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INTERNATIONAL TELECOMMUNICATION UNION

CCIR

INTERNATIONAL
RADIO CONSULTATIVE
COMMITTEE

RECOMMENDATIONS AND REPORTS OF THE CCIR, 1982

(ALSO QUESTIONS, STUDY PROGRAMMES,
RESOLUTIONS, OPINIONS AND DECISIONS)

XVth PLENARY ASSEMBLY
GENEVA, 1982

VOLUME XII

TRANSMISSION OF SOUND BROADCASTING AND TELEVISION SIGNALS OVER LONG DISTANCES (CMTT)



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**PLAN OF VOLUMES I TO XIV
XVTH PLENARY ASSEMBLY OF THE CCIR**

(Geneva, 1982)

VOLUME I	Spectrum utilization and monitoring.
VOLUME II	Space research and radioastronomy.
VOLUME III	Fixed service at frequencies below about 30 MHz.
VOLUME IV-1	Fixed-satellite service.
VOLUME IV/IX-2	Frequency sharing and coordination between systems in the fixed-satellite service and radio-relay systems.
VOLUME V	Propagation in non-ionized media.
VOLUME VI	Propagation in ionized media.
VOLUME VII	Standard frequencies and time signals.
VOLUME VIII	Mobile services.
VOLUME IX-1	Fixed service using radio-relay systems.
VOLUME X-1	Broadcasting service (sound).
VOLUME X/XI-2	Broadcasting-satellite service (sound and television).
VOLUME XI-1	Broadcasting service (television).
VOLUME XII	Transmission of sound broadcasting and television signals over long distances (CMTT).
VOLUME XIII	Vocabulary (CMV).
VOLUME XIV-1	Information concerning the XVth Plenary Assembly: Minutes of the Plenary Sessions. Administrative texts. Structure of the CCIR. Lists of CCIR texts.
VOLUME XIV-2	Alphabetical index of technical terms appearing in Volumes I to XIII.

All references within the texts to CCIR Recommendations, Reports, Resolutions, Opinions, Decisions, Questions and Study Programmes refer to the 1982 edition, unless otherwise noted; i.e., only the basic number is shown.

DISTRIBUTION OF TEXTS OF THE XVTH PLENARY ASSEMBLY OF THE CCIR IN VOLUMES I TO XIV

Volumes I to XIV, XVth Plenary Assembly, contain all the valid texts of the CCIR and succeed those of the XIVth Plenary Assembly, Kyoto, 1978.

1. Recommendations, Reports, Resolutions, Opinions, Decisions

1.1 Numbering of these texts

Recommendations, Reports, Resolutions and Opinions are numbered according to the system in force since the Xth Plenary Assembly.

In conformity with the decisions of the XIth Plenary Assembly, when one of these texts is modified, it retains its number to which is added a dash and a figure indicating how many revisions have been made. For example: Recommendation 253 indicates the original text is still current; Recommendation 253-1 indicates that the current text has been once modified from the original. Recommendation 253-2 indicates that there have been two successive modifications of the original text, and so on. 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-1.

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(1) Published separately.

1.3.1 Note concerning Reports

The individual footnote "Adopted unanimously" has been dropped from each Report. Reports in this volume 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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1.6 Decisions

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1.6.1 Note concerning Decisions

Since Decisions were adopted by Study Groups, use was made of the expression “Study Group..., Considering” and the expression “Unanimously decides”, replaced by “Decides”.

2. Questions and Study Programmes

2.1 Text numbering

2.1.1 Questions

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.

2.1.2 *Study Programmes*

Study Programmes are numbered to indicate the Question from which they are derived if any, the number being completed by a capital letter which is used to distinguish several Study Programmes which derive from the same Question. The part of the Study Programme number which indicates the Question from which it is derived makes no mention of any possible revision of that Question, but refers to the current text of the Question as printed in this Volume.

Examples:

- Study Programme 1A/10, which would indicate that the current text is the original version of the text of the first Study Programme deriving from Question 1/10;
- Study Programme 1C/10, which would indicate that the current text is the original version of the text of the third Study Programme deriving from Question 1/10;
- Study Programme 1A-1/10, would indicate that the current text has been once modified from the original, and that it is the first Study Programme of those deriving from Question 1/10.

It should be noted that a Study Programme may be adopted without it having been derived from a Question; in such a case it is simply given a sequential number analogous to those of other Study Programmes of the Study Group, except that on reference to the list of relevant Questions it will be found that no Question exists corresponding to that number.

References to Questions and Study Programmes within the text are made to the basic number as well as for other CCIR texts.

2.2 *Arrangement of Questions and Study Programmes*

The plan shown on page II indicates the Volume in which the texts of each Study Group are to be found, and so reference to this information will enable the text of any desired Question or Study Programme to be located.

VOLUME XII

TRANSMISSION OF SOUND BROADCASTING AND TELEVISION SIGNALS
OVER LONG DISTANCES

(CMTT)

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Rec. 570	B	107	Report 647-2	C	177
Rec. 571-1	C	136	Report 648-2	C	193
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Report 314-5	B	88	Report 822-1	D	201
Report 411-4	B	104	Report 823	E	219
Report 412-3	E	209	Report 965	A	47
Report 488-3	E	212	Report 966	A	54
Report 491-2	C	143	Report 967	A	72
Report 493-3	C	164	Report 968	D	199
Report 496-3	C	129	Report 969	E	217
			Report 970	E	219

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**INDEX OF THE TEXTS PUBLISHED IN OTHER VOLUMES BUT
CONTAINING INFORMATION OF INTEREST TO
TRANSMISSION OF SOUND BROADCASTING AND
TELEVISION SIGNALS OVER LONG DISTANCES (CMTT)**

Text	Title	Volume
Report 526	Measures of voice transmission performance applicable for electro-magnetic compatibility analysis	I
Question 18/1	System design for maximizing the efficiency and utility of spectrum use	I
Question 58/1	Frequency sharing between various services	I
Question 62/1	Lightning protection of radio equipment	I
Decision 56	Transfer of technology	I
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Report 208	Form of the hypothetical reference circuit and allowable noise standard for frequency-division multiplex telephony and television in the fixed-satellite service	IV
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Report 375	Noise objectives for sound-programme circuits 2500 km long provided by means of analogue radio-relay systems	IX
Report 931	Study of the effects of co-channel interference in the TV channels of FM radio-relay systems	IX
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Report 289	Preferred characteristics for the simultaneous transmission of television and a maximum of four sound channels on analogue radio-relay systems	IX
Recommendation 405	Pre-emphasis characteristics for frequency modulation radio-relay systems for television	IX
Recommendation 276	Frequency deviation and the sense of modulation for analogue radio-relay systems for television	IX
Recommendation 444	Preferred characteristics for multi-line switching arrangements of analogue radio-relay systems	IX
Recommendation 562	Subjective assessment of sound quality	X
Recommendation 468	Measurement of audio-frequency noise voltage level in sound broadcasting	X
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Question 53	Methods of synchronizing various recording and reproducing systems	X
Opinion 51	Study of digital techniques by CCIR Study Groups and the CMTT	X
Study Programme 2G/10 and 11	Composite 625-line signal for television broadcasting from satellites	X/XI

Text	Title	Volume
Study Programme 2H/10 and 11	Characteristics of receiving systems of the broadcasting-satellite service (sound and television)	X/XI
Decision 51	Unified standards for the accommodation of several audio signals in terrestrial and satellite television channels	X/XI
Report 802	Additional broadcasting services using a television or narrow-band channel	XI
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Study Programme 3A/11	Subjective assessment of the quality of television pictures	XI
Study Programme 25G/11	Digital encoding of colour television signals	XI
Study Programme 25H/11	Filtering and sampling for digital encoding of colour television signals	XI
Study Programme 25J/11	Reduction in the bit rate in the digital coding of colour television signals	XI
Study Programme 25K/11	Protection against errors, jitter and slip for digital television signals	XI
Study Programme 25L/11	Deriving digital component signals from analogue composite signals (and vice versa)	XI
Study Programme 25M/11	Quality parameters and measurement and monitoring methods to be used in the studio complex and in direct broadcasting from terrestrial transmitters and from satellites using digital or analogue-and-digital modulation	XI
Question 29/11	Broadcasting of still pictures and other information intended for the public using a television or narrow-band channel	XI
Study Programme 29D/11	Specifications for a programme delivery service	XI
Study Programme 29E/11	Specifications for a new "audiography" broadcasting service	XI
Study Programme 29F/11	Concerning broadcasting of still-picture television signals using a television or narrow-band channel	XI
Question 35/11	Synchronization necessary for the satisfactory reception of sound and picture signals	XI
Question 40/11	Methods of synchronizing various recording and reproducing systems	XI
Opinion 38	Exchange of monochrome and colour television programmes via satellites	XI
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Resolution 66	Terms and definitions	XIII
Decision 19	Terms and definitions	XIII

TRANSMISSION OF SOUND BROADCASTING AND TELEVISION SIGNALS OVER LONG DISTANCES

CMTT

Joint CCIR/CCITT Study Group for Television and Sound Transmissions

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.

1978-1982-1986 *Chairman:* Y. ANGEL (France)

Vice-Chairman: W. G. SIMPSON (United Kingdom)

INTRODUCTION BY THE CHAIRMAN OF THE CMTT

1. CMTT activities from 1978 to 1982

1.1 During the study period 1978-1982, the CMTT held two meetings in Geneva:

- Interim meeting from 30 September to 17 October 1980,
- Final meeting from 17 September to 7 October 1981.

The volume of documentation involved was as follows:

Technical contributions to meetings: 191

Texts prepared at meetings: 182

The work of the CMTT results in the submission of 70 texts to the Plenary Assembly, including 18 new ones, a proposal to delete 8 obsolete texts and the retention of 35 texts.

As regards the classification of subjects, the CMTT wishes to keep the same subdivisions as in the previous period, namely:

- 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 and 23)
Transmission of signals with multiplexing of video, sound and data and signals of new systems.

1.2 During its interim and final meetings, the CMTT set up *Working Parties* operating as follows:

Working Party: Terms of reference:

Chairman:

Interim meeting:

CMTT-AA	Q.13 and 22	Mr. L. C. Gooddy (Canada)
CMTT-AN	Q.14 and part of Q.21	Mr. D. W. Osborne (United Kingdom)
CMTT-B	Section B	Mr. L. Golding (United States of America)
CMTT-C	Sections C and D and part of Q.21	Mr. G. Zedler (Federal Republic of Germany)
CMTT-T	Terminology, editing	Mr. R. Gardiner (United Kingdom)

Final meeting :

CMTT-AB	Q.13, 22, Section B	Mr. P. Wickliffe (United States of America)
CMTT-AN	Q.14, part of Q.21	Mr. D. W. Osborne (United Kingdom)
CMTT-C	Sections C and D, part of Q.21	Mr. G. Zedler (Federal Republic of Germany)
CMTT-T	Terminology, editing	Mr. R. Gardiner (United Kingdom)

It will be seen that Working Parties AA and B of the interim meeting merged into a single group, since there was less work to be done on analogue transmission for television.

1.3 As in the previous period, Working Party CMTT-T took charge of the revision of texts and their vocabulary and of liaison with the Editorial Committee and the CMV.

All texts issued, including Recommendations and Questions, were reviewed by Working Party CMTT-T as a further assurance of uniformity.

Mr. R. Gardiner (United Kingdom), who was responsible for English, was joined by:

- Mr. Ranquet (France) at the interim meeting for the French text
- Mr. Diaz (Spain) at the interim meeting for the Spanish text
- Mr. Victorion (France) at the final meeting for the French text
- Mr. Casado, CCIR engineer, at the final meeting for the Spanish text.

1.4 During both the interim meeting and the final meeting, Interim Working Party CMTT-1 met at the instance of its Chairman, Mr. W. G. Simpson (United Kingdom).

At these meetings, statements were made by the following:

- on television transmissions:

Mr. Yamamoto (Japan), Chairman of Working Party 11-D;
Mr. Osborne (United Kingdom), Chairman of Working Party CMTT-AN;

- on sound programme transmissions:

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

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

Of the proposals submitted which were approved by the plenary CMTT, the following may be mentioned:

- Revision and shortening of Decision 18-2 with updating of Table I and deletion of Annex II, the contents of the last sub-paragraph of § 3 of the annex being substituted for “DECIDES 6”. The revised text becomes: Decision 18-3.
- Deletion of Opinion 55, which no longer adds any useful information to that contained in Decision 18-3.
- Suggestion that the CMV should be represented in IWP/CMTT-1 in the interests of the best possible alignment of the terminologies employed by Study Groups 10 and 11 of the CCIR, by the CCITT and by the CMTT.
- Since there was found to be some uncertainty regarding the correct use of “IDN” and “ISDN”, a request that Study Group XVIII of the CCITT should examine the definitions of these terms and clarify them.

The Reports of the IWP/CMTT-1 meetings will be found in Docs. CMTT/220 and 369.

2. Collaboration with other Study Groups

2.1 Relations with Study Groups 10 and 11

In addition to the liaison work undertaken by IWP/CMTT-1 (see § 1.4), Mr. Yamamoto gave, at the request of the Chairmen of Study Group 11 and CMTT, a report on the progress of the work on digital television in Working Party 11-D. Mr. W. G. Simpson, Vice-Chairman of the CMTT, reported in turn in Study Group 11 on CMTT activities regarding television.

Further, a small informal Working Party comprising representatives of the two Study Groups discussed, during the final meetings, collaboration between Study Group 11 and the CMTT on digital television, the only subject of concern at the present time. The conclusions reached (Doc. CMTT/371), which were drafted by Mr. C. J. Phillips (United Kingdom), convenor of the informal Working Party, note that the Questions and Study Programmes of Study Group 11 and of the CMTT are in line with the allocation of responsibilities that the Working Party itself would recommend.

Mr. Phillips suggests, however:

- that Study Group 11 should make an addition to one of the notes of Study Programme 25J/11 concerning the reduction of the bit rate;
- that CMTT should, if necessary, establish a Study Programme parallel to Study Programme 25J/11.

After discussion of this latter point, the CMTT considered (Doc. CMTT/381) that Study Programme 14A/CMTT already covered the question adequately and that it would be sufficient to add an indication to the effect that the studies would be conducted in collaboration with Study Group 11.

Since Doc. CMTT/371 also marginally concerns Study Group 10, this Study Group was included, together with Study Group 11 and CMTT, in the document heading.

2.2 *Relations with the CCITT*

Relations between the CCITT and the CMTT are now clearly established and there is an active exchange of information.

- (a) Certain draft Recommendations prepared by the CMTT are transmitted for submission to the CCITT Plenary Assembly with a view to adoption by it.
- (b) During the study period 1978-1982, the CMTT examined about 50 pages of documents emanating from the CCITT and introduced their substance into its own texts. For its part, it addressed 21 documents to the CCITT providing information and/or raising questions for CCITT Study Groups.
- (c) A Report on CMTT activities was presented with the VIIth CCITT Plenary Assembly (November 1980) by Mr. W. G. Simpson, Vice-Chairman of the CMTT.

2.3 *Special Rapporteurs*

At its interim meeting in 1980, the CMTT began to appoint Special Rapporteurs. Thanks to this arrangement, it was able at its final meeting to receive reports on the work done in 1981 by Study Group XVIII of the CCITT, by Group RA/TV of Study Group IV of the CCITT and by the CMBD (Docs. CMTT/285, 283 and 284). The interest shown by the CMTT as a result of this first experiment has induced it to expand the procedure, so that at the end of the final meeting, the situation is as follows (Doc. CMTT/381):

Working Group RA/TV of Study Group IV of the CCITT:	Mr. Gardiner (United Kingdom)
Study Group XV of the CCITT:	to be appointed
Study Group XVIII of the CCITT:	Mr. Wery (Canada)
Joint Study Group CMBD:	Mr. Bonhoff (France)
Joint Study Group CMV:	Mr. Gardiner (United Kingdom) Mr. Poncin (France)

For the moment the duties of these Special Rapporteurs are still of an informal nature, and their main function is to transfer information between the CMTT and the Study Group for which they have been appointed.

3. **Salient points**

3.1 *Analogue signals*

In this field, existing Recommendations and Reports deal with most of the problems raised by international sound and video transmissions. Current CMTT activities consist in improving and updating these texts.

Some of them seem likely in the future to be applicable to digital as well as analogue transmissions. This is true, for example, of Recommendation 603, which replaces Report 487 and defines the hypothetical reference chain as equivalent to five hypothetical reference circuits in cascade. This Recommendation, which will be brought to the attention of the CMBD, also states that the single reference chain referred to suffices to cover all the foreseeable needs.

3.2 *Digital signals*

CMTT activities in this field, which is the field of the future, are rapidly expanding. They do, however, depend on those of Study Groups 10 and 11 of the CCIR.

It was thanks to the work of Study Group 11, resulting in the adoption of Recommendation 601, that it was possible for the CMTT to bring together the guiding principles set forth in Recommendation 604 on the transmission of a television signal presented in digital form. According to this Recommendation, fully digital circuits should be used for preference and it is to these that the Recommendation applies.

As for mixed analogue-digital circuits, which will probably be used during a transitional period, a draft Recommendation was prepared but could not be adopted in the absence of general agreement. The substance of that text has been appended as Annex 1 to Report 646-2 and the main objections which prevented agreement are given in Annex 2 to this Report. For the rest, Report 646-1 has been thoroughly revised and amplified, particularly § 4 on "Coding methods".

As regards the digital transmission of sound programme signals, it had been hoped that a single Recommendation might be adopted if the different views expressed in Report 647-1 could be brought closer together. Unfortunately, this was not the case, and an important note summarizing the discussions has been annexed to Report 647-2. At its final meeting, it will be remembered, Study Group 10 encountered similar difficulties.

4. Status of texts

Care was taken to classify the pink documents in logical order with a view to their submission to the Plenary Assembly and their publication in a new CCIR Volume XII.

The lists below give the essential information on the status of these texts.

Abbreviations:

I: text adopted or modified at the interim meeting

F: text adopted or modified at the final meeting

NOC: text kept unchanged since the XIVth Plenary Assembly

MOD: text updated

SUP: text deleted

REC.: Recommendation, Rep.: Report, Q.: Question, S.P.: Study Programme

For simplicity's sake, the indication "CMTT" and the index numbers have been omitted.

RECOMMENDATIONS AND REPORTS

Section A

Pink No.

1004	Rec. 567	(I,F)	—	MOD
1005	Rep. 815	(I,F)	—	MOD
—	Rec. 868		—	NOC
1006	Rep. 637	(I)	—	MOD
1007	Rep. 816	(I,F)	—	MOD
1008	Rep. 965	(I,F)	—	new text concerning satellite transmission
1009	Rep. 636	(I)	—	MOD
—	Rep. 635		—	NOC
1010	Rep. 966	(I,F)	—	new text calling for further study on standards converters
—	Rep. 487	(I)	—	SUP deleted and replaced by the following text. See § 3.1.
1011	Rec. 603	(I,F)	—	new text defining a hypothetical reference chain. See § 3.1.
1012	Rec. 604	(F)	—	new text on digital transmissions. See § 3.2.
1013	Rep. 646	(I,F)	—	MOD
1014	Rep. 817	(I,F)	—	MOD
1015	Rep. 967	(I,F)	—	new text on impairments and methods of protection in digital transmission. Associated with Rep. 646.

Section B

1016	Rec. 473	(I,F)	—	MOD
1017	Rep. 314	(I)	—	MOD
1018	Rep. 818	(I)	—	MOD
1019	Rep. 639	(I)	—	MOD
1020	Rec. 569	(I,F)	—	MOD
1021	Rep. 411	(F)	—	MOD
—	Rec. 570		—	NOC
1022	Rep. 643	(I,F)	—	MOD
—	Rep. 819		—	NOC

Section C

1023	Rec. 502	(I,F)	— MOD
1024	Rec. 503	(I,F)	— MOD
1025	Rec. 504	(I,F)	— MOD
1026	Rec. 505	(I,F)	— MOD
1027	Rep. 496	(F)	— MOD
1028	Rec. 571	(I)	— MOD
1029	Rep. 497	(I,F)	— MOD
—	Rep. 491		— NOC
—	Rep. 820		— NOC
1030	Rep. 493	(I,F)	— MOD
—	Rep. 498	(F)	— SUP obsolete text, deleted (Doc. CMTT/313)
1031	Rec. 605	(F)	— new text on the estimation of transmission performance of sound-programme circuits shorter or longer than the hypothetical reference circuit
1032	Rep. 821	(I,F)	— MOD on the basis of preceding draft Recommendation
1033	Rec. 474	(I)	— MOD
1034	Rep. 647	(I,F)	— MOD (see comments, § 3.2)
1035	Rec. 606	(I)	— new text fixing the sampling frequency at 32 kHz
1036	Rep. 648	(I,F)	— MOD
—	Rep. 649	(F)	— obsolete text deleted; its contents now included in Rep. 647-2

Section D

1037	Rep. 968	(F)	— new text on the measuring of bit-error distortion
—	Rep. 640	(F)	— SUP obsolete text, deleted
—	Rep. 642		— NOC
1038	Rep. 822	(I,F)	— MOD
—	Rep. 623		— NOC

Section E

—	Rep. 412		— NOC
—	Rec. 572		— NOC
1039	Rep. 488	(I)	— MOD
1040	Rep. 969	(I,F)	— new text on the measurement and test procedures for special signals
1041	Rep. 970	(I)	— new text on the limitation of outband components of video signals
—	Rep. 823		— NOC

QUESTIONS AND STUDY PROGRAMMES

Question 13 (Docs. 1042-1046)

Q.		— NOC
S.P.	A, C, D, E	— MOD
	B	— NOC
	F	— new: standards converters used in transmission

Question 14 (Docs. 1047-1051)

Q.		— MOD
S.P.	A	— MOD
	B, C	— NOC
	D	— new: impairment of quality on digital circuits
	E	— new: service criteria on ISDN

Question 15 (Docs. 1052-1056)

Q.		— MOD
S.P.	A, D, F	— NOC
	B, C, E, G	— MOD

Question 16 (Docs. 1057-1058)

- Q. — MOD
 S.P. A, B — NOC
 C — new: availability of digital television transmission systems

Question 17

- Q. — NOC
 S.P. C, D, F — NOC
 A, B, E — SUP

Question 18 (Docs. 1059-1063)

- Q. — MOD
 S.P. A — MOD
 B — NOC
 C — new: multiplexing of sound programme signals on digital circuits
 D — new: performance of digital circuits
 E — new: sound programme transmissions on ISDN

Question 19 (Doc. 1064)

- Q. — MOD
 S.P. A, B, C, D — NOC

Question 20 (Docs. 1065 and 1066)

- Q. — MOD
 S.P. A — NOC
 B — MOD

Question 21 (Docs. 1067-1069)

- Q. — MOD
 S.P. A, B, E — NOC
 C, D — MOD

Question 22 — NOC*Question 23* (Doc. 1070)

new: methods for ensuring privacy of correspondence in television

OTHER TEXTS

- Decision 18* (Doc. 1071) — MOD (see § 1.4)
Opinion 41 — NOC
Opinion 55 — SUP (see § 1.4)

5. Acknowledgements

On behalf of the Vice-Chairman and himself, the Chairman expresses his gratitude to participating administrations for their contributions and to delegates for their efforts, which led to notable progress in CMTT's work. He also thanks Mr. R. C. Kirby, Director of the CCIR, who provided very favourable and convenient working conditions for the CMTT. All staff members and services of the CCIR also deserve thanks, and the Chairman congratulates Mr. Sonoki, Counsellor, on his excellent contribution to the organization of the work.

He warmly thanks the Chairmen of Working Parties and Sub-Working Parties who, though often working in difficult conditions, skillfully guided the debates from which the CMTT texts originated. He expresses his gratitude to the Special Rapporteurs for having agreed to undertake their tasks and for the Reports they prepared and submitted at the final meeting.

Lastly, he expresses his gratitude to Mr. T. H. Cloud, who performed the invidious task of serving as Secretary to the CMTT Plenary Session and on behalf of all participants congratulates him on the quality of his reports.

SECTION CMTT A: TELEVISION TRANSMISSION STANDARDS AND PERFORMANCE OBJECTIVES

Recommendations and Reports

RECOMMENDATION 567-1

TRANSMISSION PERFORMANCE OF TELEVISION CIRCUITS DESIGNED FOR USE IN INTERNATIONAL CONNECTIONS

(Question 13/CMTT)

(1978-1982)

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, television transmissions over 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.

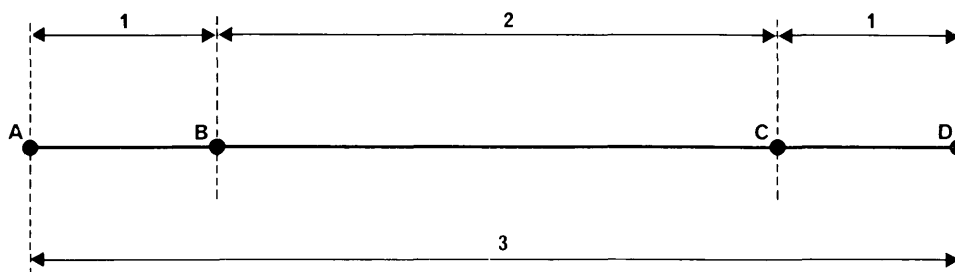
Note. — References and Bibliography are detailed at the end of the particular Part or Annex to which they are relevant.

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

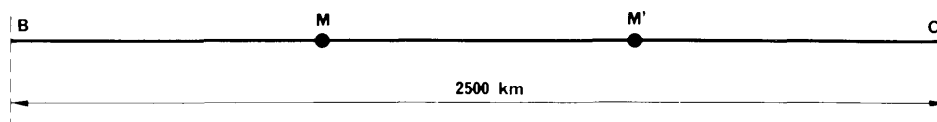
- The international television circuit, BC, comprises a chain of national and international television links. The precise locations (e.g. within buildings), to be regarded as the points B and C, will be designated by the authorities concerned.
- The local television circuit CD connects point C, the receiving 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 coaxial 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.

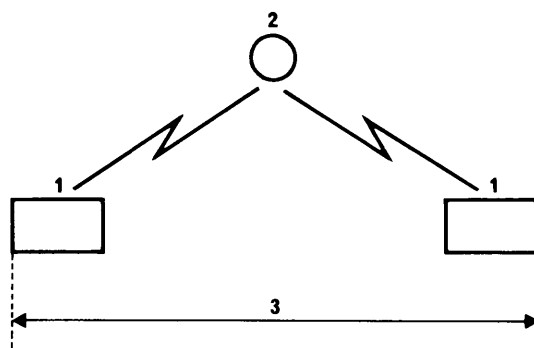


FIGURE 2b – Hypothetical reference circuit for television transmissions over a system in the fixed satellite service

- 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 breezeaway,
- 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 .

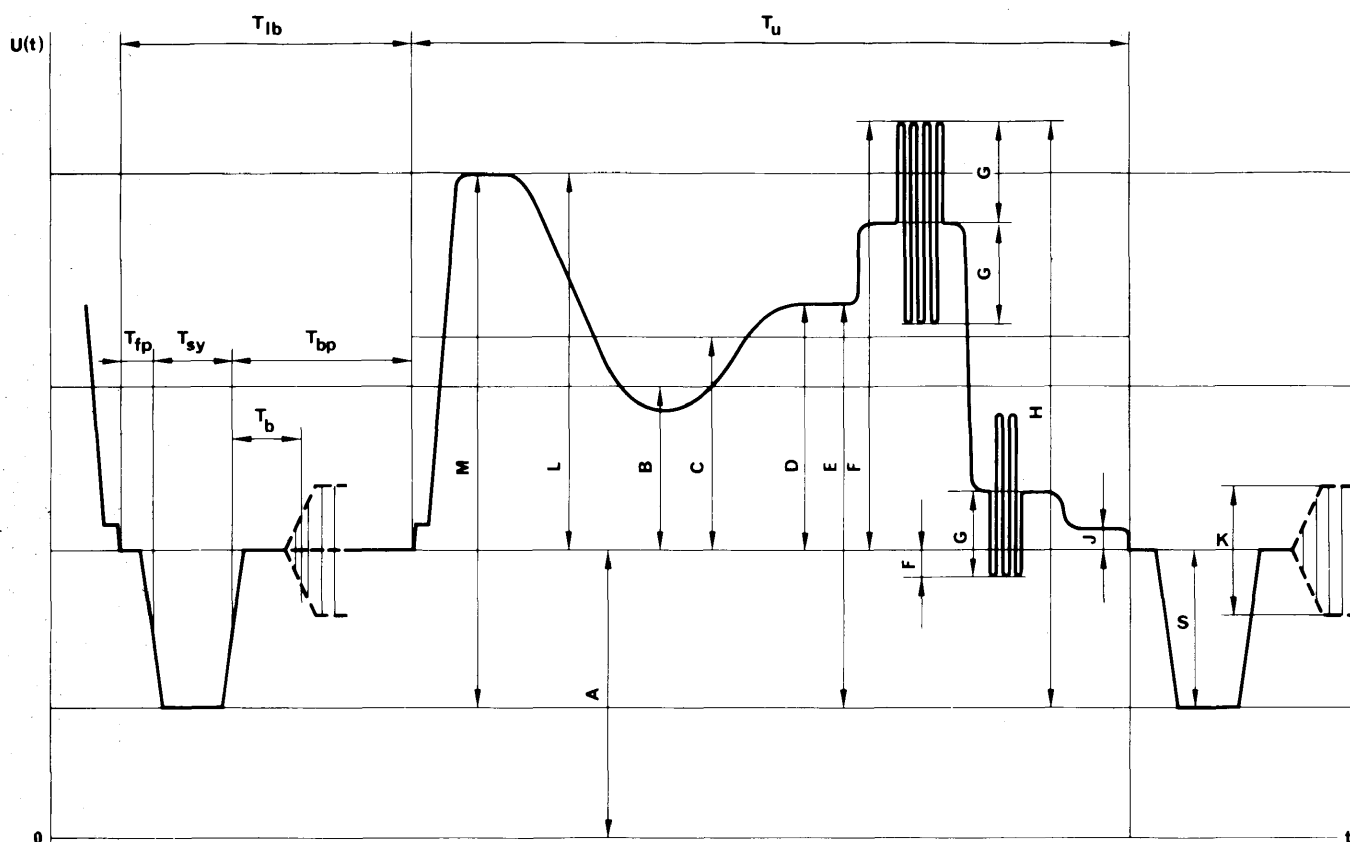


FIGURE 3 – One line of a composite colour video signal

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:

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

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

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

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*

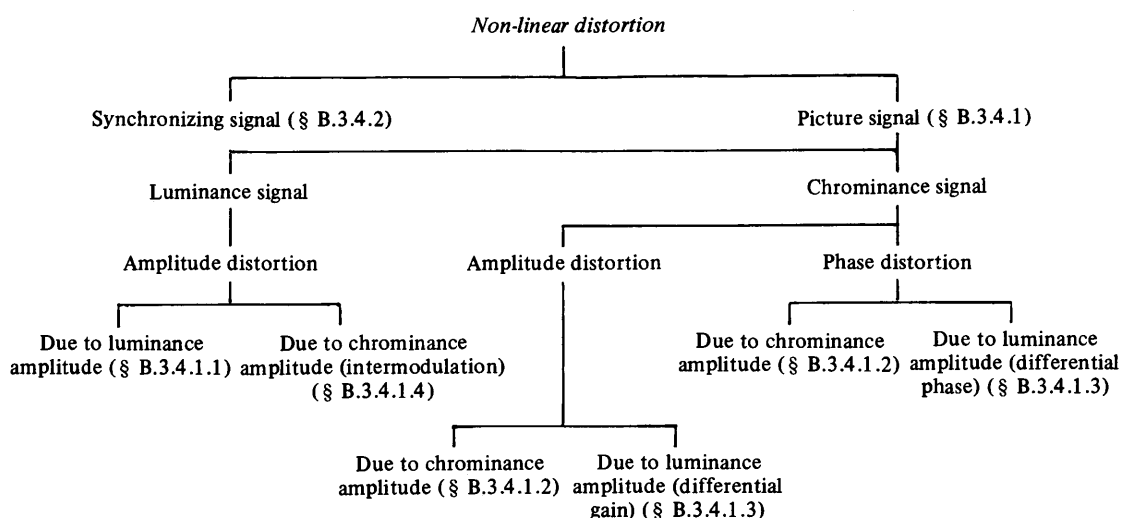
In a television circuit the transmission characteristic may not be completely linear. The extent of the non-linear distortion which is produced will depend primarily on:

- 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.

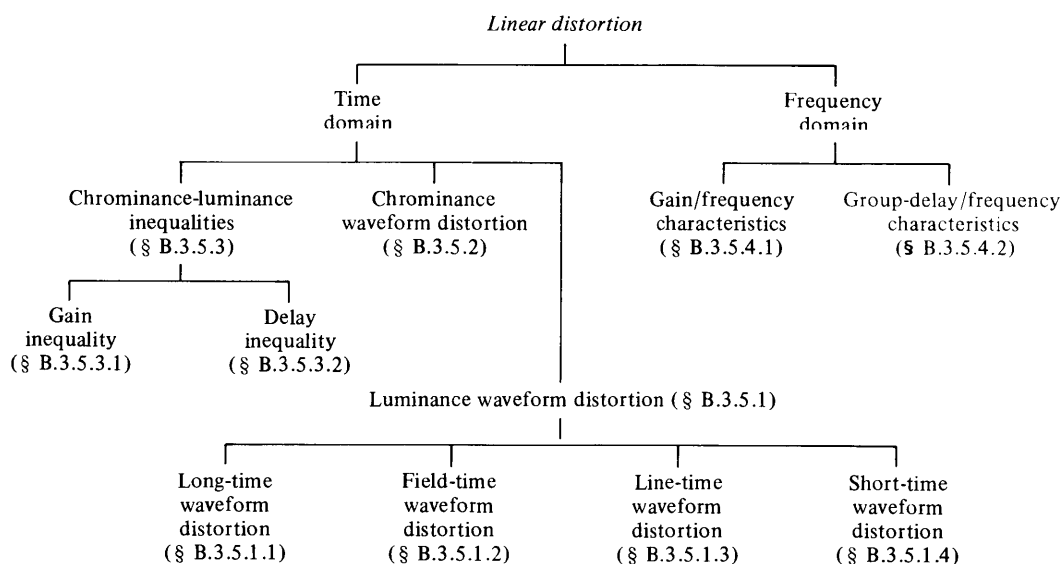
B.3.5 *Linear distortion*

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

At points of video interconnection, the input and output signal amplitude will be specified. The actual amplitudes, as defined by *L* in Fig. 4, will be measured.

C.3 Measurements of transmission performance

C.3.1 Insertion gain

The signal element used is *B3* for 625-line systems and *B3* or *B2* 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.

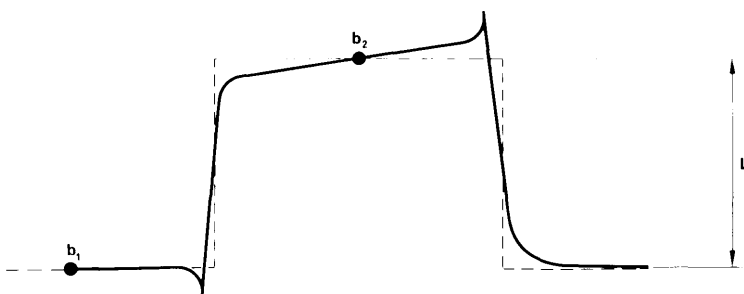


FIGURE 4 – Insertion gain measurement

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.

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),

$k_i = \frac{2i-1}{3}$ 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 G_2 .

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 D_2 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:

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

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:

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

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*, *G1* or *G2* after suppressing the incoming colour sub-carrier. It is defined as the difference between the luminance amplitude in element *G1*, or in the last section of element *G* or *G2* (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 volts/second.

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 μ s (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 μ s.

C.3.5.1.3 *Line-time waveform distortion*

Line-time waveform distortion is measured with element *B3* (Fig. 7) for 625-line systems and *B3* or *B2* (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 μ s are neglected in this measurement.

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

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.

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 on 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.

C.3.5.2 *Chrominance waveform distortion*

This distortion is measured on the chrominance component of the composite pulse *F*. The measurement method is still under study.

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* and *B3*.

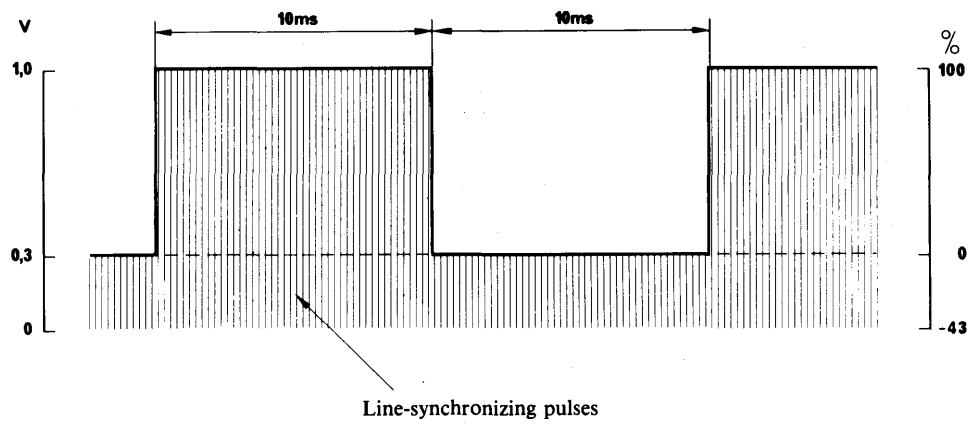


FIGURE 5 – *Signal A for 625-line circuits*

Note. — This signal may contain field-synchronizing pulses.

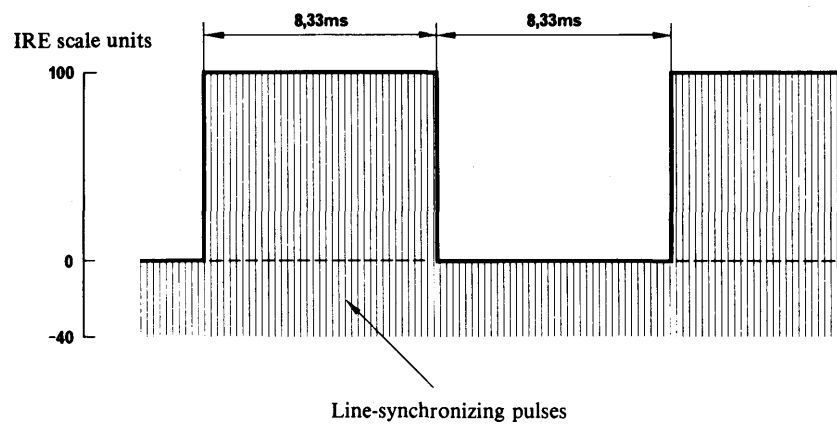


FIGURE 6a – *Signal A for 525-line circuits*

Note. — This signal may contain field-synchronizing pulses.

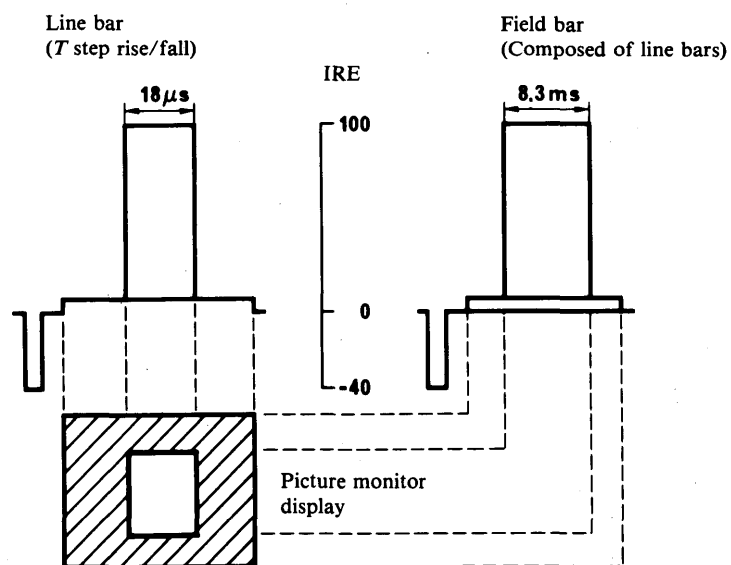


FIGURE 6b – *The window signal for 525-line circuits*

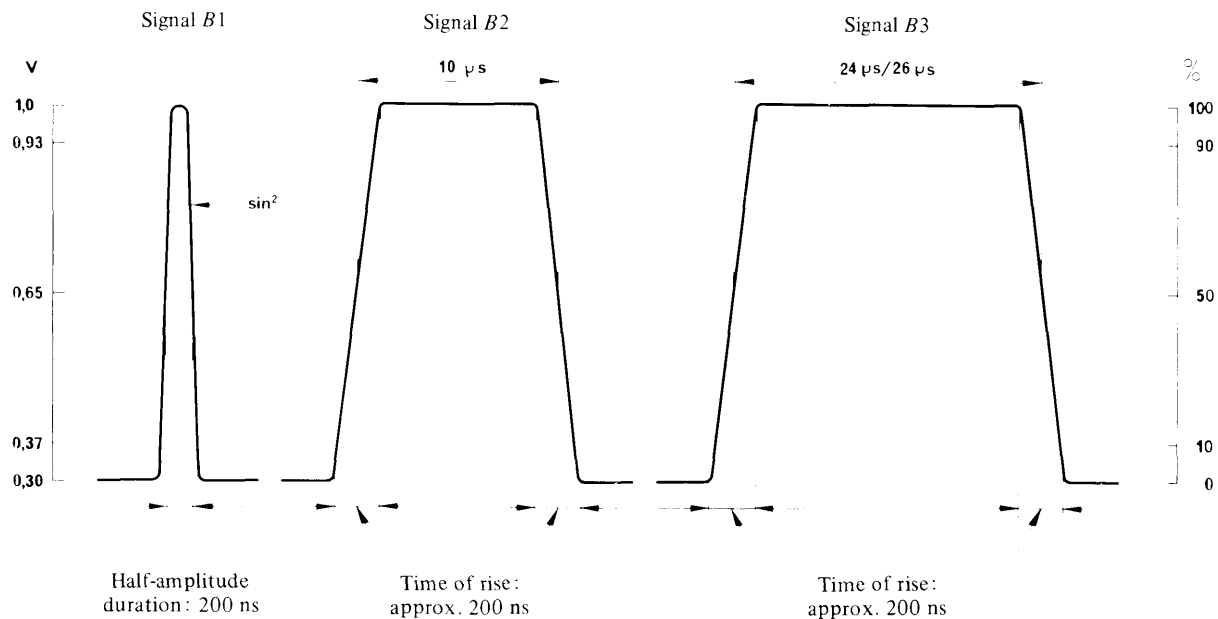


FIGURE 7 — Signal B for 625-line circuits

Note 1. — In some OIRT countries, a half-amplitude duration of 160 ns is used for B1 and a time of rise of 80 ns for B2.

Note 2. — In France, the normal time of rise of B2 and B3 is approximately 110 ns.

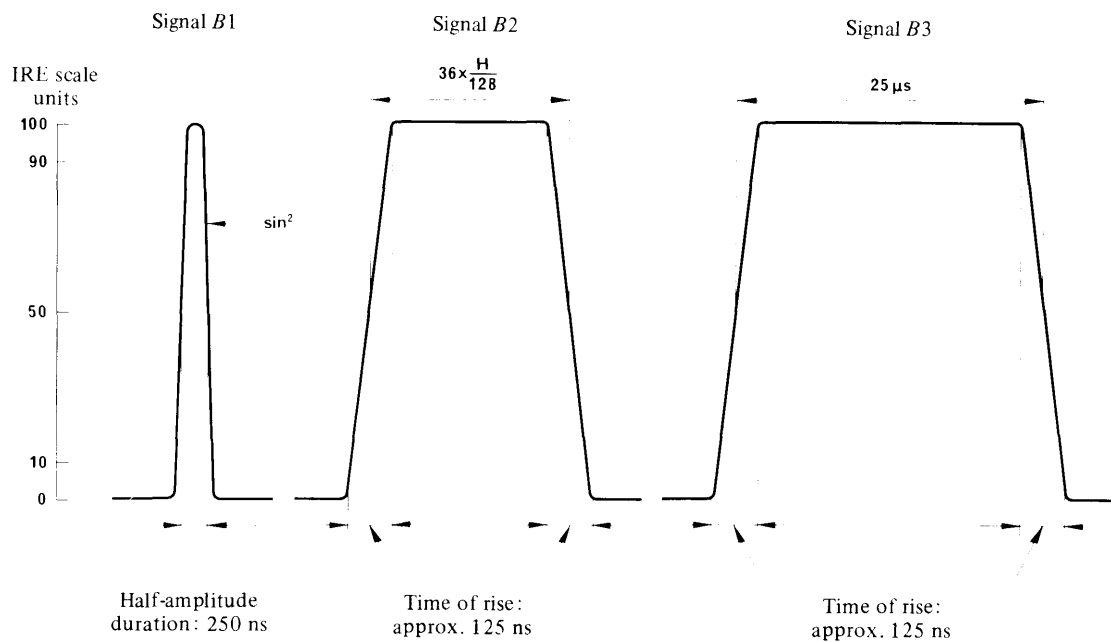


FIGURE 8 — Signal B for 525-line circuits

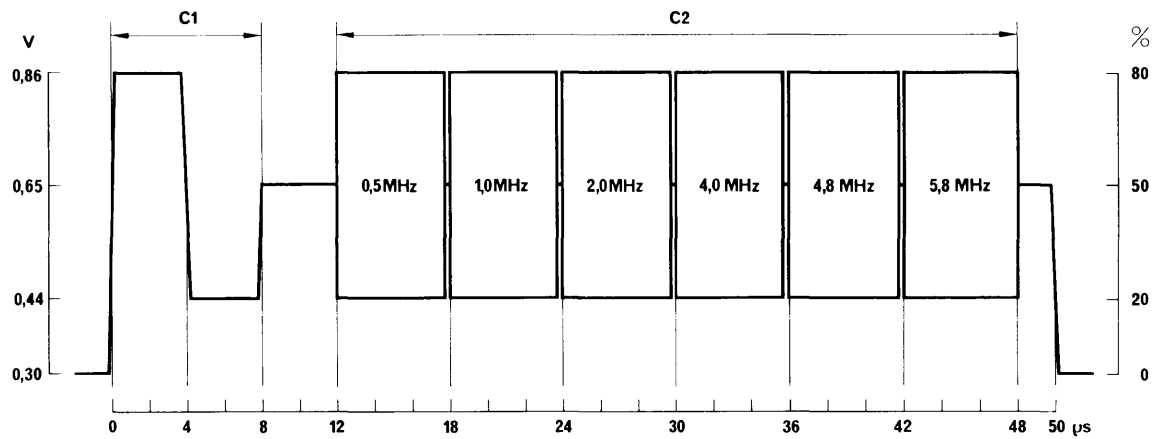


FIGURE 9 – Signal C for 625-line circuits

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

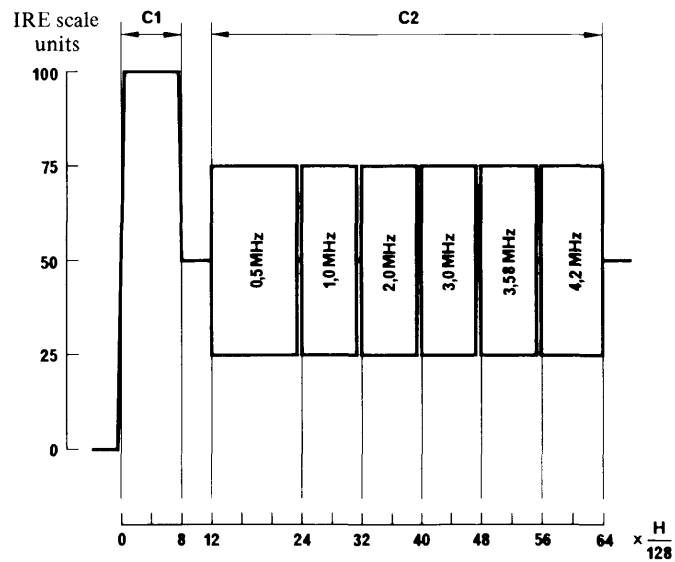


FIGURE 10 – Signal C for 525-line circuits

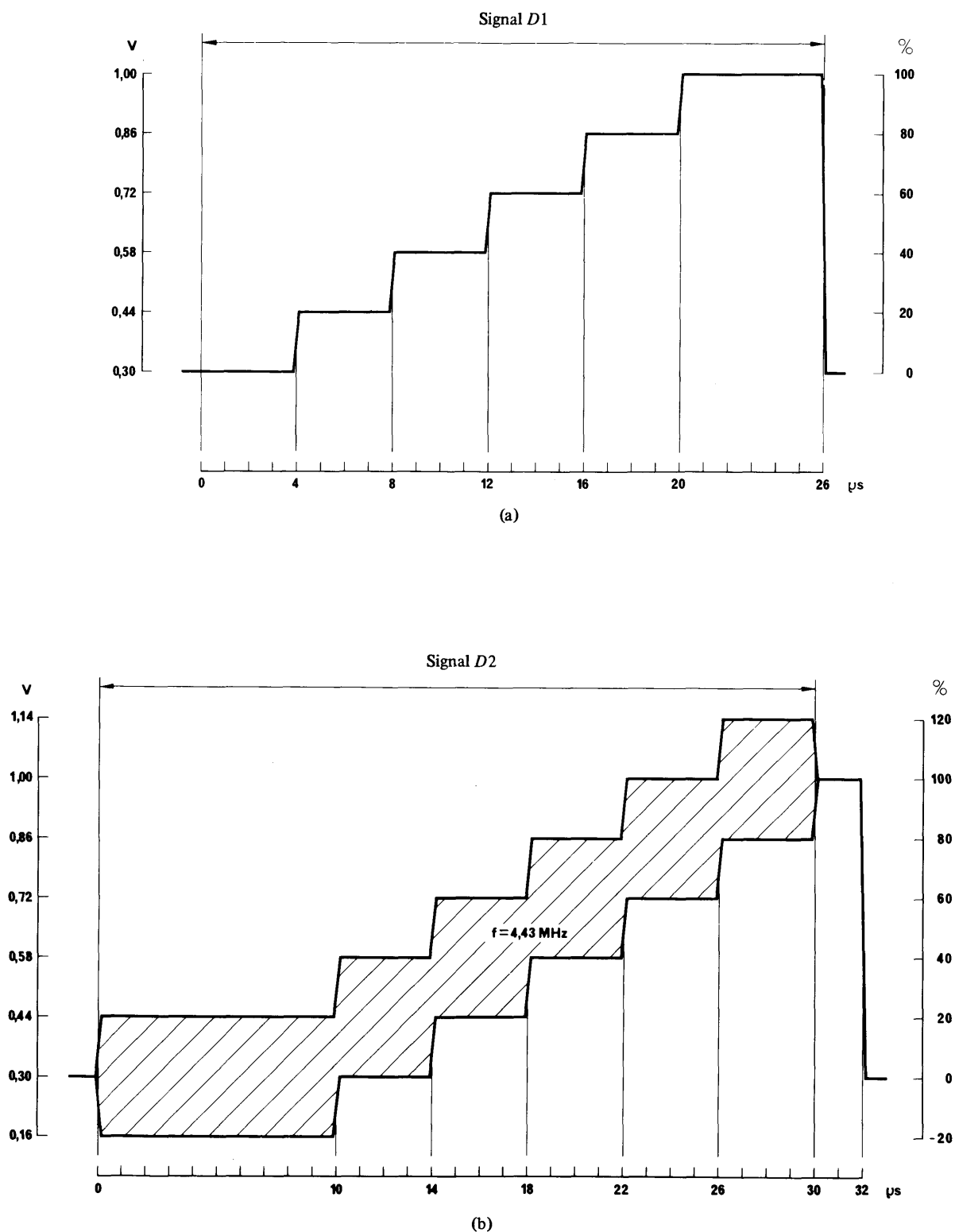


FIGURE 11 — Signal D for 625-line circuits

Note. — In full-field test signals, each tread of the staircase may have a duration of 8.66 μ s.

