIATED WITH AN ANALOGUE TELEVISION SIGNAL BY MEANS OF TIME-DIVISION MULTIPLEX IN THE LINE SYNCHRONIZING PULSE

(Study Programme 21B/CMTT)

(1978)

The CCIR,

CONSIDERING

(a) that if the sound and picture components of a television signal are transmitted over different circuits or by different methods, there may be an error of timing between the signals at the receiving end;

(b) that one method of solving this problem would be to transmit both signals over the same circuit by multiplexing;

(c) that it is desirable when sound and video signals are transmitted by time division multiplex to standardize on a particular solution,

UNANIMOUSLY RECOMMENDS

that for the transmission of one sound-programme signal together with an analogue television signal by time-division multiplex, a pulse-code modulation system should be used, in which the most important principles are as follows:

1. the sound-programme signal should be sampled at twice the rate of the video-line frequency (see Note 3);

2. the samples should be encoded in binary PCM form suitably companded into 10 bit words;

3. alternate coded samples should be delayed by half a line period, one 10 bit word from each pair of samples being complemented, and the digits of the two samples should be interleaved so that similarly numbered digits from the two samples appear consecutively;

4. the individual digit pulses should be shaped into pulses of sine-squared form, of duration and spacing such that 21 pulses can be inserted into the line synchronizing pulse. In television systems employing equalizing pulses in the field blanking interval, the duration of the appropriate equalizing pulses should be lengthened at the coder, and restored to normal at the decoder;

5. the combined and shaped pulse groups (20 bits) together with a marker pulse (similarly shaped) should be inserted in the line synchronizing pulses so that the marker pulse followed by the least significant digits are nearest to the leading edge of the synchronizing pulse, and the most significant digits are nearest to the trailing edge of the synchronizing pulse. The pulse amplitude should extend from sync bottom to peak white;

6. pre- and de-emphasis should be used, the curve in accordance with CCITT Recommendation J.17 being suitable for this purpose. The low-frequency gain should be fixed in conjunction with the compandor to minimize any impairment due to programme-modulated noise which may occur with critical programme material;

7. protection should be provided so that in the event of the decoded samples being in error due to either the video waveform or the audio pulses being non-standard, any resulting audio disturbance is minimized. For example, this may be achieved by means of holds, i.e. by the repetition of previous samples for a short duration of errors, and by muting the audio output for a longer duration of errors, or if too many holds occur in a short time;

8. for three codecs in tandem, the characteristics should conform to Recommendation 505 (see Note 3);

9. the video performance of the system should be such that a single coder and decoder introduces negligible distortion, compared with the limiting values for the television hypothetical reference circuit, given in Recommendation 567;

10. the quality of the decoded sound obtained from one codec should not be adversely affected if the television circuit between the coder and the decoder is such that its performance is equivalent to a long international television connection (see Recommendation 603); equalized if necessary.

Note 1 – Circuits using sync bottom clamps are not suitable for use with this system.

Note 2 – For use over satellite circuits which involve 17.5 MHz bandwidth operation at IF, it is desirable to reduce the pulse amplitude by 6 dB.

Note 3 – It may be difficult to obtain a response extending to 15 kHz as the sampling rate is twice line-frequency (see § 1 above). Hence the highest frequency at which the hypothetical reference circuit amplitude/frequency response is specified may be modified to 14 kHz.

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