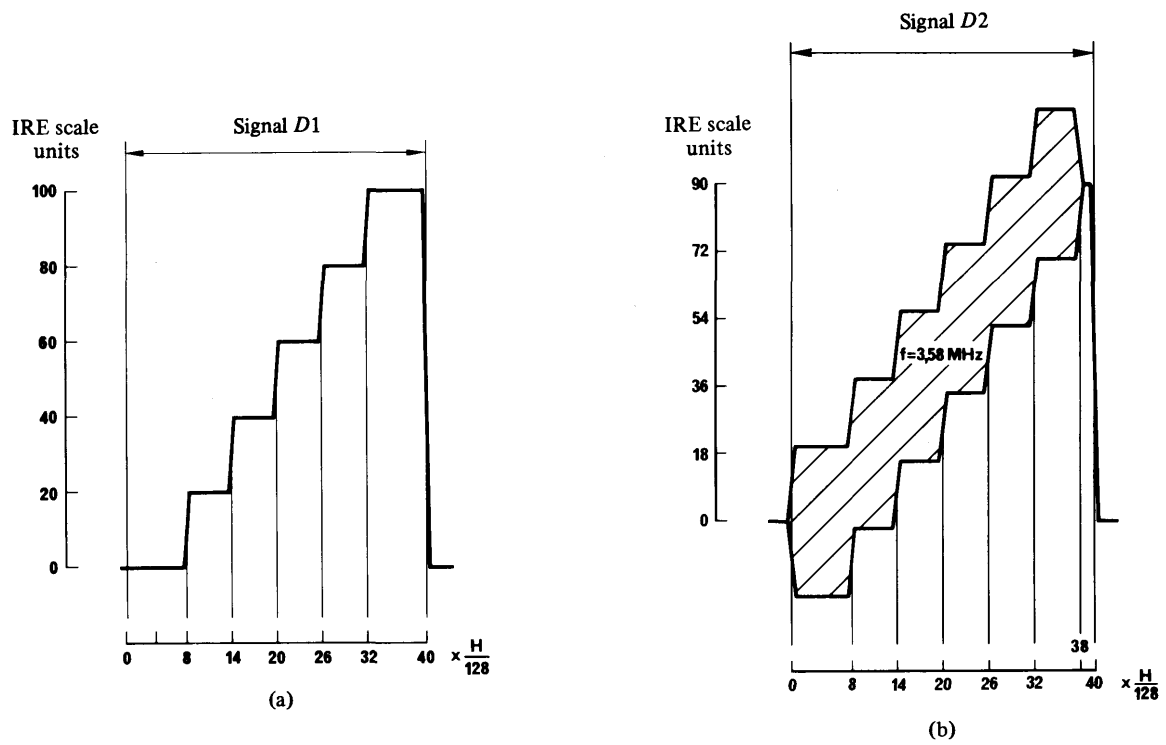


FIGURE 12 – Signal D for 525-line circuits

- Note 1. – Scale refers to tread levels.
 Note 2. – Sub-carrier amplitude is ± 20 IRE units.

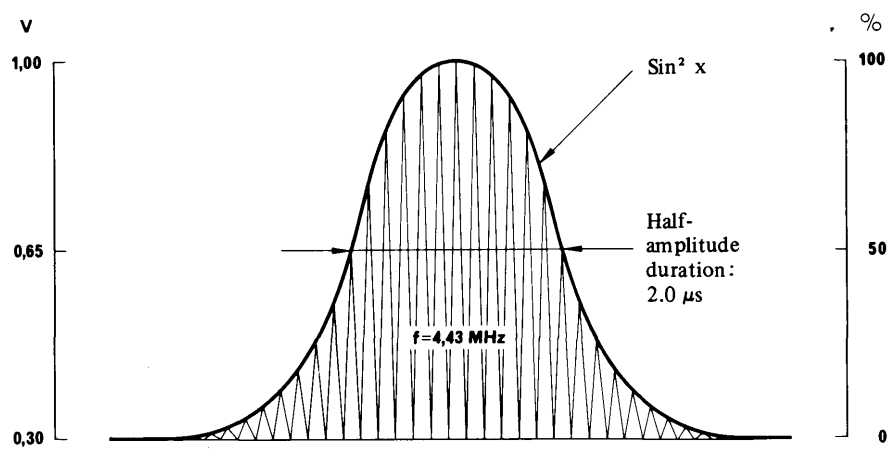


FIGURE 13 – Signal F for 625-line circuits

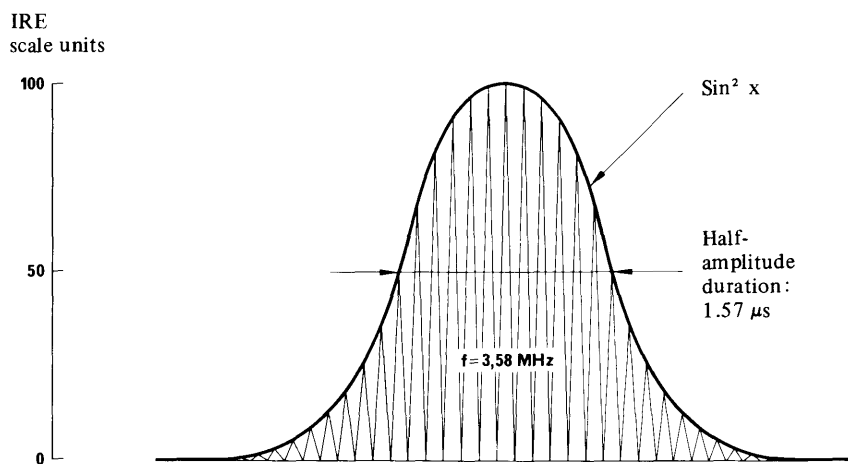


FIGURE 14 – Signal F for 525-line circuits

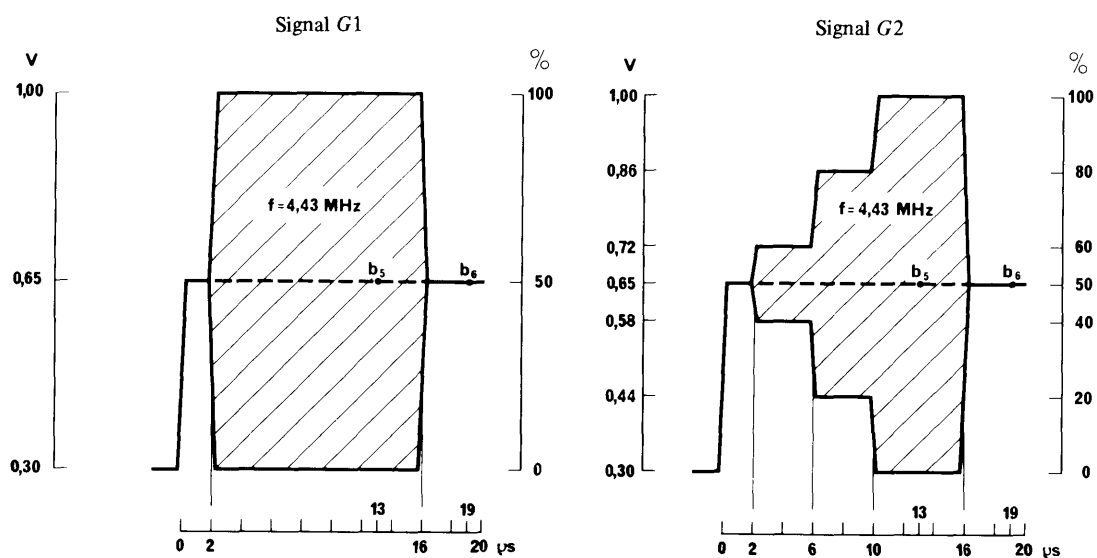


FIGURE 15 – Signal G for 625-line circuits

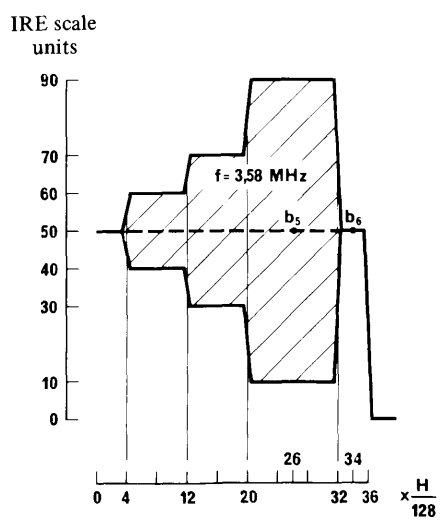


FIGURE 16 – Signal G for 525-line circuits

ANNEX II TO PART C
DESIGN OF FILTERS USED FOR MEASUREMENTS

1. Low-pass filter for use in noise measurements

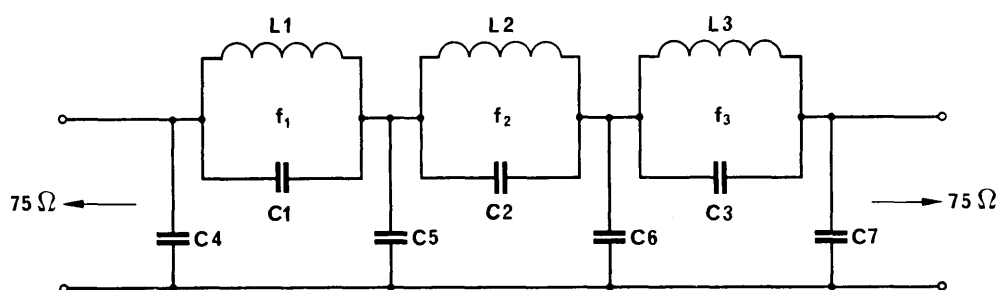


FIGURE 17 – Low-pass filter diagram

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 $\pm 2\%$.

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.

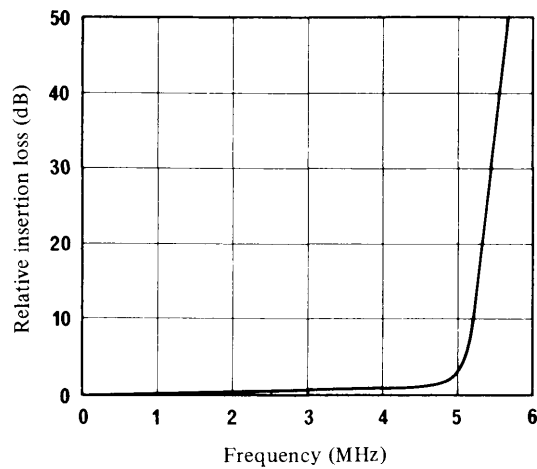


FIGURE 18 – Low-pass filter characteristic

2. Combined high-pass, low-pass filter ($f_c = 10\text{ kHz}$)

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.

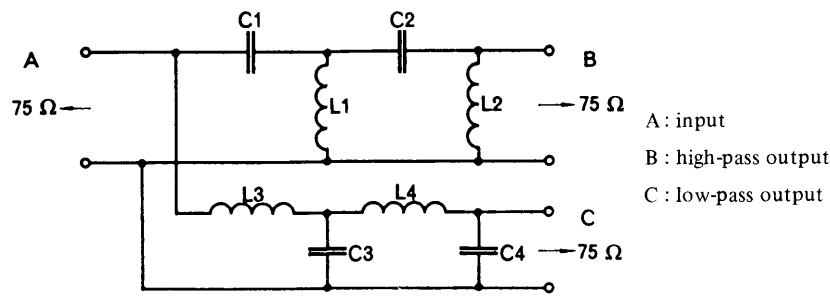


FIGURE 19 – Combined filter design diagram

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.

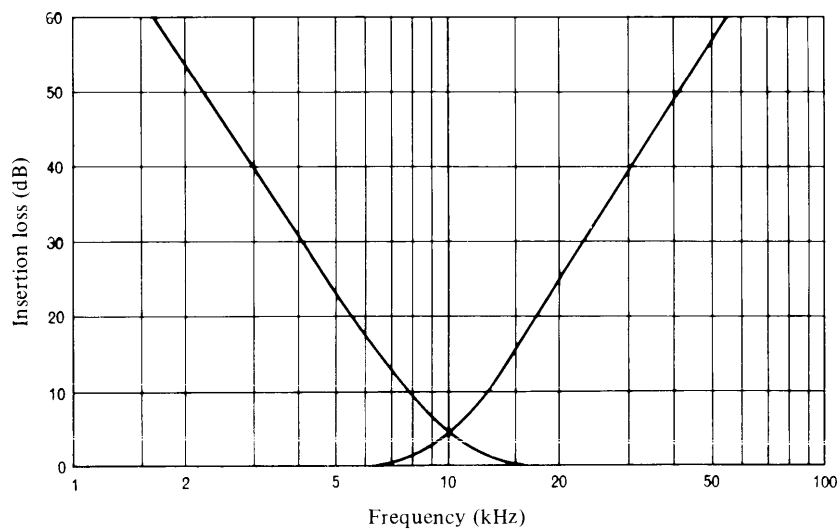


FIGURE 20 – Combined filter characteristic

3. Unified weighting network for random noise

3.1 Network configuration

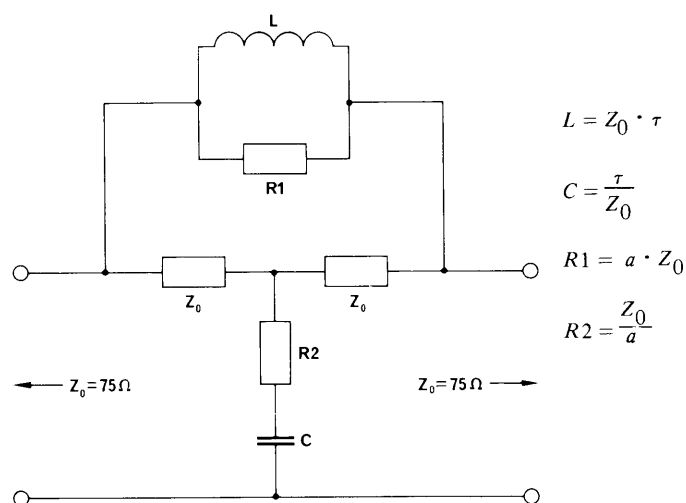


FIGURE 21 – Network diagram

3.2 Insertion loss A

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

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

where:

$\tau = 245 \text{ ns}$; $a = 4.5$

($A_\infty \rightarrow 14.8 \text{ dB}$)

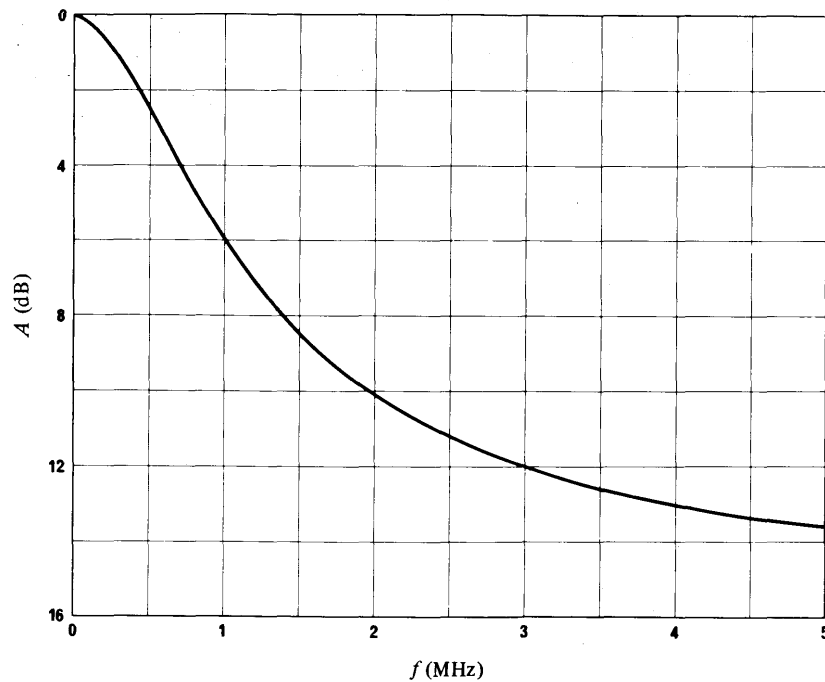


FIGURE 22 – Unified weighting characteristic

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

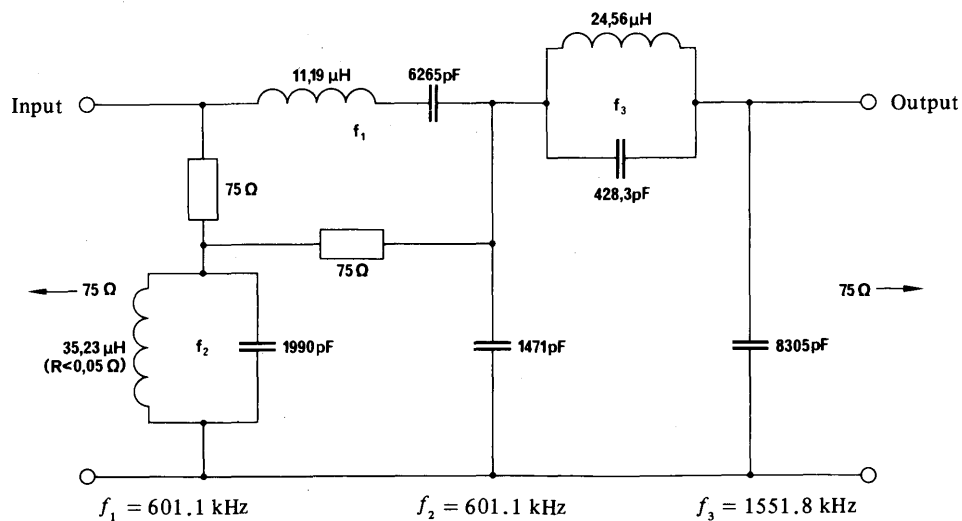


FIGURE 23 – Non-constant resistance network diagram

Note 1. – Capacitor and resistor tolerances $\pm 1\%$.

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

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

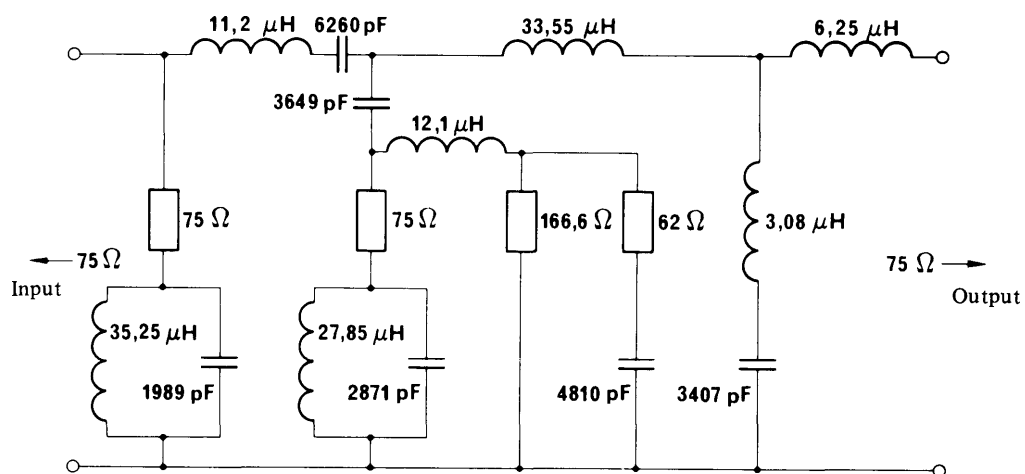
4.2 *Constant resistance form*

FIGURE 24 – Constant resistance network diagram

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

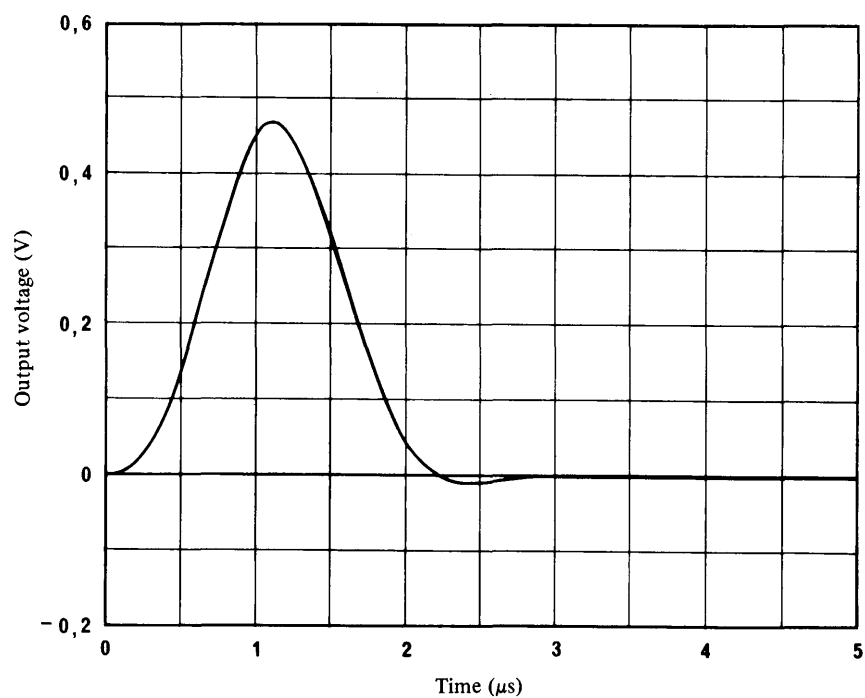
4.3 *Step response of staircase differentiating network*

FIGURE 25 – Transient response of the network

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

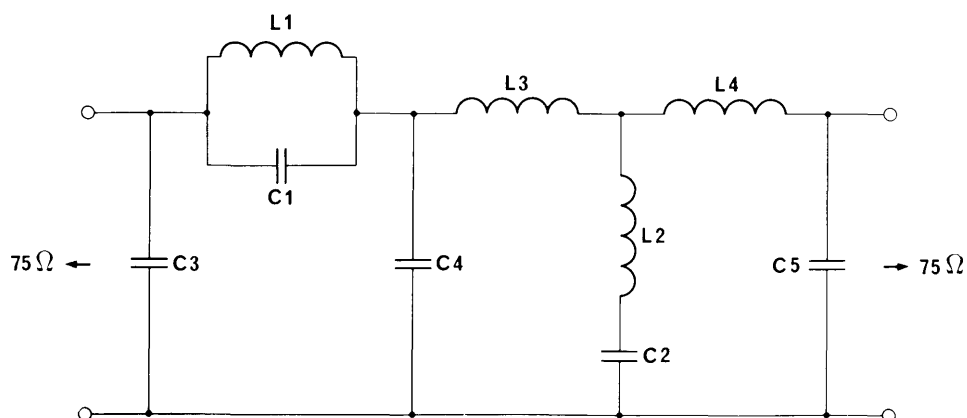


FIGURE 26 – Thomson filter diagram

TABLE OF VALUES

Component	Value ($f_{\infty} = 3.3 \text{ MHz}$)
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 reference 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 (*b₂* in Figs. 27 and 28) and a specified reference point (*b₁* 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 of Part C, but in some cases the measuring instrument may have to be preceded by alternative 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.

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 *b₄* and *b₃* (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 (*b₁* 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).

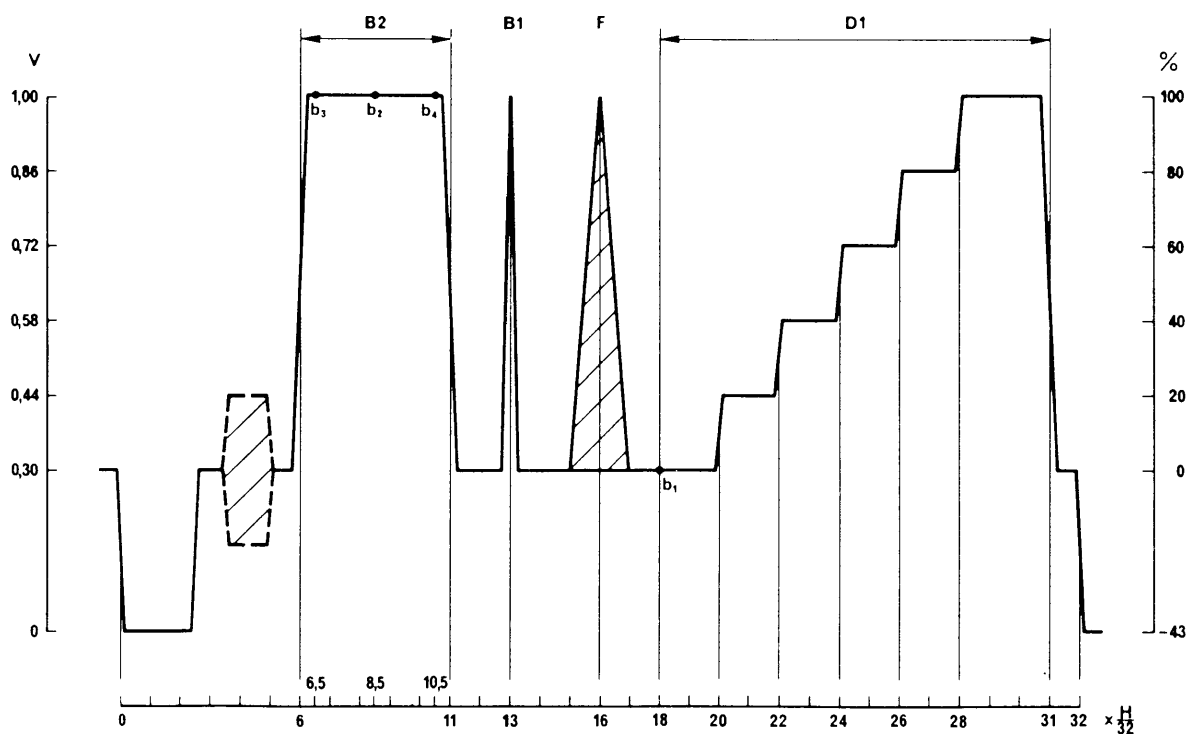


FIGURE 27 – Line 17 for 625-line signals

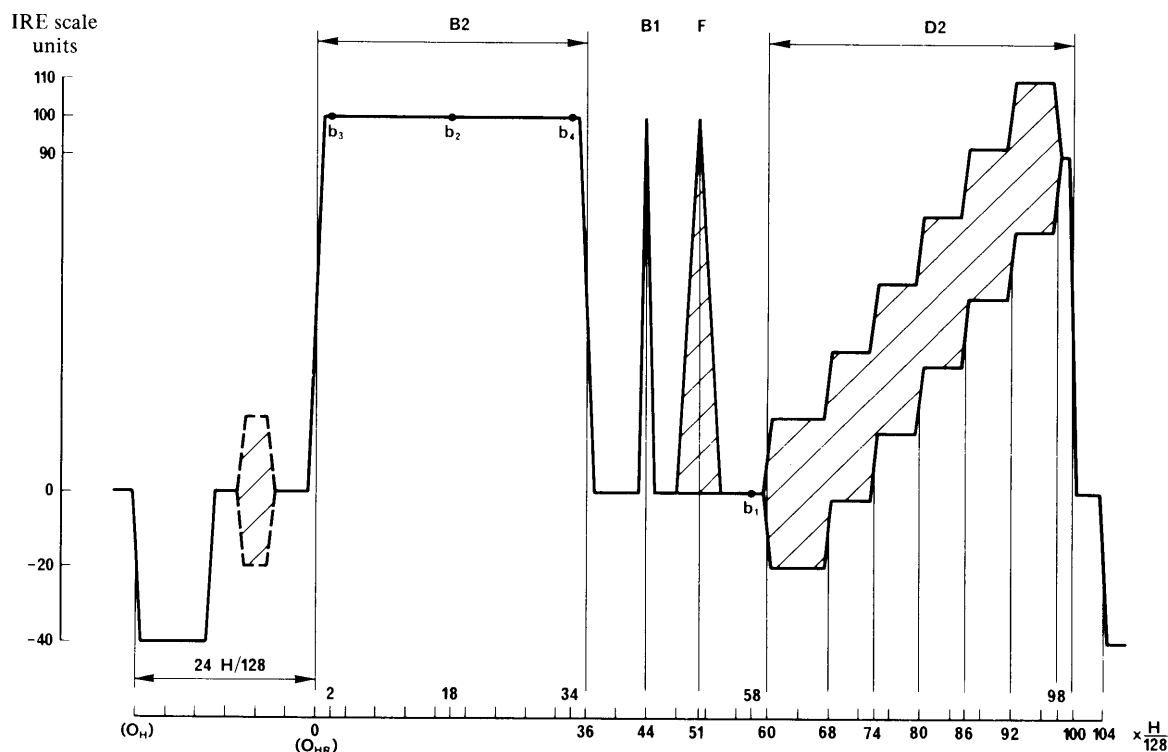


FIGURE 28 – Line 17/field 1 for 525-line signals

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)).

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 are in-band interferences, 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 $75\ \Omega$ resistive, or balanced to earth with a nominal value of $124\ \Omega$ 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.

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. – Measurements for colour television systems using the unified weighting 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 signals. 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 $\pm 3\%$ at low or high average picture levels. The figure applicable to a test signal sent at 3 dB above normal amplitude is $\pm 6\%$ 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 $\pm 10\%$ at high or low average picture level. The figure applicable to a test signal sent at 3 dB above normal amplitude is $\pm 20\%$ 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 $\pm 6\%$.

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 $\pm 3\%$. 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 $100 \pm 12\%$.

The pulse lobes should lie within the limits shown in Fig. 29a for 625-line systems and in Fig. 29b for 525-line systems.

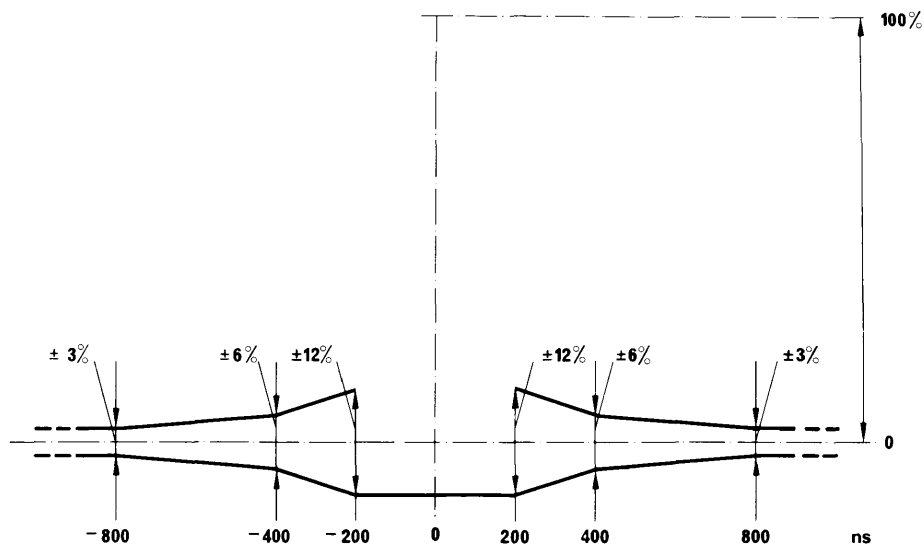


FIGURE 29a — Mask for response to test signal B1 (625-lines)

(Half-amplitude duration 200 ns)

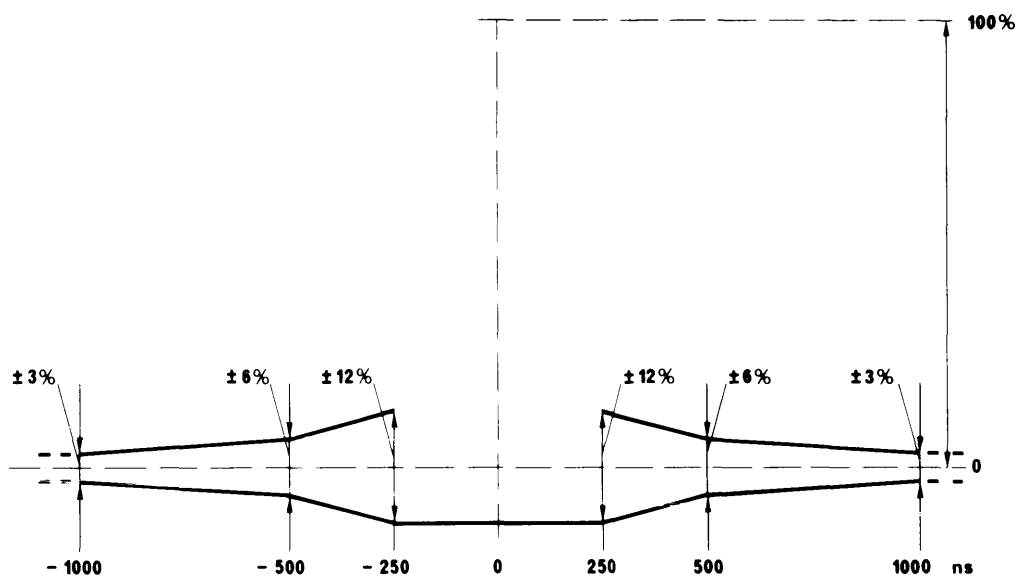
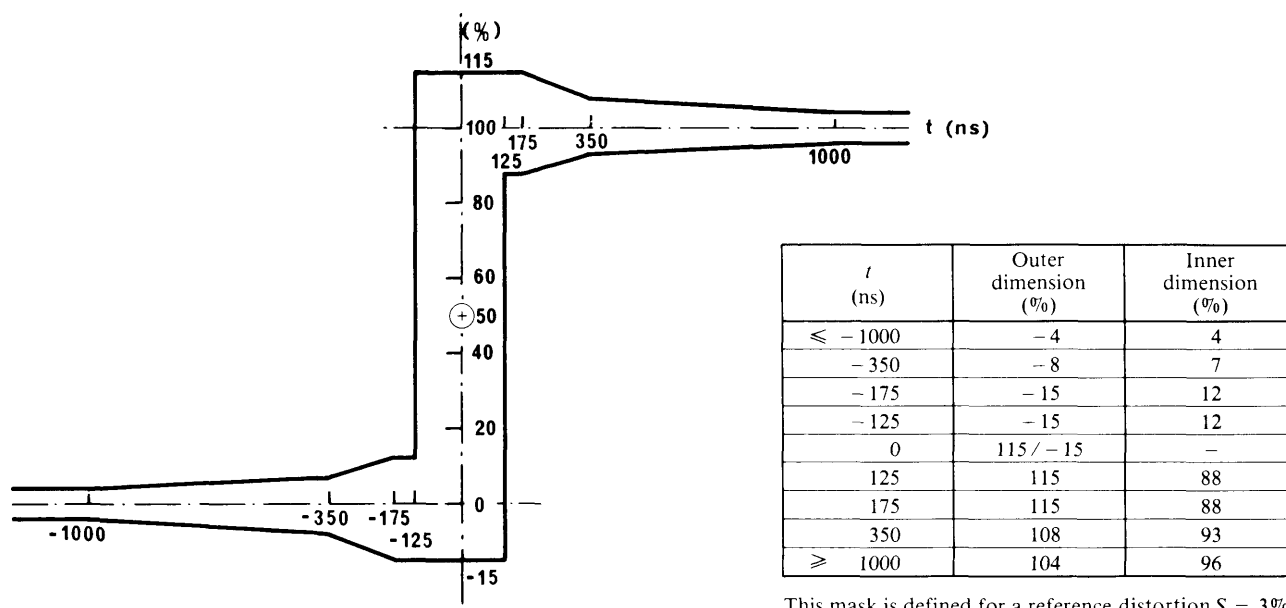


FIGURE 29b – Mask for response to test signal B1 (525-lines)
(Half-amplitude duration: 250 ns)

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



This mask is defined for a reference distortion $S = 3\%$.

FIGURE 29c – Mask for response to test signals B2/B3 (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:

$$\text{Outer dimension: } (100+) \text{ or } (0-) \quad \frac{100 \cdot S \cdot A}{100 - S \cdot A} \%_0$$

$$\text{Inner dimension: } (100-) \text{ or } (0+) \quad \frac{100 \cdot S \cdot A}{100 + S \cdot A} \%_0$$

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 1

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

D.3.5.2 Chrominance waveform distortion

It is not possible to quote figures for chrominance waveform distortion at this time.

D.3.5.3 Chrominance-luminance inequalities

D.3.5.3.1 Gain inequality

Gain inequality should not exceed $\pm 10\%$.

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

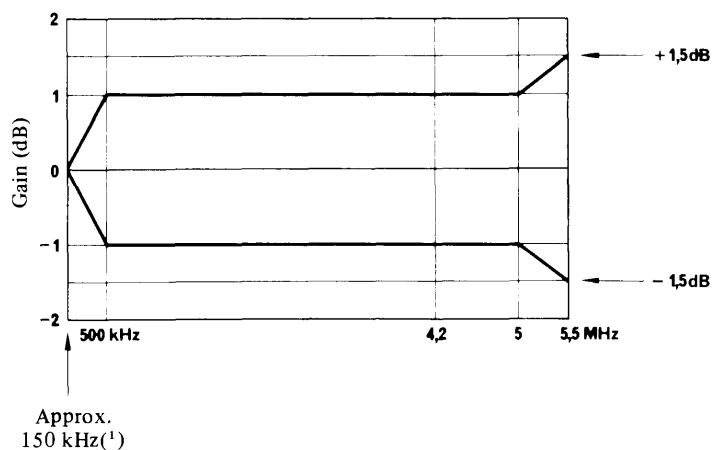


FIGURE 30 – Limits for gain/frequency characteristic

(1) May be referred to element C1.

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

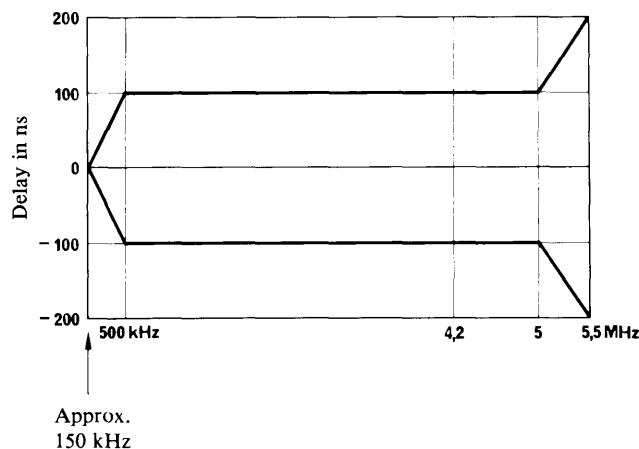


FIGURE 31 – Limits for delay/frequency response characteristic

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

REFERENCES

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).

[1978-82]: a. CMTT/74 (Canada); b. CMTT/227 (Japan).

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 II and III 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.

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 III except for "continuous random noise".

If D_3 : design objective as expressed in this Recommendation, or the parameter derived therefrom and indicated in Table II, 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 III: $h = 1$ 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 II.

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.

Note. — Further information on this additional law is to be found in [CCIR, 1970-74a].

TABLE II

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 III

§ of Part B	Characteristic	D_3 expressed in	$h^{(1)}$	Notes
3.1	<i>Insertion gain (error)</i>	dB	2	
	<i>Insertion gain variations</i>	dB	2	
3.2.1	<i>Continuous random noise</i>			1;8
3.2.2	Low-frequency noise	dB	No law	—
3.2.3	<i>Periodic noise</i> Power-supply hum Single frequency	{ Noise voltage	2 2	2;7 3
3.2.4	<i>Impulsive noise</i>	Noise voltage		4
3.3	<i>Crosstalk</i>	Crosstalk voltage	3/2	
3.4.1	<i>Non-linear distortion of the picture signal</i> Luminance Chrominance gain Chrominance phase Chrominance-to-luminance intermodulation Differential gain Differential phase	% % Degrees % % Degrees	3/2 3/2 3/2 2 3/2 or 2 3/2 or 2	— — — 9 9
3.4.2	<i>Non-linear distortion of the synchronizing signal</i> Steady state distortion	%	3/2	
3.5.1	<i>Linear waveform distortion</i> Field-time Line-time Short-time overshoot and ringing Rise-time Long-time waveform distortion	% % Mask μ s	1 2 No law No law	
3.5.3	<i>Luminance-chrominance inequalities</i> Gain inequality Delay inequality	% ns	2 2	5 5
3.5.4	<i>Steady state characteristics</i> Gain/frequency Delay/frequency	dB μ s	3/2 3/2	6 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 $n \leq 3$.

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 $h = 3/2$.

E.2.4 Tables and formulae 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 IV 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 V 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 IV

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 V

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).

BIBLIOGRAPHY

D'AMATO, P. [April, 1976] The determination of tolerances for chains of television circuits. *EBU Rev. Tech.*, 156.

REPORT 815-1

**TRANSMISSION PERFORMANCE OF SATELLITE UP-LINKS FOR
TELEVISION BROADCASTING**

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

(1978-1982)

The performance requirements for broadcast satellite up-links were examined by the CMTT at the Final Meeting as a result of a request from the Director, CCIR, in a letter RF(XIV)0-3 dated 21 February 1977.

Bearing in mind the possible interface conditions between the up-link and the broadcast transmitter, which may not be at baseband, and the fact that the up-link comprises only a part of a circuit in the fixed-satellite service, the CMTT wishes, as a possible aid to further studies regarding broadcast satellite up-link transmission performance, to comment as follows:

Recommendation 567 contains the definition of the hypothetical reference circuit for television transmission in the fixed-satellite service. In addition, Report 965 contains some transmission performance values for circuits over systems in the fixed-satellite service and some design values for a European regional system.

Since the hypothetical reference circuit for the fixed satellite service contains only a single section it would be desirable that the up-link of a broadcast satellite be considered as a fractional part of the hypothetical reference circuit for the fixed-satellite service. For practical purposes, an earth station-satellite-earth station loop can be considered equivalent to the hypothetical reference circuit, and the performance of the loop can be measured if the earth station is equipped with a suitable measuring receiver. In view of the probable configuration of the broadcast satellite, e.g. the method of interface between the up-link and the broadcast transmitter, the up-link can be considered, for most of the video transmission parameters, equivalent to half of a hypothetical reference circuit for the fixed-satellite service. On the basis of the list of characteristics given in Report 965, attention should be paid to Part E of Recommendation 567 which indicates the method to be applied to determine the transmission performance of circuits with fewer sections than the hypothetical reference circuit.

With regard to the noise performance of the up-link, it is thought that the provision contained in the second paragraph of § 3.3 in Annex 8 to the Final Acts of the World Administrative Radio Conference on the broadcasting-satellite service (WARC-BS-77), should be taken into account for any signal-to-noise ratio requirement which may be derived for the up-link on the basis of information in Recommendation 567. The provision of § 3.3 in Annex 8 to the Final Acts of WARC-BS-77 states that "the reduction in the quality in the down-link * due to thermal noise in the up-link is taken as equivalent to a degradation in the down-link carrier-to-noise ratio not exceeding 0.5 dB for 99% of the worst month".

At present a digital up-link seems less likely than an analogue up-link for two reasons:

- for a current type of satellite transponder the broadcast signal would be modulated in the same way as the up-link signal;
- present technology is such that the decoder required to handle a low bit rate, high quality television signal from a broadcast satellite would be relatively expensive.

However, Report 952 indicates that very stringent protection ratios may be required in planning the up-link (10 dB higher than the values for the down-link). This in turn may necessitate different methods of transmission for the up-link and for the down-link, and therefore a need for demodulation and remodulation in the satellite. Two methods for transmission for the up-link are envisaged:

- frequency modulation with a greater deviation;
- digital modulation.

In the future, digital satellite broadcasting may be possible if advancements in integrated circuit technology lead to low-cost decoders for television receivers. Digitally-coded broadcast signals are particularly attractive for such applications as subscription television since encryption and other processes can be carried out more easily when the signals are in digital form.

* Strictly speaking the broadcasting path.

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 (U.S.S.R.).

REPORT 637-2

SIGNAL-TO-NOISE RATIO IN TELEVISION

Effect of the de-emphasis network, alone or in combination with a weighting network

(Question 13/CMTT)

(1974-1978-1982)

1. Introduction

In television, the effect of the unified weighting network on the signal-to-noise ratio, for both "white" noise and "triangular" noise, is given as a theoretical weighting factor in Recommendation 567, Part C, Annex II. The appropriate weighting factor, in decibels, is added to the signal-to-unweighted noise ratio to obtain the signal-to-weighted noise ratio in a given transmission system.

For those transmission systems which use pre-emphasis, the corresponding de-emphasis network will also affect the signal-to-noise ratio. Where a weighting network is also used the signal-to-noise ratio is again affected.

This Report presents conversion factors, in decibels, which may be used to convert values of signal-to-noise ratio obtained without de-emphasis or weighting, to values of signal-to-noise ratio where the noise considered is weighted, or de-emphasized, or both weighted and de-emphasized. The emphasis characteristic is given in Recommendation 405. For unified weighting the network is in accordance with Recommendation 567. Conversion factors, appropriate to weighting networks shown in Recommendations 421-3 and 451-2 (Geneva, 1974), are also presented. These factors were in use prior to the introduction of the weighting network given in Recommendation 567. Further study is required to determine the effect on the signal-to-noise ratio of the pre-emphasis network used in cable carrier systems for television transmission. The characteristics of the pre-emphasis network used in the 18 MHz and 60 MHz cable systems are described in CCITT Recommendation J.77.

2. Conversion factors

Table I specifies the de-emphasis and weighting characteristics which apply to 525-line and 625-line television systems.

The de-emphasis characteristic for all systems is given by:

$$h(f) = k^{\frac{1}{2}} \left[1 + (f/f_a)^2 \right]^{\frac{1}{2}} / \left[1 + (f/f_b)^2 \right]^{\frac{1}{2}} \quad (1)$$

Similarly, the weighting characteristics are given by:

Unified weighting:

$$g_3(f) = \left[1 + (f/f_2)^2 \right]^{\frac{1}{2}} / \left[1 + (f/f_1)^2 \right]^{\frac{1}{2}} \quad (2)$$

For all systems except system M (prior to the introduction of the unified network):

$$g_1(f) = 1 / \left[1 + (f/f_1)^2 \right]^{\frac{1}{2}} \quad (3)$$

For system M (prior to the introduction of the unified network):

$$g_2(f) = \left[1 + (f/f_3)^2 \right]^{\frac{1}{2}} / \left\{ \left[1 + (f/f_1)^2 \right]^{\frac{1}{2}} \times \left[1 + (f/f_2)^2 \right]^{\frac{1}{2}} \right\} \quad (4)$$

For the expressions shown above all parameters are specified in Table I. The nominal upper video frequency limit, f_c , is specified for each system in Table I.

TABLE I

System	k	f_a (MHz)	f_b (MHz)	f_1 (MHz)	f_2 (MHz)	f_3 (MHz)	f_c (MHz)
M	10	0.875	0.187	0.270	1.37	0.390	4.2
B, C, G, H	12.59	1.565	0.313	0.482	—	—	5
I (luminance)	12.59	1.565	0.313	0.796	—	—	5.0
I (chrominance)	12.59	1.565	0.313		see Annex I		6.0
D, K, L	12.59	1.565	0.313	0.482	—	—	6
Unified weighting	—	—	—	0.5315	2.923	—	—

In general, the conversion factor F , whether for emphasis, weighting or emphasized weighting, is given by:

$$F = -10 \log \frac{\int_{f_l}^{f_c} H_1^2 df}{\int_{f_l}^{f_c} H_2^2 df} \quad \text{dB} \quad (5)$$

Where f_c and f_l are the nominal upper and lower frequency limits respectively, limiting the noise bandwidth during measurements. Nominally, it is assumed that $f_l = 0.01$ MHz.

H_1 is the spectral characteristic of the noise after passage through the network in question, and H_2 is the spectral characteristic of the input noise.

Thus,

$$\frac{H_1}{H_2} = h(f) \text{ or } g(f) \text{ or } h(f) g(f) \quad (6)$$

and $H_2 = 1$, or f , or $h(f)$, or $f \cdot h(f)$ for white noise, triangular noise, de-emphasized white noise and de-emphasized triangular noise respectively.

The complete theoretical expressions for the conversion factors, $F(\text{dB})$, are given in Annex II to [CCIR, 1970-74a], in Annex II to [CCIR, 1970-74b] and in Annex II to [CCIR, 1974-78].

Fig. 1 is a block diagram showing the conversion factors F .

The numerical values of the conversion factors as they apply to various 525-line and 625-line television systems are given in Table II.

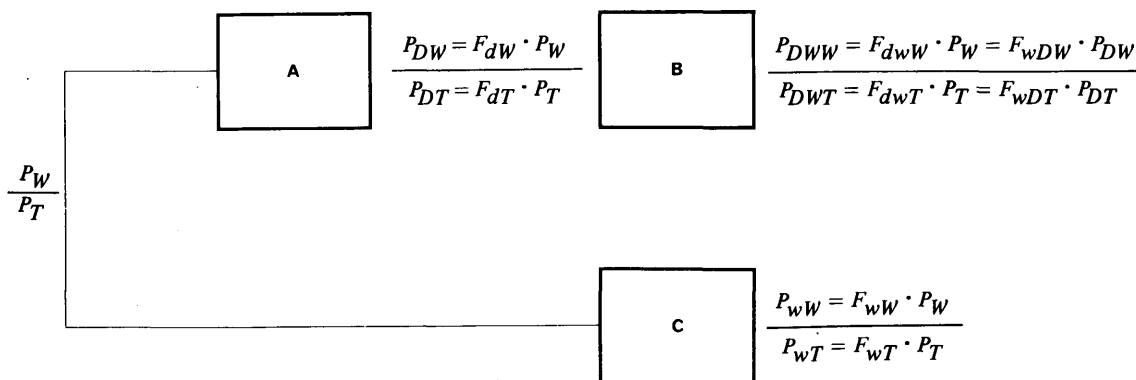


FIGURE 1 – Block diagram with conversion factors F

For definitions of terms see Table II

A: de-emphasis network

B: } weighting networks
C: }

TABLE II

Measurement bandwidth (MHz)	Weighting	De-emphasis factors (dB)		Weighting factors (dB)				Weighting plus de-emphasis factors (dB)	
		(a) White noise $F_{dW}^{(1)}$	(b) Triangular noise $F_{dT}^{(1)}$	(c) White noise F_{wW}	(d) Triangular noise F_{wT}	(e) De-emphasised white noise $F_{wDW}^{(1)}$	(f) De-emphasised triangular noise $F_{wDT}^{(1)}$	(g) White noise $F_{dwW}^{(1)}$	(h) Triangular noise $F_{dwT}^{(1)}$
4.2 4.2 5.0	M ⁽²⁾ Unified Unified	525-lines		6.2 6.8 7.4	10.3 11.5 12.2	2.7 2.7 3.3	9.9 10.9 11.7	2.3 2.3 3.3	12.8 13.8 14.8
		-0.4	2.9						
		-0.4	2.9						
5.0 5.0 5.0/6.0 ⁽²⁾	B, C, G, H I (luminance) I (chrominance)	625-lines		8.5 6.5 4.8	16.3 12.3 0.8	3.1 2.3 9.4	14.3 10.9 1.3	1.0 0.2 7.3	16.3 12.9 3.3
		-2.1	2.0						
		-2.1	2.0						
5.0 6.0 6.0/5.0 ⁽²⁾	Unified D, K, L Unified	-2.1	2.0	7.4 9.3 8.2	12.2 17.7 14.6	3.0 3.5 3.2	11.2 15.8 13.3	0.9 1.9 1.6	13.2 18.1 15.6
		-1.6	2.3						
		-1.6	2.3						
6.0	Unified	-1.6	2.3	8.0	12.8	3.3	12.0	1.7	14.3

⁽¹⁾ (a) + (e) = (g)

(b) + (f) = (h)

⁽²⁾ The first figure is the measurement bandwidth for the unweighted noise and the second figure is the measurement bandwidth for the de-emphasised noise, weighted noise and/or weighted de-emphasised noise.

REFERENCES

CCIR Documents
[1970-74]: a. CMTT 10 (Canada) and Corr.1; b. CMTT 225 (U.S.S.R.).
[1974-78]: CMTT 64 (U.S.S.R.).

ANNEX I

SYSTEM I (CHROMINANCE) WEIGHTING NETWORK CHARACTERISTICS

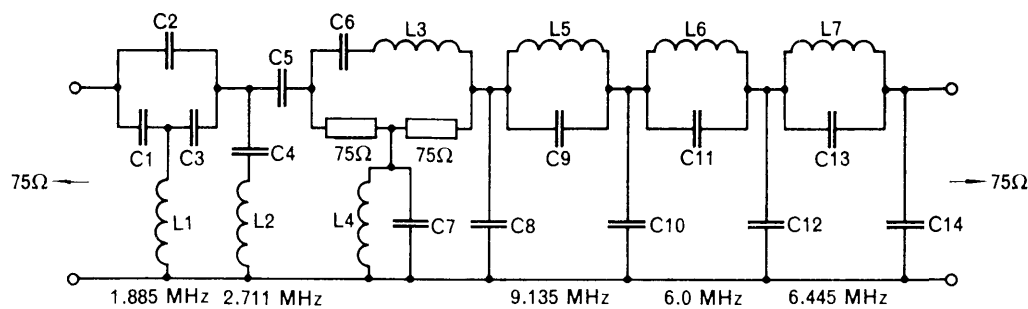


FIGURE 2 – Weighting network design diagram
(non-constant resistance)

TABLE OF VALUES

Component	Value	Tolerance	Component	Value	Tolerance
C1	496.0	± 1 %	C12	311.4	± 1 %
C2	89.47		C13	619.2	
C3	292.1		C14	187.5	
C4	715.8		L1	2.960	Note 2
C5	1239.0		L2	4.814	
C6	194.3		L3	6.650	
C7	1182		L4	1.093	
C8	385.7		L5	2.149	
C9	141.3		L6	0.7476	
C10	418.6		L7	0.9846	
C11	941.2				

Note 1. – Inductances are given in μH , capacitances in pF .
Note 2. – L3 is adjusted to resonate with C6, and L4 with C7 at 4.428 MHz. L1, L2, L5, L6 and L7 are adjusted to give maximum insertion loss at the appropriate indicated frequencies.
Note 3. – The Q -factor of each inductor should be equal to, or greater than, 100 between 3 MHz and 6 MHz.
Note 4. – The insertion loss is equal to, or greater than, 35 dB at frequencies above 6 MHz.

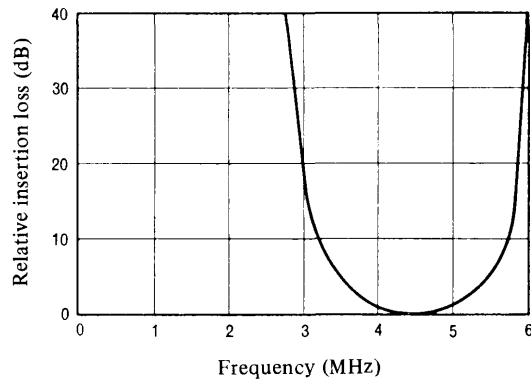


FIGURE 3 – Characteristics of the weighting network

REPORT 816-1

CHARACTERISTICS, METHODS OF MEASUREMENT AND DESIGN OBJECTIVES FOR INTERNATIONAL TELEVISION CIRCUITS

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

(1978-1982)

1. Introduction

The Joint CCIR/CCITT Study Group for Television and Sound Transmission (CMTT) has reached agreement on a large number of aspects of the problems which occur when transmitting television signals of various standards over international circuits. These agreements are recorded in Recommendation 567.

However, this Recommendation recognizes the need for further study of a number of aspects, and it may not cover any new aspects which arise in the future. Examples of outstanding requirements are:

- to further the study of performance objectives, test methods and test signals already quoted in Recommendation 567, in order to give an answer to outstanding questions and to achieve better unification of the transmission for 525- and 625-line television systems;
- to stimulate the study of any aspects not investigated so far, e.g. the correlation of design objective values for different parameters, the statistical variation of transmission performances with time, transient performance, etc.

This Report is intended to facilitate consideration of material which may be contributed in this respect, so that the desirability of transferring this information to Recommendation 567 may be established.

It has been found convenient to identify this material under the same headings as in the Recommendation, noting that this may include information on the behaviour of actual circuits which has a bearing on the establishment of measurement methods and performance objectives. Contributions of this nature are to be encouraged.

The sub-division of the Report is as follows:

- Part A: Definition of a connection and circuits
- Part B: Definitions of parameters
- Part C: Measurement methods and test signals
- Part D: Design objectives and tolerances applicable to the hypothetical reference circuits
- Part E: Performance of circuits shorter or longer than the hypothetical reference circuit.

Additional parts may be required in the future if contributions are made to aspects which have not been covered above.

2. Summaries of relevant information

A. *Definition of a connection and circuits*

No new information is available at present.

B. *Definitions of parameters*

B.1 The definitions and corresponding test methods described in Recommendation 567 for noise do not take into account the fact that noise may, in certain circumstances, be a function of the television signal being transmitted. This problem is considered in a contribution on truncation noise [CCIR, 1974-78a].

C. *Measurement methods and test signals*

C.1 Discussion of the need to continue consideration of interference effects of test signals, particularly full field signals. Texts in which possible interfering effects of test signals should be noted are suggested [CCIR, 1974-78b];

C.2 Proposal and rationale for unifying the time duration of the luminance bar for insertion test signals (ITS) and full field measurements is given in [CCIR, 1974-78c];

C.3 Proposed assembly of test signal elements to allow measurement of linear waveform distortions and chrominance-luminance intermodulation in a single line period [CCIR, 1974-78d];

C.4 In the United Kingdom a variation on test signal element C, shown in Figs. 9 and 10 of Recommendation 567 has been developed. This consists of a series of composite pulses, (each based on the principle illustrated in Fig. 13 of Recommendation 567, but with different carrier frequencies), to form a new signal, generally known as the multipulse.

This development offers two advantages:

- it makes possible the assessment of relative delay as well as gain,
- it obviates the interaction between adjacent bursts which can occur with the current signal because of a lack of specification of burst edge rise time.

This is suggested as a proposed alternative to, and possible eventual replacement for, the existing 625-line multiburst signal in Recommendation 567, and possibly in the longer term, for line 18 of the 625-line insertion test signal in Recommendation 473. Additional information on this development is contained in [CCIR, 1978-82a; Holder, 1977];

C.5 A technique for the measurement of line synchronizing pulse edge jitter is described in [CCIR, 1978-82b]. Certain television processing equipments such as video tape recorders are sensitive to this effect and administrations are invited to suggest limiting values of jitter which might be permitted. Jitter in digital television transmission systems is being dealt with in Report 967.

D. *Design objectives*

D.1 Design objectives for chrominance non-linearity and transient distortion of the synchronizing signal are suggested in [CCIR, 1974-78e];

D.2 The impact of non-linear distortion on television signals which have been modified by high frequency pre-emphasis has been studied [CCIR, 1978-82c]. It has been concluded that the measurement of luminance non-linearity with test signals D_1 or D_2 does not provide an adequate indication of the extent of the non-linear distortion which may seriously affect other luminance and chrominance signal components when high frequency pre-emphasis is used (e.g. in radio-relay and satellite circuits). Accordingly, there should be some recognition of this shortcoming when considering the objectives for the luminance non-linearity parameter which is measured according to the definitions contained in Recommendation 567. Further study is encouraged to define a test procedure for luminance non-linearity when high-frequency pre-emphasis is used.

D.3 In a contribution to the CMTT [CCIR, 1978-82d] following tests in the United Kingdom on terrestrial circuits there is an indication that the major impairment of the teletext signal occurs in the chain of television broadcast transmitters and transposers, and that a television programme distribution network, generally complying with Recommendation 567 and operated with normal maintenance directed towards the television parameters, will contribute only a small part of the total teletext signal impairment.

However, studies and field trials using satellite circuits carried out in France [CCIR, 1978-82e] indicate that truncation noise near to the FM reception threshold may impair the reception of data in the field blanking intervals of television signals, although the picture quality is still acceptable. In addition, when the frequency demodulator includes a phase-lock loop, special design precautions are necessary to obtain satisfactory data reception.

D.4 In [CCIR, 1978-82f] the results are given of interference caused by return signals, having the same carrier frequency as the wanted signal, in television channels of terrestrial FM radio-relay systems.

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

No new information is available at present.

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[1978-82] a. CMTT/19 (United Kingdom); b. CMTT/238 (UER); c. CMTT/76 (Canada); d. CMTT/27 (United Kingdom); e. CMTT/265 (France); f. CMTT/279 (U.S.S.R.).

REPORT 965

TRANSMISSION PERFORMANCE OF TELEVISION CIRCUITS OVER SYSTEMS IN THE FIXED-SATELLITE SERVICE

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

(1982)

1. Introduction

Since satellite circuits are designed and operated under limitations such as satellite equivalent isotropically radiated power (e.i.r.p.), occupied radio-frequency bandwidth, etc. it is necessary at this time to allocate values to some performance parameters that differ from the design objectives for the hypothetical reference circuit set out in Section D of Recommendation 567.

The values below are given as a reference for designers to assist them in determining the performance of chains which include circuits employing a satellite.

2. Maintenance values for the INTELSAT satellite system

Maintenance values for the INTELSAT satellite system are given in CCITT Recommendation N.62).

Additionally, the INTELSAT satellite system operations guide gives the following values for the signal-to-noise ratio (unified weighting):

525/60 Systems, full transponder: 53.3 dB
625/50 Systems, full transponder: 50.1 dB
525/60 Systems, half transponder: 48.7 dB
625/50 Systems, half transponder: 47.1 dB

3. Practical design values for a regional system

Practical design values for a European regional system are given in Table I.

In this table, column (a) gives values for the earth station/satellite/earth station segment. They have been justified by a series of tests with the European Space Agency (ESA) Orbital Test Satellite (OTS), which was launched in May, 1978 [EBU — Interim EUTELSAT, 1981].

Column (b) gives values for a terrestrial link connecting the earth station to a national technical control centre (CNCT) and which typically has a length of about 100 km. The values given are based largely on those in Recommendation 567 and would apply for lengths up to about one third of a hypothetical reference circuit.

Column (c) gives the resultant values for the whole chain from CNCT to CNCT, including terrestrial links, earth stations and the satellite. Wherever possible, the laws of addition specified in Recommendation 567 have been used.

TABLE I

§ of Part D of Rec. 567	Parameters	(a)	(b)	(c)
2.1	Nominal impedance	75 Ω		
2.2	Return loss	30 dB		
2.3	Non-useful d.c. component	0.5 W		
2.4	Nominal signal amplitude	1 V		
3.1	Insertion gain	0 $\pm$ 0.25 dB	0 $\pm$ 0.3 dB	0 $\pm$ 0.5 dB
3.1.1	Insertion gain variation (1 s)	$\pm$ 0.1 dB	$\pm$ 0.2 dB	$\pm$ 0.3 dB
3.1.1	Insertion gain variation (1 h)	$\pm$ 0.25 dB	$\pm$ 0.3 dB	$\pm$ 0.5 dB
3.2.1	Signal-to-continuous random noise	53 dB ⁽¹⁾	58 dB	51 dB
3.2.3	Signal-to-periodic noise (0-1 kHz)	50 dB	45 dB	39 dB
3.2.3	Signal-to-periodic noise (1 kHz - 6 MHz)	55 dB	60 dB	53 dB
3.2.4	Signal-to-impulsive noise	25 dB	25 dB ⁽²⁾	25 dB ⁽²⁾
3.3	Crosstalk between channels (undistorted)	58 dB	64 dB	56 dB
3.3	Crosstalk between channels (differentiated)	50 dB	56 dB	48 dB
3.4.1.1	Luminance non-linear distortion	10%	2%	12%
3.4.1.2	Chrominance non-linear distortion (amplitude)	3.5%	2%	5%
3.4.1.2	Chrominance non-linear distortion (phase)	4°	2°	6°
3.4.1.3	Differential gain (x or y)	10%	5%	13%
3.4.1.3	Differential phase (x or y)	3°	2°	6°
3.4.1.4	Chrominance-luminance intermodulation	$\pm$ 4.5%	$\pm$ 2%	$\pm$ 5%
3.4.2.1	Steady-state sync pulse non-linear distortion	+ 5% - 10%	$\pm$ 5%	+ 10-15%
3.4.2.2	Transient sync pulse non-linear distortion	20%		
3.5.1.2	Field-time waveform distortion	6%	2%	10%
3.5.1.3	Line-time waveform distortion	3%	2%	4%
3.5.1.4	Short-time waveform distortion (pulse/bar)	100 $\pm$ 12%	100 $\pm$ 6% ⁽²⁾	100 $\pm$ 18% ⁽²⁾
3.5.1.4	Short-time waveform distortion (pulse lobes)	3% ⁽³⁾	1.5% ⁽³⁾	4.5% ⁽³⁾
3.5.3.1	Chrominance-luminance gain inequality	$\pm$ 10%	$\pm$ 6%	$\pm$ 13%
3.5.3.2	Chrominance-luminance delay inequality	$\pm$ 50 ns	$\pm$ 60 ns	$\pm$ 90 ns
3.5.4.1	Gain/frequency characteristic (0.15 - 6 MHz)	$\pm$ 0.5 dB	$\pm$ 0.5 dB ⁽⁴⁾	$\pm$ 1.0 dB ⁽⁴⁾
3.5.4.2	Delay/frequency characteristic (0.15 - 6 MHz)	$\pm$ 50 ns	$\pm$ 50 ns ⁽⁴⁾	$\pm$ 105 ns ⁽⁴⁾

(1) In cases where the receive earth-station is colocated with the broadcaster's premises, a relaxation of up to 3 dB in video signal-to-weighted-noise ratio may be permissible. In this context, the term "colocated" is intended to represent the situation where the noise contribution of the local connection is negligible.

(2) Law of addition not specified in Recommendation 567.

(3) The pulse lobes are contained within a mask of the type shown in Fig. 29a in Recommendation 567. The figure in column (a) is the amplitude of the mask in Fig. 29a for times ≤ -800 ns and ≥ 800 ns. The figures in columns (b) and (c) are corresponding amplitudes of scaled versions of the mask.

(4) Highest frequency: 5 MHz.

REFERENCE

EBU — Interim EUTELSAT [1981], Analogue television transmission tests with OTS.

REPORT 636-2

**LONG-TIME WAVEFORM DISTORTION
IN LONG-DISTANCE TELEVISION CIRCUITS**

(Study Programme 13B/CMTT)

(1974-1978-1982)

1. Introduction

The d.c. component of a television signal is not preserved when the signal is transmitted over an a.c.-coupled television circuit. A sudden change in d.c. component, such as results at the output of a television picture source when a bright scene immediately follows a dark scene, is equivalent to introducing a voltage step to the input to the transmission circuit. The blanking of the television signal restricts the maximum amplitude of this step to approximately 80% of the luminance signal nominal value. The response to this step at the output of the circuit depends upon the low-frequency cut-off characteristics of the various equipments in the circuit and is normally of a damped quasi-oscillatory nature on a time scale measured in seconds. This effect is commonly referred to as bouncing or bumping, but a better term is "long-time linear waveform distortion".

2. Effects of damped oscillation

The quasi-oscillatory response may not be serious in itself provided that the damped oscillation added to the television signal produces no unwanted side effects and can be removed by the clamp at the broadcast transmitter. The oscillation does, however, increase the dynamic voltage range occupied by the signal and this must be allowed for by system designers so that it does not give rise to significant transient non-linear distortion of the signal.

Probably the most important transient non-linear distortion effects are compression of the synchronizing pulses and amplitude and phase variations of the chrominance signal. Amplitude and phase variations of the luminance signal can also occur. These distortions have steady-state counterparts, as shown in Part B of Recommendation 567 and it is suggested that they should be termed "transient synchronizing pulse non-linearity", "transient luminance non-linearity", "transient chrominance gain non-linearity" and "transient chrominance phase non-linearity" respectively.

It is necessary to consider these distortions when deciding upon the dynamic voltage range which is likely to be delivered to the input of a chain, the resultant dynamic range allowable at the output of this chain, and the consequent range at any intermediate video interconnection points. In deciding the range allowable at the output, consideration must be given not only to the transient distortion characteristics of the chain itself, but also to those of the equipment to be fed by the chain.

3. Growth of oscillation

The way in which the oscillation grows as the number of equipment components (e.g. amplifiers) connected in cascade increases, is dependent upon their design. There is evidence to suggest that with ideal design the peak amplitude of the damped oscillation becomes asymptotic to a maximum value of 40% of the input voltage step as the number of items of equipment is increased [Comber and McDiarmid, 1972]. Non-ideal design can, however, result in the amplitude increasing without limit as the number of items is increased; a state of affairs which requires the introduction of additional clamps at intermediate points.

A system design philosophy should therefore have the following characteristics:

- it should ensure that non-linear distortions are kept within certain limits;
- it should be applicable to both future and existing heterogeneous networks whose various types of amplifiers show a different behaviour as regards long-time waveform distortion;
- it should ensure that, if sub-networks established and maintained separately are interconnected, non-linear distortion does not increase without control. The same applies to circuits whose configuration deviates from that of the hypothetical reference circuit, i.e. circuits having, for instance, more than three video-to-video sections.

Such a philosophy is described in [CCIR, 1970-74].

4. Measurement and specification

Test signals suitable for use in measuring long-time waveform distortion (and its possible side-effects) can be constructed from elements shown in Recommendation 567. Methods of performance specification are still under study, but the following suggestions may be of help:

4.1 The test signal construction should be such that its average picture level (APL) can be stepped between high and low values.

Note. — A value of 10% or 12½% is accepted as a low average picture level and a value of 90% or 87½% as a high average picture level.

4.2 Long-time linear waveform distortion can be assessed by observation of the excursion of the received test-signal blanking-level following upon a step in the average-picture level, e.g., by means of a d.c. coupled, long-persistence oscilloscope. Probably the most important parameter to be measured is the peak amplitude of any overshoot beyond the final steady-state level, (expressed as a percentage of luminance signal nominal value). Two other parameters can be measured:

- the time taken for any oscillation to decay to a specified value;
- the slope at the beginning of the phenomenon expressed in volts per second.

If the circuit under test is very non-linear it may be necessary to reduce the change in APL, or the test signal level, in order to carry out the above test. Where this is necessary the limits proposed in § 4.6 for the linear component should be changed in direct proportion to the reduction of change in APL.

4.3 Transient synchronizing pulse non-linearity can also be observed on a long persistence oscilloscope, but in this case measurement can be facilitated by first passing the received test signal through a differentiating network of the type suggested for the measurement of luminance non-linearity in Recommendation 567. The distortion is defined as the maximum instantaneous departure from nominal amplitude following a step in APL (expressed as a percentage of nominal synchronizing pulse amplitude).

4.4 Transient luminance non-linearity can be measured as in § 4.3 above, when element *D1* (Recommendation 567) is included in the test signal. The distortion is defined as the maximum instantaneous difference between the largest and smallest pulse amplitude in any of the resultant trains of five pulses following a step in APL (expressed as a percentage of the largest).

4.5 Transient chrominance non-linearity can be measured by including the 3-level chrominance elements *G* or *G2* (Recommendation 567) in the test signal. The distortions (gain and phase) are defined as the maximum instantaneous values obtained, following a step in APL, when measured in accordance with § 3.4.1.2 of Part C of Recommendation 567.

4.6 The magnitude of distortion for the hypothetical reference circuit requires further study, but the following values have been suggested as limits:

- long-time linear waveform distortion: 40-60%,
- transient synchronizing pulse non-linearity: 20%,
- transient chrominance gain non-linearity: 16%,
- transient chrominance phase non-linearity: 8°.

4.7 It is suggested that, in estimating the performance of circuits differing in length from that of the hypothetical reference circuit (see Part E of Recommendation 567), a value of 3/2 should be used for *h*. In some long distance national circuits where all the video amplifiers are of uniform design with near-ideal characteristics, it has been suggested [CCIR, 1978-82] that a value of 2 should be used for *h* to determine the performance for circuits longer or shorter than the hypothetical reference circuit. The document also suggests that for circuits where *h* = 2 the limit for the long-time waveform distortion of an hypothetical reference circuit should be 40%.

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[1974-78]: CMTT/19 (United Kingdom); CMTT/24 (OIRT).

REPORT 635-1

ALLOCATION OF TOLERANCES FOR COLOUR TELEVISION

(Study Programme 13C/CMTT)

(1974-1978)

The subject of Study Programme 13C/CMTT is the allocation of the total tolerances among the component parts of an overall television system from the picture source up to and including the receiver, taking into account the statistical behaviour of the performance of the equipment and the fact that the subjective quality of a colour television picture is affected by the objective performance of the component parts of the system as detailed.

Studies on some aspects of this problem are in progress in Study Group 11, as shown in Report 405 and studies on other aspects have also been reported [CCIR, 1970-1974a; Lari *et al.*, 1974]. In particular such studies concern:

- the influence on picture quality of some types of distortion simultaneously present on 625-lines PAL colour-coded signals
- the addition laws of some distortions *
- the addition law of noise.

On the basis of the results of these studies, an attempt has been made to allocate the tolerances of some parameters among the component parts of a broadcasting television system. The system considered is shown in Fig. 1 and consists of a:

- television studio,
- television hypothetical reference circuit,
- hypothetical reference circuit for an active communication-satellite system,
- television hypothetical reference circuit,
- broadcasting transmitter,
- domestic receiver.

The only parameters taken into account are the following:

- differential phase,
- differential gain,
- chrominance-luminance gain inequality,
- chrominance-luminance delay inequality,
- signal-to-noise ratio.

It is assumed that the above-mentioned distortions are simultaneously present.

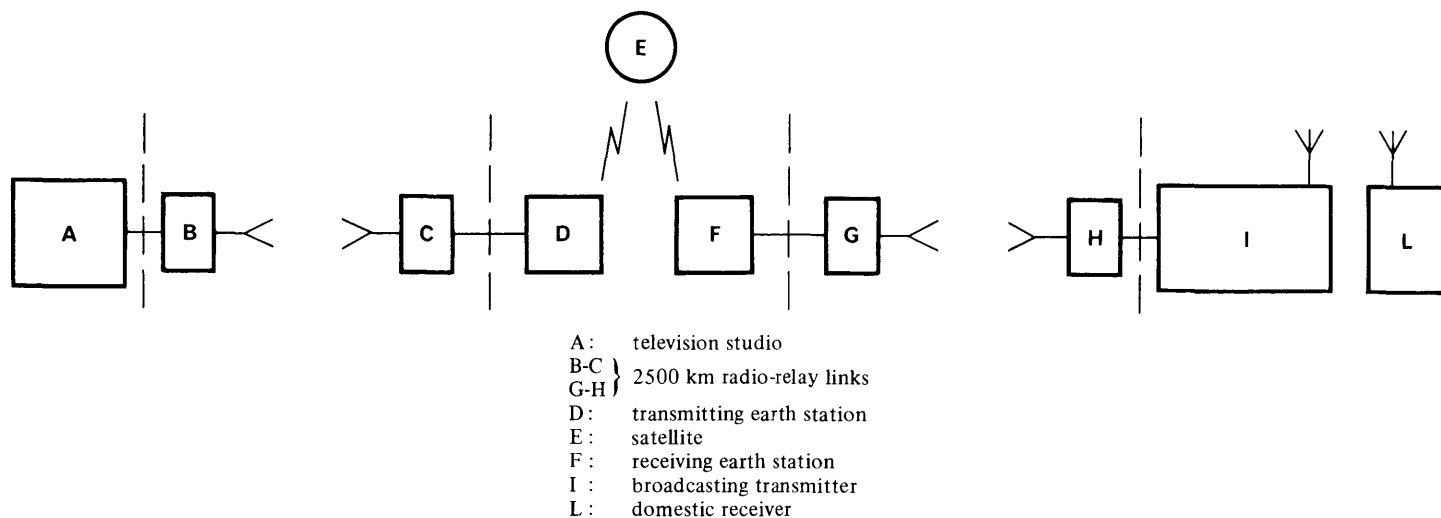


FIGURE 1 – Television system considered

* The variance of each of certain parameters of the entire television chain is equal to the sum of the variances of the individual component parts (connected in cascade).

The global mean value of a given parameter at the downstream end of the television chain is the sum of the means of the individual component parts (connected in cascade).

The assumed tolerances and the related calculation hypotheses, based on experience, are those quoted in Table I and refer to television system G.

Noise has been weighted according to the weighting characteristic shown in [CCIR, 1970-74d].

TABLE I – Tolerances for 625-line system G

Parameter → Component part ↓	Differential phase (degrees)	Differential gain (%)	Chrominance-luminance gain inequality (%)	Chrominance-luminance delay inequality (ns)	Signal-to-noise ratio (weighted) (dB)
Studio	± 5 ⁽¹⁾	± 5 ⁽¹⁾	± 5 ⁽¹⁾	± 10	48
Terrestrial circuit (2500 km)	± 5 ⁽¹⁾	± 10 ⁽¹⁾	± 10 ⁽¹⁾	± 50	56 ⁽²⁾
Satellite circuit	± 4 ⁽¹⁾	± 8 ⁽¹⁾	± 5 ⁽¹⁾	± 30	56
Broadcasting transmitter	± 5 ⁽¹⁾	± 10 ⁽¹⁾	± 15 ⁽¹⁾	± 50	—
Domestic receiver	± 6 ⁽⁵⁾	± 10 ⁽⁵⁾	— ⁽³⁾	± 100	52 ⁽⁴⁾

⁽¹⁾ Statistically variable and not exceeded at least for the 80% of any month.

⁽²⁾ Exceeded at least for 80% of any month.

⁽³⁾ It is assumed that the receiver distortion is equalized by manual chrominance control.

⁽⁴⁾ It is assumed a field-strength of 80 dB(μV/m) and a noise figure of 10 dB.

⁽⁵⁾ This value, assumed to be constant, is added to the square root of the sum of the squares of the values given above (in the same column).

On the basis of such hypotheses and according to the distortion and noise addition laws [CCIR, 1970-74b and c], the resulting overall parameter values at the end of the system are the ones quoted in Table II.

TABLE II – Overall parameter values

Differential phase (degrees)	17 ⁽¹⁾
Differential gain (%)	31 ⁽¹⁾
Luminance-chrominance gain inequality (%)	22 ⁽¹⁾
Luminance-chrominance delay inequality (ns)	290 ⁽²⁾
Weighted signal-to-noise ratio (dB)	45 ⁽³⁾

⁽¹⁾ Not exceeded at least for 80% of any month.

⁽²⁾ Exceeded at least for 80% of any month.

⁽³⁾ Worst case, arithmetic addition.

The picture quality at the end of the broadcasting system has been determined considering, firstly, the case in which the domestic receiver employs an automatic chroma control (ACC), which equalizes the luminance-chrominance gain inequality of the overall system, and, secondly, the case in which the domestic receiver is not equipped with such a device.

The following results were obtained for the picture quality at the end of the system:

- domestic receiver with ACC: grade 3.6 *
- domestic receiver without ACC: grade 3.3

Such picture quality is obtainable for at least 41% of any month.

In both cases, domestic receivers with or without automatic chroma control, the picture quality obtainable at the end of the system for at least 80% of any month is *slightly over grade 2*. This last result shows that when the pictures are sufficiently impaired by some distortions, the effect of a further distortion has practically no influence on their quality.

The parameter values which determine the grade 2 quality are shown in Table III.

TABLE III – Overall parameter values for grade 2 quality

Differential phase (degrees)	23 ⁽¹⁾
Differential gain (%)	41 ⁽¹⁾
Luminance-chrominance gain inequality (%)	33 ⁽¹⁾
Luminance-chrominance delay inequality (ns)	290 ⁽²⁾
Weighted signal-to-noise ratio (dB)	44.5 ⁽²⁾

⁽¹⁾ Not exceeded at least for 95% of any month.

⁽²⁾ Exceeded at least for 95% of any month.

⁽³⁾ Worst case, arithmetic addition.

It has been shown [Allnatt, 1975] that in a well-designed system, the variability in magnitude of an individual impairment has, in practice, a negligible effect on the overall picture quality of the system, compared with variations in the standards of judgment of the individual viewer and between viewers, more especially the former. Although viewers can note that the most prominent impairment may not always be of the same kind, variations in opinion as to the level of picture quality are almost entirely due to the variability of the viewers themselves. It can be concluded therefore, that, in a well-designed system, the variability of impairments is of much less importance than their mean values.

Thus, if for a particular type of impairment, a scale of objective magnitude can be found which possesses the property of simple summability, it should be necessary to consider only the average values of the impairments contributed by the individual component links in a system. However, the performance of links is customarily set in terms of "tolerances", i.e. impairment magnitudes which are not to be exceeded, rather than average levels of performance.

As a preliminary to investigations of the relationship between the tolerances for individual links on the one hand and the average performance of a complete system on the other, it is helpful to classify impairments into 3 groups, roughly according to the manner in which they add when contributions to a given type of impairment arise in different parts of a system. A simple description follows. Other classifications of a more limited nature have been proposed for slightly different purposes [Podemski, 1973 and d'Amato, 1974].

Class 1 impairments will be defined as those whose magnitudes can be expressed unidimensionally in terms of simple values having either a positive or a negative sign and capable of being summed algebraically. Gain, gain inequality and delay inequality are well known examples. In suitable circumstances the impairment from one component of a system may exactly offset that from another.

Class 2 impairments can also be expressed in terms of summable values, but negative values cannot arise. Random noise is the most common example. The combined impairment of given type from a system can never be less than that arising in any of its components. Periodic noise would generally be of Class 2, but periodic noise of given waveform, always in the same phase relationship apart from sign, would fall into Class 1.

* The adopted quality scale is the following:

- 5 excellent
- 4 good
- 3 fair
- 2 bad
- 1 very bad

Class 3 impairments are the most complex, varying both in magnitude and in character, but which nevertheless are normally expressed as a simple quantity. Waveform distortion, differential gain, and differential phase are examples. Some Class 3 impairments can, in principle, be analyzed in terms of a number of Class 1 and/or Class 2 impairments if the definition and method of measurement are changed. For example, in Recommendation 567, differential gain is defined in terms of the greatest departures of sub-carrier amplitude, (from that at blanking level) as the luminance varies from blanking level to white level at the more unfavourable of two average picture levels. If this single result were replaced by 10 separate results (5 luminance levels at 2 average picture levels), then each of these could be treated (approximately) as a Class 1 impairment. Unfortunately this approach does not only unduly complicate the measurement process but also produces figures which cannot be related to any existing subjective data. Further study of how to treat Class 3 impairments is clearly required.

The classification which has been proposed is a general one, which should cover all situations and provide a starting point for studies relating mean values of impairments in a complete television chain to tolerances in the individual parts. The difficulties in dealing with Class 3 impairments are inherent in the complex nature of these impairments and are not due to the method of classification proposed.

Theoretical results concerning the influence of some types of distortion on SECAM colour coded signals, obtained by computer analysis, have been reported in [CCIR, 1974-78].

Administrations are invited to send contributions on this subject.

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REPORT 966 *

TRANSMISSION PERFORMANCE OBJECTIVES FOR TELEVISION STANDARDS CONVERTERS EMPLOYING FIELD STORES

(Study Programme 13F/CMTT)

(1982)

Since television standards converters employing field stores may be encountered in very long-distance circuits, as represented by the hypothetical reference chain, there is a need to specify the transmission performance objectives for such standards converters. It has been suggested [CCIR, 1978-82] as a basis for further consideration, that the objectives for a hypothetical reference circuit should also apply to such standards converters. However, while the design objectives for a hypothetical reference circuit are defined in Recommendation 567, it is possible that some of the measurement methods in Recommendation 567 will have to be modified and/or that additional methods will have to be devised to enable the complete transmission performance of standards converters to be measured. Further study is required.

REFERENCES

- CCIR Documents*
[1978-82]: CMTT/57 (Japan).

* This Report should be brought to the attention of Study Group 11.

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 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.

BIBLIOGRAPHY

CCIR Documents

[1978-82]: CMTT/26 (United Kingdom); CMTT/40 (France); CMTT/56 (Japan); CMTT/75 (Canada).

* This Recommendation which cancels Report 487-2 should be brought to the attention of CMBD.

RECOMMENDATION 604 *

DIGITAL TELEVISION TRANSMISSION – GENERAL PRINCIPLES

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

(1982)

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 digital television transmission, for example:
 - to carry signals to production centres where post-production processing may take place;
 - to distribute television programmes;
- (d) that it is highly desirable to ease 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 hierarchical levels 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;
3. that for each application, for which examples are given in CONSIDERING (c), a single transmission coding technique (see Note) should be used for the 525-line television signals and likewise for the 625-line television signals;
Note. – 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.
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 recommended by the CCITT.

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

REPORT 646-2 *

**DIGITAL OR MIXED ANALOGUE-AND-DIGITAL TRANSMISSION
OF TELEVISION SIGNALS**

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

(1974-1978-1982)

1. Introduction

The CMTT is studying digital coding methods, digital standards and monitoring methods which are suitable for the transmission of television and sound-programme signals over long distances. It is also studying the transmission impairments, due to the transmission links, which are tolerable.

Study Group 11 of the CCIR is making similar studies in relation to the processes carried out in the studio complex and for direct broadcasting from terrestrial transmitters and from satellites. Report 629 of Study Group 11 outlines the basic principles of coding and summarizes several proposals. A matter of particular interest is the discussion concerning component coding and its advantages in television studios. The move in favour of separate component coding for colour television signals (see Report 629 and Recommendation 601) has important implications for the CMTT [CCIR, 1978-82a], which include the possibility that a requirement for more than one transmission standard may emerge. Some of these implications are discussed in this revised version of Report 646-1.

In Recommendation 567, the CMTT has laid down the requirements for the all-analogue transmission of television signals. It is recognized that there may be a transition first to mixed analogue and digital circuits, and later to all-digital circuits, although in some cases all-digital circuits may be used from the start.

In Decision 18, the CMTT has proposed that systems employing digital or mixed digital and analogue methods should have overall subjective performances not worse than the equivalent analogue systems.

Recommendation 604 has set out the general principles for all-digital transmission of television signals.

In the course of the final CCIR meeting (Geneva, 1981) the consensus was that it was too early for specific coding and multiplexing methods to be defined, because no practical equipment or conclusive experimental evidence was available. Nevertheless, it was felt that the general statements in Recommendation 604 would give useful guidelines for the case when digital component signals are transmitted through all-digital circuits.

It has been stressed in Recommendation 604 that, when establishing transmission circuits presented with a digital signal, mixed analogue-and-digital paths should be avoided if possible and it was unanimously agreed that preference should be given to all-digital transmission circuits.

A proposed Recommendation covering mixed analogue-and-digital transmission was considered but was not adopted because of the fundamentally divergent opinions that are expressed in Annex II of this Report. The proposal is contained in Annex I of this Report as it is a valuable record of the aspects considered and is a basis for further study.

The further objective of this Report is to find a basis for more detailed proposals for coding and transmission techniques within the principles set out in Recommendation 604 and Annex I of this Report. In seeking this basis it should be noted that there is no intention in this Report to consider modifications to Recommendation 567 concerning analogue transmission.

One very important aspect, the imperfections which can occur in the actual digital transmission process and the consequent impairments which can arise, is discussed only briefly within the text of this Report. A more detailed discussion is the subject of a separate companion Report 967.

Finally, it should be noted that in this Report 646 a digital television circuit is defined as one in which the television signal is in the digital domain throughout the circuit, including terminal and intermediate points. A mixed analogue and digital circuit is defined as one in which the television signal is in the digital domain in some parts and in the analogue domain in other parts, these parts being in tandem.

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

2. Signals to be transmitted and multiplexing considerations

It is desirable that long distance television transmission circuits which utilize digital or mixed analogue-and-digital techniques should be able to carry the colour television signals in component or composite (NTSC, PAL or SECAM) form. These signals may be accompanied by others, which are programme-related, which are for monitoring and measurement or which belong to supplementary services.

2.1 *Television picture signals*

The television picture signals to be transmitted may be analogue or digitally coded in component or composite form. The parameters of the composite television signals are described in Report 624. The filtering characteristics and optimum sampling rates will be defined under Study Programme 25F/11. The characteristics of the digitally encoded television signals are under study in Study Programmes 25A/11 and 25B/11.

Information is necessary on what types of electronically-generated colour picture signals need to be transmitted, in order to allow better specification of the range of picture material that proposed coding techniques will be required to handle.

2.2 *Sound signals*

The digital television transmission systems should be able to provide at least two high quality programme sound channels for stereophonic or independent use. The necessity to transmit additional commentary signals should be studied.

2.3 *Data signals*

Methods are being studied (Study Programmes 15A/CMTT and 21B/CMTT) on how best to transmit data signals by inserting them into the blanking intervals of the colour television signal. These data signals can be used to support the television programme or to carry supplementary services such as investigated under Study Programmes 29D/11 and 29E/11.

These data signals may be digitized together with the picture signal or may be separated from it prior to its being coded into digital form.

2.4 *Test signals*

Digital television transmission systems should be able to transmit test signals for monitoring and measuring codec performance and digital as well as mixed analogue-and-digital circuits. These problems are discussed in detail in Report 819.

2.5 *Organization of a digital multiplex*

The signals mentioned in § 2.1 to 2.4 are to be multiplexed to one digital stream. The resulting multiplex structure clearly depends upon the properties of each signal to be transmitted, but it must also take into account the constraints imposed by the transmission path and the television programming facilities.

Doc. [CCIR, 1978-82b] describes a multiplex structure which includes within a digital frame of fixed length picture signals, sound-data signals and signals for monitoring and measurement. The signals are multiplexed in two stages:

- service stage: components of a television signal are multiplexed into one bit stream; high quality sound and commentary signals and the digital data of new services are combined in a 2048 kbit/s stream;
- transmission stage: the two previous streams, together with the signals for monitoring and measurement are multiplexed plesiochronously.

To ensure some flexibility for a multiplexing system, it seems desirable to adopt a frame structure which is independent of the coding methods of picture or sound and to ensure a plesiochronous insertion of the various tributaries at the transmission stage. Results of experiments are reported in [Ranquet, 1978].

Doc. [CCIR, 1978-82c] reports on an experimental system where a DPCM coded composite NTSC signal (Standard M) is multiplexed with a PCM coded sound signal. The sampling frequency of the sound signal is locked to the line frequency of the television signal. Justification technique is used for the adaptation of the multiplexed signal to the transmission system.

Doc. [CCIR, 1978-82d] draws attention to the multiplex structure, in particular of the three video signal components. It is shown that interleaving successive sequences of the luminance and colour-difference samples of each television line has operational and maintenance advantage at the expense of somewhat more complicated hardware compared with interleaving individual luminance and colour-difference samples.

3. Digital bit rates

Choices of coding methods, error protection methods, multiplex structures, and bit rates are closely interdependent. The CMTT considers that a complete television multiplex (including the video signal, test, sound and data signals, etc.) should produce a bit rate at the hierarchical levels studied by the CCITT.

The third order hierarchical levels (i.e., bit rates of 32 064, 44 736 and 34 368 kbit/s) are considered as the most appropriate for long-distance television transmission when for example the transmission cost has to be reduced and when the video signal will not be used for down-stream processing. The EBU states in [CCIR, 1974-78a] that the bit rate of 34 368 kbit/s is a desirable objective for use in the European digital hierarchy.

Study Group 11 is currently studying proposals for component-coding standards for studios (see Report 629 and Recommendation 601) and the CMTT notes that these studies should take the hierarchical levels into account, as there might be a need for transmission between production centres where the high quality and the component-coded nature of the signal should be preserved. In such cases, higher transmission hierarchy levels may be necessary, for example in Europe at the 139 264 kbit/s level [CCIR, 1978-82a, e and f].

For all hierarchical levels, information is needed on interface arrangements and the net bit rates available for the broadcast signal itself. It is also important to know whether any constraints must be applied to the composition of the bit stream of digital television signals fed into digital transmission networks.

Information on these subjects is needed urgently by the CMTT as discussed in Doc. [CCIR, 1978-82g].

Doc. [CCIR, 1978-82h] states that, in order to be compatible with existing protection and alarm systems, in the case of 44 736 kbit/s hierarchy level, the actual available bit rate for digital television signals is 44 407 kbit/s.

4. Digital coding for transmission

4.1 General

Decision 18 states that, for coding methods including redundancy reduction, emphasis should be given in Study Group 11 to the requirements for studios and direct digital broadcasting, while in the CMTT, emphasis should be given to the requirements for long-distance transmission. Further, it should be noted that for the CMTT studies, the greatest interest lies in coding methods which will lead to transmission bit-rates which can be accommodated within the hierarchical levels recommended by the CCITT.

A knowledge of the coding standard used in the studio is of importance for the design of the transmission code since, for example, the transmission signal may be directly transcoded from the studio signal, and the transmission coding could have an effect on any desired "processibility" downstream of the transmission link in an exchange of programme material between production centres. Coding standards for studio applications are discussed in Reports 629 and 962.

Most administrations in both 625-line and 525-line areas are now in accord that component coding offers significant advantages for studios [CCIR, 1978-82e, i, j, k, l and f]. These include the possibility of a single studio standard for PAL and SECAM, a 625-line standard with many common parameters to a 525-line studio standard, and the possibility that an extensible family of compatible digital coding standards, corresponding to different production needs, could be developed, as envisaged in Study Programme 25B/11.

Doc. [CCIR, 1978-82f] refers to experiments with a 625-line component coding system with a data rate of 216 Mbit/s, which, by use of bit rate reduction techniques described in [Wilson, 1981], can be accommodated into the 140 Mbit/s digital transmission hierarchy level. Doc. [CCIR, 1978-82j] gives an example of an extensible family of digital production standards for NTSC with data rates of 336 Mbit/s, 224 Mbit/s, 168 Mbit/s and a lower level. In such a case, different conversion schemes would probably be needed for transmission to convert to an appropriate hierarchy level.

Recommendation 601 gives the basic principles of coding standards which include component coding, a spatially static sampling pattern, some degree of co-siting of samples, and a compatible number of samples per line for all television systems. Such a proposal could be considered for transmission coding after further study.

4.2 Use of composite and component coding systems

Composite and separate-component methods each have advantages for particular applications.

Digital transmission of the signal in the composite form may be advantageous where:

- the input to the circuit is provided in the composite form by the signal sources (e.g. in the studio) and
- the colour system used by the receiving country, in case of an international circuit, is the same as that used by the source country;
- the transmission path consists of mixed analogue and digital sections [CCIR, 1978-82a].

Under these conditions, the composite-signal approach would avoid the need to decode the signal (e.g. to Y , U , V or Y , I , Q) at the input to the digital section and to recode to the composite form again at the output; processes which could impair picture quality. However, reversible decoding and recoding of composite PAL and NTSC signals is possible (see § 4.2.3).

Digital transmission of the signal in separate-component form may be advantageous where:

- the input to the circuit is provided in separate component form by the signal sources (e.g. in the studio);
- the component coding is adopted generally for studios and the inherent advantages of component signals for studios must be preserved over the transmission link, in order to allow downstream processing at a receiving studio;
- the country receiving the signals via an international circuit uses a colour system different from that used in the source country;
- the transmission path is entirely digital, which is expected to be the long-term situation.

These advantages include the elimination of transcoding, a process that can cause loss of picture quality in certain cases. In Docs. [CCIR, 1974-78a and 1978-82m], the EBU recommends that, for digital programme exchange between its members, a signal based upon the separate coding of the luminance and colour signals be used. (The colour signals are not necessarily the same as the components normally associated with a PAL or SECAM signal, but they can easily be derived from them.)

4.2.1 *Composite coding methods*

Primary coding of composite PAL or NTSC signals usually involves:

- sampling the signal at a frequency above the Nyquist limit and which is an integer multiple of the colour sub-carrier frequency (f_{sc}), typically $3 f_{sc}$ or $4 f_{sc}$, or of the line frequency;
- quantizing the signal into PCM form with 8 bits per sample.

However, in Doc. [CCIR, 1978-82a], it is suggested that 9-bit quantizing would be advantageous in high bit rate transmission to provide transparent circuits for all forms of composite signals, including insertion test signals.

Experimental results concerning the cascading of NTSC PCM codecs are given in Docs. [CCIR, 1974-78b and 1978-82n]. The latter document gives results of an experiment where up to 10 PCM codecs were cascaded using a high quality signal source. It is concluded that a sampling frequency of $4 f_{sc}$ rather than $3 f_{sc}$ is desirable for studio equipment and that quantizing with 8 bits per sample is adequate when the number of codecs in tandem is less than 6. For more than 8 cascaded codecs, 9 bits quantizing is considered desirable.

This is an important factor in establishing performance objectives for hypothetical reference circuits and chain as discussed in § 6 of this Report.

Transmission over long distances may require bit-rate reduction methods to be used in order to reduce transmission costs; the choice of technique will depend upon relative performance and equipment cost.

Three basic bit-rate reduction methods may be employed:

- removal of horizontal and/or vertical blanking intervals;
- reduction of sampling frequency;
- reduction of the number of bits per sample.

4.2.1.1 *Removal of blanking intervals*

This technique can provide bit savings of up to about 20% without introducing any impairment to picture quality. However, because blanking intervals waveforms are deterministic signals, the saving may be dependent on the bit rate reduction technique used: for interframe coding, the measurement results reported in Doc. [CCIR, 1978-82o] range between 1.8% and 8%. This document shows that it is possible to remove the colour-burst signals from all but two lines in every frame, with no perceptible impairment to

picture quality; one is the representative value for the odd lines and the other is that for the even lines. Such a removal technique has been successfully applied to two composite predictive coding systems for the NTSC signal [Sawada and Kotera, October 1978a, and CCIR, 1978-82o].

Blanking interval removal can also be effective in reducing the effects of transmission errors on synchronization.

When the vertical interval is removed, separate processing of insertion test signals and additional programme signals (e.g. teletext) is required.

4.2.1.2 *Reduction of sampling frequency*

In the case of PAL and NTSC, it is possible to reduce the sampling frequency under the Nyquist limit: Doc. [CCIR, 1974-78c] describes a technique for sampling the PAL signal at a frequency precisely equal to $2f_{sc}$. Rossi [1978] proposes a similar method for NTSC where again the sampling frequency is $2f_{sc}$, the samples being derived from original $4f_{sc}$ samples. A phase shift of 180° every alternate time sequential television line is necessary in order to permit the removal of alias components. This technique is used in the system described in the reference [Rossi, 1980]. In both NTSC and PAL sub-Nyquist systems, comb-filters are required to remove alias components and to reconstruct a standard signal.

4.2.1.3 *Reduction of the number of bits per sample*

In general, the following techniques may be used singly or in combination:

- predictive coding (differential PCM);
- entropy coding;
- transform coding.

The principles of these techniques are described in Report 629.

Differential PCM has so far emerged as the most popular method. The prediction process required can be classified into two groups: the first one is "intrafield" which is based only on the reduction of spatial redundancy; the second one is "interfield" or "interframe", which is based on the reduction in temporal redundancy as well as in spatial redundancy.

It is claimed in [CCIR, 1978-82p] that the intrafield coding would be usable at a bit rate greater than 32 Mbit/s, while, at lower bit rates, it is very difficult to attain satisfactory picture quality by the intrafield coding. In such a case, the interfield coding and/or interframe coding techniques will be effective alternative means. These techniques are also very effective to attain better picture quality at a bit rate greater than 32 Mbit/s. In Doc. [CCIR, 1978-82q] it is stated that interfield coding can save a bit rate of about 10 Mbit/s, for an equivalent picture quality, in comparison with a conventional intrafield coding technique operating at about 40 Mbit/s. The document also states that an interfield DPCM coder operating at 34 Mbit/s produced no perceptible impairment compared with the original PCM encoded picture.

The reference [Hatori and Yamamoto, 1979] describes intrafield predictive codecs operating directly on composite NTSC signals which achieve bit rates of 32 and 44 Mbit/s. An alternative coding method for NTSC composite signals employing the Hadamard transform technique is discussed in [CCIR, 1974-78d]. The document [CCIR, 1978-82r] shows that, although there is no significant difference in coding performance between the DPCM and the transform coding for most of the natural television images at the same bit rate, the DPCM gives better picture quality for pictures containing many saturated colour components.

The document [Sawada and Kotera, 1978a] presents a 32 Mbit/s composite intrafield DPCM coding system for NTSC, which employs two-dimensional prediction and dual word length coding (entropy coding).

The document [Hatori *et al.*, 1979] presents a codec based on interfield and intrafield adaptive prediction, which makes it possible to transmit a colour television signal, including sound and forward-error-correction codes, at 30 Mbit/s.

The document [Koga *et al.*, 1979] describes a codec utilizing an adaptive intraframe/interframe predictive coding which encodes an NTSC composite colour television signal at a bit rate ranging from 20 up to 30 Mbit/s.

The reference [Rossi, 1980] describes a 28 Mbit/s system for electronic news-gathering applications using intra-frame differential PCM technique.

4.2.2 *Component coding methods*

Primary coding of the components of a colour television signal involves the definition of sampling frequencies for luminance and colour difference signals ($R - Y$, $B - Y$, or I , Q) of sampling patterns and of the number of bits per sample, when the signal is converted into PCM form.

For studio applications these parameters are under study in Study Programme 25F/11 and some of the proposals made up to now are contained in Docs. [CCIR, 1978-82e, i, j and l] as well as Report 962.

These parameters may be used directly for some transmission applications (for example for high quality transmission with a bit rate of 140 Mbit/s).

The transmission of component coded signals over long distances may, as in the case of composite coding, require bit-rate reduction methods and the same three basic methods (removal of blanking, reduction of sampling frequency and reduction of the number of bits per sample) are applicable.

4.2.2.1 *Removal of blanking intervals*

The same arguments quoted in § 4.2.1.1 apply in the case of component transmission.

4.2.2.2 *Reduction of sampling frequencies*

In the case of component coding, the reduction of the sampling frequency to a sub-Nyquist rate is a very flexible and easy to handle method for bit-rate reduction. It may be used for one or more of the three components (luminance and two colour-difference signals). In such a case, it is thought that the same basic principles as for studio parameters (spatially static sampling patterns and co-siting of luminance and colour samples) should be retained. Comb-filtering techniques apply to component as well as to composite signals [Messerschmid, 1969].

4.2.2.3 *Reduction of the number of bits per sample*

A special method applicable to colour-difference signals is non-linear PCM quantizing which may be used to render the visibility of quantizing errors more uniform throughout the colour space [CCIR, 1974-78e]. Other methods mentioned in § 4.2.1.3 apply to components as well as to composite signals. DPCM has been successfully applied separately to the luminance signal and to the two colour-difference signals [CCIR, 1974-78f, g, and h; Sabatier, 1977; Boyer, 1978]. A system is described in [CCIR, 1974-78i] in which DPCM is applied to the luminance signal and to two other signals ($U + V$, $U - V$) that are derived from the colour-difference signals [Heitmann, 1976]. A system is described in [CCIR, 1974-78h] in which the digitized separate-component signals are time-multiplexed to form one signal, that is then converted to the DPCM form. Doc. [CCIR, 1978-82s] shows that, in comparison with the above-mentioned purely intrafield DPCM systems, an adaptive inter/intra-frame prediction algorithm can increase the coding gain by about 1 bit/sample. A design method for DPCM quantizers taking into account visual properties, by considering activity and masking functions, is described in reference [Pirsch, 1981].

A form of hybrid DPCM coding scheme, in which both PCM and DPCM samples are transmitted, is described in reference [Wilson, 1981]. The results of tests using such a system mentioned in Doc. [CCIR, 1978-82t] show that the impairment is imperceptible with the average number of bits per sample reduced to 5¾.

The application of transform coding to the luminance component of the separate components is described in [CCIR, 1974-78j; Motsch, 1977]. Good results were obtained using a Hadamard transform block consisting of four rows each containing four elements. The system employed adaptive coding of the transform signal and it is claimed that this approach could provide an alternative to DPCM when using separate-components coding. A related form of such coding known as adaptive group coding is described in references [Tsukkerman *et al.*, 1981, Sardyko and Tsukkerman, 1977 and Tsukkerman, 1980].

Entropy coding has been applied to the NTSC signal in the separate-components form [CCIR, 1974-78h; Sawada and Kotera, 1978b] and to the transform signal representing the luminance component of a separate component signal [CCIR, 1974-78j; Motsch, 1977].

4.2.3 *Separation of the composite signal*

In some cases, it may be necessary to code the colour television signal using separate luminance and colour-difference components when the source signal is in composite form. A high-quality comb-filter for the purpose of decoding the PAL composite signal is given in Doc. [CCIR, 1970-74a]. However, [Sawada and Kotera, 1978b] reports difficulty in reliably separating the components by means of analogue comb-filter circuit techniques, and describes a two-stage process of analogue and digital comb-filtering.

Alternatively, it is possible to employ processes which, provided certain constraints are observed, permit an incomplete but fully reversible decoding of a PAL or NTSC composite signal [CCIR, 1978-82u]. Provided the constraints are observed and the partially separated “components” are not subjected to distortion, they can be recoded to give the original composite signal with no loss. Such a technique could further the evolution from mixed analogue-and-digital transmission to all-digital separate-components transmission.

Doc. [CCIR, 1970-74b] presents the results of a study on the effectiveness of colour separation as a function of the initial sampling parameters of an NTSC composite signal. It indicates that the sampling rate of $4f_{sc}$ affords the best algorithms, a close second choice is $3f_{sc}$ phase alternating line encoding (PALE); but straight PCM encoding at $3f_{sc}$ is a poor third choice.

However it is claimed in reference [Clarke, 1981] that all performance limitations with composite PAL analogue to digital *YUV* convertors are unrelated to the choice of sampling-frequency (provided that this is at the Nyquist criterion or above), whether or not the latter is simply related to the sub-carrier frequency.

4.3 *“Digital-to-digital” conversion methods*

Terrestrial and satellite transmission circuits for television may consist of a number of digital sections in tandem. Adjacent digital sections may have dissimilar requirements (for example different bit rates according to different hierarchies of two different world regions) which will necessitate means to convert from one set of digital coding parameters to another, as for example, when changing the digital sampling rate of the television picture signals from one rate to another. The same problem may appear at the interface between a digital studio and a digital transmission link.

These conversions can employ a digital-to-analogue converter followed by an analogue-to-digital converter but it is generally recognized that a fully digital method would be preferable in terms of quality and reliability.

Several digital-to-digital converters for System M/NTSC are outlined in [CCIR, 1974-78k], which draws attention to the fact that, using only a modest amount of digital circuitry, conversion of the sampling rate of $3f_{sc}$ to $4f_{sc}$ is possible. It is further claimed that a completely reversible $3f_{sc} - 4f_{sc} - 3f_{sc}$ conversion is possible without incurring any distortion beyond that caused by the quantization accuracy of the filter co-efficients.

5. **Performance characteristics**

5.1 *General*

It is necessary to distinguish between impairments which can arise in analogue sections, and those attributable to digital sections. The admissible tolerances for distortions in a digital section should be so fixed that the subjective impairment of picture quality at the end of a hypothetical reference circuit is not worse than in the case of a hypothetical reference circuit which consists only of analogue sections.

5.2 *Distortions common to both digital and analogue circuits*

These impairments are dealt with in Recommendation 567.

5.3 *Distortions peculiar to digital transmission*

Impairments which can arise in digital transmission fall into two categories:

5.3.1 Distortion caused by coding methods and coding equipment. This can be identified by operating coding and decoding equipments back-to-back. Depending on the coding method, impairment can arise from:

- the sampling process and filtering, (e.g. comb filtering),
- the initial amplitude quantization,
- bit-rate reduction processes.

A matter of considerable importance in making comparisons between different methods of coding is to derive techniques for relating the subjective performance of a codec to its objective parameters. Some preliminary studies of this problem are reported in [Kretz *et al.*, 1978]. The reference [CCIR 1978-82v] indicates a method in which the coding distortions are first classified into several categories of disturbance based upon the subjective deterioration of the picture and then objective measures are derived for each classified category. One method in which the performance of the coding method is defined by the time percentage satisfying the required picture quality is described in [Koga *et al.*, 1979].

Document [CCIR, 1978-82w] suggests that a useful technique, for measurement purposes, is to evaluate the difference between the input and output signal of a digital codec. It is suggested that subtraction in digital form is better than in analogue format.

5.3.2 Distortion introduced in the actual digital transmission. Ideally, a digital transmission circuit should convey the applied digital signal so that the received signal is indistinguishable from the original. However, in practice impairments can occur which fall into the following categories:

- that due to timing perturbations in the digital signal – usually known as jitter,
- that due to bit errors,
- that due to bit slip.

These three forms of impairment can occur to varying degrees according to the type of transmission circuit, e.g., cable, optical fibre, terrestrial radio and satellite. They are discussed more fully in Report 967 together with methods of protection.

6. Hypothetical reference circuits and chains

6.1 Introduction

In order to develop performance requirements for digital and mixed analogue and digital long distance transmission of television signals it is essential that definitions of hypothetical reference circuits and hypothetical reference chains be formulated. This formulation should provide a useful tool for the development of performance specifications for the above types of circuits and for the design of equipment to be used in real networks. Any definition of hypothetical reference circuits and chains for the digital and mixed analogue-and-digital cases should be compatible with existing definitions of analogue hypothetical reference circuits.

Since the mixed analogue and digital and all-digital transmission circuits must be compatible with the all-analogue case, reference should be made to Recommendation 567 for this case for both the terrestrial and fixed-satellite service circuits.

During the 1974-1978 Study Period, two proposals were made relating to the definition of hypothetical reference circuits [CCIR, 1974-78l and m]. The essential features of these proposals are summarized in the following paragraphs.

6.2 The compatibility requirement

As noted above, there is general agreement that compatibility with existing definitions for the analogue hypothetical reference circuit is a requirement in defining the hypothetical reference circuit for the mixed analogue and digital circuit.

Acceptance of the need for compatibility restricts the options available for the mixed analogue and digital case. Firstly, the terrestrial hypothetical reference circuit must be made up of three parts. Secondly, the analogue sections must be the same as already defined in Recommendation 567, making the length of each analogue section (2500/3) km. However, options remain giving either full or partial compatibility and these are discussed below.

This form of the hypothetical reference circuit is consistent with ideas expressed by CCITT Study Group XVIII (formerly Special D). While this Study Group has not yet made recommendations for hypothetical reference circuits for high capacity digital systems, they have made a Recommendation G.721 for a 64 kbit/s digital path through such systems. This Recommendation also defines a 2500 km length circuit divided into three sections for the all-digital telephony network.

6.2.1 Partial compatibility

Some administrations consider that, with digital sections, it is both possible and desirable to increase the length of the hypothetical reference circuit and, in [CCIR, 1974-78l], alternative figures of 2500 km and 1500 km for each section were proposed. If the need for a longer circuit is accepted while the

need to maintain compatibility with existing analogue plant is recognized, the increase in length must be obtained by using longer digital sections in the mixed analogue and digital case. This means that the length (x) of the hypothetical reference circuit is given by:

$$x = m (2500/3) + (3 - m) d \quad (1)$$

where

m : number of analogue sections and has values 0, 1, 2 or 3, and

d : length of each digital section.

In considering proposals of this kind it should be noted that, in practice, the length cannot be chosen independently of the performance standards required. In the future, digital television is likely to be carried on high-capacity digital bearers which are shared with other services and the performance standards of such bearers will be determined by the requirements of the major user which will be telephony.

6.3 *Hypothetical reference circuit for a system in the fixed-satellite service*

The path earth station-satellite-earth station is normally all-analogue or all-digital so that problems about mixed analogue and digital operation do not arise.

Therefore, the principle developed in Recommendation 567 can be followed here by having the same design objectives and tolerances for a hypothetical reference circuit whether it is terrestrial or a system in the fixed-satellite service.

6.4 *Number of analogue-to-digital conversions*

Noting that the number of A/D and D/A conversions should be kept to a minimum to minimize the accumulation of distortion and the possibility of using digital-to-digital converters for interfacing between different digital standards (§ 4.2 of this Report), the maximum number of A/D and D/A conversions required in the mixed analogue and digital hypothetical reference circuit is two (where an analogue section separates two digital sections). In other configurations only one conversion is required — except for all-analogue transmission.

It is therefore proposed that the proposal previously made in [CCIR, 1974-78] be adopted, and the hypothetical reference circuit defined, as containing a maximum of two A/D and D/A conversion processes.

6.5 *Hypothetical reference chains*

Recommendation 603 discusses hypothetical reference chains for analogue circuits. This Recommendation contains proposals for a specific configuration of a hypothetical reference chain comprising five hypothetical reference circuits in cascade. The proposals in this Report can serve as the basis for further study with respect to the mixed analogue and digital and all-digital cases. The study may concentrate on the additional considerations, such as the maximum number of A/D and D/A conversions, which have to be made when dealing with mixed analogue and digital chains, but which are not made when considering only the all-analogue chain.

Experimental results concerning the cascading of NTSC PCM codecs are given in [CCIR, 1974-78b and 1978-82n]. A summary of the results reported in the latter document is given in § 4.2.1 of this Report.

6.6 *Summary*

The ideas outlined above are considered by some administrations to provide a basis for an early Recommendation of limited scope defining some of the more basic features of hypothetical reference circuits and chains involving progression from analogue transmission, through mixed analogue and digital transmission to all-digital transmission. There are many areas requiring further study, some of which are listed in the next sub-section, and it may be a long time before adequate answers are available to these outstanding questions. For this reason it is believed that an early Recommendation of limited scope, which may require subsequent revision as experience is gained in digital techniques, would be of considerable help in establishing international transmission of television by digital methods.

6.7 *Some areas of further study*

The following items are some areas requiring further study in order to formulate hypothetical reference circuits and chains for digital and mixed analogue and digital transmission.

6.7.1 The length of the hypothetical reference circuit being x kilometres should be studied to determine the proper values of x . For the existing analogue hypothetical reference circuit x equals 2500 km.

6.7.2 The length of a hypothetical reference chain, which is n times the length of a hypothetical reference circuit should be studied to determine the proper value of n .

6.7.3 Should all terminal and intermediate points be digital or should they be all analogue or should there be defined mixtures of these points?

6.7.4 If sections (parts of a chain) contain analogue-to-digital and/or digital-to-analogue converters at which end of the section (parts of a chain) should they be located?

6.7.5 How many analogue-to-digital conversions should be permitted in the hypothetical reference circuit?

6.7.6 How many analogue-to-digital conversions should be permitted in a reference chain made up of n hypothetical reference circuits?

Note. — It does not necessarily follow that the number of conversions in the chain is n times the number of conversions considered in § 6.7.5.

6.7.7 Should more than one hypothetical reference circuit (chain) be formulated?

6.7.8 As television and sound signals are likely to go over common bearer circuits, the question arises as to whether the sound-programme hypothetical reference circuit for digital and mixed analogue and digital transmission should be the same as the one formulated for the television signal.

6.7.9 What signal format or formats should be defined for use at terminal and intermediate points in the hypothetical reference circuit or chain? This question refers both to analogue and digital signals.

6.7.10 Should digital-to-digital converters be incorporated in the hypothetical reference circuit and chain?

6.7.11 Should the n parts of a reference chain be individually lined up and then interconnected, without requiring any form of overall adjustment or correction?

6.7.12 Should the reference chain contain special equipment for standard television conversion or for synchronizing pulse regeneration or for the insertion of special signals in the blanking intervals?

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ANNEX I

PROPOSAL FOR A NEW RECOMMENDATION WHICH THE CMTT WAS NOT ABLE TO ADOPT UNANIMOUSLY

The CCIR,

CONSIDERING

- (a) that although television studios may eventually adopt entirely digital operation based on separate component coding (according to Recommendation 601) existing analogue operation using composite signals will continue for a considerable time;
- (b) that a transitional period may occur during which television signals will be transmitted through a circuit comprising analogue and digital sections in tandem;
- (c) that to ease the international transmission of television programmes during the transitional period the method of coding should be transparent to the colour system standard (e.g. NTSC, SECAM, PAL or an analogue form of colour components) and to ancillary and additional signals (e.g. insertion test signals, teletext etc.);
- (d) that during the transitional period television signals should be transmitted with minimum impairment through a mixed analogue and digital path;
- (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,

RECOMMENDS

1. that, when carrying a television signal presented in analogue form, preference should be given to all-analogue paths. However, in cases where mixed analogue-and-digital paths are not avoidable, the principles given in this Recommendation should be followed;

2. that, for mixed analogue-and-digital paths, the television signal should be coded as uniformly quantized PCM with a super-Nyquist sampling rate and with not less than 8 bits per sample;
3. that, when transmitting a television programme comprising video, sound, ancillary and additional signals over digital sections, it should be preserved in its original form.

ANNEX II

MAJOR POINTS OF OBJECTIONS WHICH PREVENTED AGREEMENT TO THE PROPOSALS IN ANNEX I

- (a) Composite television signals in digital form should not be used for the international exchange of television programmes.
- (b) A recommendation for mixed analogue and digital transmission could impede progress towards the international exchange of television programmes over all-digital paths.
- (c) The proposal in Annex I would lead to inefficient utilization of transmission capacity.
- (d) The use of bit rate reduction techniques for composite signals should be more clearly excluded for the present.

REPORT 817-1 *

INFORMATION RECEIVED FROM CCITT STUDY GROUP XVIII REGARDING DIGITAL TRANSMISSION SYSTEMS

(Questions 14/CMTT and 18/CMTT)

(1978-1982)

1. Hypothetical reference connections

It is noted that the CCITT have defined hypothetical reference connections for the digital network in Recommendation G.104, and that these are based principally on telephony applications. It is also noted that at an early stage, the need is foreseen for hypothetical reference connections to be established for the integrated services digital network. These will be useful when studying impairments arising from:

- digital errors,
- slips,
- jitter,
- delay,
- wander,
- short interruptions.

2. Hypothetical reference digital paths

It is noted that the CCITT have defined hypothetical reference digital paths for the following (in Recommendation G.721):

- 2 Mbit/s system,
- 8 Mbit/s system,
- 34 Mbit/s system,
- 140 Mbit/s system,
- digital sections normally operating at and below 6312 kbit/s and associated with 1544 kbit/s primary multiplex equipment,
- digital sections normally operating at and above 6312 kbit/s and associated with 1544 kbit/s primary multiplex equipment,
- very long paths in digital sections normally operating above 6312 kbit/s and associated with 1544 kbit/s primary multiplex equipment.

Other hypothetical reference digital paths are under study.

* This Report should be brought to the attention of Study Group XVIII of the CCITT.

3. Digit rate hierarchies

It is noted (from Recommendations G.751 and G.752 *et al.*) that the CCITT has recommended two bit rate hierarchies:

- 1544 – 6312 – 32 064 or 44 736 kbit/s
- 2048 – 8448 – 34 368 – 139 264 kbit/s

It is understood that non-hierarchical bit rates might possibly be included if necessary, for example, for digitally coded television signals. However the use of hierarchical bit rates is to be preferred.

4. Transmission performance of digital networks

4.1 Error performance

The CCITT has in its Recommendation G.821 proposed performance objectives for 64 kbit/s connections in the ISDN used among others for voice traffic or as a "Bearer Channel" for data type services. The 64 kbit/s connection referred to is a subscriber-to-subscriber all-digital Hypothetical Reference Connection (HRX) of 25 000 km in length.

The performance objective is specified for periods when the connection is available and is stated in terms of an error performance parameter, which is defined as the percentage of averaging periods T_0 during which the bit error ratio (BER) exceeds a threshold value. The percentage is assessed over a much longer time interval T_L . For voice services the specified value for T_0 is one minute and the percentage of available minutes for which the BER is worse than 1×10^{-6} is less than 10%. For data services T_0 is one second and the percentage of available seconds for which the BER is greater than 0 is less than 8%, i.e. the percentage of "error free seconds" (EFS) is 92%.

CCITT Study Group XVIII proposes to amend Recommendation G.821 to include the definition of non-availability as follows:

A connection is considered to exhibit a complete failure (or is not available) when the BER in each second is worse than 1×10^{-3} for periods equal to or exceeding 10 consecutive seconds [CCIR, 1978-82a, b].

For the time being it is postulated that the permissible end-to-end impairment will be allocated on the basis of approximately 50-60% to the local exchange-local exchange part of the connection and 20-25% to each of the local exchange-subscriber parts. However, it should be noted that this allocation is only intended as a guideline and for some national networks other allocations might be appropriate. The latter is likely to be acceptable so long as that part ultimately allocated to the national portion is not exceeded.

The CCITT recognizes [CCIR, 1978-82c] that Recommendation G.821 deals with performance objectives for a 64 kbit/s HRX and therefore does not directly apply to broadcast quality TV or other higher bit rate services. In respect of the transmission of these signals it is assumed that television signals will not encounter the lower order parts of the network (below 34 Mbit/s) where a substantial portion of the total impairment of the HRX will be encountered [CCIR, 1978-82b]. It is generally expected that existing and planned designs of higher bit rate systems, with the possible exception of satellite systems, will be adequate for TV. Nevertheless, it seems appropriate to define an HRX for TV and this done, to establish error performance objectives using the error performance parameter defined above. The severe subjective effects of loss of TV synchronization suggest that a much shorter averaging interval (T_0), and a much lower percentage allowance of intervals above threshold than those cited above may be appropriate. The final choice of values is likely to depend on the TV coding technique ultimately recommended.

As stated in Annex 5 of CCITT Question 9/XVIII [CCIR, 1978-82d], studies are continuing to establish appropriate error ratios at the higher hierarchical bit rates. Depending on the outcome of the Study Group XVIII studies it is expected that the design objectives will be in the order of those shown in the following Table:

TABLE I

Digital line system			Hypothetical reference digital path		Expected error ratio design objective (except allowance for multiplexers)
Recommendation	Capacity (Mbit/s)	Medium	Figure	Length (km)	
G.916	34 368	Coaxial pair	Fig. 1	2 500	2.5×10^{-8}
G.918	139 264	Coaxial pair	Fig. 2	2 500	2.5×10^{-8}

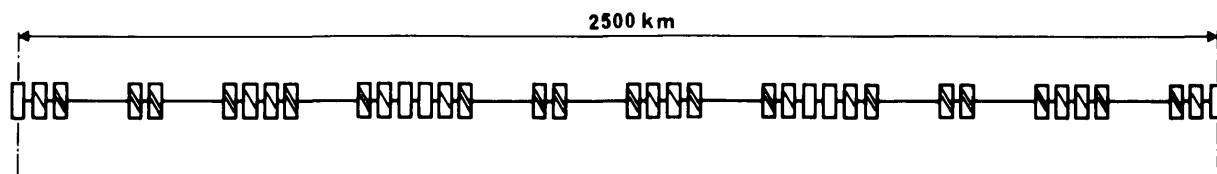


FIGURE 1 – *Hypothetical reference digital path for 4-MHz systems which could also be used for 34-Mbit/s systems*
(from Recommendation G.721, Fig. 3)

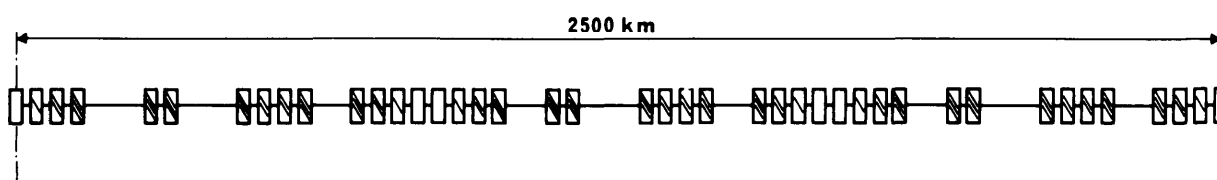


FIGURE 2 – *Hypothetical reference digital path for 140 Mbit/s systems*
(from Recommendation G.721, Fig. 4)

4.2 Slip rate objectives

In its Recommendation G.822 the CCITT has proposed an end-to-end slip rate performance for telephone and non-telephone services based on a 64 kbit/s digital connection. The slip rate objectives are stated with reference to the Hypothetical Reference Connection (HRX) of 25 000 km in length.

It is assumed that International Exchanges (CT's) are interconnected by international links which are operating plesiochronously, using clocks with accuracies as specified in Recommendation G.811. It is recognized that one slip in 70 days per plesiochronous interexchange link is the resulting maximum theoretical slip rate, taking into account clock accuracies according to Recommendation G.811 only, and providing the transmission and switching requirements remain within their design limits.

One proposal to CCITT Study Group XVIII when this subject was under study was that when satellites are used, there could be a possibility of slipping in multiframe blocks rather than in frames. The magnitude (2 ms) of the necessary buffer stores would more than cater for the satellite Doppler shift and consequently would reduce the number of slips to less than those predicted, therefore virtually eliminating slip due to plesiochronous operation [CCIR, 1978-82e].

In a practical international end-to-end connection the slip rate may for various reasons exceed the value computed from "n" plesiochronous interexchange links.

Slip occurs due to digital processing of non-synchronized bit streams in synchronous devices such as digital switches and synchronous multiplexers.

At this stage, digital switching at speeds above the basic rate is undefined. Multiplexing using justification techniques is asynchronous and should not introduce slip [CCIR, 1978-82e].

4.3 Jitter requirements

The reply to Question 1/XVIII.C, CCITT, 1977-80 [CCIR, 1978-82e] states that the CCITT made substantial progress on the definition of jitter requirements. The outcome of the studies is incorporated in additions to the relevant equipment Recommendations and revisions of Recommendation G.703. Nevertheless, further studies are still required of the overall network considerations relating to jitter.

It now seems reasonable to presume that provided the appropriate network considerations for jitter are met then there is little probability that voice services will be significantly impaired by jitter.

CCITT Study Group XVIII also states that more sensitive services such as sound programme and visual services, particularly broadcast television, may need jitter reduction over and above that attained by the transmission network. This subject requires further study in collaboration with CMTT.

5. Maintenance of digital networks

It is noted that the CCITT defines in Recommendation G.704 a maintenance philosophy for integrated digital networks. Maintenance principles, alarm indications and consequent actions, upstream failure indications and automatic-changeover of equipment are discussed in this Recommendation.

6. Customer access to the ISDN

6.1 ISDN definition

CCITT proposes a new definition for the ISDN as follows:

“A network evolved from the telephony IDN that provides end-to-end digital connectivity to support a wide range of services, including voice and non-voice services, to which users have access by a limited set of standard multipurpose customer interfaces” [CCIR, 1978-82a].

The evolving ISDN may include at later stages switched connections at bit rates higher and lower than 64 kbit/s [CCIR, 1978-82f].

6.2 ISDN information types

A number of information types are suggested by the CCITT for the purpose of characterizing different service feature requirements for the ISDN. Type “*u*” information is of interest to the CMTT and is defined as follows:

Type u : digital information above the rate defined by type *w* (see Note)

u_1 — digital information at a rate up to the primary hierarchical level (1544/2048 kbit/s)

u_2 — digital information at a rate greater than u_1 up to the secondary hierarchical level (6312/8448 kbit/s)

u_3 — digital information at a rate greater than u_2 (up to 140 Mbit/s)

Note. — *w* information is composed of $n \times 64$ kbit/s with n in the order of 2 to 3.

6.3 Available bit rate of the access to the ISDN

The CCITT advises the users of wideband services that in the digital network for leased services the full bit rate could be made available at the hierarchical levels (see Note); however certain restrictions may exist at the 1544 kbit/s rate [CCIR, 1978-82f].

For switched services it cannot be decided at this point in time whether the full bit rate of a particular hierarchical level can be made available for wideband services such as sound programme transmission or TV transmission. This will be influenced among other aspects by a possible requirement for setting up switched connections for these services [CCIR, 1978-82f].

Note. — Practical systems at the 44 736 kbit/s hierarchical level will impose restrictions on the available bit rate [CCIR, 1978-82g].

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REPORT 967 *

**DIGITAL TELEVISION:
TRANSMISSION IMPAIRMENTS AND METHODS OF PROTECTION**

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

(1982)

1. Introduction

This Report is concerned with the impairments which can arise from imperfections in transmission systems carrying digitally-coded television signals.

Ideally, a digital transmission system should convey the applied digital signal so that, apart from a constant time delay, the received signal is indistinguishable from the original. However, in practice, impairments can occur which fall into the following categories:

- those due to timing perturbations in the digital signal – usually known as jitter;
- those due to bit errors;
- those due to bit slips.

These three forms of impairment can occur to varying degrees according to the type of transmission circuit, e.g. metallic cable, optical fibre cable, terrestrial radio and satellite.

This Report discusses the impairments which can arise from transmission defects and methods of protection against them. Desirable transmission performance objectives and ways of specifying them are also considered.

2. Jitter

Jitter can be introduced by the digital transmission system especially when highly-structured, picture-dependent bit sequences are present as in some forms of video coding, e.g. PCM. Unless scrambling techniques are used to break up such sequences, jitter can be such that synchronizing and demultiplexing processes in the receiving equipment may be seriously disturbed with consequent severe impairment. Provided the demultiplexing can be achieved, however, the phase-locked loops usually involved in the demultiplexing process can give a large degree of jitter reduction.

Residual jitter may still be present on the analogue signal after the final process of digital-to-analogue conversion and this must be kept to a suitably low level to avoid perceptible impairment to the displayed picture. The acceptable level of jitter will depend on the type of video signal, i.e. composite PAL, SECAM or NTSC, or the separate luminance and colour-difference components comprising a colour video signal.

The permissible amount of jitter for a PAL colour signal has been studied [CCIR, 1970-74a] taking account of the phase offset likely to occur in a PAL domestic receiver and a limit of 0.5 ns r.m.s. has been suggested for random timing jitter. The results of further studies of the permissible jitter with other types of video signal are of interest. The more stringent requirements of broadcast television signals in relation to telephony signals is recognized in Doc. [CCIR, 1978-82a].

Techniques for the measurement of line synchronizing pulse edge jitter are considered in Report 816.

3. Transmission errors**3.1 Circuit performance**

The error-rate performance objectives of digital transmission circuits having a bit-rate capacity suitable for broadcast quality television signals are being studied but have not yet been specified by the CCITT.

* This Report which is associated with Report 646, should be brought to the attention of CCITT Study Group XVIII.

The CCITT have so far proposed performance specifications for 64 kbit/s voice traffic connections or data-type bearer channels and figures quoted in CCITT Recommendation G.821 include, for example, a mean bit-error-ratio of worse than 1×10^{-6} which is for less than 10% of available minutes for telephony. The document [CCIR, 1978-82b] suggests that for 34 Mbit/s channels it can be assumed that bit-error-ratios do not exceed 1×10^{-4} for a significant time. This is consistent with Doc. [CCIR, 1978-82c] which discusses the general difficulty of specifying error performance objectives and suggests that a long-term mean bit-error-ratio of 1×10^{-6} is an appropriate objective for a high bit rate, maximal length, hypothetical reference connection.

Care is required however in defining bit-error objectives. The parameter used in CCITT Recommendation G.821 concerning 64 kbit/s connections is the percentage of averaging periods T_0 during which the bit-error-ratio (BER) exceeds a threshold value. The percentage is assessed over a much longer time interval T_L . For telephony the specified value for T_0 is one minute and the percentage of available minutes for which the BER is worse than 1×10^{-6} is less than 10%. For data services T_0 is 1 s and the percentage of available seconds for which the BER is greater than 0 is less than 8%, i.e. the percentage of "error-free seconds" (EFS) is 92%.

The CCITT recognizes in Doc. [CCIR, 1978-82a] that the error-performance requirements for broadcast quality television signals are much more stringent than those for telephony and data services and that a much shorter averaging interval for T_0 and a much lower percentage allowance of intervals above a given threshold are appropriate.

Further study is required by the CMTT concerning suitable values for the parameters adopted by the CCITT. In order to avoid the serious subjective effects of loss of television synchronization it seems likely that the averaging interval (T_0) must be much shorter and of the order of tens of microseconds [CCIR, 1978-82d].

The parameter "mean BER" is not in itself sufficient to specify the error-performance requirement for broadcast quality digital television transmission. Information on error distribution patterns, (e.g. burst behaviour), is also needed if appropriate error correction systems are to be designed. It is possible that bursts of errors for example, could be within a mean BER tolerance, yet not be capable of correction by an error protection system which nevertheless has an adequate performance for essentially random errors.

In respect of error patterns, both the error spread effect of a multilevel transmission code (or other modulation or code-conversion process) and the presence of a self-synchronizing scrambler/descrambler must be taken into account.

The attention of CCITT Study Group XVIII should be drawn to the difficulties for broadcast television signals which the error-spread from these two sources creates in terms of a requirement for multiple error correcting codes and/or the need to introduce a number of identical error correctors to cope with error bursts. Study Group XVIII should also be asked to consider the possibility of ways of reducing error-spread by minimizing the length of scrambler necessary to achieve adequate dispersion of the spectral components of a coded colour television signal, or alternatively to consider the use of RESET scramblers and avoid inherent error-spread of the self-synchronizing type.

The CMTT should examine the advantages, such as the possibility of using RESET scramblers, which may derive from compatibility between frame structures for digital television transmission and those specified for the hierarchical levels recommended by the CCITT.

3.2 *Susceptibility of digital television signals to bit errors*

The error performance required of the digital transmission path is somewhat dependent upon the precise form of source coding and type of broadcast signal, but, in general, it is much higher than that for telephony signals. For vision signals, a generally acceptable criterion would probably be that the mean bit-error-ratio (BER) should not exceed 1×10^{-9} assuming that error events are essentially random in nature. Such an error rate would be acceptable for the most susceptible coding method [CCIR, 1978-82d].

The provision of relatively simple error-correcting codecs could, however, reduce the 1×10^{-9} error criterion for vision signals to about 1×10^{-6} with an excess transmission bit rate not exceeding 10%. It is also possible that the criterion may be further relaxed to about 1×10^{-4} by implementing more complex error-correction procedures.

Permissible bit-error-ratios for certain coding methods are given in [CCIR, 1970-74b and c] and lie within the range 1×10^{-4} to 1×10^{-9} . Docs. [CCIR, 1974-78a; Guionnet and Sallio, 1978] consider certain component-coded signals, and [BBC, 1974a] certain composite-coded signals (PCM, DPCM and hybrid DPCM/PCM). [CCIR, 1978-82b] describes comparative investigations on different DPCM systems, designed to bit-rate reduce

625-line PAL to 34 Mbit/s, which show admissible channel error-ratios ranging from 1×10^{-6} to 1×10^{-8} , depending on the system. [CCIR, 1978-82e] reports that after component coding for bit rate reduction to 34 Mbit/s, the luminance signal is more sensitive to transmission errors than the colour difference signals (grade 4.5 on the impairment scale reached for bit error ratios of 3×10^{-7} and 1×10^{-5} respectively). On the other hand, a simple correction system operating only on the luminance signal brings, at the same subjective quality level, a gain of about 10 to 30 on the error ratio with a system redundancy of 1.9%. [CCIR, 1978-82f] describes an interfield and intrafield adaptive prediction method, for 30 Mbit/s, which has an inherent permissible error-ratio tolerance of about 1×10^{-8} .

3.3 Error protection systems

Error protection systems may be divided broadly into two categories:

- *Concealment*: a technique whereby an erroneous sample word is detected by means of an additional parity bit (or bits). The erroneous sample is then replaced by an approximation to its value derived from adjacent correctly-received samples.
- *Correction*: a technique whereby erroneous bits are again detected by means of additional bits. In this case however, the additional bits are configured so that the erroneous bits can be located and actually corrected.

Each technique can be used separately or combined together. Error-correction usually requires more complex hardware than error concealment. Simple correction techniques such as Hamming codes with low redundancy can provide adequate margin of improvement for coding systems which are not too sensitive to error; but for other systems multiple error correction techniques and/or the use of more redundancy within the transmitted bit rate may be necessary.

The Documents [CCIR, 1974-78b] and [BBC, 1974b] describe detailed studies on certain error-correction systems. [CCIR, 1974-78b] discusses probable error rates and the use of low-redundancy Hamming codes. The Document [BBC, 1974b] gives a simplified description of error-correcting codes and in particular discusses the relative merits of Hamming and Wyner-Ash codes; it concludes that these two codes are comparable in performance. A practical Wyner-Ash corrector is also described together with objective measurements of its performance.

The Document [CCIR, 1974-78c] points out that the use of error correcting codes additionally allows continuous monitoring of the performance of the digital circuit in use, in the presence of the television programme.

An error detection and correction scheme, handling burst and random errors, is described in [CCIR, 1978-82b and Stammnitz, 1980]. The scheme has been implemented for a transmission rate of 34 Mbit/s with 3% redundancy. The residual error ratio attainable with a random error ratio of 1×10^{-4} is 1×10^{-8} . The burst correcting capability is 26 bits.

Documents [CCIR, 1978-82f] describes the performance of a BCH (255:239) double-error-correcting code when applied to digital television transmission through a satellite. The tests showed that a random bit-error-ratio of 1×10^{-5} can be reduced to 1×10^{-11} . In this system a back-up concealment technique was available in the inter-frame DPCM codec which provided acceptable picture quality at a channel bit-error-ratio as high as 1×10^{-4} .

Normally error detection and correction require the addition of further digital signals. Goldberg and Rossi [1981] describe a technique requiring no overhead bits to detect and correct errors in PCM television signals having super-Nyquist sampling rates. The technique uses a bandpass filter at one-half the sampling rate to detect energy when errors occur. Error correction in the four most significant bits of the PCM samples has been shown to make a large improvement in the subjective quality of the picture.

4. Transmission bit slip

A bit slip is a discontinuity in the bit stream arriving at the receiving terminal of a digital transmission system. Two kinds of slip are defined as follows:

4.1 Uncontrolled slip

The uncontrolled slip is the loss or gain of a digit position or a set of consecutive digit positions resulting from an aberration of the timing processes associated with transmission or switching of a digital signal.

In digital television transmission a bit slip would probably result in re-framing and re-locking of phase-locked loops at the demultiplexers. Since such processes are likely to take up to about 1 s they would be deemed an interruption. Bit slips would therefore be acceptable only if they were suitably infrequent, say not exceeding once per month.

4.2 Controlled slip

The controlled slip is a deliberate and irretrievable loss or gain of a set of consecutive digit positions in a digital signal to enable the signal to accord with a rate different from its own.

The effects of controlled slips on digital television transmissions are covered in Report 817.

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GOLDBERG, A. A. and ROSSI, J. P. [1981] Digital television error correction without overhead bits. *Television Technology in the 80's*, SMPTE, Scarsdale, NY 10583. 15th Annual SMPTE Television Conference, San Francisco, CA, USA, 80-88.

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STAMMNITZ, P. [1980] Error protection of 34 Mbit/s DPCM encoded TV signals with multiple error correcting BCH codes. Proc. International Zurich Seminar, IEEE Catalogue No. 80, CH 1521-4 COM.

CCIR Documents

[1970-74]: **a.** CMTT/146 (United Kingdom); **b.** CMTT/192 (Japan); **c.** CMTT/223 (CCITT Study Group Special D).

[1974-78]: **a.** CMTT/106 (CCIR, Report 629-1); **b.** CMTT/62 (France); **c.** CMTT/234 (France).

[1978-82]: **a.** CMTT/109 (Study Group XVIII, CCITT); **b.** CMTT/16 (Germany (Federal Republic of)); **c.** CMTT/107 (Study Group XVIII, CCITT); **d.** CMTT/12 (EBU); **e.** CMTT/268 (France); **f.** CMTT/47 (Japan).

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

Recommendations and Reports

RECOMMENDATION 473-3 *

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)

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;
- 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;

* CCITT Recommendation N.67 has been amended to bring it into agreement with this Recommendation.

- in the case of PAL transmissions, the chrominance sub-carrier of the insertion signals is locked at $60 \pm 5^\circ$ 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 $4.433\,618\,75\text{ MHz} \pm 10\text{ Hz}$.

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: $0.700 \pm 0.007\text{ V}$;
- rise and fall times of transitions: derived from the shaping network of the sine-squared pulse (element B_1);
- overshoot and undershoot $\leq 0.5\%$;
- tilt $\leq 0.5\%$.

2.2 $2T$ sine-squared pulse (B_1)

- peak position: $13H/32$;
- amplitude: within $\pm 1\%$ of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- half-amplitude duration: $200 \pm 10\text{ ns}$ (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 $\pm 1\%$ of the amplitude of the luminance bar (B_2) (nominal value: 0.700 V);
- half-amplitude duration: $2 \pm 0.06\text{ }\mu\text{s}$;
- perturbations of the pulse base-line, due to inherent chrominance-luminance amplitude and delay inequalities and different shape of the luminance and chrominance components: $\leq 0.5\%$ 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 $\pm 1\%$ 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.

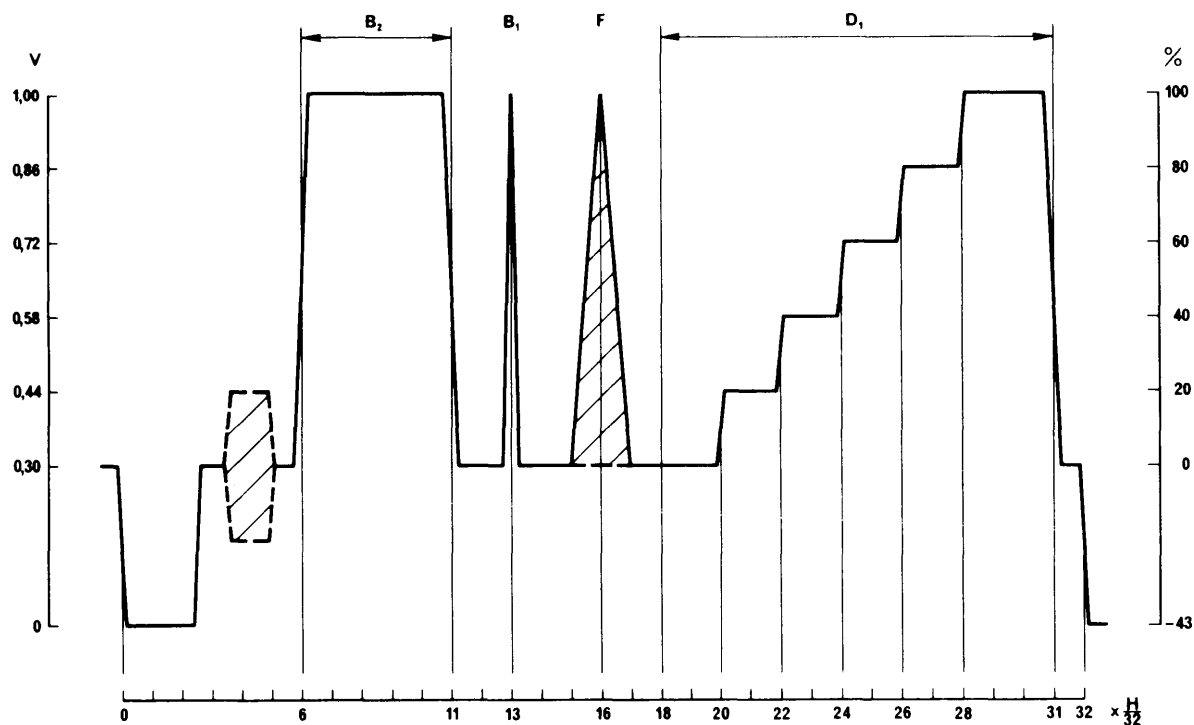


FIGURE 1 - Line 17

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 $\pm 1\%$ 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 $\pm 1\%$ of four-fifths of the amplitude of the luminance bar (B_2) (nominal value: 0.560 V);
2nd section: within $\pm 1\%$ 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:

TABLE I

Burst No.	Precise starting position(1)(2)	Frequency MHz(3)
1	$12H/32$	0.5
2	$15H/32$	1.0(4)
3	$18H/32$	2.0(4)
4	$21H/32$	4.0
5	$24H/32$	4.8
6	$27H/32$	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 $0.4 \mu s$ nor longer than $2.0 \mu s$ 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 $\pm 1\%$ 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.

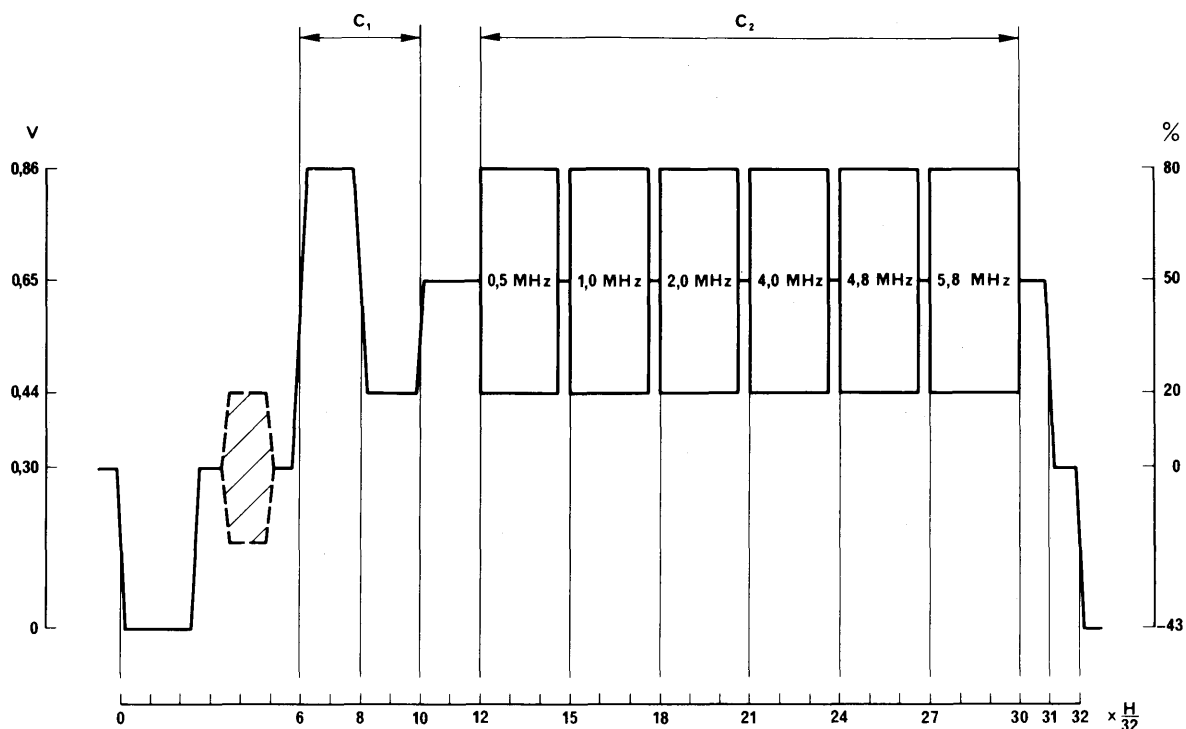


FIGURE 2 - Line 18

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 $\leq 0.5\%$;
- tilt $\leq 0.5\%$.

4.2 $2T$ sine-squared pulse (B_1)

- peak position: $13H/32$;
- amplitude: within $\pm 1\%$ 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 $\pm 1\%$ 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 $\pm 2\%$;

- inherent differential-gain distortion: $\leq 0.5\%$;
- inherent differential-phase distortion: $\leq 0.2^\circ$;
- rise and fall times of the envelope of the chrominance transition: $1\ \mu\text{s}$ approximately.

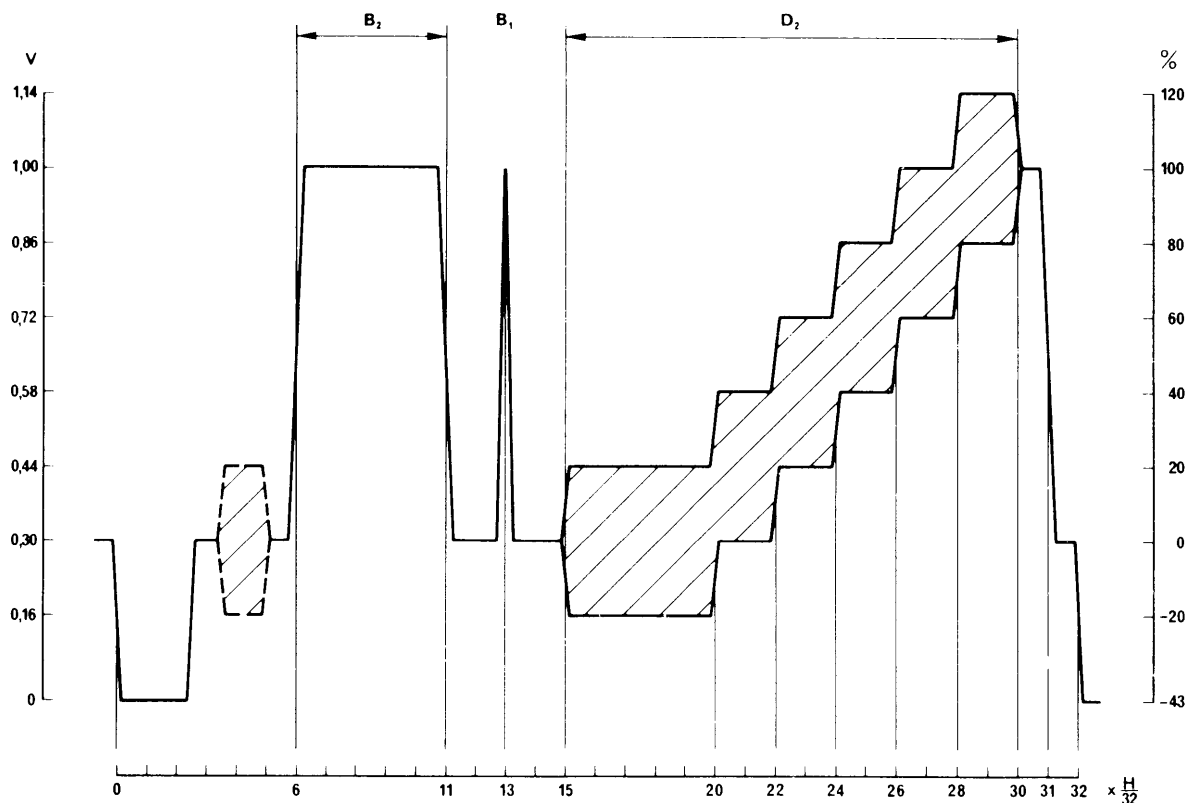


FIGURE 3 – Line 330

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 $\pm 1\%$ 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 $\pm 1\%$ 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: $1\ \mu\text{s}$ approximately;
- inherent chrominance-luminance cross-talk: $\leq 0.5\%$ 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: $\leq 2^\circ$.

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 $\pm 1\%$ of one-fifth of the amplitude of the luminance bar (B_2) (nominal value: 0.140 V);
 - 2nd section: within $\pm 1\%$ of three-fifths of the amplitude of the luminance bar (B_2) (nominal value: 0.420 V);
 - 3rd section: within $\pm 1\%$ 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: $1\ \mu\text{s}$ approximately;
- inherent chrominance-luminance cross-talk: $\leq 0.5\%$ of luminance pedestal amplitude;
- inherent phase/amplitude distortion: $\leq 0.5^\circ$;
- phase difference between the sub-carrier superimposed on the staircase in line 330 and the sub-carrier superimposed on line 331: $\leq 2^\circ$.

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 $\pm 1\%$ 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: $1\ \mu\text{s}$ approximately;
- phase difference between the sub-carrier superimposed on the staircase in line 330 and the sub-carrier superimposed on the line 331: $\leq 2^\circ$.

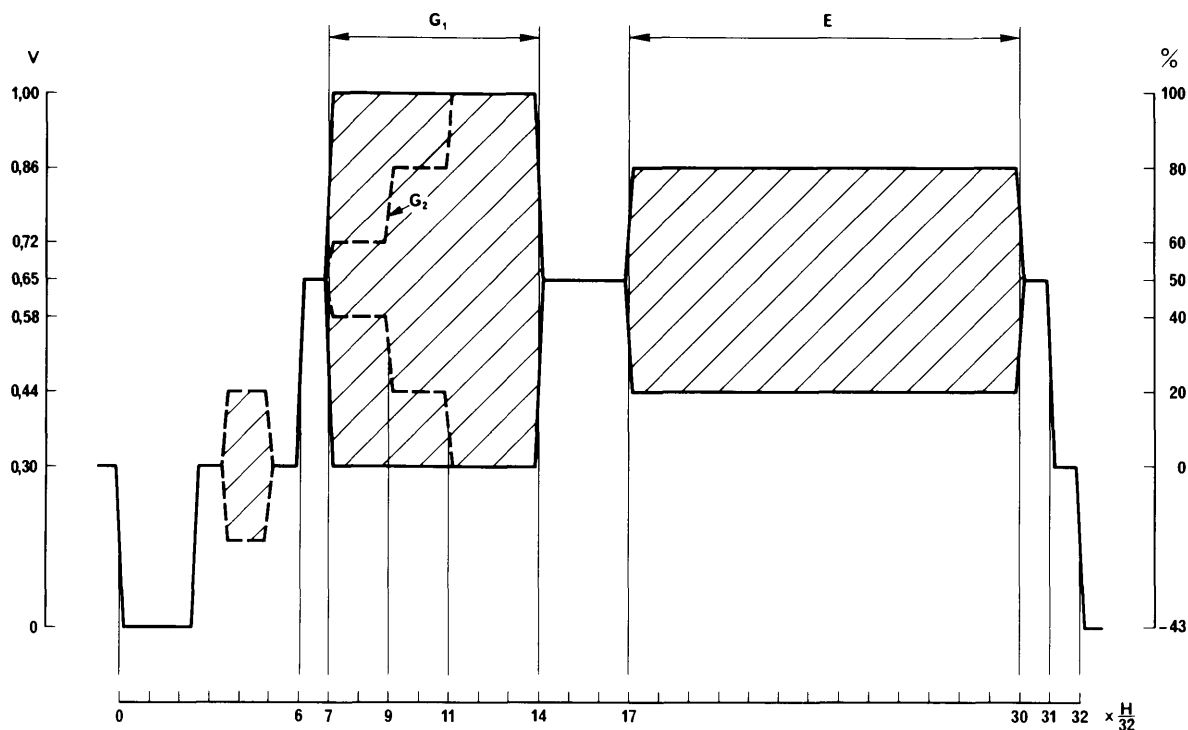


FIGURE 4 - Line 331

6. List of measurements which can be made with the defined insertion test signals

TABLE II

Characteristics measured	Waveform used	Line number
<i>Linear distortions</i>		
Insertion gain	B_2	17 and 330
Amplitude/frequency response	C_1 and C_2	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
<i>Non-linear distortions</i>		
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;
- 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) and ± 300 ns (see Note) 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. – 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)**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: $\leq 1\%$;
- tilt: $\leq 0.5\%$.

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 μ s;
- inherent chrominance-luminance amplitude inequality: $\leq 0.5\%$;
- 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 $\pm 1\%$ 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: $\leq 0.25\%$ (average picture luminance (APL): 10 to 90%);
- inherent differential-gain distortion: $\leq 0.2^\circ$ (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: $0 \pm 1^\circ$ (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)

3.1 Reference bar signal (C_1)

- position of transitions: $0H/128$ (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: $\leq 1\%$;
- tilt: $\leq 0.5\%$.

3.2 Luminance pedestal

- position of transitions: $8H/128$ and $100H/128$;
- amplitude: within $\pm 1\%$ 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 chrominance-luminance intermodulation: ≤ 0.25 IRE unit;
- inherent phase/amplitude distortion: $\leq 0.5^\circ$;
- chrominance component is to be phase-locked to programme colour burst, if present.

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 μ s, nor longer than 2.0 μ s 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

TABLE IV

Characteristics measured	Waveform used	Line number
<i>Linear distortion</i>		
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
<i>Non-linear distortion</i>		
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.

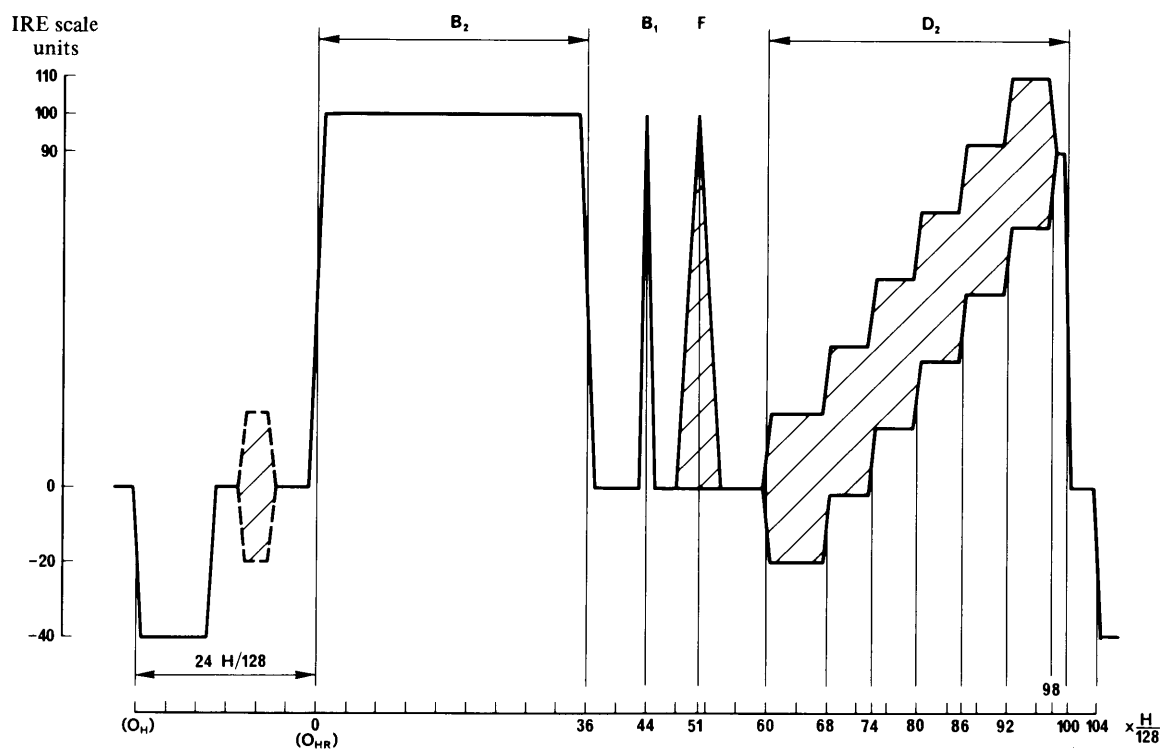


FIGURE 5 – Line 17 of Field 1

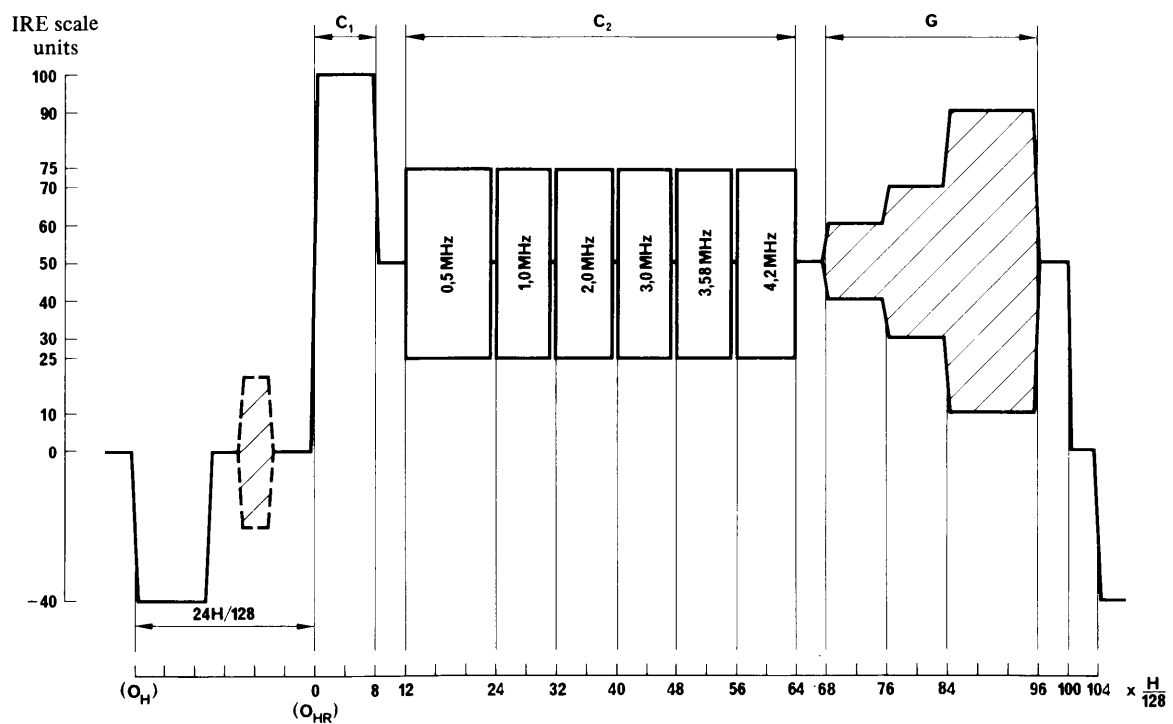


FIGURE 6 – Line 17 of Field 2

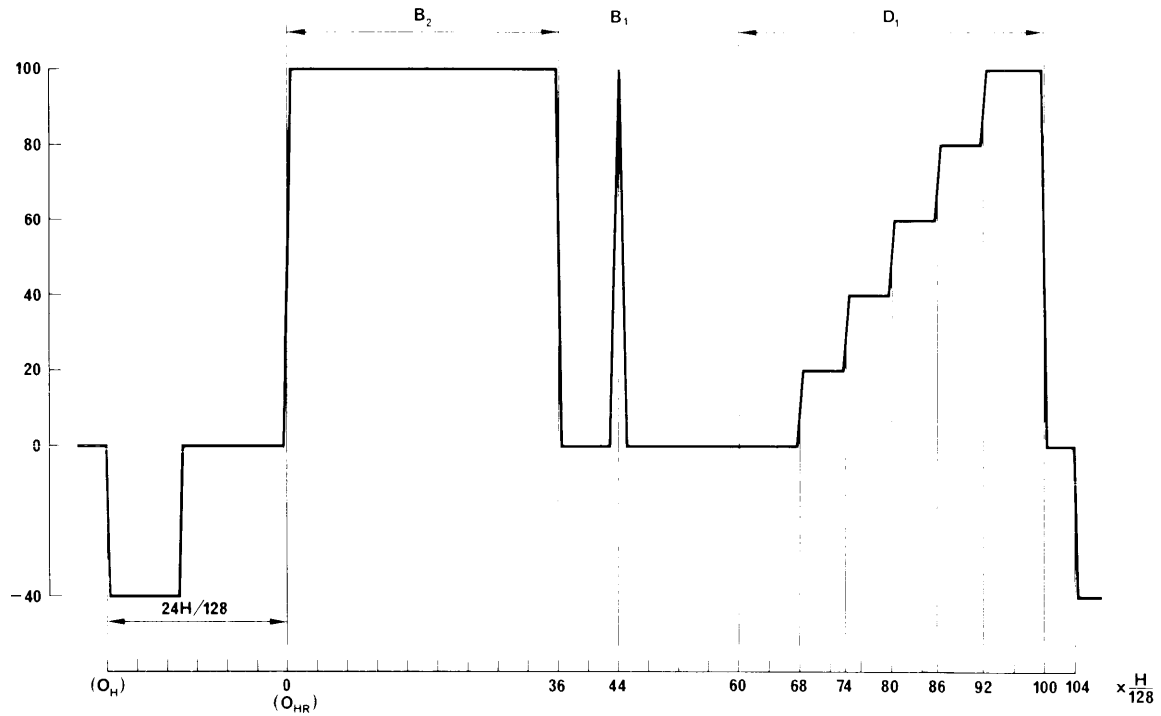


FIGURE 7 – Line 17 of Field 1

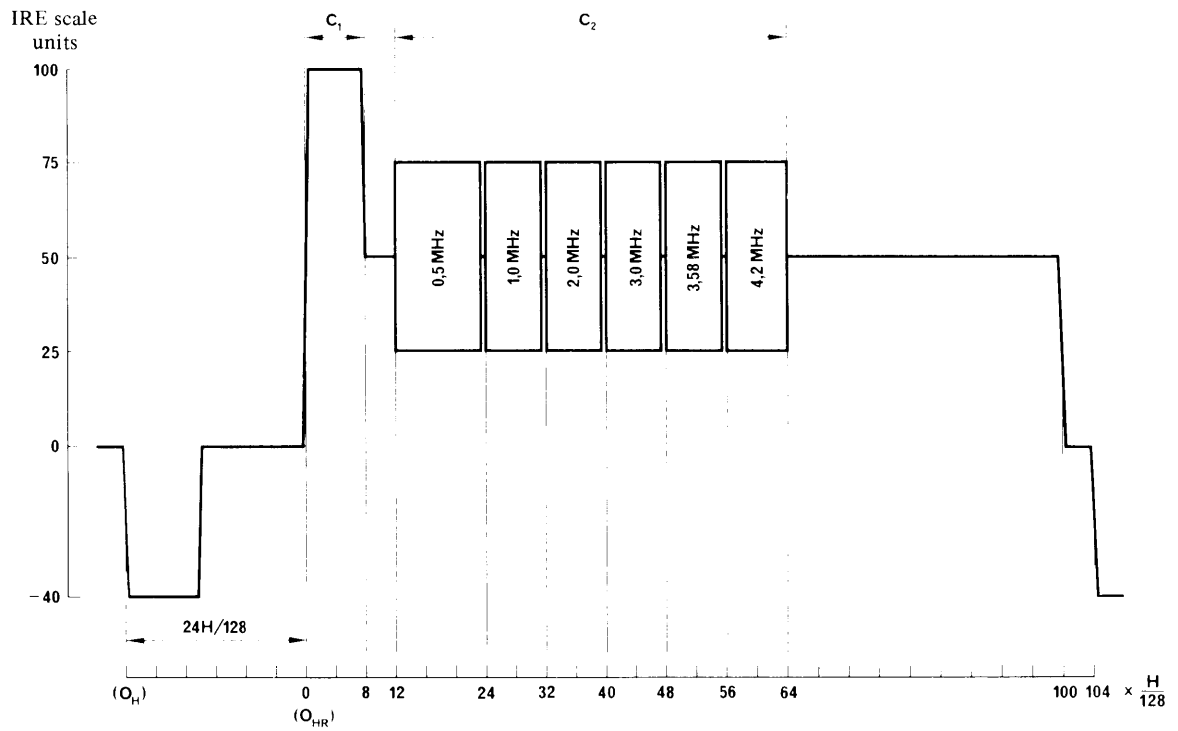


FIGURE 8 – Line 17 of Field 2

Note. – Figs. 7 and 8 are examples of insertion test signals for monochrome transmission.

REPORT 314-5

INSERTION OF SPECIAL SIGNALS IN THE FIELD-BLANKING INTERVAL OF A TELEVISION SIGNAL

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

(1966-1970-1974-1978-1982)

1. Introduction

Special signals can be inserted into and removed from the field-blanking interval of a television signal without detriment to the quality of the picture. Such insertion or removal can be done by electronic procedures commonly employed for other purposes in a television system.

Such signals can be used for:

- supervision and measurement of various transmission characteristics (insertion test signals),
- transmission of data concerning operation such as information instructions and remote control of equipment,
- transmission of sound and/or other information.

The special signals can be placed in two categories, those used for international purposes and those used for national purposes.

At the present time, international special signals are used for the supervision and measurement of some of the parameters of international circuits. Sound signals or other information in coded form may also be inserted. The international signals should be inserted by the appropriate authority at the point of origin of the international circuit and should not be removed or replaced except by the authorities at the terminal point or points of the international circuit (Recommendation 473).

At the present time, national special signals are used for many purposes including the supervision and measurement of the parameters of transmission circuits, automatic control and correction and data transmission. Sound signals or other information in coded form may also be inserted. National special signals are inserted and removed by the particular authorities concerned. In the case of international transmission, any national signals must be removed if so requested by an authority at the receiving end of the international circuit.

The addition of a dispersal signal to the television waveform during transmission may result in modification of some performance measurement results [CCIR, 1978-82a and b]. In the particular case where a triangular dispersal signal at half the field frequency is employed, the maintenance of a pre-determined relationship between its polarity and the field structure of the television waveform would lead to better repeatability of results, particularly in respect of non-linear distortion parameters measured on the international insertion test signals.

2. 625-line systems

2.1 To facilitate the uses of special signals it is necessary that all countries should accept a field-blanking interval of 25 lines duration (Recommendation 472).

2.2 Although international insertion test signals have hitherto been generally regarded as an “in-service” testing method, giving results which are less precise than those obtained by line-repetitive signals, it now seems probable that future traffic demands on the vision circuits could make it difficult to allow time for conventional tests before each programme, and it would in that case, be necessary to perform all operational measurements by means of insertion signals.

2.3 Recommendation 473, Annex I, reserves two lines in each field for international test signals, i.e., line 17 (330) and 18 (331).

2.4 It has been proposed, furthermore, [CCIR, 1966-69a] to reserve the last line of each field-blanking interval, i.e., lines 22 (335) for the measurement of noise on international circuits.

2.5 Taking the previous considerations into account, it is proposed that the diverse types of use be distributed as follows:

- data signals: lines 16 (329);
- international signals: lines 17 (330) and 18 (331);
- national signals: lines 19 (332), 20 (333) and 21 (334);
- noise measurement: lines 22 (335).

The other special signals mentioned in the Introduction could be transmitted in the remaining lines of the field-blanking interval, when available.

2.6 It may be noted that, if a simplified synchronizing waveform were adopted, additional lines might become available for special signals (see Report 626).

2.7 It has also been proposed to initiate a long-term investigation to consider the possibility of simplifying the international test signals defined in Recommendation 473, Annex I [CCIR, 1970-74a]. This could lead, in the future, to a reduction in the number of lines occupied by test signals.

Test-signal generators and automatic measuring equipment, designed to operate with a simpler signal, could become simpler and then cheaper and more reliable.

As a first contribution to this investigation, it is proposed that an insertion test signal which occupies only one line could substitute for the signals of lines 17, 330 and 331, defined in Recommendation 473, Annex I. The necessary devices to generate this signal could be derived from existing equipment.

2.8 In the European Broadcasting Area the EBU employs lines 16 and 329 for the transmission of programme source identification. Other information may also be transmitted. Bi-phase coding is used with a data rate of 2.5 Mbit/s and each data line comprises fifteen 8-bit bytes. The first byte is used for synchronization, the second byte provides a unique start/address code (10010001 11010101), the third byte identifies one field in each set of 8 successive fields so that 8 characters may be time-multiplexed in the fourth byte, which is used to identify the source by a display of two rows, each of four characters. Eleven bytes are available for transmission of other information; for example, to provide a telegraph message channel and/or display of two lines of text [CCIR, 1970-74b].

2.9 The French Administration has reported on a time-division multiplex data packet system which may be used in the field-blanking interval of a television signal. That system may be used to provide a large number of independent and transparent data channels which can then be used in many different ways, either for broadcasting or for transmission (teletext, multilingual sub-titles, commentaries and/or operational signals associated with the signal transmitted, video measurement results effected at a point at which conversion is carried out, etc.).

In addition, by simply reducing the bit-rate, this multiplex system may also be used with 525-line transmissions.

2.10 In the United Kingdom, pages of text and graphical symbols are broadcast. A service of news flashes and sub-titles, which may be displayed as inserts of the picture, is also made available to the viewer. These facilities are provided by means of a teletext system [BBC, IBA, BREMA, 1976]. This system uses non-return-to-zero (NRZ) binary signalling at 6.9375 Mbit/s during each data line. At present, lines 17, 18, 330 and 331 are used for the transmission of these data within the United Kingdom.

3. 525-line systems

3.1 Special signals are in use with the 525-line systems of Japan [CCIR, 1963; CCIR, 1966-69b], Canada [CCIR, 1963-66a; CCIR, 1966-69c] and the United States of America [CCIR, 1963-66b; CCIR, 1966-69d; CCIR, 1966-69e]. The current reservation of lines for these signals is shown in Table I.

TABLE I – 525-line systems; lines reserved for special signals⁽¹⁾

Country	National transmission testing	Other national purposes	International transmission testing
Japan	Both fields: 17, 18	Both fields: 16, 19, 20, 21	Both fields: 17
USA	Both fields: 17, 18	Both fields: 19, 20, 21	Both fields: 17
Canada	(²)	(²)	Both fields: 17

(¹) The position of the line refers to the beginning of the field-blanking period.

(²) In 1977, a review was in progress of the use of special signals in the field-blanking interval.

3.2 The international test signals are specified in Recommendation 473, Annex II.

3.3 Transmission of operational information and control signals in line 19, and occasionally other lines, is now fully operational in Japan.

- 3.4 In the United States of America, line 17 of both fields is used for both national and international test signals, the format of both being identical.
- 3.5 In the United States of America, line 19 of both fields has been reserved exclusively for, and is being used by, all the major television networks for a vertical interval reference (VIR) signal, which is intended to reduce undesired variations in colour quality throughout the television system [1970-74c].
- 3.6 Line 16 and below of both fields might be used for sound transmission, still image transmission or transmission of data or other information.
- 3.7 Major television networks in the United States of America are conducting field-trials of a source identification (SID) signal on line 20 field 1, utilising digital techniques.
- 3.8 In the United States of America, line 21 has been authorized, on a non-exclusive basis, for the transmission of a data signal at a rate of 0.5 Mbit/s to provide a service of sub-titles for the hearing-impaired.
- 3.9 In the United States of America, broadcasting organizations have been experimenting with the transmission of teletext signals on lines 15 and 16 at data rates varying from 3.7 to 6.2 Mbit/s.

REFERENCES

BBC, IBA, BREMA [1976] Broadcast teletext specification (available from the BBC or the IBA).

CCIR Documents

[1963] Geneva. 180 (Japan).

[1963-66]: a. XI/4 (Canada); b. XI/36 (USA).

[1966-69]: a. XI/9 (United Kingdom); b. CMTT/137 (XI/156) (Japan); c. XI/101 (Canada); d. XI/21 (USA); e. CMTT/14 (USA).

[1970-74]: a. CMTT/208 (Italy); b. CMTT/237 (EBU); c. 11/103 (USA).

[1978-82]: a. CMTT/20 (United Kingdom); b. CMTT/70 (USA).

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CASTELLI, E. and GIUGLIARELLI, G. [1968] Il controllo automatico delle reti di distribuzione televisive. (The automatic monitoring of the television distribution networks.) *Alta Frequenza*, Vol. 37, **7**, 633-646.

CCIR Documents

[1963]: Geneva. Annexes 15 and 16 to Doc. 11 (France).

[1970-74]: CMTT/34, CMTT/43, CMTT/183 (Germany (Federal Republic of)).

[1974-78]: CMTT/252, 11/61, 11/361 (France).

REPORT 818-1

AUTOMATIC SWITCHING SYSTEMS FOR TELEVISION CIRCUITS

(Question 15/CMTT and Study Programme 15B/CMTT)

(1978-1982)

1. Introduction

Television signals may be transmitted by means of cable, radio-relay and satellite systems. For each of these types of transmission circuits specific requirements can be stated with respect to the protection against outages of the television channel being in service. In principle the reduction of outages of the overall system is obtained by providing it with an automatic protection channel switching system, by means of which the traffic normally carried on a working television channel can be carried temporarily on a protection channel on the same route.

So far the only information which has been supplied concerns the characteristics of protection channel switching systems of terrestrial radio-relay circuits in Recommendation 444, Reports 137 and 443. Consequently the present Report deals only with the requirements of these types of protection channel switching systems. Administrations are invited to make contributions concerning the automatic switching of television circuits on cable and satellite circuits.

2. Criteria for switching-system performance on radio-relay links

The parameters important to automatic protection channel switching systems used for television transmission on radio-relay links include the following:

- switching thresholds;
- transmission impairments due to the presence of switching equipment;
- transmission impairments at the moment of the switching to the protection channel.

2.1 *Switching thresholds*

In conventional automatic protection channel switching systems, switch-over is initiated when a noise level in the noise measuring slot above the received video signal rises above a certain threshold level, and/or the received continuity pilot or intermediate-frequency carrier level falls below another fixed threshold value.

In the case of equipment failure, the switching threshold of the pilot/carrier level is not critical, since the change in the pilot/carrier level is both large and abrupt.

If there is a manual switch to a protection channel for maintenance or other operational purposes, the switching threshold is not important, since the manual switching action overrides the threshold settings.

The switching threshold required to protect the television signal from the effects of fading depends on many factors, some of which have conflicting requirements. However, in practice, as indicated in Report 137, the switching thresholds used for the noise level in telephony transmission have also been found suitable for television transmission under all switch-over conditions due to fading.

In the same Report the circumstances are discussed under which the switching threshold warrants separate consideration. For television radio-relay systems of low transmitting power the suggested threshold is a weighted signal-to-noise ratio of 43 to 48 dB. For high power systems, the suggested threshold is a weighted signal-to-noise ratio of 48 to 53 dB. Additional information is contained in [CCIR, 1978-82], which includes consideration of the nature and magnitude of impairments to the video signal which may appear prior to automatic switching during fading due to multipath propagation.

2.2 *Transmission impairments due to switching equipment*

If the automatic switching equipment is designed to meet the telephone channel requirements given in Report 137, a measurable but relatively small degradation may be introduced into the video channel.

2.3 *Transmission impairments due to insertion of another channel section*

Since the gain and phase versus frequency characteristics may vary to some degree from channel to channel, switching over to the protection path may cause some change in overall performance. The extent of this change depends upon the uniformity of the gain and phase characteristics for each radio channel which, in modern systems, shows considerable improvement, compared with older systems.

Degradation beyond normal objectives are not necessary. Indeed, in many instances distortions are in accordance with laws implying some degree of cancellation. If a channel which just meets the objectives is replaced by another channel which also meets the objectives but with opposite polarity, it is probable that the overall objective will not be achieved. However, this phenomenon can be reduced to a great degree by special maintenance procedures in which switching sections of the link, including the protection channel, are lined up in accordance with the existing CMTT Recommendations. No compensating alignment of the total link should be made. The quality of the modern radio-relay equipment is sufficient to guarantee that CMTT transmission performance requirements can be met under all switched conditions.

The CMTT should investigate the tolerances of the phenomena occurring during the switching from a working channel to a protection channel. The Study Groups concerned should interpret the results of this study with respect to television links on cable, radio-relay or satellite systems. They could be part of the present CMTT Report.

3. Summary

3.1 Automatic protection switching systems on video channels can significantly reduce outages and performance degradations in many cases.

3.2 It appears from Report 137 that there should be a difference between the setting of the switching threshold levels of low and high power radio-relay systems.

For low power radio-relay systems transmitting television the suggested values for minimum signal-to-noise ratio are 43 to 48 dB (weighted).

For high power radio-relay systems transmitting television the suggested values for minimum signal-to-noise ratio are 48 to 53 dB (weighted).

3.3 The CMTT should investigate the maximum allowable impairments of the video signal during the insertion of a protection channel on cable, radio-relay or satellite systems. From this material the Study Groups concerned could establish the criteria to be applied for the lining up of the total links and separate switching sections.

REFERENCES

CCIR Documents

[1978-82]: CMTT/77 (Canada).

REPORT 639-2

AVAILABILITY OF AN INTERNATIONAL TELEVISION CIRCUIT OR CHAIN

(Study Programme 15C/CMTT)

(1974-1978-1982)

1. Introduction

From the standpoint of availability, the transmission of television programmes differs from the transmission of other signals, such as telephony in the following respects:

- transmission is unidirectional,
- interruptions of either video or associated sound signal(s) should be deemed as interruptions of the television programme,
- the definition of an interruption to a television programme may be different from that for other signals,
- programme switching facilities are sometimes incorporated into the television transmission network,
- an interruption to a television programme will usually affect millions of viewers.

Therefore, the availability in transmission of television programmes requires special consideration.

2. Classification of impairment

Impairments of a television programme (video or sound) which can be considered an interruption may be classified as follows:

- no signal,
- wrong programme,
- excessive noise,
- level fluctuation,
- other factors, such as gross distortion.

3. Definition of an interruption to television programme services

This subject requires considerable study and some possible approaches are outlined below.

The values given in the following examples represent a compromise between those which might be desirable theoretically and those which might be deemed practicable. They can be taken as design objectives applicable to the hypothetical reference circuit.

Example 1. — An interruption is defined as a change in the insertion loss of the circuit or connection of X dB which lasts for at least y seconds. Compromise values of 6 dB and one second for X and y have been suggested.

Example 2. — An interruption is defined as a reduction of the signal-to-random-noise ratio to Z dB, or worse, weighted in accordance with Recommendation 567, which lasts for at least w seconds. Compromise values of 35 dB and one second for Z and w have been suggested.

Since interruptions may be caused by many different phenomena as described in § 2, it may be that the most practical method of defining an interruption is by means of subjective assessment.

Example 3. — It is suggested that an impairment, which, if it were present continuously, would produce a subjective grade of 1 or 2 on the 5-grade impairment scale (see Report 405, and Recommendation 500), should be deemed to be an interruption according to a table of the form given below. In the Table I, given as an example, account is taken of the duration of the impairment and of the number of such impairments in a given time. Consideration should be given to the fact that the subjective effect of some impairments may last longer than the phenomena which are causing them.

TABLE I

Duration of impairment	Rate of recurrence before condition is deemed to be an interruption
$\leq x \mu\text{s}$	1 per minute
$> x \mu\text{s}$ but $\leq y \text{ ms}$	1 per hour
$> y \text{ ms}$ but $\leq z \text{ s}$	1 per day
$> z \text{ s}$	Never ⁽¹⁾

(1) In this case the circuit is considered to be out of service.

Compromise values of 5 ms, 1 s and 3 s for x , y and z have been suggested.

The definition of an interruption to sound programme services which is given in Report 642 may also be applicable to television with its associated sound programme.

4. Example of availability in actual circuits

[CCIR, 1974-78], shows that the actual availability during 1975-1976 of the television transmission network in Japan was 99.989% when normalized to a 2500 km circuit including 9 programme switching equipments. Unavailability of 0.008% was due to interruptions in the transmission portion and the remaining 0.003% was due to interruptions in the programme switching equipment. Factors contributing to this high availability include the following:

- simultaneous transmission of video and sound signals;
- programme switching using computers;
- re-routing by means of circuits used occasionally, in case of main circuit failures.

5. Availability objective

This subject requires further study.

REFERENCES

CCIR Documents

[1974-78]: CMTT/193 (Japan).

RECOMMENDATION 569-1

DEFINITIONS OF PARAMETERS FOR SIMPLIFIED AUTOMATIC MEASUREMENT
OF TELEVISION INSERTION TEST SIGNALS

(Study Programme 15D/CMTT and Question 15/11)

(1978-1982)

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, occurring 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 μ s before, and 1 to 2 μ s 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:

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

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:

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

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 occurring 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.

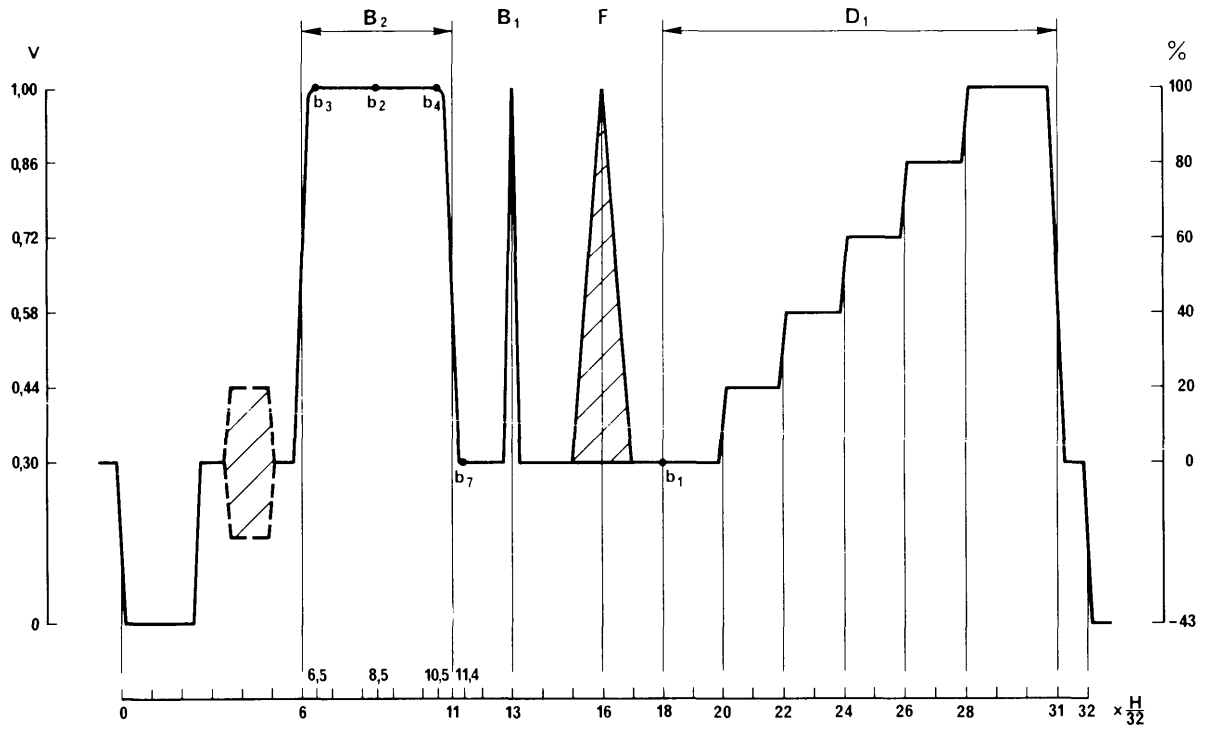


FIGURE 1 – Line 17 for 625-line signals

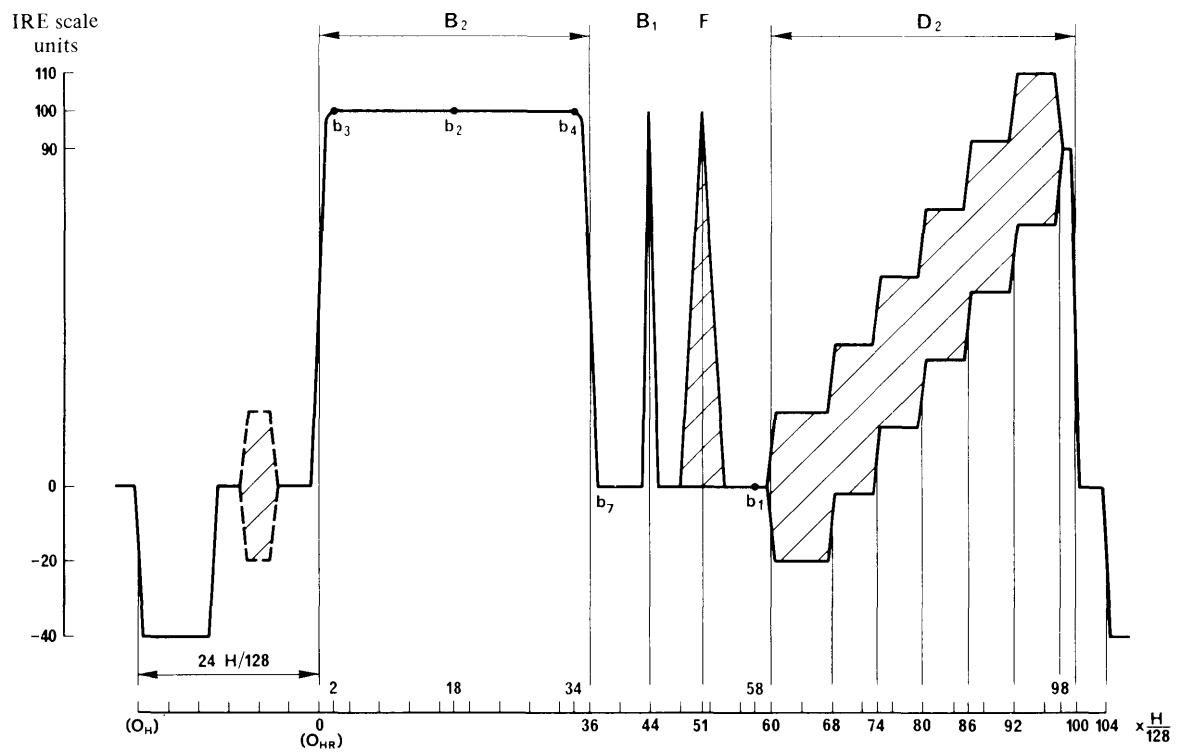


FIGURE 2 – Line 17 of field 1 for 525-line signals

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:

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

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.

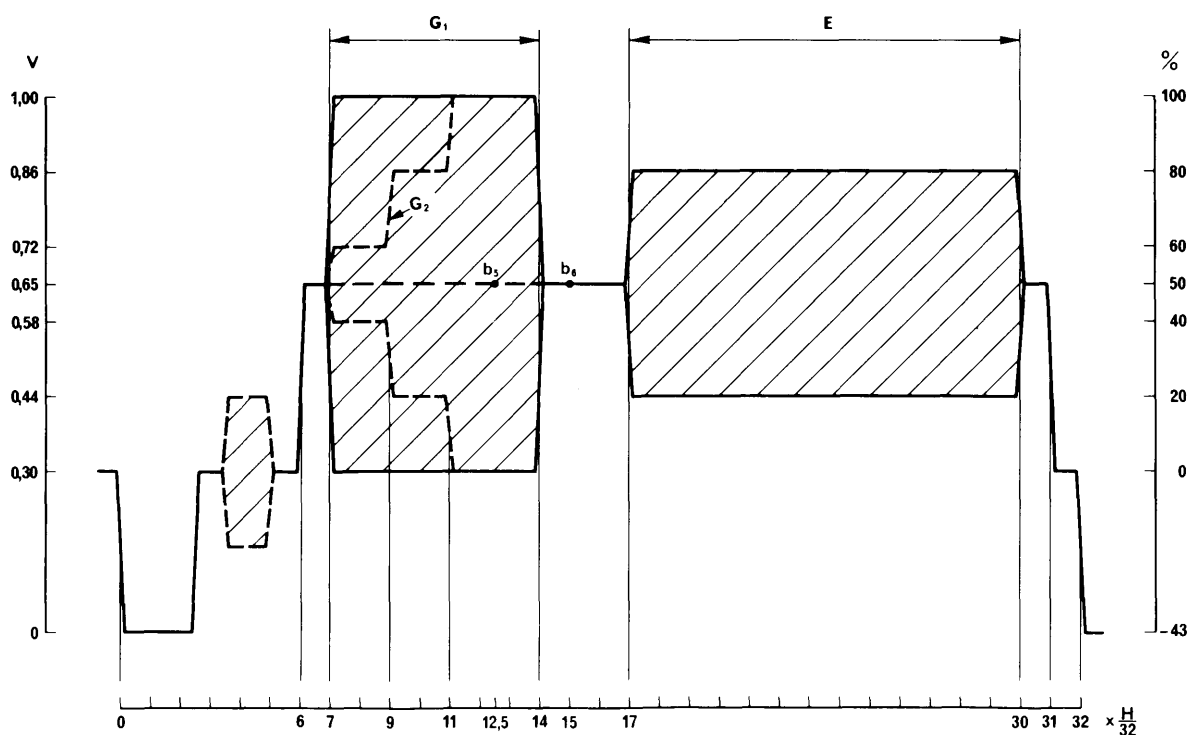


FIGURE 3 — Line 331 for 625-line signals

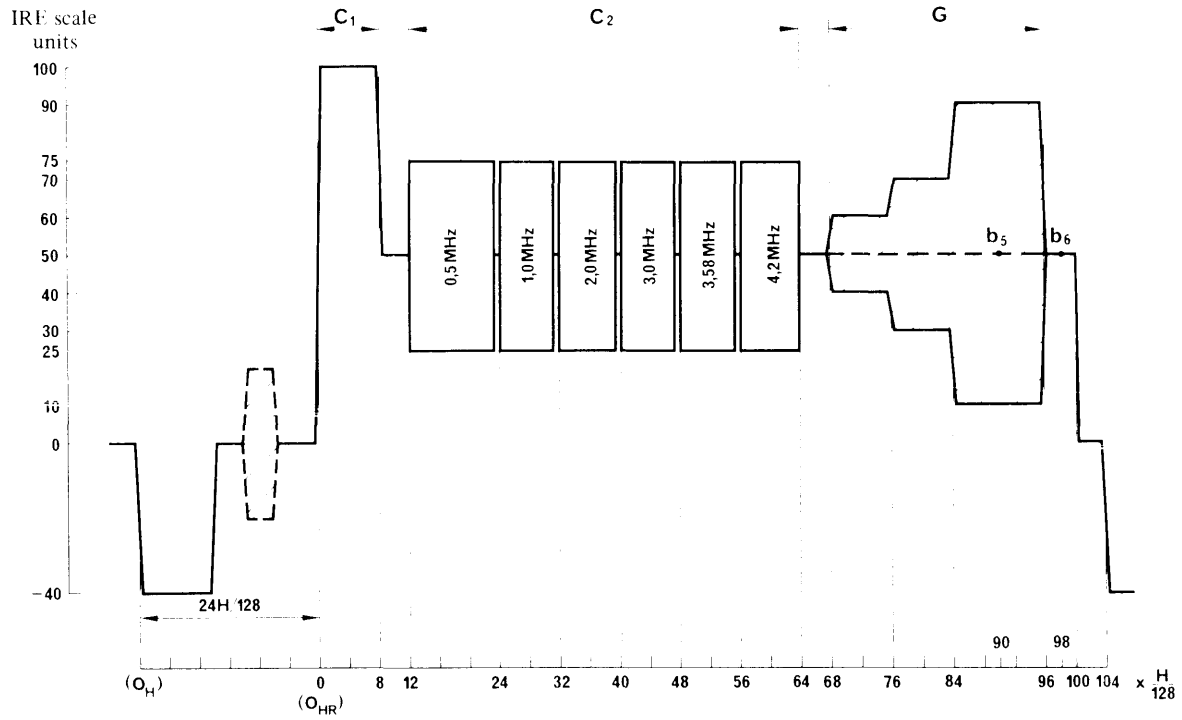


FIGURE 4 - Line 17 of field 2 for 525-line signals

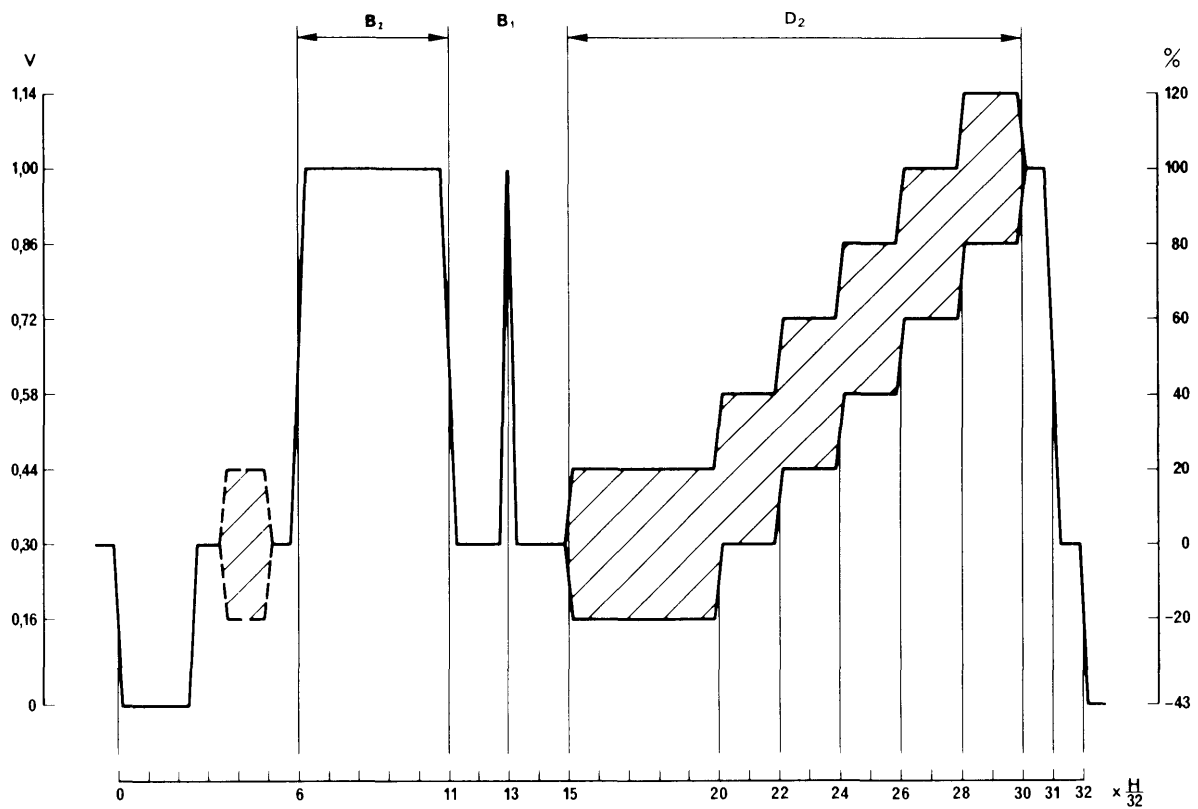


FIGURE 5 - Line 330 for 625-line signals

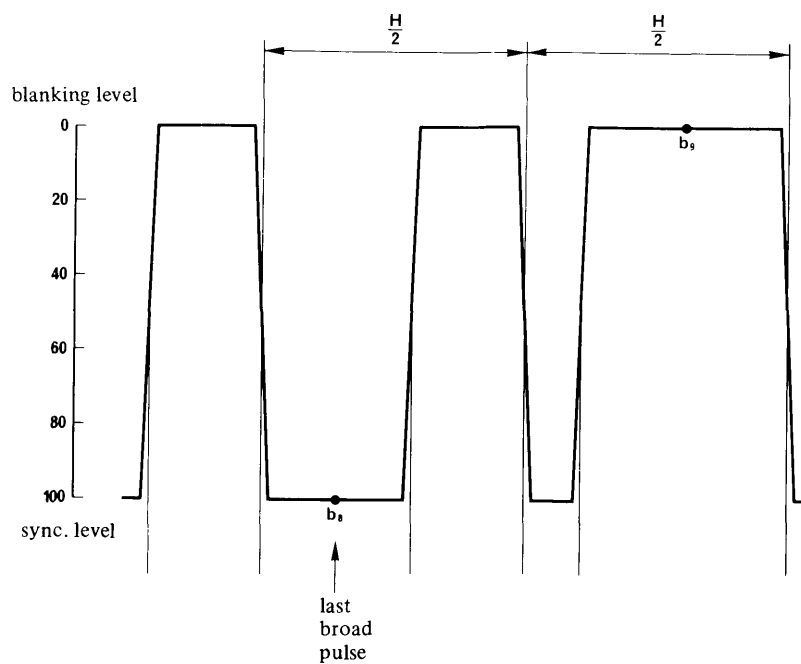


FIGURE 6 – Measurement points for sync. amplitude error during equalizing pulse period

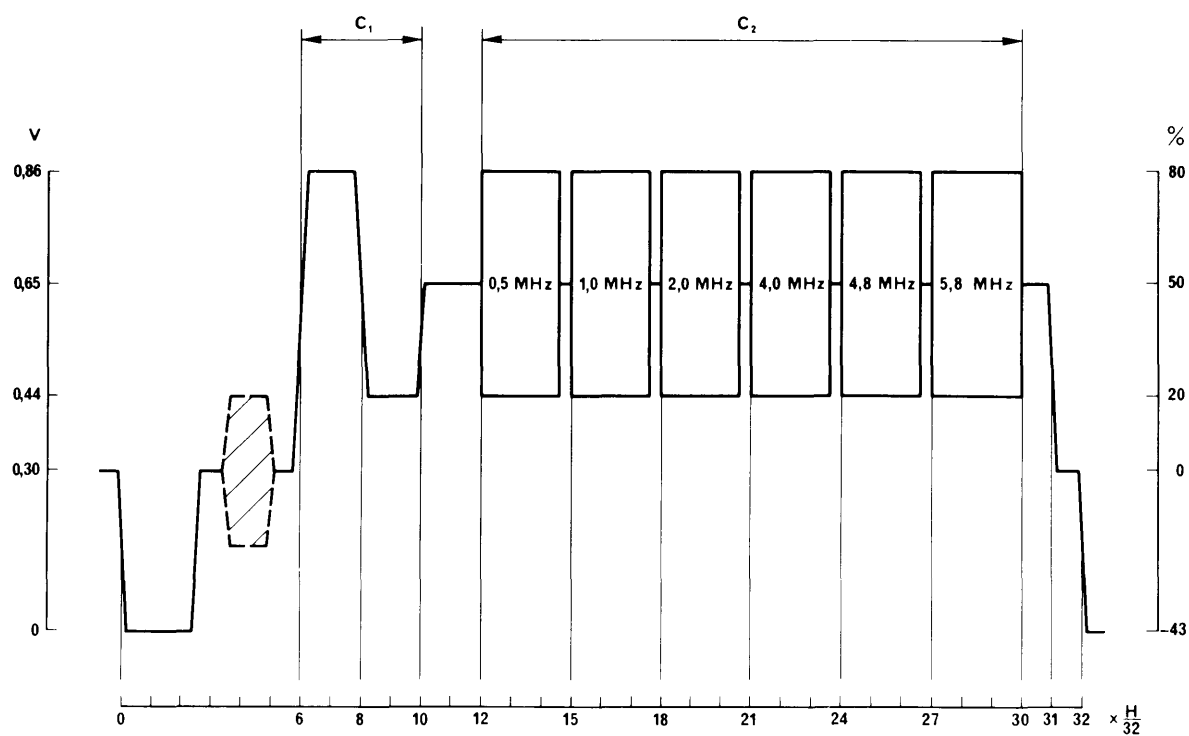


FIGURE 7 – Line 18 for 625-line signals

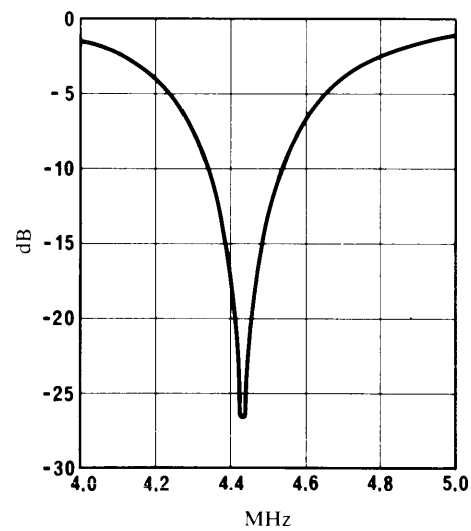


FIGURE 8 – Response of sub-carrier notch filter for 625-line signals

3 dB bandwidth: 600 kHz
 Attenuation at 4.43 MHz ≥ 26 dB

TABLE 1 – 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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REPORT 411-4

**AUTOMATIC MEASUREMENT AND MONITORING OF THE
PERFORMANCE OF TELEVISION CHAINS**

(Study Programme 15D/CMTT)

(1966-1970-1974-1978-1982)

1. Introduction

Many Administrations are now applying automatic techniques to the monitoring and control of transmission quality of their television networks. The use of insertion test signals such as those specified in Recommendation 473 makes it possible to perform automatic measurements during programme transmissions and to determine which part of a chain introduces impairments. Additional measurements may have to be made for those parameters which are not tested by the insertion test signal.

Current investigations into automatic measurements are concentrated on the signal elements included in these insertion test signals, especially (for 625-line systems) on the signal assemblies specified for lines 17 and 330 and on the noise-measurement lines 22 and 335.

Long-term measurements [CCIR, 1970-74a] seem to show that it is not essential to measure all the parameters defined in Recommendation 569 in order to manage a network effectively. Thus for the Eurovision Network, apparatus measuring six parameters only is proposed for day-to-day supervision [EBU, 1976].

Automatic measurement of video test waveforms also permits special measurements which could be made manually only with difficulty [Rhodes, 1977].

The automatic analysis of video waveforms may be accomplished by two basic methods – analogue and digital:

1.1 Analogue methods

- measurements of the waveform made directly;
- conversion of the waveforms by sampling into a low bandwidth analogue, followed by measurements on that analogue signal.

1.2 Digital methods

- conversion of samples of the waveforms into a digital bit-stream by quantization and coding followed by digital processing.

Considerable progress has been made in these methods of analysis which, when combined with systems for the acquisition, storage, processing and transmission of data, can provide information to assist the management of networks.

2. Analogue methods of measurement**2.1 Direct analogue**

Early experiments were made with a number of different test signals; line repetitive waveforms as well as special national insertion test signals. [CCIR, 1963-66a] described a method in which a number of test signals were inserted sequentially into the appropriate field blanking interval. [CCIR, 1963-66b] described an automatic monitoring device which was divided into a waveform distortion measuring unit and a signal-to-noise ratio measuring unit. The following measurements were included:

- amplitude of video signal;
- amplitude of synchronizing signal;
- $2T$ pulse/bar ratio;
- streaking on bar;
- test signal amplitude.

An alarm signal could be emitted when predetermined thresholds were exceeded.

An experimental installation for continuous automatic monitoring of a television network consisting of radio links and transmitters has been in service in Italy since 1968. Measurements were made on a national insertion test signal consisting of sine-wave bursts of different frequencies superimposed on different luminance levels [Castelli and Giugliarelli, 1968].

Later developments of automatic measuring equipment tended to make use of the signal elements described in Recommendation 473. Several sophisticated techniques have been developed and these are equally applicable to measurement [CCIR, 1970-74b and c] or automatic correction and control purposes [CCIR, 1970-74d; Shelley and Smart, 1971; D'Amato, 1976].

Techniques for the automatic measurement of signal-to-noise ratio in an unused line of the field blanking interval are described in [D'Amato and Tabone, 1974; Grosskopf, 1971]. Another method has been described which uses only a small part of the line to obtain a measurement of signal-to-noise ratio, and so avoids the need to reserve a complete line for this purpose [CCIR, 1974-78a].

All these investigations showed that in principle there are no difficulties in measuring automatically all the types of signal distortions listed in Recommendation 569.

2.2 *Waveform conversion into a low frequency analogue signal*

The insertion test signals can readily be spectrum-converted into a low frequency signal suitable for transmission on audio and telephone circuits and for recording on chart recorders or display on slow scan oscilloscopes. Various methods can be used for this conversion. For example, stroboscopic spectrum conversion allows sufficiently accurate reproduction of the shapes of the video insertion test waveforms. It is also possible to check the waveform of the synchronizing and blanking pulses and, if markers are used, the duration of all components of the special signals and synchronizing pulses. The design parameters of such a system have been considered [Krivosheev, 1962 and 1976]. Information is available on the use of two speeds of read-out [CCIR, 1963-66c; ITA, 1963 and 1964]. As the rate of change of video signal amplitude is very fast in some parts of the test line it is advantageous to vary the sampling interval. The preferred characteristics for systems of this type are as follows:

- duration of sampling scans (switching alternately): 2 μ s and 64 μ s;
- conversion time for each sampling scan: 10 s;
- conversion time for a complete test line: 20 s.

3. **Digital methods of measurement**

An experimental system based on the use of a small general-purpose computer recently operated for a period in the United Kingdom. This system could provide an important part of the fully-integrated monitoring system for controlling groups of unattended transmitting stations and for making routine network measurements. The measurements were made automatically in time sequence. The "dwell" time on each source was varied automatically from a few seconds up to about one minute, depending upon the level of noise and/or co-channel interference which was present. The test signal employed was generally in accordance with Recommendation 473 except that no use was made of the multi-burst. The results were stored on a small disc file hence they could be read for display on a visual display unit, for the production of paper records, or for transmission to other centres [Anderson and McKenzie, 1970; IBA, 1976].

Another method using digital processing techniques has been developed in Japan [CCIR, 1970-74e]. In this case, the digital processing is preceded by an analogue process, which includes an automatic gain control of the analogue signal, a separation of the superimposed sub-carrier waveforms from the luminance signal and clamping circuits, for keeping the input signal within the dynamic range of the high speed analogue-to-digital converter. In this method, which refers to 525-line television signals, the evaluation of the multi-burst signal is included.

The availability of microprocessor integrated circuits has led to the development of dedicated microcomputers capable of computing any distortion suffered in transmission by insertion test signals.

In the United Kingdom some of the techniques evolved during the experiment described earlier have been incorporated in the design of an automatic measuring equipment using an integrated circuit microprocessor. This equipment allows wholly digital measuring techniques to be applied to on-site test line parameter analysis and noise measurement [Watson, 1976; Steele, 1975].

This approach results not only in greater versatility but also economies in both size and cost over comparable analogue equipment capable of taking executive corrective switching action. In this equipment sampling proceeds stroboscopically at field rate, as in the case of the graphic devices described in § 2.2, so that conversion of the analogue television signal to digital form is performed economically. The analysis executive programmes are stored in solid state programmable read-only memory devices in which the stored instructions can be changed, simplifying subsequent software modifications.

The recent application in the USA of high speed analogue-to-digital and digital-to-analogue integrated circuits has made it possible to digitize video signals economically in real time, i.e., non-stroboscopically, at up to four times the colour sub-carrier frequency. Real time analogue-to-digital conversion avoids the need to take into account changes in transmission distortion — especially non-linear distortions — brought about by variations in average picture level during the measurement time interval which, in stroboscopic methods, may extend over many frames [Rhodes, 1977].

The availability of solid state digital memories with capacities large enough to digitally store all information on a number of television lines, permits digital noise reduction, or averaging, to reduce the masking effects of continuous random noise on the measurements.

Automatic measurements systems based upon real-time waveform digitization, semi-conductor mass memory and a dedicated (on-site) microcomputer can measure all transmission parameters of full field or insertion test signals in accordance with Recommendation 567.

4. The use of measurement information

4.1 Automatic correction of transmission distortions

The use of insertion test signals facilitates the use of automatic devices by means of which most of the typical linear transmission distortion may be corrected. Descriptions are given elsewhere [CCIR, 1970-74d; Shelley and Whythe, 1972] of suitable devices which comprise a combination of derivative equalizers and transversal echo-equalizers.

The French Administration has recently developed a combined automatic correction and measuring equipment in which certain linear distortions of the signal are calculated from the amount of correction which needs to be applied. This method should permit a more precise evaluation of linear distortions of the "streaking" type and facilitate discrimination between distortions of quite different form which may affect the $2T$ pulse [CCIR, 1974-78b; Marquet, 1976].

4.2 Television measurement information systems

The use of digital methods for television measurements provides the necessary basis for the optimization of measurement procedures and the transfer and processing of television monitoring/measurement information by digital computers. It is reasonable to combine independent measurement facilities to form comprehensive television measurement information systems. Such systems provide generalized automation by measuring signals which have been specially transmitted and by converting the measurement information by means of logical and mathematical processes so as to facilitate the recording, storage, transmission and read-out of the data in a convenient form [Anderson and McKenzie, 1970; Krivosheev, 1968; CCIR, 1966-69a and b; Shelley and Smart, 1971].

A television measurement information system [CCIR, 1978-82] may be centralized and based on the hierarchical principle. Such a system can comprise a territorially decentralized complex of control, measuring and auxiliary equipment. Standard interfaces permit the units to perform the information exchange functions. This will allow interaction with a computer and with facilities for recording and display equipment.

The automated system is able to monitor the efficiency of TV channels and ensure diagnostic control. The overall control of a TV channel is effected by monitoring the performance characteristics of representative parameters, on the basis of a set of operational limits.

Diagnostic control can be exercised by making measurements at the appropriate points of the TV channel. Diagnostic control may be exercised by the performance of automatic failure detection functions.

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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;

* This Recommendation should be brought to the attention of Study Group XV/CCITT, as well as Study Groups 4 and 9 of the CCIR.

(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.

REPORT 643-2 *

STANDARD TEST SIGNAL FOR CONVENTIONAL LOADING OF A TELEVISION CHANNEL **

(Study Programme 15E/CMTT)

(1974-1978-1982)

Study Programme 15E/CMTT relates to the situation where one or more television signals and one or more sound signals share a common path. In assessing the noise performance of the sound channel, account must be taken of the signal which may be present at the time on the vision channel. To provide a standard basis of measurement it is desirable to load the vision channel with a conventionally accepted signal.

* The Director of CCIR is requested to bring this Report to the attention of CCIR Study Groups 4 and 9, CCITT Study Group XV, CMBD and IEC Technical Committee No. 12.

** Further information is given in Report 970.

The signal should preferably be one which is already in general use or which can readily be made available to users and manufacturers of such transmission systems. It should also, in its effect on the sound channel, be typical of a wide range of vision signals likely to be used in practice, including test signals which may be applied while the sound channel is carrying programmes.

On the other hand, the final reply to Study Programme 15E/CMTT should indicate that, where any test signal is of such a character that it can give rise to exceptional interference with other traffic in progress over a common transmission system, it is undesirable that this signal should be applied so as to actually cause significant interference in this way. The need for broadcasters to test television circuits adequately should however be recognized and restrictions upon the use of any test signals should be given by CCITT Study Group IV.

A contribution from the Federal Republic of Germany [CCIR, 1970-74a] draws attention to the high level of noise, particularly single tone interference, which is produced in the sound channel when a linearity test signal consisting of a staircase or sawtooth signal alternating with three lines carrying black or white level is applied to the vision channel. The interference is caused by the high spectral components at approximately 4, 8 and 12 kHz, which are transferred to the sound channel by intermodulation.

It points out that even if the total noise in the sound channel, including intermodulation noise from the most critical vision test signal meets the requirement for weighted noise power quoted in Report 496, there will remain a deficit of 15 dB to 20 dB compared with the requirement for single tone interference in the same Report.

Contribution [CCIR, 1970-74b], tabulates the results of practical measurements on a number of radio links carrying vision and sound channels in conformity with Recommendation 402. The noise performance of each sound channel is related to various test signals applied to the corresponding vision channel. From an inspection of the results reported, it is not possible to pick out one test signal as typical in its effect over the range of links examined.

It is clear from the contribution described above that the standard linearity test signal would provide a most searching test and would probably not be acceptable as a conventional loading. *

Measurements using PAL and SECAM colour slides and colour bar transmissions, reported by the OIRT in [CCIR, 1974-78], indicate that the main part of the interference from a PAL or SECAM television signal on radio-relay systems falls in the region of the second harmonic of the colour sub-carrier. In addition there is in the range of the sound sub-carrier a spectrum due to intermodulation, the repetition rate of which corresponds to line frequency (PAL system) or half the line frequency (SECAM system). The amount of interference to the sound sub-carrier in these measurements was least when the sound sub-carrier frequency was a multiple of the line frequency.

Specific colour bar test signals appear to meet most of the criteria listed in this Report and are specified in Recommendation 570 for analogue colour television circuits.

A contribution [CCIR, 1978-82] draws attention to a CCITT Study Group XV statement (reproduced in Annex I) that it would be useful to complement such signals by a signal representing a loading not exceeded for about 80% of the time, and suggests that all the requirements in this Report could be met by a single signal which varies in a cyclic manner over a period of, say, one minute. However, the specific colour bar test signals of Recommendation 570 were chosen as being reasonably representative of normal television signals and not in any way extreme, and no evidence to the contrary has been produced. Also, given that extreme television programme signals (e.g. sports results captions and background in saturated colours) can exist for substantial periods of time, the usefulness of the concept of a cyclically varying signal must first be considered. Further study is also required to determine suitable signals for analogue monochrome transmission and digital colour and monochrome transmission.

* [CCIR, 1970-74a] suggests that, on radio systems which combine sound and vision channels, it may be desirable to limit the use of such linearity test signals to very short and infrequent test periods. If therefore there is a requirement for more frequent measurement of differential gain and phase whilst the sound channels are in use, some other method may perhaps be chosen. The insertion test signals of Recommendation 473 may be appropriate in such a case.

It should be noted that the contributions made to these studies so far relate to the problems of FM/FDM microwave links and that considerable further studies may need to be carried out in connection with TV in a 12 MHz cable system (as specified in CCITT Recommendation J.73) or in 18 MHz or 60 MHz systems as specified in CCITT Recommendation J.77 and further studied in CCITT Question 20/XV. Also requiring further study is a limitation on the level of out-of-band components of any test signals proposed. This might be along the lines of § 3 of Recommendation 463.

The rise time of the luminance component of the colour bar signal may be of importance. This should be a matter for further study in Study Programme 21D/CMTT.

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[1974-78]: CMTT/25 (OIRT).

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ANNEX I

EXTRACT FROM THE PRELIMINARY REPLY TO QUESTION 20/XV (CCITT) – TELEVISION TRANSMISSION ON 60 MHz SYSTEMS

It is noted that the adoption of a colour-bar test signal is proposed as provisional only and that further study is to be asked for by Report 643. Study Group XV welcomes this standpoint since so far only one administration has reported any results of using this form of loading signal on coaxial-cable line systems.

However, it is to be noted that a single test measurement with such a signal gives relatively little information on the behaviour of a system when carrying actual traffic. When a system carries multiplex telephony, the large number of independent sources means that the properties of the combined busy hour signal are relatively stable in spite of random variations of loading in the individual channels; and hence measurement with a single conventional value of loading is sufficient. Since each individual television signal represents a large part of the load on a system, it can be useful to have information on the statistical behaviour of such signals. All the test signal alternatives given in Recommendation 570 represent a condition which is relatively rare, i.e. such a loading is perhaps not exceeded for 99% of the time. A condition representing a loading not exceeded for more than 80% or so of the time would be a useful complement to those already given. It is to be hoped that CMTT will be able to specify such a test signal on the basis of available information of actual television signal statistics. CCITT Study Group XV proposes to study this point.

REPORT 819

**APPLICATION OF INSERTION TEST SIGNALS TO DIGITAL
AND MIXED ANALOGUE-AND-DIGITAL TELEVISION CIRCUITS**

(Questions 13/CMTT and 18/CMTT;
Study Programmes 14A/CMTT, 15A/CMTT,
16A/CMTT and 16B/CMTT

(1978)

1. Introduction

Insertion Test Signals (ITS) are now widely used for the measurement of performance and the monitoring, control and correction of characteristics of international analogue television circuits. Particulars of these signals are specified in Recommendation 473. Recommendation 473 also states 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". The lines in the field-blanking interval on which they are inserted are defined in Report 314. This Report also allocates a "quiet line" for the measurement of noise on 625-line transmissions.

The existing test signals, which are based on the full-field signals in Recommendation 567 were developed to explore the types of distortion which are characteristic of analogue transmission systems. It is possible that in the long-term international circuits will employ digital bearers exclusively, but it is probable that digital bearers will be introduced in stages and that there will therefore be a period during which mixed analogue-and-digital television circuits have to be considered. Two requirements may be identified:

1.1 *Forward transmission of existing analogue ITS through digital sections*

It may not be economic to produce a digital system which will directly transmit the existing ITS with as low a distortion figure, in all respects, as the analogue system which it replaces.

1.2 *Methods of testing the digital section itself*

The existing ITS may not fully explore all the types of distortion which are characteristic of the particular type of digital system employed.

In connection with the problem of § 1.1, it should be borne in mind that, as stated above, the test signals, methods of measurement and specification of distortion limits are all related to analogue techniques and that, consequently, the fact that a digital circuit might give rise to a higher distortion figure for one parameter than did its analogue equivalent, does not necessarily mean that it would introduce greater subjective impairment on a television picture. Decision 18 states only that "the first objective for systems employing digital or digital and analogue methods should be that the overall subjective performance is not worse than the equivalent analogue system". A similar idea is expressed in Recommendation 604 and Report 646.

Whilst digital transmission sections on international circuits are not yet a reality, some digital equipments are already in service, e.g. as timing correctors, source synchronizers and standards converters.

2. Ability of digital systems to transmit the present ITS

Various types of digital system which are currently being developed are described in principle in Report 646. The ability of a digital codec to transmit the present form of ITS will depend upon the coding technique employed. The following sub-sections consider respectively the characteristics of simple PCM systems and of systems employing redundancy reduction techniques.

2.1 *PCM systems*

The coding technique which seems least likely to impair an existing ITS is that of PCM of the composite colour signal at a sampling rate in excess of the Nyquist limit. However, significant distortion of the ITS and of the quiet line will be caused by the process of amplitude quantization, resulting in an apparent increase in

non-linear distortion and in the value of noise measured. For example, a single 8 bits/sample PCM codec adjusted to handle a maximum signal excursion of 176 IRE units may cause differential gain and phase values of 2.2% and 1.3° * respectively [D'Amato, 1972; Thompson, 1975; Felix, 1976]. It should be noted however, that these are peak values, and that measured quantities depend upon the phase of the sub-carrier with respect to the sampling signal and upon the instantaneous luminance value.

Therefore the current 5 riser staircase test signal with superimposed sub-carrier, as described in Recommendation 567, Part C, can give widely inconsistent results through arbitrary positioning of its treads relative to the quantizing levels of the PCM codec.

The subjective effect of this distortion is quite different from that of similar magnitudes of non-linearity produced by an analogue system. Typically, a picture signal of varying sub-carrier phase and luminance will cause the instantaneous values of differential gain and phase to cycle sample by sample through several magnitudes within the peak values quoted above, and with varying sign, so that subjectively a much reduced average quantity will be perceived. The matter of defining and measuring quantizing non-linearity in terms of differential gain and phase requires further study.

A further difficulty of measuring the performance of a digital section is that the present quiet line method of measuring noise is unlikely to reliably indicate the effects of quantization. It is necessary to measure quantization noise in the presence of a varying signal and to avoid the possibility of a constant luminance value falling between quantizing levels of the PCM codec.

Finally, the increase of distortion and noise resulting from the addition of further codecs is at present somewhat unpredictable, but this is not likely to exceed that given by a law of root sum of squares.

2.2 *Digital systems using redundancy reduction techniques*

Simple PCM may be uneconomic for long distance digital transmission of television signals, so that redundancy reduction techniques may be used to achieve a reduced bit-rate. In general, this will result in additional impairment to the present ITS, but the form of this impairment will depend largely on the coding technique employed.

One form of impairment may be considered analogous to a slew-rate limitation, and it seems likely that such systems will result in significant distortion of the $2T$ sine-squared pulse and the composite chrominance-luminance pulse in the present ITS. Impairments of this type are likely to be particularly serious in systems employing differential coding, where the prediction incorporates information from the previous line, or in sub-Nyquist sampling, because at present each insertion test signal occurs on only one line in each field.

2.3 *Methods of overcoming digital transmission difficulties*

Since future coding systems will be designed primarily for efficient picture transmission, their test signal handling capability may have to be considered separately.

One method of overcoming the difficulties mentioned in the previous section would be to generate and insert a new ITS at the output of each digital circuit. The disadvantage of this, on a composite analogue-digital chain, would be that the new ITS would give no information on the performance of the digital section nor on the performance of any analogue circuits upstream thereof.

An alternative method would be to arrange for the coder to make a more detailed examination of the ITS lines only, and then to transmit the resultant extra data during the time slots occupied by other (unused) lines in the field-blanking interval. Yet another method would be to make a detailed examination of, say, every tenth ITS (the coder being arranged to ignore the ITS on the intervening nine fields) and then to transmit the resultant excess data shared out into the time slots allocated for ITS during the next nine fields.

A further possibility might be to reallocate the lines in the field-blanking period such that the same source ITS was repeated on several adjacent lines, thus minimizing problems that may be encountered in coding systems using two dimensional prediction, or sub-Nyquist sampling with comb filters.

It should also be borne in mind that the interface between the analogue and digital sections of a mixed circuit is by no means clear. In particular, an analogue-to-digital converter contains closely associated analogue and digital circuit elements, which may not be physically separable, and it may be desirable that distortions in the analogue parts (e.g. in the sample and hold circuit) are suitably monitored by the ITS.

* The apparent value will be a function of the level of superimposed sub-carrier on the test signal. The examples given are for the level specified by the CCIR for international testing, i.e. 40 IRE units peak-to-peak.

Further study is required to determine which, if any, of the above methods should be recommended for future systems.

3. ITS for digital sections

3.1 *Suitability of present ITS for testing digital systems*

It seems probable that the distortions most likely to occur on a digital transmission circuit are:

- (a) those due to sampling and filtering in the codec;
- (b) those due to quantization in the coder;
- (c) those due to bit-rate reduction in the coder;
- (d) those due to jitter on the bearer circuit;
- (e) those due to bit errors on the bearer circuit.

As has been pointed out in § 2 above, the type of digital system which will be economic for use on long television circuits is likely to introduce considerable distortion of an ITS of the present type, even when operated under optimum conditions. It follows that it may well be difficult to assess distortions of the types listed above on such a signal. In this connection it should be noted that causes (a), (b) and (c) above may result from equipment design deficiencies as well as from malfunctions.

3.2 *Possibility of a new ITS for testing digital systems which code the composite signal*

There seems to be general agreement that a saw-tooth waveform is very useful for testing composite PCM codecs and it seems reasonable that this should form the basis of a new ITS intended to test a complete digital system. The superimposition on this saw-tooth of a burst of colour sub-carrier might well provide useful additional information on the operation of the system, particularly if its amplitude was made such as to just fill the "headroom" specified for the system. Also, because a "white reference" is always found useful in practice, the saw-tooth could be flattened at the top.

Finally, it would be useful to ensure that the superimposed sub-carrier is not synchronized to the sampling signal since all quantized values of differential gain and phase can then be observed.

The Document [CCIR, 1974-78a] recommends such a saw-tooth with sub-carrier test signal for use in 525-line television circuits containing digital sections, in place of a test signal, which comprises a 5 riser staircase with superimposed sub-carrier. The precise form of a similar signal for use in 625-line television circuits requires further study.

The Document [CCIR, 1974-78b] suggests the use of a sine-squared pulse of half amplitude duration equal to half the active line period, with a superimposed colour sub-carrier as a test signal for the measurement of quantization noise [Krivosheev and Vilenchik, 1977]. Both the sub-carrier and the sine-squared pulse can be removed by filtering to permit the noise to be measured.

The design of any new signal should ideally be such that, if it has to be passed over a mixed analogue and digital chain, it should not be so distorted by the analogue sections, as to mask any distortions introduced by the digital sections.

It might be possible to prepare a limit graticule on the principle described in Recommendation 567, Part D, which could be used, either directly on the new ITS for the complete system, or, after passing through a high-pass filter to eliminate the low frequency component, to quantify some or all the types of codec distortions listed in § 3.1 above. It might also be possible to utilize this ITS in order to assess the magnitude of any bit errors on the bearer circuit. This subject requires further study.

4. ITS for the all-digital circuit

In the case of an all digital international circuit having digital rather than analogue input and output interfaces, i.e. a circuit without codecs, it may be desirable to include within the input bit stream a purely digital ITS in order to monitor the performance of the digital bearer, including code converters, digital multiplexers, etc.

An ITS of this type could also be employed on mixed chains if arrangements were made to decode it at each digital to analogue interface, and then to transmit the result as a number, say error rate, in the form of a data stream on a spare line in the on-going analogue video signal.

The need for such an all digital ITS, and the form which it should take, requires further study.

5. Conclusions

5.1 ITS for analogue sections

The present insertion test signals (including, where appropriate, a line for noise measurement), are widely used for the purposes of measurement, supervision and control of the performance of analogue circuits. Because these signals are internationally agreed, and because of the existence of a large amount of equipment related to this form, it would be desirable to retain the present signals to serve the purposes mentioned above on the analogue sections of mixed analogue and digital circuits. This will only be practicable if the digital sections of these circuits are designed not to impair test signals of the present form. While this is theoretically possible (e.g. by treating ITS in a manner different from that in which the picture signal is treated), it may in practice prove to be uneconomic to provide. In this case it may be necessary to make modifications to the existing ITS (e.g. by replacing the 5-riser staircase by a saw-tooth) that will enable them to pass through the digital sections while still providing useful information on the performance of the analogue sections.

5.2 ITS for digital sections using PCM for the composite signal

The present insertion test signals are unlikely to be satisfactory for the measurement, supervision and control of the performance of digital sections in mixed circuits. For these purposes therefore, it is probable that new analogue test signal elements, such as those mentioned in this Report, will have to be added. Alternatively, it may be possible to modify the existing ITS to serve both purposes. The interpretation to be placed on the results of measurements carried out on such "dual purpose" signals requires further study.

5.3 ITS for digital sections employing redundancy reduction techniques

Before deciding on the form which any modifications or additions to the current ITS should take, it will be necessary to consider the characteristics of the different types of digital coding which are likely to be used in the future, e.g. multi-dimensional predictive coding, separate component coding, etc.

5.4 ITS for the all-digital circuit

The best method of monitoring the performance of bearer circuits which are entirely digital from input to output may be to include a purely digital ITS in the input bit stream.

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SECTION CMTT C: TRANSMISSION STANDARDS AND PERFORMANCE OBJECTIVES FOR SOUND CHANNELS

Recommendations and Reports

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.

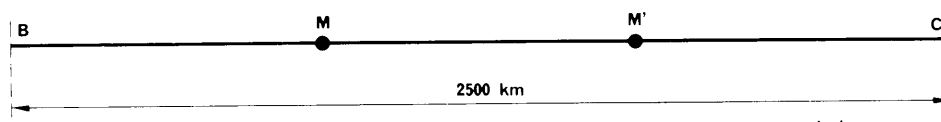


FIGURE 1 – *The hypothetical reference circuit for sound programme transmissions over a terrestrial system*

* 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.

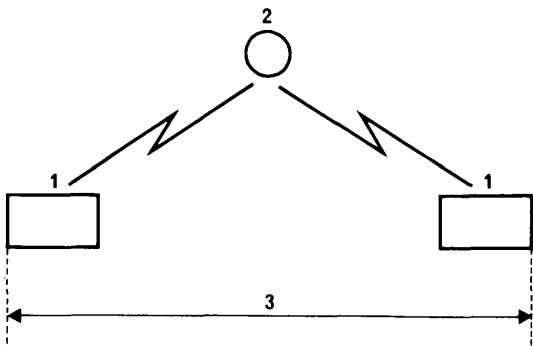


FIGURE 2 – *Hypothetical reference circuit for sound-programme transmissions over a system in the fixed-satellite service*

- 1: Earth station
- 2: Space station
- 3: Hypothetical reference circuit

ANNEX 1

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.

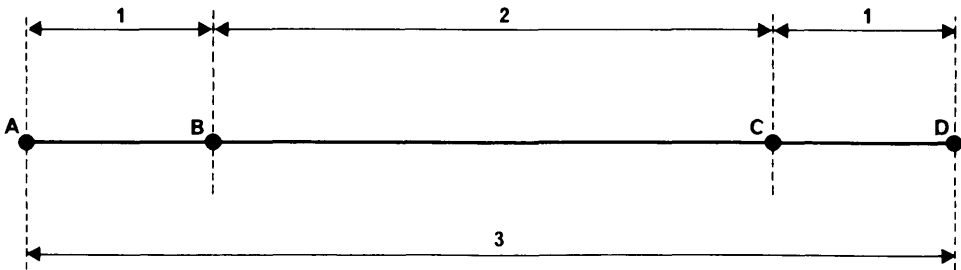


FIGURE 3 – *An international sound-programme connection*

- 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 Yellow 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 503-2 *

PERFORMANCE CHARACTERISTICS OF NARROW-BANDWIDTH SOUND-PROGRAMME CIRCUITS **

Circuits of medium quality for monophonic transmission (Question 17/CMTT)

(1974-1978-1982)

The CCIR

UNANIMOUSLY RECOMMENDS

the following characteristics for narrow-band monophonic sound-programme circuits.

1. Application

This Recommendation applies to new 5 kHz type and 7 kHz type analogue or digital circuits. The requirements shall be met by 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. — 6.4 kHz type narrow bandwidth sound programme circuits are still being used in some countries. For the performance characteristics refer to Recommendation 503-1 (Kyoto, 1978).

Note 2. — Additional characteristics for digital circuits are under study.

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-Z, balanced

The open-circuit output level shall not decrease more than 0.3 dB within the nominal frequency range, if the output is terminated by the specified test load.

Note. — 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 dBm0)	
Initial adjustment	0 $\pm$ 0.5 dB
Variation over 24 hours not to exceed	$\pm$ 0.5 dB

* This Recommendation 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 and permitted reactance need further study.

3. Performance of the HRC

3.1 Gain/frequency response

Reference frequency		1 kHz
5 kHz type	7 kHz type	
0.07 ... < 0.2 kHz	0.05 ... < 0.1 kHz	+1 to -3 dB
0.2 ... < 4 kHz	0.1 ... < 6.4 kHz	±1 dB
4 ... 5 kHz	6.4 ... 7 kHz	+1 to -3 dB

3.2 Group delay variation

Difference between the value of group delay at the following frequencies and the minimum value:

5 kHz type	7 kHz type	$\Delta\tau_g$ (ms)
—	0.05 kHz	≤ 80
0.07 kHz	—	≤ 60
—	0.1 kHz	≤ 20
5 kHz	—	≤ 15
—	6.4 kHz	≤ 5
—	7 kHz	≤ 10

Between the points defined above the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

3.3 Maximum weighted noise level

The measurement to be made with an instrument conforming to Recommendation 468.

	Idle channel	Programme-modulated (see Note 3)	
5 kHz-type circuits	-32	—	dBq0ps
7 kHz-type circuits	-44	-32	dBq0ps

Note 1. — The weighting network as per CCITT Recommendation P.53B (Green Book) is now obsolete, and further use of instruments conforming to that specification is deprecated.

Note 2. — Unweighted measurements are not considered to be useful.

Note 3. — Programme-modulated noise occurs only in companded systems. The only applicable CCITT Recommendation is Recommendation J.34, which refers to 7 kHz-type systems only. The recommended noise value is based on a test signal of 9 dBm0s at 60 Hz, which is suppressed by a highpass filter with the characteristics $f_0 \leq 400$ Hz and $a \geq 60$ dB at 60 Hz.

3.4 Single tone interference

Level of any individual tone $\leq (-73 - \psi)$ dBm0s

Where ψ is the weighting as per Recommendation 468 at the particular frequency.

3.5 Disturbing modulation by power supply

The ratio of the level of a sine-wave test signal applied to the sound-programme circuit to the highest level unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains rectifiers shall be ≥ 45 dB. The value required with higher frequency a.c. primary sources and when invertors and d.c.-d.c. convertors are used, has to be determined. (Study Programme 17F/CMTT; see also Report 495-1, Vol. XII, Geneva 1974.)

3.6 Non-linear distortion

3.6.1 As measured with fundamental signals at +9 dBm0 the total harmonic distortion (THD) shall be:

5 kHz-type circuits	7 kHz-type circuits	THD
0.07 ... < 0.1 kHz	0.05 ... < 0.1 kHz	≤ 2% (-25 dBm0)
0.1 ... 2.5 kHz	0.1 ... 3.5 kHz	≤ 1.4% (-28 dBm0)

Note. — If THD cannot be measured directly, compliance is considered to be fulfilled if a calculated value k meets the requirement:

$$k = \sqrt{k_2^2 + k_3^2}$$

where

k_2 : second order harmonic coefficient,

k_3 : third order harmonic distortion coefficient (see § 3.6.2 below).

3.6.2 With exciting signals at 0.8 kHz and 1.42 kHz, each at +3 dBm0, the third order difference tone at 0.18 kHz shall be $\leq 1.4\%$ (−34 dBm0).

3.7 *Error in reconstituted frequency*

The error in reconstituted audio frequency, when frequency translating equipment is used, shall be ≤ 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.

3.8 *Intelligible cross-talk*

3.8.1 The near-end and far-end cross-talk attenuations between two sound-programme circuits, or between a telephone circuit (disturbing circuit) and a sound-programme circuit (disturbed circuit) shall be at least 74 dB for the range of 0.5 to 3.2 kHz. Below 0.5 kHz and above 3.2 kHz, the limit is reducing at a rate of 6 dB per octave from an initial value of 74 dB at 0.5 kHz and 3.2 kHz respectively.

3.8.2 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. An explanation for the relation between the relative levels for sound-programme circuits and telephone circuits is given in the Annex A to Recommendation J.22.

Note 2. — The attention of administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or in the low frequency range (e.g. below about 100 kHz) in certain carrier systems on coaxial cables. 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 a telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk attenuation of 58 dB from the sound programme circuit into the telephone circuit is acceptable.

Note 4. — The attention of administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk limits between two sound-programme circuits simultaneously occupying the go and return channels respectively, of a carrier system (the most economical arrangement), because in those circumstances they occupy the same position in the line-frequency band (see Recommendation J.18).

Note 5. — The value indicated is based on the assumption that sine-wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6. — The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

3.9 *Amplitude linearity*

When a 1 kHz input signal is stepped from -6 dBm0 to $+6$ dBm0, the output level shall increase by 12 ± 0.5 dB.

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[1978-82]: CMTT/68 (OIRT).

RECOMMENDATION 504-2 *

PERFORMANCE CHARACTERISTICS OF 10 kHz-TYPE SOUND-PROGRAMME CIRCUITS **

(Question 17/CMTT)

(1974-1978-1982)

The CCIR,

CONSIDERING

- (a) that 10 kHz-type circuits are not quite adequate for modern high-quality transmission;
- (b) that such circuits are more than adequate for medium-quality transmission;
- (c) that they have been extensively used in the past;
- (d) that some administrations may for various reasons still wish to continue to provide circuits of this type;
- (e) the necessity to reduce the number of different standards in the interest of not forcing the industry to diversify production and other efforts unnecessarily;
- (f) that such reduction as in (e) would be expected to assist in keeping system costs low,

CONCLUDES

that the 10 kHz-type circuit is non-preferred for new systems in general, but may still be justified in certain cases (for example extension of already implemented systems),

UNANIMOUSLY RECOMMENDS

1. that whenever possible 7 kHz type or 15 kHz type sound programme circuits be considered for new systems rather than the 10 kHz type;
2. however, when it is necessary to choose the 10 kHz type for new systems, the circuit shall meet requirements set forth in Annexes I and II of this Recommendation.

ANNEX I

PERFORMANCE CHARACTERISTICS OF 10 kHz-TYPE SOUND-PROGRAMME CIRCUITS

1. **Application**

The requirements below apply to new 10 kHz-type analogue monophonic circuits and shall be met by 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.

* This Recommendation is a revision by the CMTT of CCITT Recommendation J.22, 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.

Note 1. — For real audio-frequency circuits it may not be possible to meet the requirement for intelligible cross-talk. When this occurs administrations concerned have to agree on an appropriate relaxed value.

Note 2. — For real circuits involving frequency division multiplex systems (FDM) it may not be possible to meet the requirements for intelligible cross-talk and noise. When this occurs administrations concerned have to agree on appropriate relaxed values. See Annex II for further information.

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-Z, balanced

The open-circuited 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.

Note. — The reactive part of the source impedance must be restricted to 100 Ω maximum (provisional value) within the nominal frequency range.

2.3 Levels

Input maximum programme level	+9 dBm0s
Insertion gain (1 kHz at –12 dBm0s)	
Initial adjustment	0 ± 0.5 dB
Variation during 24 h not to exceed	± 0.5 dB

3. Performance of the HRC

3.1 Gain/frequency response

Reference frequency	1 kHz
Frequency (Hz)	Relative level (dB)
50 ... < 100	+1.7 to –4.3
100 ... < 200	+1.7 to –2.6
200 ... < 6 000	± 1.7
6 000 ... < 8 000	+1.7 to –2.6
8 000 ... 10 000	+1.7 to –4.3

3.2 Group delay variation

Difference between the value of group delay at the following frequencies and the minimum value:

f (Hz)	$\Delta\tau_g$ (ms)
50	≤ 80
100	≤ 20
10 000	≤ 8

Between the points defined above the tolerance limit varies linearly in a linear-delay/logarithmic-frequency diagram.

3.3 Maximum weighted noise level

The measurement to be made with an instrument conforming to Recommendation 468:

–39 dBq0ps.

Note 1. — The weighting as per CCITT Recommendation P.53B (Green Book) is now obsolete, and further use of instruments conforming to that specification is deprecated.

Note 2. — Unweighted measurements are not considered to be meaningful.

Note 3. — For circuits on carrier systems, it is not always possible, in the absence of special precautions, to meet the limits recommended in this paragraph (see Annex II).

* The tolerance and permitted reactance need further study.

3.4 *Single tone interference*

Under study. (The subjective assessment of single tone interference on high quality circuits will be carried out by a method described in Report 623).

3.5 *Disturbing modulation by power supplies*

The ratio of the level of a sine wave test signal applied to the sound programme circuit to the highest level unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains rectifiers shall be ≥ 45 dB. The value required with higher frequency a.c. primary sources and when invertors and d.c.-d.c. convertors are used, has to be determined. (Study Programme 17F/CMTT; see also Report 495-1, Volume XII, Geneva, 1974.)

3.6 *Non-linear distortion*

As measured with fundamental signals at + 9 dBm0 the total harmonic distortion (THD) shall be:

f (kHz)	THD
0.05 ... < 0.01	$\leq 3\%$ (– 21 dBm0)
0.01 ... 10	$\leq 2\%$ (– 25 dBm0)

Note. — Precautions should be taken in the measurement of harmonic distortion on circuits equipped with pre-emphasis networks. (See CCITT Recommendation N.21.)

3.7 *Error in reconstituted frequency*

The error in reconstituted audio frequency, when frequency translating equipment is used, shall be ≤ 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.

3.8 *Intelligible cross-talk*

3.8.1 The near-end and far-end cross-talk attenuations between two sound-programme circuits, or between a telephone circuit (disturbing circuit) and a sound-programme circuit (disturbed circuit) shall be at least 74 dB.

3.8.2 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. An explanation for the relation between the relative levels for sound-programme circuits and telephone circuits is given in Annex II.

Note 2. — The attention of administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or in the low frequency range (e.g. below about 100 kHz) in certain carrier systems on coaxial cables. 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 a telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk attenuation of 58 dB from the sound-programme circuit into the telephone circuit is acceptable.

Note 4. — The attention of administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk limits between two sound-programme circuits simultaneously occupying the go and return channels respectively, of a carrier system (the most economical arrangement), because in those circumstances they occupy the same position in the line-frequency band. (See Recommendation J.18).

Note 5. — The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6. — The effect of cross-talk from a sound programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

3.9 *Amplitude linearity*

When an 1 kHz input signal is stepped from -6 dBm0 to $+6$ dBm0, the output level shall increase by 12 ± 0.5 dB.

ANNEX II

VALUES OF NOISE EXPECTED IN PRACTICE ON 2500 KM CIRCUITS

Estimated noise power levels

The following table shows the noise values arising when sound-programme circuits (using pre-emphasis and de-emphasis in accordance with CCITT Recommendation J.17) are set up in place of three telephone channels, each of which conforms to the general noise objectives given in CCITT Recommendation G.222. The assumptions made for the purpose of the noise calculations are shown at the end of this Annex.

TABLE I

	One-minute mean value	
	for not more than 20% of a month	for not more than 0.1% of a month
Noise power level weighted with the network of CCITT Rec. P.53 B (Green Book)(1)	-44.5 dBm0ps	-37.5 dBm0ps

(1) This network is now obsolete.

Note. — The increased noise level shown as occurring for less than 0.1% of a month, applies when the carrier circuit is established over a radio-relay system.

When 10 kHz-type sound-programme circuits which include emphasis and de-emphasis networks, are set up on a carrier system it is recommended that, for reasons of overload, the relative level at 1000 Hz on such a circuit at a zero-relative level point, (deduced from the level diagram of telephone circuits set up on the same 12-circuit group), should lie between a maximum of -1.5 dB and a minimum of -4.5 dB.

The level of -1.5 dB could be considered as normal. A further 3 dB adjustment to permit a decrease down to -4.5 dB should, however, be included to cover the case of exceptional overloading if operational experience shows that in fact this is necessary.

Note. — Certain problems connected with the use of pre-emphasis on carrier systems have not yet been satisfactorily resolved. These are:

- the limitation of the level of testing tones, which is of concern to CCITT Study Group IV;
- the effect of pre-emphasis on the harmonic distortion requirements which the programme circuit should meet at high frequencies. *

* Measurements of harmonic distortion on programme circuits having pre-emphasis must be treated with reserve. This point is being studied by the CCITT.

Use of companders

Provided the compressor and the expander are of the same make, it is possible to obtain overall transmission characteristics as regards noise, which conform to the CCITT Recommendations for the 2500 km hypothetical reference circuit, without introducing other factors that might impair transmission performance. The CCITT is now examining recommendations on the compressor and the expander, considered separately, so as to achieve the same result.

Assumptions and conventional terms

The expression dBm0ps is used to indicate noise power levels in a sound-programme circuit which are psophometrically weighted and measured in decibels relative to 1 mW at a point of zero-relative level in that circuit. The CCITT practice is to quote noise level for sound-programme circuits relative to "peak programme" or "maximum voltage" which is defined as a voltage of 2.2 V r.m.s. (measured at the terminals of an impedance of 600 Ω) at a point of zero-relative level, i.e. 9 dB above telephone test level, and the signal-to-noise ratio objective of 57 dB is thus equivalent to a noise power level of -48 dBm0ps.

The value for not more than 20% of a month was calculated for 10 kHz-type circuits on the following assumptions:

- Noise on one telephone channel (including the multiplex equipment) according to CCITT Recommendation G.222, weighted for telephony:	- 50 dBm0ps
- Bandwidth correction for 10 kHz:	+ 5 dB
- Suppression of weighting for telephony (in the case of a uniform-spectrum noise):	+ 2.5 dB
- Improvement due to pre-emphasis * (see CCITT Recommendation J.17):	- 9 dB
- Effect of the relative level shifted by -1.5 dB at 800 Hz:	+ 1.5 dB
- Weighting for sound-programme transmissions	+ 5.5 dB
Total	- 44.5 dBm0ps

The value for not more than 0.1% of a month was calculated on the basis of the noise variations to be expected on a radio-relay link used mainly for providing telephone circuits and conforming with CCITT Recommendation G.222.

RECOMMENDATION 505-2 **

CHARACTERISTICS OF 15 kHz-TYPE SOUND-PROGRAMME CIRCUITS ***

Circuits for high quality monophonic and stereophonic transmissions

(Question 17/CMTT)

(1974-1978-1982)

The CCIR

UNANIMOUSLY RECOMMENDS

that, taking account of the definition in § 1 below, high quality monophonic and stereophonic sound-programme transmissions should satisfy the requirements laid down in § 2 and 3 below.

1. Definition

When the hypothetical reference circuit (HRC) defined in Recommendation 502, (equivalent to CCITT Recommendation J.11) is composed of three "sound-programme carrier sections" the requirements indicated below shall be met.

* Set to have zero loss at 800 Hz.

** This Recommendation, which is a revision by the CMTT of CCITT Recommendation J.21, 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.

For estimation of the performance of circuits shorter or longer than the HRC, see Recommendation 605 and Study Programme 17D/CMTT.

Note. — For further work, Report 496 may be consulted. This Report also draws attention to certain differences between the above Recommendation and one drawn up by the OIRT.

2. Requirements at audio interconnection points

2.1 *Measurement of characteristics*

When making measurements of the characteristics of a circuit, these should be made with the output terminated with a 600 Ω non-reactive load.

2.2 *Impedance and matching conditions*

The audio-frequency input impedance should be 600 Ω balanced; the tolerance on this value is a matter for further study.

It is provisionally recommended that the output impedance be balanced with respect to earth and be so low that the output level in the nominal transmission range does not decrease by more than 0.3 dB if the open-circuit output is loaded with 600 Ω . This output impedance is intended for connection to a nominal load impedance of 600 Ω .

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.

For amplifiers which are intended for direct connection to audio programme lines, the reactive part of the output impedance should be restricted. A maximum value of 100 ohms for the series reactance part of the output impedance at frequencies in the transmitted range is provisionally recommended.

2.3 *Relative level*

The relative level on a sound-programme circuit at the audio-frequency amplifier output should be fixed at +6 dBrs.

3. Performance of the hypothetical reference circuit for 15 kHz-type sound-programme circuits

The values given correspond to circuits operating with analogue techniques and are expected to be met on such transmission systems. Parameters concerning digital techniques are under study (see § 4 below).

3.1 *Parameters for monophonic programme transmission*

3.1.1 Nominal bandwidth: 0.04 to 15 kHz.

3.1.2 Insertion gain at 0.8 or 1 kHz: this parameter should be measured at a sending level equivalent to -12 dBm0 as specified by the CCITT for setting up sound-programme circuits.

3.1.2.1 Adjustment error: not to fall outside the range ± 0.5 dB.

3.1.2.2 Variation during 24 hours: not to exceed ± 0.5 dB.

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organizations to insert additional trimming attenuators.

3.1.3 The gain/frequency response referred to 0.8 or 1 kHz should comply with the limits indicated:

0.04 to 0.125 kHz:	+0.5 to -2.0 dB
0.125 to 10 kHz:	+0.5 to -0.5 dB
10 to 14 kHz:	+0.5 to -2.0 dB
14 to 15 kHz:	+0.5 to -3.0 dB.

For the combined effect of three modulator and demodulator equipments, a tolerance of ± 0.5 dB from 0.125 to 10 kHz is considered the closest that can be met by equipments in practice. If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

This response should be measured using a test level of -12 dBm0s.

3.1.4 The difference between group delay at the given frequency and the minimum value group delay should not exceed the following limits:

0.04 kHz:	55 ms
0.075 kHz:	24 ms
14 kHz:	8 ms
15 kHz:	12 ms.

3.1.5 *Maximum weighted noise levels*

These parameters are defined in terms of a weighting network and a quasi-peak measuring instrument in accordance with Recommendation 468.

When using radio-relay systems, the values given should not be exceeded for more than 20% of any month. For 1% and 0.1% of any month, limits 4 dB higher and 12 dB, respectively, seem to be acceptable.

3.1.5.1 *Idle channel noise*

– 42 dBq0ps.

3.1.5.2 *Programme-modulated noise (expanded noise)*

– 30 dBq0ps.

This noise value can 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 ($f_0 \leq 400$ Hz, $a \geq 60$ dB/60 Hz) before the measuring set.

Programme-modulated noise may only occur on sound programme circuits which are equipped with compandors (for example types of circuits corresponding to CCITT Recommendation J.31).

Report 493 indicates that if a compandor is used, then with some programme material an improved signal-to-noise ratio is necessary to avoid objectionable effects.

3.1.6 *Single-tone interference*

The single-tone interference, measured selectively, should not exceed $(-73 - \Psi)$ dBm0s, in which Ψ is the correction for the frequency being measured, given by the weighting characteristics in Recommendation 468.

For sound-programme transmissions over carrier systems, occurrence of carrier leaks can be expected. For this reason, stop filters may be provided in the carrier frequency path which can be switched in, if required, to suppress the tones otherwise audible in the upper frequency range from 8 to 15 kHz. For a hypothetical reference circuit, a 3 dB bandwidth of less than 3% for stop filters, referred to the mid-frequency, is recommended. The use of stop filters influencing frequencies below 8 kHz should be avoided.

3.1.7 *Disturbing modulation by power supply*

The ratio of a sine-wave measuring signal applied to the sound-programme circuit to the highest level unwanted side-component due to modulation caused by interference from conventional a.c. (50/60 Hz) line power supply sources should be greater than 45 dB (in accordance with Recommendation 474). The value for higher frequencies has to be determined, see Study Programme 17F/CMTT.

3.1.8 *Non-linear distortion*

There are certain difficulties in giving a general recommendation on non-linearity, due to restrictions imposed by CCITT on the levels and durations of test tones, see especially CCITT Recommendations N.21 and N.23. Pending progress with other test methods, the following tests are recommended.

3.1.8.1 Harmonic distortion factors measured selectively with single-tone test signals:

- at +9 dBm0s in the frequency range lower than 4 kHz and
- at +6 dBm0s in the frequency range higher than 4 kHz

(including 4 kHz) should not exceed the following limits:

TABLE I

Frequency of test-tone (kHz)	Total harmonic distortion (%)	Second harmonic and third harmonic measured selectively (%)
0.04 to 0.125	1	0.7
0.125 to 7.5	0.5	0.35

The duration for which a single-tone is to be transmitted at this level should be restricted in accordance with the appropriate CCITT Recommendations of the N series.

3.1.8.2 The difference-tone * factors selectively measured with double-tone test signals each:

- at +3 dBm0s in the frequency range lower than 4 kHz and
- at 0 dBm0s in the frequency range from 4 kHz and above

shall not exceed the following limits:

3.1.8.2.1 Frequencies 0.8 and 1.42 kHz corresponding to those prescribed in CCITT Recommendation O.31, 3rd order difference-tone measured at 0.18 kHz: 0.5%.

3.1.8.2.2 Frequencies 5.6 and 7.2 kHz for a 2nd order difference-tone measured at 1.6 kHz: 0.5%.

3.1.8.2.3 Frequencies 4.2 and 6.8 kHz for a 3rd order difference-tone measured at 1.6 kHz: 0.5%.

The measurements under § 3.1.8.2.2 and 3.1.8.2.3 are intended for baseband transmissions on physical circuits only and on modulation equipment in local loops.

3.1.8.3 Distortion products measured by shaped noise: under study. See Report 640.

3.1.9 *Error in reconstituted frequency*

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.

3.1.10 *Intelligible cross-talk ratio*

3.1.10.1 The intelligible cross-talk ratio from other sound-programme circuits or from a telephone circuit into a sound-programme circuit should be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and should not be less than the following values:

0.04 kHz:	50 dB
0.04 to 0.5 kHz:	oblique straight-line segment on linear-decibel and logarithmic-frequency scales
0.5 to 5 kHz:	74 dB
5 to 15 kHz:	oblique straight-line segment on linear-decibel and logarithmic-frequency scales
15 kHz:	60 dB.

3.1.10.2 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. An explanation for the relation between the relative levels for sound-programme circuits and telephone circuits is given in the Annex to Recommendation J.22.

Note 2. — The attention of administrations is drawn to the fact that it is in some cases difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or in the low frequency range (e.g. below about 100 kHz) in certain carrier systems on coaxial cables. If sub-standard is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

* Attention is drawn to the fact that in transmission systems using compandors, a 3rd order difference-tone may occur which exceeds the specified limit of 0.5%. This may occur when the difference between the two fundamental frequencies is less than 200 Hz. Thus, the components due to 3rd order distortion will have frequencies which correspond to the difference between the two test frequencies. However, in these cases the subjective masking is such that a distortion up to 2% is acceptable.

Note 3. — When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

Note 4. — The attention of administrations is drawn to the fact that, because of cross-talk which may occur in terminal modulating and line equipment, special precautions may have to be taken to meet the above cross-talk limits between two sound-programme circuits, simultaneously occupying the go and return channels respectively, of a carrier system (the most economical arrangement) because in those circumstances they occupy the same position in the line-frequency band (see Recommendation J.18).

Note 5. — The value indicated is based on the assumption that sine wave test signals are used. The use of the test signal as described in Recommendation J.19 is under study.

Note 6. — The effect of cross-talk from a sound-programme circuit into a telephone circuit is not a question of secrecy, but rather of subjective disturbance by an interfering signal whose character is noticeably different from random noise or babble.

The frequency offset adopted for some sound-programme equipment allows a reduction of cross-talk from a telephone circuit into a sound-programme circuit. However, in the reverse direction, this reduction of cross-talk remains only for speech material but is practically ineffective for music material.

3.1.11 *Error in amplitude/amplitude response*

When the level of a 0.8 or 1 kHz test signal is changed from +6 to −6 dBm0s or vice versa, the level difference at the receiving end should not lie outside the range 12 ± 0.5 dB. This level change of the test signal corresponds to that prescribed in CCITT Recommendation O.31.

3.2 *Additional parameters for stereophonic programme transmission*

3.2.1 The difference in gain between A and B channels should not exceed the following values:

0.04 to 0.125 kHz:	1.5 dB
0.125 to 10 kHz:	0.8 dB
10 to 14 kHz:	1.5 dB
14 to 15 kHz:	3 dB.

3.2.2 The phase difference between the A and B channels should not exceed the following values:

0.04 kHz:	30°
0.04 to 0.2 kHz:	oblique straight-line segment on linear-degree and logarithmic-frequency scales
0.2 to 4 kHz:	15°
4 to 14 kHz:	oblique straight-line segment on linear-degree and logarithmic-frequency scales
14 kHz:	30°
15 kHz:	40°.

3.2.3 The cross-talk ratio between the A and B channels should 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 should 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 II.

TABLE II

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

4. Transmission performance of the hypothetical reference circuit for 15 kHz-type sound-programme circuits with particular reference to digital methods of transmission

This section will deal with parameters for digital systems. Report 647 and Study Programme 18A/CMTT refer.

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.

REPORT 496-3 *

CIRCUITS FOR HIGH-QUALITY MONOPHONIC AND STEREOPHONIC TRANSMISSIONS

(Question 17/CMTT and Study Programme 19B/CMTT)

(1970-1974-1978-1982)

1. Characteristics of circuits for high-quality monophonic and stereophonic transmissions over long distances

1.1 The main documents used to establish the quality characteristics of circuits for long-distance monophonic and stereophonic transmissions are listed in the bibliography.

- With regard to the characteristics of these circuits as a whole, reference should be made to Recommendation 505 which specifies all the values to be respected.
- [CCIR, 1970-74] which reproduces a similar recommendation drawn up by the OIRT Technical Commission specifying the characteristics of the hypothetical reference circuit, contains some differences compared with the CMTT Recommendation which have been included in a recapitulatory table (Table I).

The most important differences between the two series of values relate to the unweighted noise level, the phase difference between channels *A* and *B* and intelligible cross-talk in the case of a monophonic transmission.

It should also be noted that the value given for cross-talk ratio of a stereophonic transmission circuit excludes the separate use of each of the two circuits for monophonic transmission.

The effect of cross-talk introduced by the stereophonic circuit on overall cross-talk has been studied in Italy [CCIR, 1966-69]. From the curves in Fig. 1 taken from this document the admissible cross-talk at the input of a transmitter-to-receiver chain with specific amplitude and phase characteristics can be determined.

1.2 With regard to the mode of transmission of stereophonic signals, the CMTT, in view of the results of service trials carried out by certain countries, considers that long-distance stereophonic transmission should be effected exclusively by either of the following methods:

- separate transmission of signals *A* and *B*;
- transmission of signals *M* and *S* in the form of the stereophonic multiplex signal modulating the broadcasting transmitter.

1.3 Attention must be drawn to the need to specify additional parameters when pulse-code modulation systems are used to convey sound-programme signals. The CMTT is aware that the quality tolerances of 15 kHz-type sound-programme circuits provided by means of pulse-code modulation may be better than those given in Recommendation 505 (see also Report 649). A study of the additional parameters to be specified and the appropriate quality tolerances will be made when the CCIR Study Groups concerned have indicated the likely range of fundamental characteristics for analogue to digital converters for programme circuits, e.g. sampling rate, number of digits per sample, type of coding, companding characteristics. As this range of fundamental characteristics will be affected by decisions concerning basic PCM systems for telephony and the higher order hierarchy for higher speed systems, the CMTT considers that due account should be taken of the requirements of programme circuits when these matters are decided.

* For the definition of absolute power, relative power and noise levels, see Recommendation 574.

TABLE 1 – Limiting values of parameters of high-quality circuits for monophonic and stereophonic transmission

Parameter	O.I.R.T. Recommendation No. 67	Recommendation 505
1. Level stability (dB)	± 1.0 (long term)	± 0.5 (daily)
2. Amplitude/frequency response (dB): 10 to 14 kHz 14 to 15 kHz	+ 0.5 to – 2.0	+ 0.5 to – 2.0 + 0.5 to – 3.0
3. Group delay time difference (ms): 0.040 kHz 15 kHz		
4. Noise level with new weighting filter (dBm0s): (1) (2) unweighted (1) (2) (3) impulsive noise (4) (dBm0s)	Not yet defined – 46	– 47 – 41
5. Single tone interference measured selectively (dBm0s)(5)	Not taken into consideration	– 73 – Δps (5)
6. Overall harmonic factor (%):(6) 0.125 to 7.5 kHz 0.125 to 5.0 kHz	0.5	0.5
7. Cross-talk ratio <i>A</i> and <i>B</i> Single overall measurement using a weighted noise signal(7)	Not taken into consideration	Not specified
8. Intelligible cross-talk attenuation (dB): 0.04 kHz 0.04 to 0.5 kHz 0.5 to 5 kHz 5 to 15 kHz 15 kHz	78	50 oblique segments 74 oblique segments 60
9. Amplitude/amplitude response (dB):(8) 1 kHz 0.04 to 15 kHz		
10. Difference in gain <i>A/B</i> channels (dB): 0.04 to 0.125 kHz 0.125 to 10 kHz 10 to 15 kHz 10 to 14 kHz 14 to 15 kHz	2 1 2	1.5 0.8 1.5 3
11. Phase difference <i>A/B</i> (degrees): 0.04 kHz 0.2 to 4 kHz 14 kHz 15 kHz	20 10 20	30 15 30 40

Notes to Table 1:

(1) The value given is to be met when the r.m.s. value is measured with the new weighting network mentioned in Recommendation 468. This should be modified if the quasi-peak indicator is used.

(2) The noise level N_n expressed in dBm0s can be derived from the indicated noise level N_{ni} expressed in dBm and the relative level N_r expressed in dBr at the measurement point by the formula:

$$N_n = N_{ni} - N_r$$

the ratio ΔN_n between the maximum signal level + 9 dBm0 and the noise level N_n can be derived as follows:

$$\Delta N_n = 9 - N_n$$

(3) The unweighted measurement is made with the band limitation (31.5 Hz to 16 kHz) given in Annex II to Recommendation 468.

- (4) The method of measurement and the tolerance will have to be studied. CISPR Publication No. 1 contains information which may be useful in this respect.
- (5) Δps : correction for the frequency being measured in accordance with the new weighting characteristic described in § 2.2 of this Report. The value measured (selectively) will need correction in accordance with the new weighting characteristic. There is doubt as to the use of the weighting characteristic for this purpose and this method of expressing the requirement is provisional prior to receiving the results of subjective tests.
- To obtain the given value when, for example, there is interference from a carrier frequency, a rejection filter may need to be inserted. A study should be made of the parameters to be specified for this filter so that its adverse effects on parameters Nos. 3 and 4 are reduced to a minimum.
- The following two studies on this subject have been published:
- BÜCKLEIN, R. [1962] Audibility of irregularities in the frequency response in acoustical transmission (in German). *Frequenz*, Vol. 16, 3, 103-108.
- FLOHRER, W. [1968] The deterioration of the naturalness of speech due to gaps in the transmission frequency band (in German). *Frequenz*, Vol. 22, 6, 175-178.
- (6) The CMTT considers it necessary to measure non-linear distortion for frequencies above 5 kHz. The method of measurement is under study (see Report 640-1 (Kyoto, 1978)).
- (7) Certain countries have elaborated for broadcasting purposes a method of measuring cross-talk using weighted noise (see Report 497). A study should be made to see whether this method could be applied to links.
- (8) The purpose of this parameter is to take into account the effect on the signal of the use of compandors in links. Methods of measurement will have to be studied.

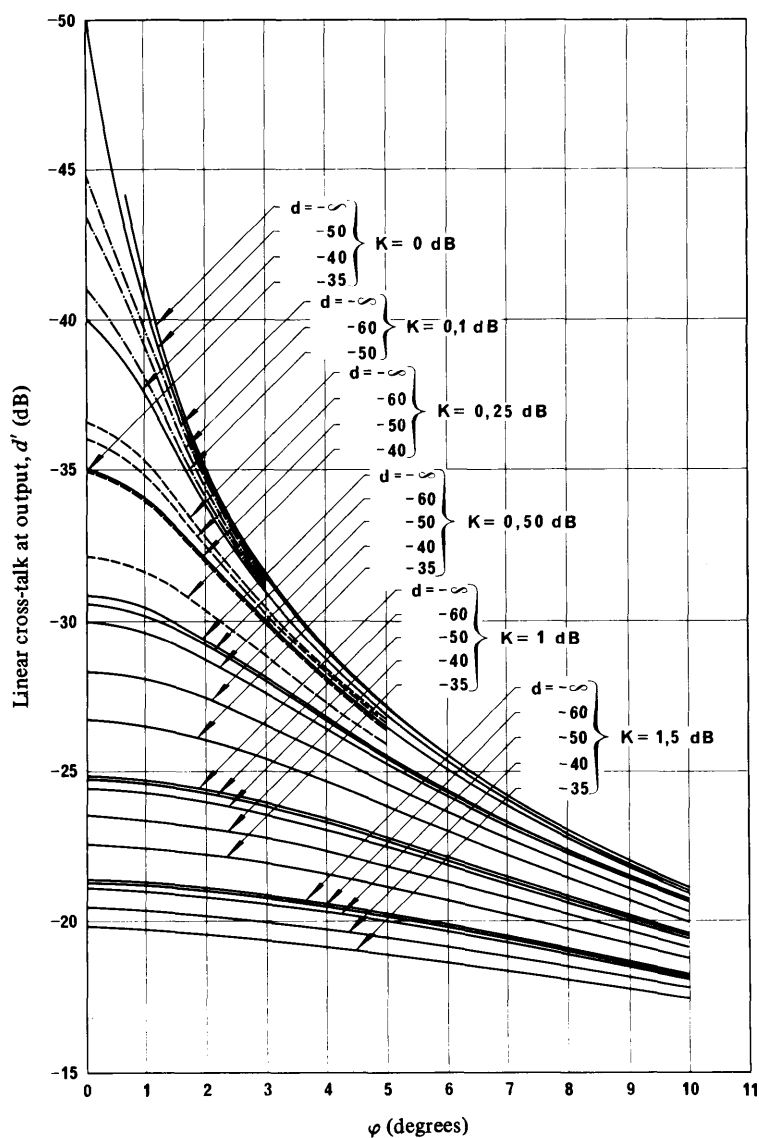


FIGURE 1

d : cross-talk between A and B signal channels at the input M and S systems
 K : amplitude difference between M and S signals due to transmission system
 φ : phase difference between M and S signals due to transmission system
 d' : cross-talk between A and B signal channels at the output of M and S systems

2. Listening tests and measuring methods for high quality sound-programme circuits

2.1 *Subjective effects of cross-talk and noise*

2.1.1 *Subjective tests*

The CMTT believes that it is necessary to carry out subjective listening tests within the CCIR as an essential part of the studies relating to new methods employed in the transmission of sound-programmes over long distances, and in the setting of standards for such methods.

The general method proposed for subjective listening tests is given in Report 623.

2.1.2 *Test noises for subjective tests*

Certain types of noise and cross-talk are commonly encountered on analogue transmission systems and it may be of assistance in carrying out subjective noise tests to utilize tape recordings of sample noises which have been made available by the CCIR, (upon application). The test noises available upon these tapes are as shown below:

- (a) hum (50 Hz + harmonics);
- (b) cross-talk of vision on sound channel on a multi-channel FDM system. (Vision signal supplied from a generator to provide a pattern of horizontal and vertical lines);
- (c) direct current signal (clicks) from a teleprinter (see Note);
- (d) unintelligible cross-talk (speech);
- (e) intelligible cross-talk (speech in German) (see Note);
- (f) noise from tape recording;
- (g) empty studio noise;
- (h) 3 kHz sine-wave tone (steady);
- (j) 10 kHz sine-wave tone (steady);
- (k) white noise;
- (l) triangular noise;
- (m) voice frequency dialling;
- (n) intelligible cross-talk (music) (see Note);
- (o) data transmission signals (see Note).

Note. — The cross-talk interference deriving from the sources indicated is obtained by simulating the cross-talk path using a network to provide a 6 dB per octave slope giving a rising power with rising frequency.

Noise from exchange switching operations will be provided if a suitable sample can be obtained.

Referring to noise item (e), this should be replaced by an item of subjectively the same loudness in the appropriate language and should have a 6 dB per octave slope on the frequency characteristics. The views are sought of CCIR Study Group 10 and CCITT Study Group IV on the list of noises to be used.

2.2 *The measurement of noise on sound-programme systems*

The CMTT has recommended that noise measurements in sound-programme systems should be made with an instrument in accordance with Recommendation 468. In recognition of the fact that using this instrument readings different from those obtained by measurement using the earlier instrument given in CCITT Recommendation P.53 will be obtained, and also that for a time organizations may wish to use the new weighting characteristic but might not have available the instrument portion defined in Recommendation 468; the following information is given concerning the effects of the noise weighting characteristic when used in conjunction with a true r.m.s. indicating instrument.

2.2.1 *Comparison of measurements (see Note) of Gaussian noise made with the new weighting network and with the previous one*

The CMTT has decided to apply the new weighting characteristic for measurement of audio-frequency noise defined by Recommendation 468 for measurement of noise on sound-programme circuits. If it is assumed that only two typical kinds of noise can occur on analogue transmission paths of modern sound-programme circuits, namely white and triangular noise, it is possible to calculate corresponding

weighting factors by integration after summing the spectral noise distribution and the weighting characteristic. It is useful for many practical cases to compare noise values resulting from the new weighting characteristic with those measured with the old weighting characteristic (defined in CCITT Recommendation P.53). It is also of practical interest to take into account the effects that the pre/de-emphasis characteristic, given by the CCITT in Recommendation J.17 has upon the measured noise values.

Note. — It should be noted that for the purposes of this comparison a true r.m.s. measuring instrument is assumed. An exact correspondence with the values given in Table II may not be obtained with other types of instrument.

Table II gives the calculated results of making weighted and unweighted noise measurements using the old and new weighting networks upon sound-programme circuits of 5 kHz, 10 kHz and 15 kHz bandwidth established over bearer circuits which contribute white noise or triangular noise, with and without pre/de-emphasis. Derived from these results and also tabulated are:

- the “weighting factors” for the two weighting networks — defined as the difference in readings between the weighted and unweighted noise measurement, other conditions being unchanged;
- the “reading conversion factor” for the two weighting networks — defined as the difference between the readings obtained with the new network and the old network, other conditions being unchanged;
- the “improvement in signal/noise ratio effected by the use of pre/de-emphasis” — defined here as the difference in weighted measurement due to the introduction of pre/de-emphasis under conditions designated by (b) in the notes to Table II.

Notes to the table explain conditions which apply for each case considered.

For convenience only, reference conditions are used in which the level of unweighted noise without pre/de-emphasis is assumed to be 0 dBm at a 0 dBr point at the output of the sound-programme circuit irrespective of the bandwidth and the type of noise. This enables readings of level to be shown as dBm, whereas “factors” which give the relationship between readings are shown as dB.

2.2.2 *Pre/de-emphasis used on sound-programme circuits*

The CCITT recommends (in Recommendation J.17) that only a single pre-emphasis curve should be used whenever pre-emphasis is applied to sound-programme circuits in group links. For convenience the loss/frequency functions defining the pre/de-emphasis curves are given here, (for fuller information see CCITT Recommendation J.17).

The shape of the pre-emphasis curve can be calculated from the following formula:

insertion loss between nominal impedances =

$$10 \log \frac{75 + \left(\frac{\omega}{3000}\right)^2}{1 + \left(\frac{\omega}{3000}\right)^2} \quad \text{dB}$$

where ω is the angular frequency in radians per second corresponding to frequency f .

The de-emphasis network should have a complementary curve.

TABLE II

Spectral distribution of noise from system	Upper frequency limit of transmission of the circuit under conside- ration (kHz)	Level of noise measured at a 0 dBr point at the output of the circuit					Reading conversion factors (difference between readings obtained with new network and old network) {(5) – (7)} (dB)
		Unweighted measurement (dBm)	Weighted measurements				
			New CCIR weighting network		Old CCITT weighting network		
				Weighting factor of network {(4) – (3)} (dB)		Weighting factor of network {(6) – (3)} (dB)	
(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)
White noise	5	(a) 0 (b) – 1.3	+ 7.9 dBm + 1.0 dBm	+ 7.9 + 2.3	+ 6.0 dBm – 0.3 dBm	+ 6.0 + 1.0	+ 1.9 + 1.3
			(c) 6.9 dB		(c) 6.3 dB		
	10	(a) 0 (b) – 3.8	+ 9.9 dBm + 1.0 dBm	+ 9.9 + 4.8	+ 6.0 dBm – 1.8 dBm	+ 6.0 + 2.0	+ 3.9 + 2.8
			(c) 8.9 dB		(c) 7.8 dB		
	15	(a) 0 (b) – 5.2	+ 8.5 dBm – 0.5 dBm	+ 8.5 + 4.7	+ 4.2 dBm – 3.6 dBm	+ 4.2 + 1.6	+ 4.3 + 3.1
			(c) 9.0 dB		(c) 7.8 dB		
Triangular noise	5	(a) 0 (b) – 7.2	+ 10.1 dBm + 1.9 dBm	+ 10.1 + 9.1	+ 7.8 dBm – 0.2 dBm	+ 7.8 + 7.0	+ 2.3 + 2.1
			(c) 8.2 dB		(c) 8.0 dB		
	10	(a) 0 (b) – 9.9	+ 10.9 dBm + 0.7 dBm	+ 10.9 + 10.6	+ 5.3 dBm – 4.3 dBm	+ 5.3 + 5.6	+ 5.6 + 5.0
			(c) 10.2 dB		(c) 9.6 dB		
	15	(a) 0 (b) – 10.8	+ 6.6 dBm – 3.7 dBm	+ 6.6 + 7.1	+ 0.1 dBm – 9.6 dBm	+ 0.1 + 1.2	+ 6.5 + 5.9
			(c) 10.3 dB		(c) 9.7 dB		

Notes. - Conditions applying to each case considered in Column (3):

(a) No emphasis and de-emphasis employed. (This applies to any value shown in the rows marked (a));

Note. - The level of 0 dBm is chosen purely as a reference against which other values relating to the same bandwidth of circuit and type of noise can be displayed. It does not represent the actual level expected to be measured on a real-sound programme circuit, nor does it denote equality in the actual noise level measured on the circuits shown as having different bandwidths of different types of noise.

(b) De-emphasis employed in accordance with CCITT Recommendation J.17 and set to have 1.5 dB gain at 800 Hz (1.5/800). Since emphasis has the complementary characteristic, the output of the de-emphasis network is still a 0 dBr point. (This applies to any value shown in the rows marked (b));

(c) The individual values marked (c) show the improvement in weighted signal noise ratio obtained by the use of the emphasis and de-emphasis described in note (b).

The pre-emphasis curve calculated from this formula passes through the following points:

TABLE III

f (kHz)	Insertion-loss (dB)
0	18.75
0.05	18.70
0.2	18.06
0.4	16.48
0.8	13.10
2	6.98
4	3.10
6.4	1.49
8	1.01
10	0.68
∞	0

The measured pre-emphasis and de-emphasis curves should not depart by more than ± 0.25 dB from the theoretical curves when the measured levels at 800 Hz are made to coincide with the theoretical levels.

Note. — The formula given above defines only the “insertion-loss/frequency” characteristic. The level at which the modulated programme signal is given to the group link is different for the various types of sound-programme equipments.

2.3 *The use of limiters on sound-programme circuits*

The use of limiters at the interface between the broadcasting studio and the sound-programme circuit is not at present covered by a Recommendation, but a proposed text for a possible Recommendation which might be applicable to both analogue and digital transmission is annexed to Report 647. Further study is required and contributions on this subject are requested from administrations.

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RECOMMENDATION 571-1 *

A CONVENTIONAL TEST SIGNAL SIMULATING SOUND-PROGRAMME SIGNALS
FOR MEASURING INTERFERENCE IN OTHER CHANNELS **

(Study Programme 19D/CMTT)

(1978-1982)

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.

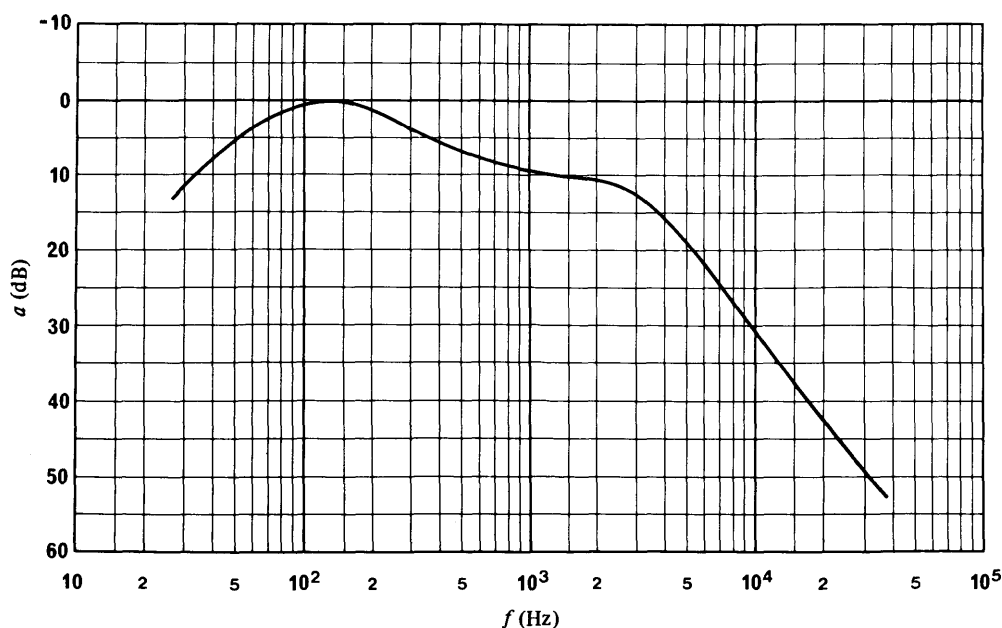


FIGURE 1 — Insertion-loss/frequency

* 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.

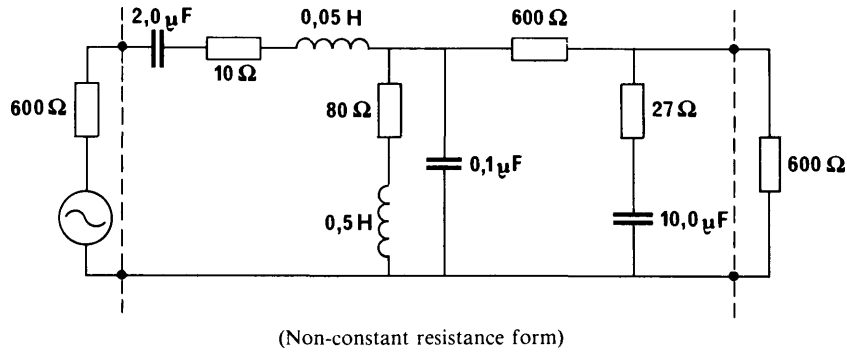


FIGURE 2

TABLE I

Frequency (Hz)	Relative insertion-loss (dB)	Tolerance ($\pm$ 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

ANNEX I

Study Group XV of the CCITT had put some questions as regards this Recommendation and the CMTT has worked out their answers. As those questions and the answers may be helpful for anyone who applies the conventional test signal for carrying out measurements of any kind, they are given below:

Question:

- (a) For the measurement of cross-talk from a sound-programme circuit to a telephone circuit, could the signal described in Recommendation 571 be used, considering the different bandwidth and possible frequency shift?

Reply:

- The intelligible cross-talk ratio is based on selective measurements in the telephone circuit when the sinusoidal signals are transmitted in the sound-programme circuit within the frequency range of 0.3 to 3.4 kHz. In CCITT Recommendation J.21 a minimum ratio of 65 dB is defined.
- The unintelligible cross-talk ratio should be ascertained by measuring the increase of noise in the telephone circuit by loading the disturbing sound-programme with the simulated test signal defined in Recommendation 571. As for this increase no tolerable values are recommended up to now, the CMTT proposes such values based on a maximum noise contribution produced by interference of -65 dBm0p. Depending on the basic noise level in the telephone circuit the following increased values can be tolerated:

TABLE III

Basic noise level (dBm0p)	- 75	- 70	- 65	- 60	- 55	- 50
Tolerable increase of noise level (dB)	10.4	6.2	3	1.2	0.4	0.1

Question:

- (b) What is the equivalent value for 65 dB ratio (given in Recommendations J.21, J.22 and J.23) using sinusoidal tones, when measuring with the recommended new test signal?

Reply:

The answer to this question is included in the proposal for the measurement of the ratio for the total cross-talk caused by intermodulation given in the answer to Question (a).

Question:

- (c) Can the signal defined in Table II from the point of view of the mean loading it would impose on transmission systems and in the light of Recommendations N.12 and N.13 be regarded as acceptable for unrestricted use over complete sound-programme circuits of any constitution?

Reply:

The conventional test signal simulating sound-programme signals defined in CCIR Recommendation 571/CCITT Recommendation J.19 in all aspects can be regarded as acceptable for unrestricted use over sound-programme circuits of any constitution.

REPORT 497-3 *

**A CONVENTIONAL TEST SIGNAL SIMULATING SOUND-PROGRAMME
SIGNALS FOR MEASURING INTERFERENCE IN OTHER CHANNELS ****

(Study Programme 19D/CMTT)

(1970-1974-1978-1982)

1. Introduction

This signal has been proposed to meet the requirements of CCITT Study Group IV and CCIR Study Group 9 in measuring the interference originating from a programme channel and measurable in other traffic channels. The characteristics of the signal are therefore intended to be representative of those of actual programmes, thereby making it suitable:

- for comprehensive wideband measurements of cross-talk (see Note), in conjunction with a detecting instrument appropriately weighted according to the disturbing effect in the channel concerned;
- for loading the sound-programme circuit in a manner whereby substantially the same amount of non-linear interference would be produced in a multiplex system as by actual programmes carried on the circuit.

This signal is not intended for measurement of non-linearity within one and the same sound-programme channel, for which Report 640-1 (Kyoto, 1978) should be consulted.

The results contained in the present Report are reflected in Recommendation 571.

Note. — It is understood that intelligible cross-talk will generally be subject to more stringent limitations than unintelligible cross-talk.

2. Energizing signal

The basic energizing signal is to be obtained from a generator of Gaussian white noise or of pseudo-random noise [HPJ, 1967]. Further study is necessary to specify performance for a pseudo-random generator, e.g. maximum spacing of spectral lines.

It is recognized that a Gaussian distribution of amplitudes only very approximately simulates that of actual programmes — see Report 491, Fig. 2. Nevertheless it is considered adequate for the requirements of a conventional measurement. For more exact studies, special techniques may be used to modify this distribution [Pfyffer, 1971].

3. Shaping network

The flat spectrum at the output of the generator is modified by a shaping network, whose response is shown in Fig. 1, curve A. Since it is the signals of higher-than-average level which contribute most to cross-talk and intermodulation, an attempt has been made to shape the spectrum so as to be typical of such higher-level signals rather than of the long-term mean. A summary of the studies is given in the following paragraphs.

Recent measurements, [CCIR, 1974-78a], have suggested that the shaping networks previously given for this purpose (in Report 497) overvalues the high frequency signal components, and that a better representation of actual sound-programme signals would be obtained with a modified characteristic. Two possible characteristics were given in the above-mentioned reference. One of these is representative of sound-programme signals whose 1-minute mean-power level is exceeded with a probability of 10%; the other is representative of sound-programme signals whose 1-minute mean power is exceeded with a probability of 1%; (Note: this latter curve derived from the reference [CCIR, 1974-78a] is now incorporated in Fig. 1 as curve B).

A later work [CCIR, 1974-78b] suggested that particular account should be taken of signals whose power level is exceeded with a probability of 1%, and also give details of a practical network which could be used for this purpose. The insertion-loss/frequency characteristic of this network is given in Table I and in Fig. 1, curve A. Also shown in Fig. 1 is the idealized target response (curve B). A possible circuit schematic for realizing curve A is also given.

* This Report should be brought to the attention of Study Group 10.

** — For the definition of absolute power, relative power and noise levels, see Recommendation 574.
— Consideration should be given to incorporating Report 497 with Recommendation 571 during the next study period. Contributions are invited from administrations on this point.

The effect of the above spectral-shaping characteristics is such that when the signal is subsequently passed through a sound-programme pre-emphasis network (to CCITT Recommendation J.17, 1.5/800) the output power is increased by 3.7 dB. The corresponding values for the characteristic shown (previously) in Report 497-1 (page 194 of Volume XII, Geneva, 1974) and for the characteristic given in Recommendation 559 are increases in output power of values 5.0 dB and 1.2 dB, respectively. When these values are compared with the corresponding results given for real sound-programme signals in Report 491, Table II and Fig. 5 in particular, it is noted that the 5.0 dB increase in mean power is greater than any of the reported values, and the 1.2 dB increase is exceeded with a probability of more than 50% by real programme signals when these are associated with the same pre-emphasis.

The network characteristic given in curve A of Fig. 1, and in Table I form the basis for a new Recommendation, as indicated in § 1.

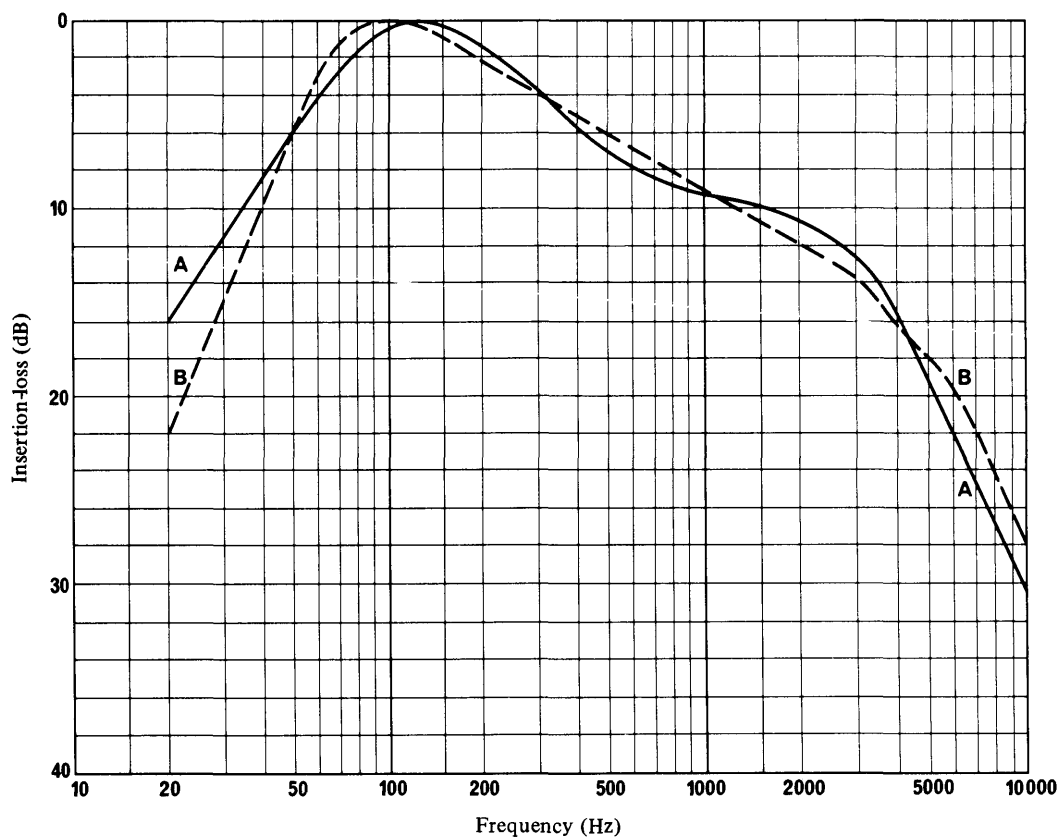
Note. — In Recommendation 559, for the objective measurement of radio-frequency protection in LF, MF and HF broadcasting, a similar test signal based on filter-shape white noise is recommended. In this case not only the insertion loss of the shaping filter is specified but also the frequency spectrum response referred to one-third octaves. With such a filter-set the shaped noise signal can be easily checked. Because of this advantage, the frequency spectrum response of the conventional test signal recommended in Fig. 1 of Recommendation 571, should also have the response referred to one-third octaves inserted. Administrations are invited to contribute on this point.

TABLE I — *Relative insertion-loss/frequency characteristic of the signal-shaping network*

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

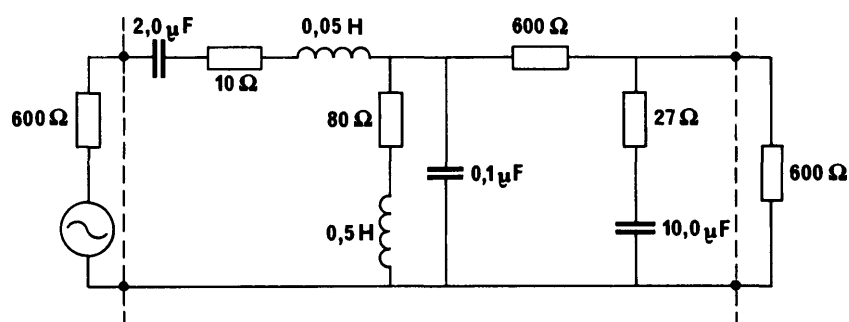
Note 1. — The integrated loss of the network complying with Table I has been calculated as 14.4 dB for white noise in the band 0 to 15 kHz.

Note 2. — The practical network shown associated with Fig. 1 has an additional basic insertion loss of 17.1 dB.



A : proposed characteristic as given in Table I

B : reproduction of curve CB from [CCIR, 1974-78a] for comparison with A.



A practical network configuration for achieving the proposed characteristic A, in the non-constant resistance form.

FIGURE 1 – Relative insertion-loss/frequency characteristic and the practical signal-shaping network, with suitable component values

Note 1. – The integrated loss of the network complying with Table I has been calculated as 14.4 dB for white noise in the band 0 to 15 kHz.

Note 2. – The practical network shown associated with Fig. 1 has an additional basic insertion loss of 17.1 dB.

4. Mean-power levels to provide appropriate circuit loading

It is intended that the conventional test signal should be applied to the sound-programme circuit at a point of zero relative level.

The Document [CCIR, 1974-78a] suggested that a suitable range of test levels (corresponding to the lower probabilities of occurrence) would lie between +2 dBm0s and +5 dBm0s. Bearing in mind the wide variation in the different results for mean-power level of sound-programme signals, as well as the need to take account of high-level signals which are exceeded for a low proportion of the time, two test levels seem to be appropriate. As suggested by [CCIR, 1974-78c], there would be advantages in changing the level of the applied test signal periodically in a way that would enable the presence of small amounts of interference arising in the disturbed circuit to be more readily recognized. It is therefore proposed that the test signal level should be cyclically changed according to the following:

TABLE II

Step	Signal level	Duration
1	-4 dBm0s	4 s
2	+ 3 dBm0s	2 s
3	no signal	2 s

Step 1, 2, 3 can be continuously repeated if necessary.

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REPORT 491-2

CHARACTERISTICS OF SIGNALS SENT OVER SOUND-PROGRAMME CIRCUITS

(Study Programme 17A/CMTT)

(1970-1974-1978)

1. Characteristics of test signals

The first point of the Study Programme 17A/CMTT (Kyoto, 1978) is concerned with the characteristics of the test signals used by broadcasting authorities on sound-programme circuits and asks if new types of test signals are to be expected.

Some administrations note that new types of test signals are to be expected.

In [CCIR, 1966-69b], it is also suggested that new test signals may be wanted, in particular for non-linearity tests and for stereophonic circuits. It also points out that, as a guiding principle, no test signal should place more stringent demands on the transmission path than actual traffic. New test signals for cross-talk and non-linearity tests are proposed for study in Report 497 and in Report 640-1 (Kyoto, 1978).

The broadcasting authorities are invited to give detailed information for this point of the Study Programme at the next meeting of CMTT.

2. Mean power and distribution of instantaneous levels of programme signals

The second point of the Study Programme 17A/CMTT (Kyoto, 1978) is concerned with the mean power of programme signals presented to transmission circuits and with the distribution of the instantaneous or quasi-instantaneous power, but it seems likely that the distribution of, for instance, the one-minute mean power will also be needed. The instantaneous power should be expressed in terms of the r.m.s. value of the equivalent sine-wave signal, which is the sine-wave signal whose peak amplitude reaches the voltage which corresponds to the instantaneous power observed.

2.1 General

In [CCIR, 1966-69a] a warning is given that programme signals presented for transmission have frequently been submitted to selected and adjustable emphasis in the studio to meet artistic requirements.

In consequence of this it may be unwise to draw broad conclusions about the power spectrum of programme signals because the power levels at higher spectrum frequencies may, for instance, have been increased substantially above the "normal" value.

An increase of this nature may tend to invalidate the basis on which the CCITT Programme Pre-emphasis Network was chosen (see CCITT Blue Book, Vol. III, 1965, pages 547-548). The network is used at the input to a carrier programme transmission channel.

"Studio Emphasis" must be accepted as a fact and the various tests must include a realistic proportion of such material. This type of programme material is particularly necessary when measuring the statistical distribution of programme power in various parts of the spectrum.

It is noted that Report 497 proposes that a shaped band of noise should be used as a test signal to simulate a sound programme. The study of spectral power distribution of real sound programmes will be of value when assessing the new test signal.

2.2 Signal level at the zero relative level point

It is understood that at the zero level point, the quasi-peak level of the programme signal will be 9 dBm, but that the instrument used for setting the level has not been widely standardized. It is essential that information must always be made available on the type of unit used for any series of measurements and also on the mode of using it.

It is further desirable that information should be made available on the instantaneous peak levels measured at the zero points of the programme circuits. This information is of interest in connection with transmission channels as well as with transmitters. It is most readily measured on an oscilloscope. (The equivalent sine-wave power of the peak is $0.5 V^2/Z_0$ where V is the peak voltage.)

2.3 Results of measurements

The information made available by various administrations has been summarized in the following paragraphs:

2.3.1 Mean power

For many purposes, the mean power level, taken for instance over one-minute intervals, is very significant. Such information is given in [CCIR, 1966-69c] in the form of the distribution over the day for two classes of programmes: Programme No. 1 with a preponderance (up to 60-65%) of speech and Programme No. 2 with a preponderance (up to 35-40%) of jazz and symphony music. Table I is taken from this document.

Analysis of the measurements carried out, showed that the lower the mean power of the excerpt, the more the dynamic range compressor increases the mean power. In the excerpts investigated, the difference between the level of the one-minute mean power with the compressor switched in and with it switched out, varied within the limits of 2 dB and 17 dB.

The transmission power, with the pre-emphasis network switched in, depends on the spectral density of the signal. In the excerpts studied, the difference between the levels of the one-minute mean power with the pre-emphasis network switched in and with it switched out, varied within ± 7 dB.

TABLE I

Averaging periods and levels	Power of the broadcasting signal at a point of zero relative level of the sound-programme circuit (dBm0)			
	Without compandor or pre-emphasis		With compandor followed by pre-emphasis	
	Programme No. 1	Programme No. 2	Programme No. 1	Programme No. 2
Day (level of long-term mean power)	-5.7	-4.5	-0.6	+0.5
Hour (most highly-probable level of mean power)	-4.6	-3.1	+0.7	+1.4
Level of one-minute mean power exceeded during 20 % of the day	-4.4	-2.5	+0.7	+2.2
Level of one-minute mean power exceeded during 1 % of the day	-1	-0.1	+3.6	+4.1
Level of one-minute mean power exceeded during 0.1 % of the day	+1.7	+2.2	+4.3	+4.9

Note 1. — When the signal, after compression and pre-emphasis, is applied to a carrier line, an attenuation of 1.7 dB is used in transferring the programme circuit zero level point to the carrier circuit zero level point.

Note 2. — The compandor has a compression coefficient of 2 and an unaffected level of +9 dBm.

Note 3. — The pre-emphasis network complies with CCITT Recommendation J.17.

Note 4. — While the measurements were taken there were no signal limiters in circuit (manual control).

2.3.2 Peak power

From [CCIR, 1966-69c], it may be seen that if a quasi-peak power level of +9 dBm0 (as measured by an instrument with an integration time of 20 ms) is not exceeded at the studio output, then the quasi-peak power measured at the input of a carrier programme circuit exceeds a value of +12 dBm0 for 8% of the time for Programme No. 1 and 17% of the time for Programme No. 2. A compandor followed by a pre-emphasis network and 1.7 dB attenuation is connected between the studio output and the carrier equipment.

The analysis was made of excerpts of a programme transmission of varied composition, lasting 5 to 30 minutes, pre-recorded on magnetic tape. These excerpts were divided into one-minute sections and, for each section, the distribution of the instantaneous voltage values was deduced by means of a 10-channel analyser. The measurements obtained were used to determine the dispersion which, on condition that the arithmetic mean of the instantaneous voltage values is equal to zero, is proportional to the one-minute mean power of the signal.

2.3.3 The carrier system for the establishment of sound-programme circuits corresponding to CCITT Recommendation J.31 (see also [CCIR, 1970-74d]) which is considered as adequate for international application, has to a large extent been used by the Administration of the Federal Republic of Germany for the setting up of 15 kHz type programme circuits which are also suitable for stereophonic transmissions. With this system, one or two sound-programme circuits are established on a primary group link instead of six telephone channels. To achieve the desired flexibility in the use of carrier systems for the establishment of sound-programme circuits, the characteristics of the additional equipment required for the reduction of noise had to be so specified that the load on the primary group link does not exceed that obtained when setting up telephone channels.

According to CCITT Recommendation G.223, the load allowed for a primary group link with 12 telephone channels can be described by the following parameters:

- nominal mean power during busy hour (long-term value):
 $-15 + 10 \log 12 \approx -4 \text{ dBm0}$
- conventional load:
 $-1 + 4 \log 12 = +3.3 \text{ dBm0}$
- peak level exceeded with a probability of 10^{-5} times:
 $+19 \text{ dBm0}$.

To be able to verify the compatibility of the sound-programme channel load with these values, the Administration of the Federal Republic of Germany invited the "Institut für Rundfunktechnik" in Hamburg to perform fundamental investigations concerning the amplitude statistics of broadcasting signals for the following practical cases:

- A : unchanged sound signal;
- B : sound signal changed by pre-emphasis network according to CCITT Recommendation J.17 (see also [CCIR, 1970-74d]) with an insertion loss of 1.5 dB at 0.8 kHz;
- C : sound signal changed by the pre-emphasis network mentioned under B, but with an insertion loss of 6.5 dB at 0.8 kHz, and by companders according to the CCITT new Recommendation as mentioned above.

A normal 24-hour programme of the "Norddeutscher Rundfunk" was selected for the investigations. In this connection, it became apparent that there is practically no difference between the amplitude statistics of various days of the week. For the purpose of analysis, the broadcasting signals to be investigated were applied to a statistical counter with 19 different thresholds via a full-wave rectifier operating without a time constant. Samples were taken 5 times per second from the rectified signal for a period of 10 μ s.

Since the amplitude statistics of the 24-hour programme do not, however, indicate the distribution existing only during short highpower parts of the programme, analyses for 10% and 1% of the time were made in addition to the determination of the long-term distribution (100% of the time). These analyses were made in such a manner that samples were only taken when, during a parallel power measurement integrating over one second, certain thresholds were exceeded which are decisive for the 10% and 1% values. (Further details are available [Jakubowski, 1971].)

The most essential results of the measurements are listed in Table II. The given mean power values are calculated from the cumulative distribution curves. Figure 1 shows the distribution curve in the cases A, B and C for 100% of the time as an example. All instantaneous values are expressed as equivalent sine-wave power levels.

From Table II, case C, it is obvious that the loading caused by two sound-programme channels, installed with the mentioned system, is comparable with the loading produced by 12 telephone channels in the primary group.

2.3.3.1 Mean power

The distribution of the one-minute mean powers is given in Table III-A; the results are taken from [CCIR, 1970-74a]. The measurements leading to the values shown were carried out on 6 different sound-programme circuits, at audio frequencies, over a total time of 40 hours. No pre-emphasis was used.

The level of the long term mean power was found to be -3.4 dBm0 .

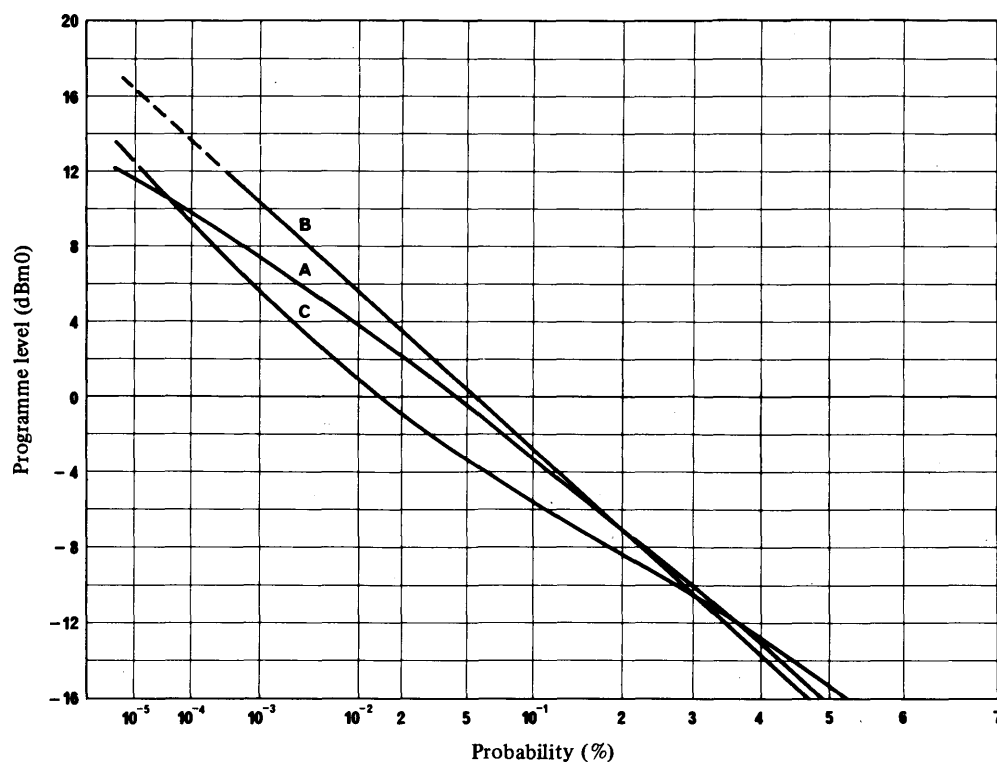


FIGURE 1 – Cumulative distribution
Cases A, B and C: Distribution of the programme level in 100% of the time

Curves A: unchanged signal
B: changed by pre-emphasis 1.5 dB/0.8 kHz
C: changed by pre-emphasis 6.5 dB/0.8 kHz and compressor

TABLE II

Time base of statistics (%)	Case (see § 2.3.3)	Mean power value (dBm0)	Instantaneous value with probability of exceeding:	
			10^{-2}	10^{-5}
			(dBm0)	(dBm0)
100	A	-4.4	+3.6	+11.4
	B	-2.8	+5.6	+16.5
	C	-5.8	+0.8	+12.4
10	A	-0.3	+6.2	+12.2
	B	+2.7	+9.8	+18.2
	C	-2.4	+3.7	+13.9
1	A	+1.9	+ 7.5	+12.9
	B	+5.5	+12.0	+19.5
	C	0	+ 5.7	+15.1

TABLE III-A – 1-minute mean power of audio frequency signal

Probability of exceeding (%)	Mean power level (dBm0)
50	–3.7
10	–1.0
1	+1.0
0.1	+2.5

2.3.3.2 *Peak power*

Information about peak power can be deduced from [CCIR, 1970-74a] and this is given in Table III-B. Studios are equipped with limiters, which operate as linear amplifiers whose gain is reduced in the event of the signal exceeding a value corresponding to an equivalent sine-wave peak power of +9 dBm0. A time constant of about 0.5 ms is associated with a gain reduction; this characteristic accounts for the fact that peak powers higher than +9 dBm0 are encountered, although with small probability.

TABLE III-B – Peak powers

Probability of exceeding	Level of equivalent sine-wave signal (dBm0)
10^{-3}	+ 9.5
10^{-4}	+ 11.2
10^{-5}	+ 12.0

These values can be deduced from the instantaneous level distribution curves.

2.3.3.3 *Distribution of instantaneous levels*

The distribution of instantaneous levels, measured over 13 hours, is shown in Fig. 2 of this Report; it is taken from [CCIR, 1970-74a]. The level of the mean power in this case was found to be –2.2 dBm0; the peak power associated with a probability of 10^{-5} is +12 dBm0. A comparison of curves A and B shows clearly the non-Gaussian character of the programme signals.

2.3.3.4 *Mean power*

In [CCIR, 1966-69d], a mean distribution is quoted for a sound programme not subjected to any process of limitation or compression. The median power level is –9.2 dBm0 and the standard deviation is 1.7 dB. Distributions of processed sound programmes are also included. Table IV is an extract from that document.

TABLE IV

Probability of exceeding (%)	Studio output (dBm0)	Partially limited and compressed signal (dBm0)	Entirely limited and compressed signal (dBm0)
Long-term mean	– 8.9	– 6.4	– 2.4
20	– 7.8	– 5.0	– 1.7
1	– 5.0	– 2.2	0

Note. – The meaning of “limited” and “compressed” is indicated in [CCIR, 1966-69d].

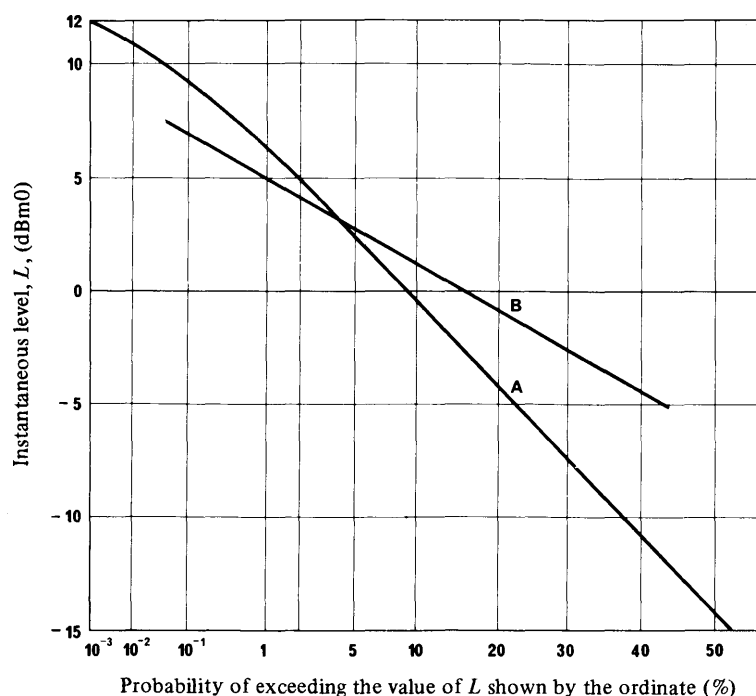


FIGURE 2 – Distribution of instantaneous level, L , expressed in equivalent sine-wave powers of the signal in a programme channel at audio frequencies (Curve A)

Measuring time: 13 hours.

Curve B indicates a normal distribution

$$L = 20 \log \left(\frac{V_{\text{inst}}}{0.775 \text{ V}} \right) - 3 \quad \text{dBm0}$$

2.3.3.5 Peak power

The following is obtained from [CCIR, 1966-69d]:

Probability of exceeding	Equivalent sine-wave power of studio output signal (dBm0)
10^{-3}	+ 7
10^{-4}	+ 9.5
10^{-5}	+ 11.5

2.3.4 From [CCIR, 1974-78] results are given for measurements made on two contrasting types of regular programme service, over a long period (> 30 hours).

The first of these services, designated R1 for the purposes of this Report, consists mainly of popular music with short announcements, news items and short interviews. The second designated R4 for the purposes of this Report, has a high proportion of talks, topical discussions, plays etc. with musical support. The results given in the following tables apply to un-emphasized signals which were controlled in the normal fashion not to exceed +8 dBm0* as indicated upon a peak programme meter** and no additional programme signal limiters were employed.

* This is the nominal peak indicated value used in the BBC.

** The characteristics of this meter are shown as item 4 in the table of CCITT Recommendation J.15.

The results obtained are shown in Figs. 3, 4, 5 and 6 and include:

- the cumulative distribution of 1-minute mean powers;
- the cumulative distribution of instantaneous powers;
- the cumulative distribution of peak powers occurring within each 1-minute period.

The results also include those for the unprocessed signal, the signal pre-emphasized according to CCITT Recommendation J.17 (1.5 dB loss at 800 Hz) and pre-emphasized and compressed in accordance with CCITT Recommendation J.31.

The values relating to any particular level of probability can be found from reference to Figs. 3, 4, 5 and 6.

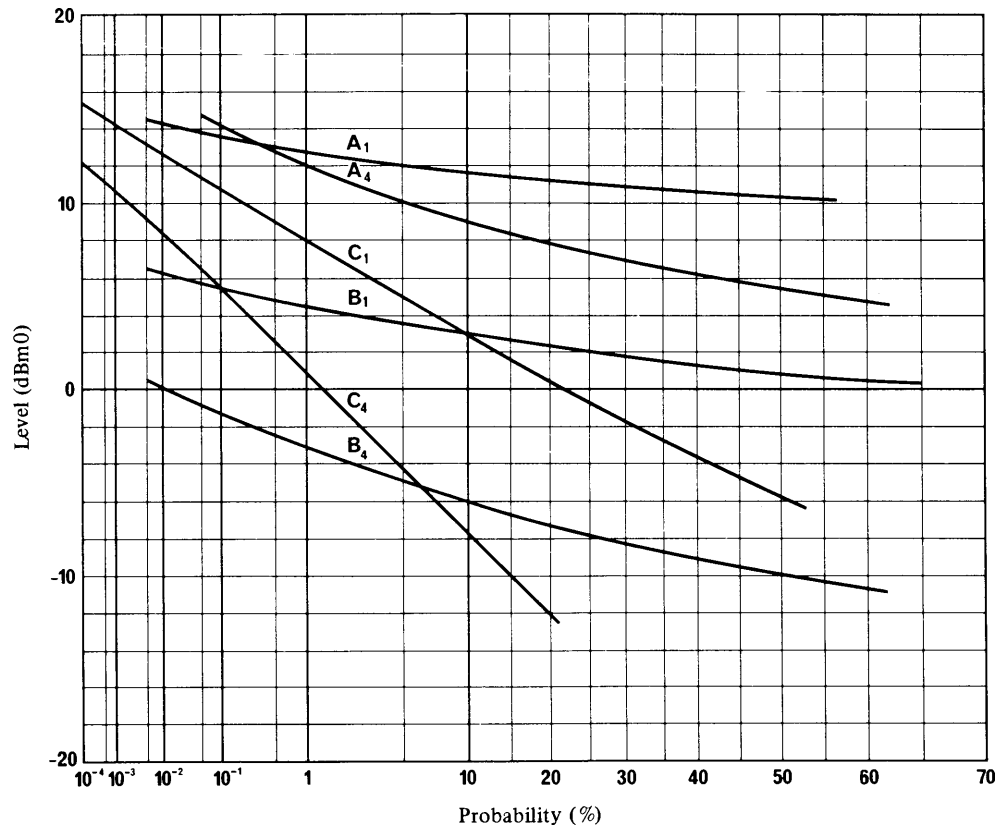


FIGURE 3 – *Characteristics of unprocessed signals*

Curves A_1 : cumulative distribution of 1-minute peak powers for programme R1

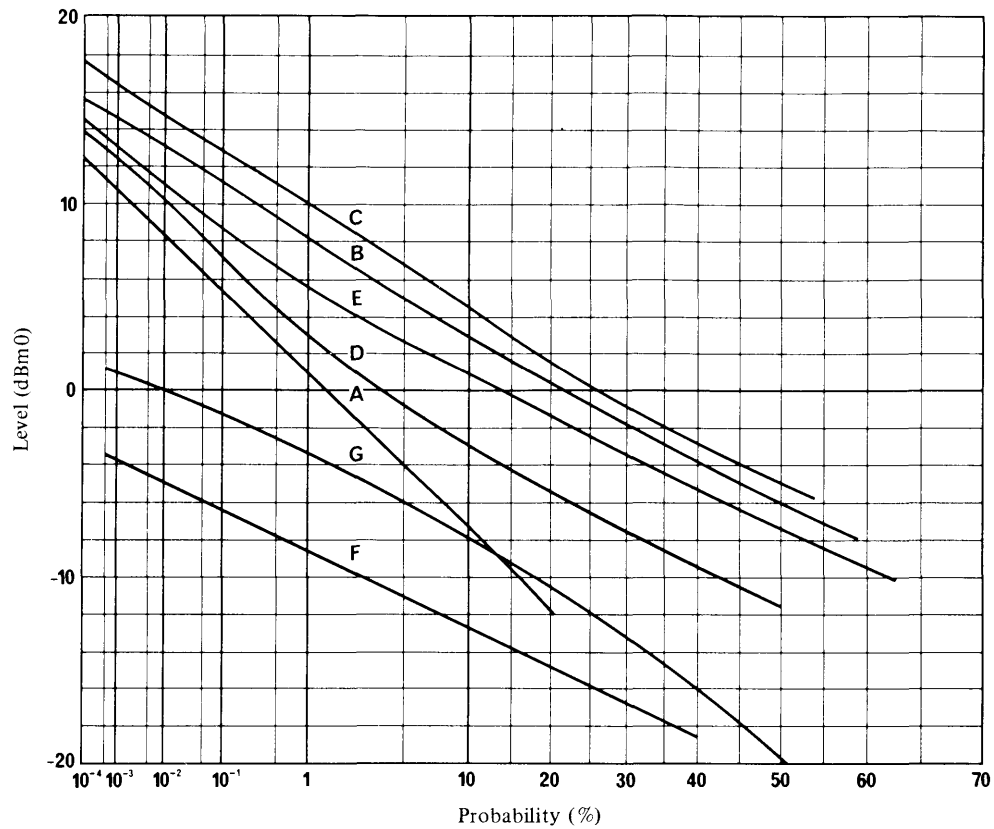
A_4 : cumulative distribution of 1-minute peak powers for programme R4

B_1 : cumulative distribution of 1-minute mean powers for programme R1

B_4 : cumulative distribution of 1-minute mean powers for programme R4

C_1 : cumulative distribution of instantaneous powers for programme R1

C_4 : cumulative distribution of instantaneous powers for programme R4

FIGURE 4 – *Instantaneous powers*

Curves A : characteristics of unprocessed programme R4

B : characteristics of unprocessed programme R1

C : characteristics of pre-emphasised programme R1 (Loss 1.5 dB at 800 Hz Recommendation J. 17 of the CCITT)

D : characteristics of programme R1, pre-emphasised and compressed in accordance with CCITT Recommendation J. 31

E : characteristics of 2 separate programmes of type R1 pre-emphasised and compressed in accordance with CCITT Recommendation J. 31

F : characteristics of signal resulting from a white noise signal applied to a single programme channel (without pre-emphasis or compression)

G : characteristics resulting from coherent white noise signals applied to two programme channels (without pre-emphasis or compression)

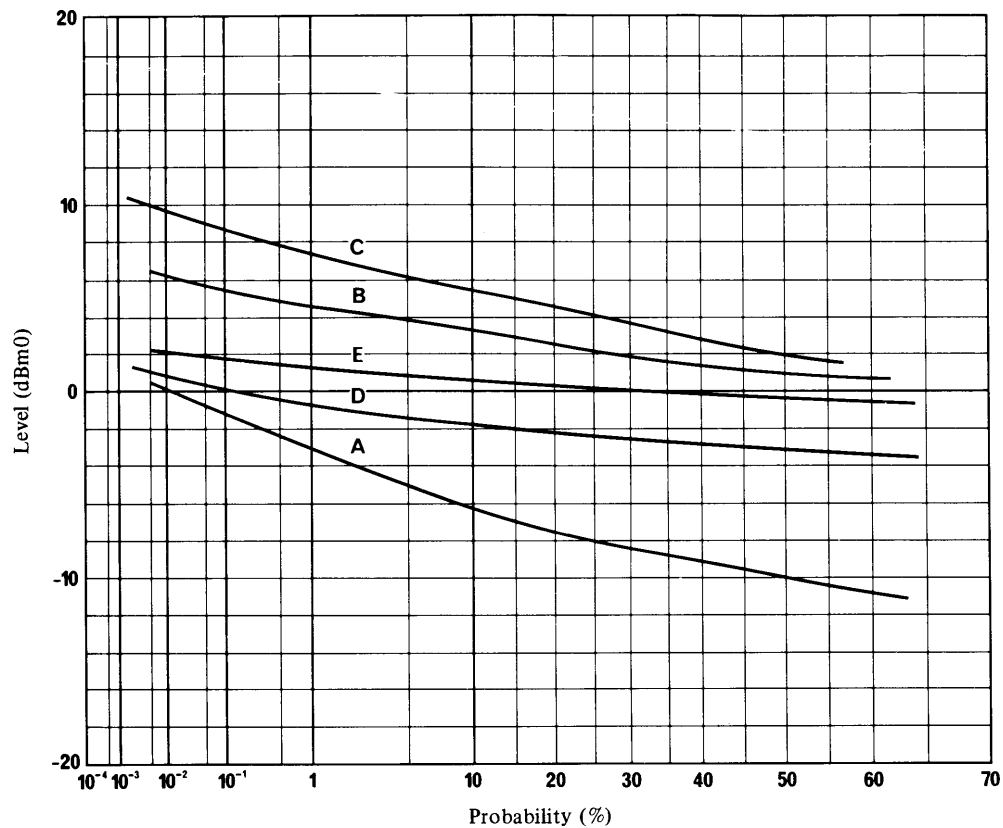


FIGURE 5 – 1-Minute mean powers

Curves A : characteristics of unprocessed programme R4

B : characteristics of unprocessed programme R1

C : characteristics of pre-emphasised programme R1. (Loss 1.5 dB at 800 Hz Recommendation J. 17 of the CCITT)

D : characteristics of programme R1, pre-emphasised and compressed in accordance with CCITT Recommendation J. 31

E : characteristics of 2 separate programmes of type R1, pre-emphasised and compressed in accordance with CCITT Recommendation J. 31

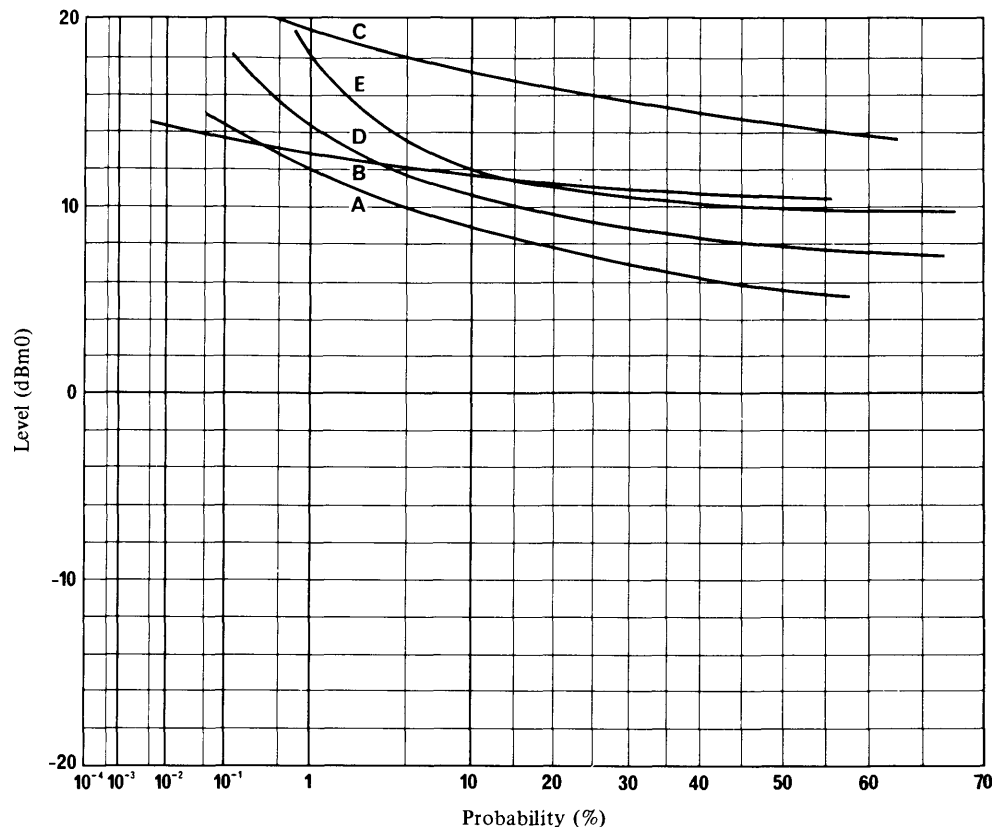


FIGURE 6 – Peak powers occurring within each 1-minute period

Curves A: characteristics of unprocessed programme R4

B: characteristics of unprocessed programme R1

C: characteristics of pre-emphasised programme R1. (Loss 1.5 dB at 800 Hz Recommendation J. 17 of the CCITT)

D: characteristics of programme R1, pre-emphasised and compressed in accordance with CCITT Recommendation J. 31

E: characteristics of 2 separate programmes of type R1, pre-emphasised and compressed in accordance with CCITT Recommendation J. 31

3. Spectral power distribution

Study Programme 17A/CMTT (Kyoto, 1978), § 6 is concerned with the spectral power distribution of sound-programme signals.

The following information has been made available.

3.1 Document [CCIR, 1966-69c] gives some spectral power distribution and Figs. 7, 8, 9 and 10 are reproduced from this source.

The spectral characteristics of the one-minute mean powers of the broadcasting signals were measured over the same one-minute programme excerpts. The measurements were made with equipment incorporating $\frac{1}{3}$ octave filters and a voltmeter with a square law detection characteristic.

The spectral characteristics were determined for 100 one-minute excerpts of different kinds of broadcasting programmes. The characteristics obtained were classified into 17 groups (according to the nature of the transmission) and the average spectral characteristics were determined for each group. Figures 7 and 8 give two distributions as an example: one for male speech and one for a symphony orchestra. Analysis of the characteristics showed that, for all types of programmes, most of the power is concentrated in the comparatively narrow band of 50 to 1200 Hz.

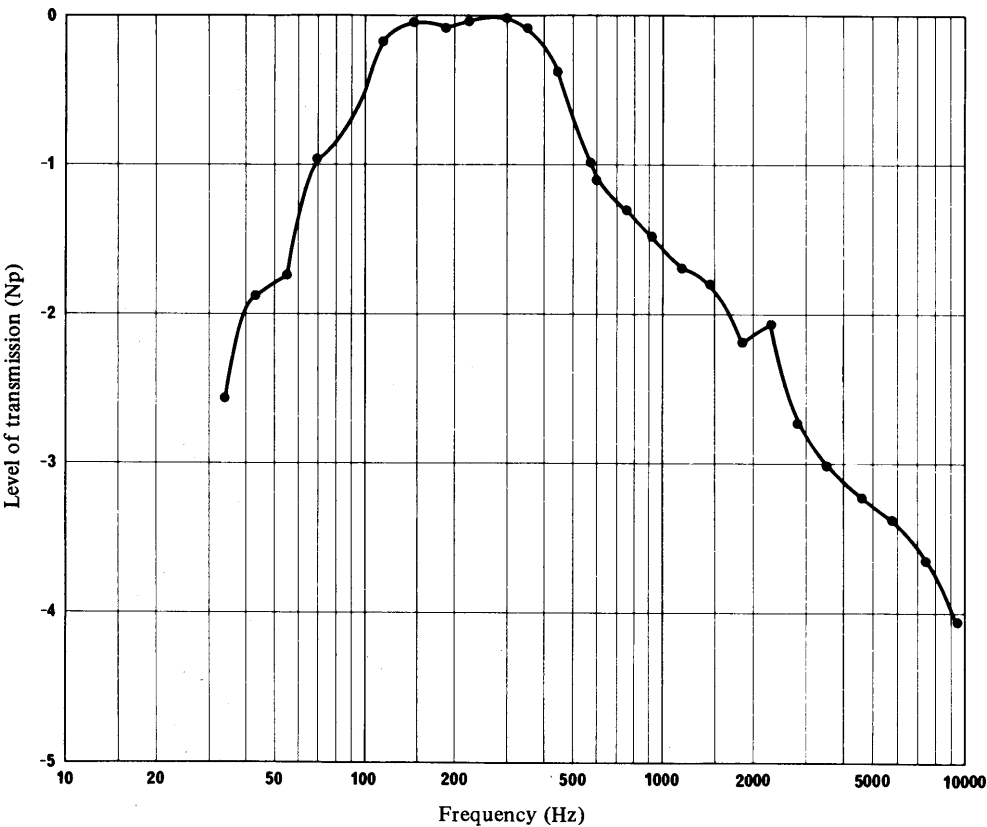


FIGURE 7 – Averaged spectral characteristics of level of transmission of male speech

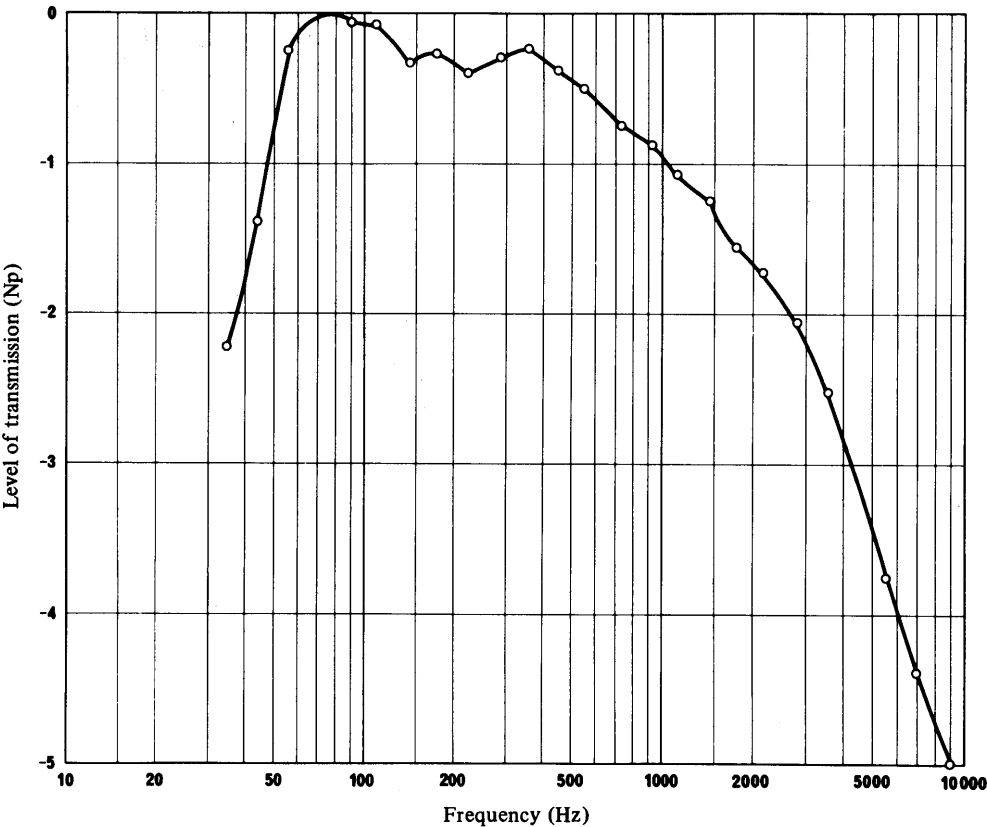


FIGURE 8 – Averaged spectral characteristics of level of transmission of a symphony orchestra

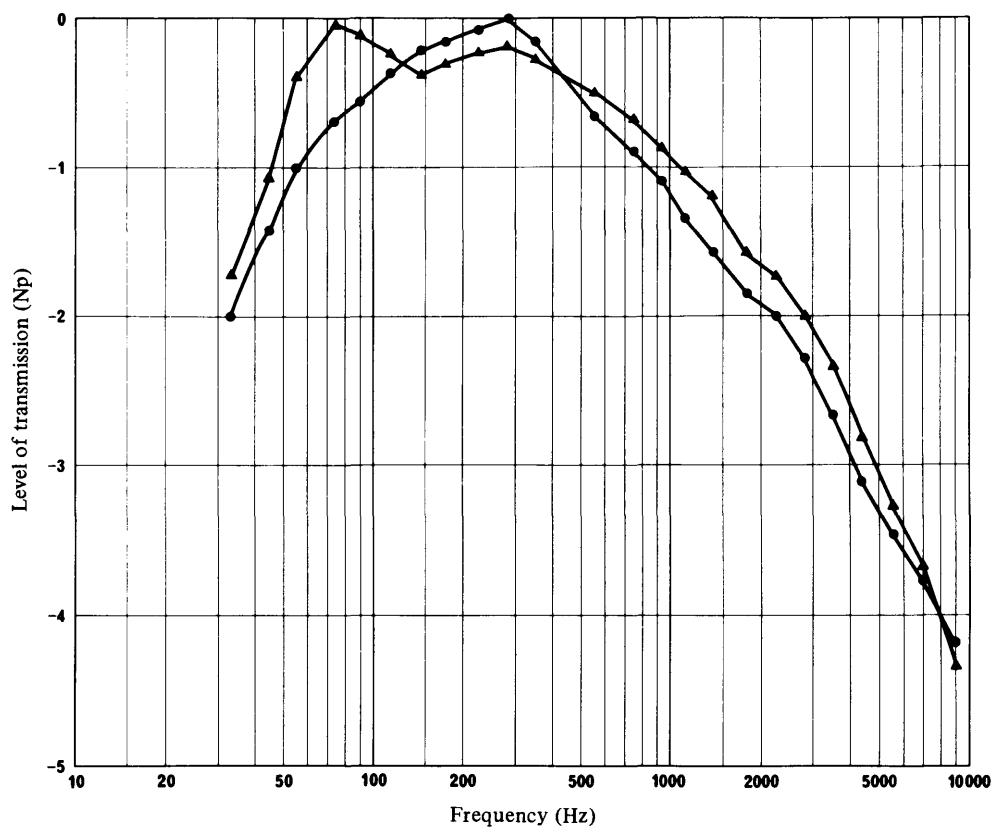


FIGURE 9 – Averaged spectral characteristics of level of transmission of daily broadcasting programme, taking account of the incidence of transmissions with different characteristics

—▲— Programme No. 1

—●— Programme No. 2

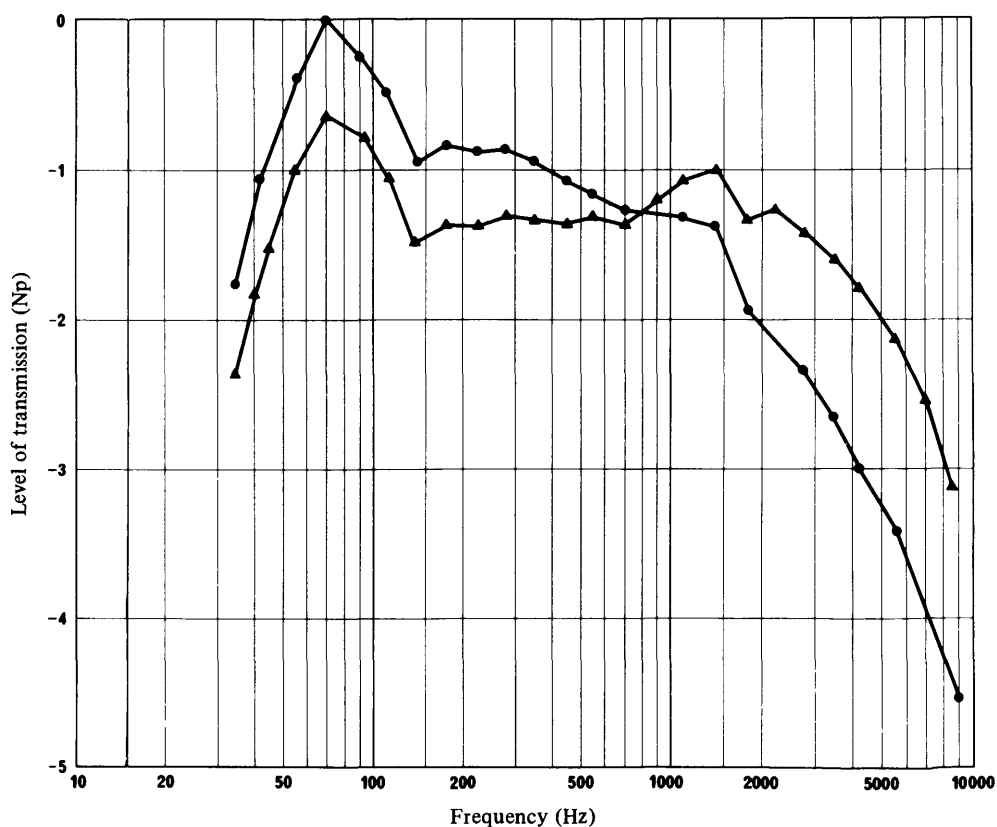


FIGURE 10 – Averaged spectral characteristics of level transmission of a light orchestra with the pre-emphasis circuit switched off and with it switched on

—●— with pre-emphasis circuit switched off

—▲— with pre-emphasis circuit switched on

The vertical scales on the four figures are not intended to indicate absolute values, but give relative spectral power density per unit bandwidth.

Measurements of the spectral characteristics of broadcasting signals in the channel with the compressor switched in showed that the compressor does not alter the spectral character of the signal.

3.2 Figures 11 and 12 show the spectral average power distribution for some different programme items recorded on magnetic tape [CCIR, 1966-69d].

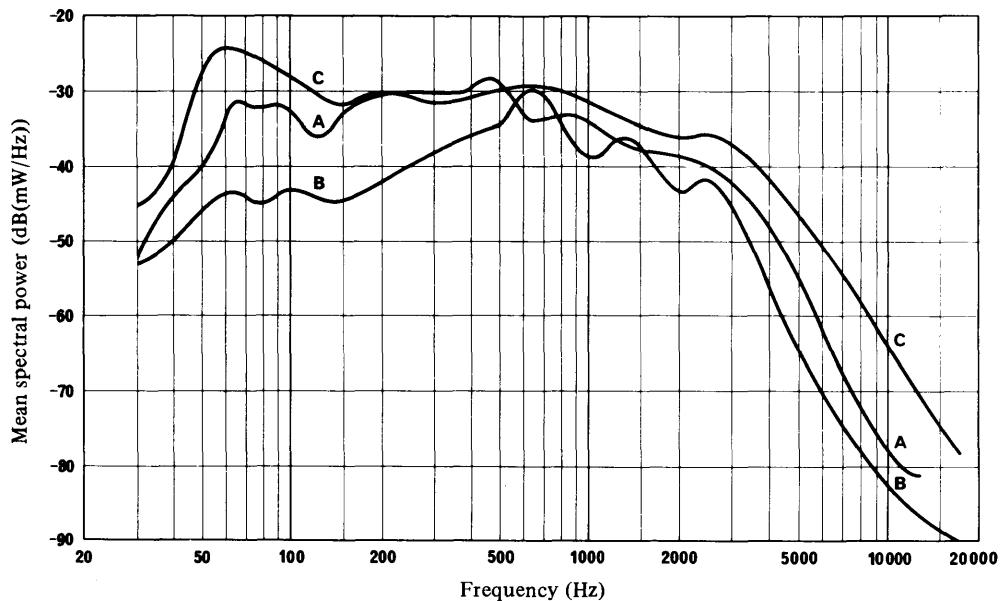


FIGURE 11 – Mean spectral power of sound signals

Curve A : classical music

Curve B : opera

Curve C : modern music

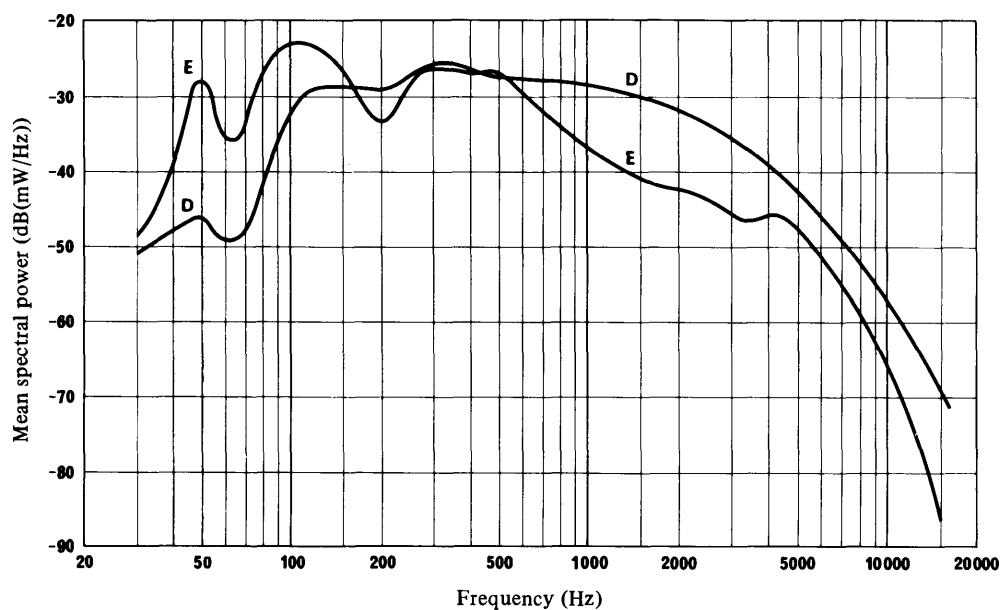


FIGURE 12 – Mean spectral power of sound signals

Curve D : jazz

Curve E : speech

Measurements were effected by means of a square-law instrument connected to a recorder able to record a signal variation speed of 250 dB/s.

The writing arm of the recorder was connected to a level analyser having 5 dB spaced thresholds, a sampling duration of 20 μ s and a sampling rate of 10 per second.

The frequency analysis of the programme items was carried out on contiguous third octave bands from 31 to 16 000 Hz (constant relative bandwidth analysis).

For each third octave band the statistical distribution of the instantaneous power of the different items has been recorded and afterwards the distribution of the mean value of the power has been calculated.

Figures 11 and 12 are in terms of power per Hz (dB(mW/Hz)).

3.3 In [CCIR, 1970-74b] it is stated that the spectral power decreases at a rate of 3 dB per octave during those parts of the sound programmes which are of high power level, when the analysis covers a one second period, but by 6 dB per octave for long-term observations.

4. Observations

4.1 The results of the statistical programme signal measurements given in § 2.3 show that the long term mean power ranges from -9 dBm0 to $+1$ dBm0. A conventional mean power level representing all kinds of sound programmes should lie somewhere between these extremes and a value of -4 dBm0 has been suggested. Further measurements are needed to confirm the usefulness of this proposed value.

4.2 The instantaneous power level which is exceeded with a probability of 10^{-5} is of particular interest because multi-channel carrier systems are designed on that basis. It is desirable to fix more precisely the time interval over which an instantaneous power should be averaged (over a few milliseconds).

The data so far considered have not generally quoted the instantaneous power in this way but the indications are that an equivalent sine-wave power of $+12$ dBm0 is unlikely to be exceeded for a probability of 10^{-5} .

4.3 It is difficult to draw firm conclusions from the spectral analysis primarily because typical programme material is so variable in content. National characteristics might also have a bearing on spectral power distribution. However, on the basis of measurements so far, it can be concluded that, even for programmes of the widest spectrum (symphony orchestra), the basic power of the signal is concentrated in the band 50-1200 Hz.

5. Proposal for the form of future measurements

5.1 A statement is required about the control of the "studio output".

The general aim is that the quasi-peak level should not exceed $+9$ dBm at a zero relative level point of the sound-programme circuit. The method of control used should be stated and the important features of any equipment used should be described.

5.2 Measurements should be expressed as relative to the zero relative level point of the sound-programme circuit. For the analysis of the "studio output", it is convenient to work in terms of one-minute samples. For a particular example of a broadcast programme, samples might be taken for each minute, or possibly alternate minutes. The results of tests should be presented in a form which shows the values exceeded for 20%, 1% and 0.1% of the time for the mean power results and 10^{-3} , 10^{-4} and 10^{-5} of the time for instantaneous peak values, expressed in equivalent sine-wave power.

5.3 The following data should be determined from each sample:

- mean power (one-minute mean);
- instantaneous peak power in terms of equivalent sine wave ($0.5 V^2/2Z_0$), where V is the peak voltage;
- the ratio of the instantaneous peak to main power.

5.4 Spectral information is also desirable and is conveniently determined also from the one-minute samples. From each sample, the following should be obtained, using $\frac{1}{3}$ octave filters according to IEC Publication 225:

- mean power (one-minute mean), corrected for filter shape;
- instantaneous peak power in terms of the equivalent sine-wave power.

Note. — To reduce the amount of detailed spectral analysis, measurements need only be made for individual spectral channels when the result is likely to contribute to the upper 20% of the distribution for the particular $\frac{1}{3}$ octave channel.

Data obtained from these measurements will be of value when determining the characteristics to be used for the test signal in Report 497.

6. New additional study method proposed (Instantaneous peak levels actually reached)

6.1 Introduction

The behaviour of some transmission systems and certain audible deteriorations in channel quality are related to the frequency with which the instantaneous signal voltage reaches or exceeds certain critical levels. The study of the instantaneous voltage value distribution normally carried out by signal sampling is therefore not enough to cover all the phenomena which are liable to occur. It can, however, be usefully supplemented by a precise study of the peak levels actually reached in successive time intervals of given duration, using the method proposed in [CCIR, 1970-74c].

6.2 Method proposed

Basically, the method consists in applying the signal, after rectification (full wave), to the input of a so-called "peak follower". At any point of time t , the voltage value at the peak follower output is equal to the maximum value reached by the voltage at the input since the previous zero resetting of the instrument.

This zero resetting is controlled at intervals of period T . Just before each zero resetting, the "peak voltage value" information is collected and classified in an amplitude analyser.

The variable parameter of this study method is the period T of the time intervals between consecutive zero resettings of the peak follower. The selection of particularly significant T values may involve, for example, the dynamic characteristics of certain components of the transmission system ("recovery time after overload", time constants of limiters, etc.) as well as certain subjective effects.

In trying out this new method, it is recommended for the time being that at least one of the three following values be used:

$$T = 1 \text{ s}; \quad T = 0.1 \text{ s}; \quad T = 0.01 \text{ s}.$$

The zero resetting time and the tracking speed of the peak follower must be such that the percentage of peak values which are not extracted or which are distorted by the use of this device is absolutely negligible.

6.3 Presentation of results

As in the other methods, the results will be presented in the form of tables and of families of cumulative probability curves. They will contain, in particular, the percentages of time segments of duration T , i.e., for example, the percentage of seconds, tenths of a second or hundredths of a second, during which the absolute signal peak level exceeds certain set levels.

For certain critical values of the level L , it will be useful to compare the percentage of time intervals of duration T during which the absolute signal peak level exceeds L , with the percentage of time during which the "instantaneous" signal power level, determined by high frequency sampling, exceeds this same level L .

7. Characteristics of monitoring and signalling tones

The fifth point in the Study Programme 17A/CMTT (Kyoto, 1978) is concerned with the characteristics of monitoring and signalling tones used by the broadcasting authorities.

Administrations and other organizations interested are invited to submit any proposals concerning the use of monitoring and signalling tones (restricted to within the nominal bandwidth) for use on sound-programme circuits, so that these proposals can be taken into account in studies under Study Programme 17A/CMTT.

8. A statistical model of sound-programme signals

Annex I gives details of a means of describing and generating a signal having definable statistical properties similar to those of sound-programme signals [Ehara, 1977a and b; Weiss, 1964].

This model may be considered to give a basis for the establishment of a new test signal.

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- [1966-69]: **a.** CMTT/22 (United Kingdom); **b.** CMTT/31 (L.M. Ericsson); **c.** CMTT/153 (U.S.S.R.); **d.** CMTT/158 (Italy).
- [1970-74]: **a.** CMTT/27 (Switzerland); **b.** CMTT/36 (Germany (Federal Republic of)); **c.** CMTT/47 (France); **d.** CMTT/52 (Germany (Federal Republic of)).
- [1974-78]: CMTT/35 (United Kingdom).

ANNEX I

A STATISTICAL MODEL OF SOUND-PROGRAMME SIGNALS

1. Introduction

A number of data have been reported concerning amplitude distribution of sound-programme signals. These were usually represented by graphs or sometimes expressed by approximating formulae individually, but difficulties were often met in application because of the variety of measurements. This paper describes a unifying statistical model for the general understanding of such measurements; its probability density function and statistical property agree closely with the experimental results both for speech and music. The model here presented suggests a test signal which offers a means of measuring the dynamic characteristics of transmission equipment.

2. A statistical model of sound-programme signal

A sound-programme signal $U(t)$ may be regarded as the product of an audio signal $V(t)$ with a constant r.m.s. level and the signal $W(t)$ which represents a slowly fluctuating r.m.s. value of the signal $U(t)$, i.e., we have,

$$U(t) = V(t)W(t) \quad (1)$$

It has been ascertained that the $V(t)$ signal commonly follows the Gaussian distribution,

$$g(v) = \frac{1}{\sqrt{2\pi}} \exp\left(-\frac{v^2}{2}\right) \quad (2)$$

and has harmonious and consonant spectral components.

On the other hand, regarding the p.d.f. (probability density function) of the fluctuating r.m.s. value $W(t)$, denoted by $h(w)$, it was found that $h(w)$ can be approximated by the M -distribution which is well-known as the intensity distribution of radio-wave fading. The M -distribution is defined as follows:

$$h(w) = \frac{2m^m}{\Gamma(m)} w^{2m-1} \exp(-mw^2) \quad (3)$$

The shape of this distribution changes with the value of m as shown in Fig. 13.

Then the p.d.f. $f(u)$ of the $U(t)$ signal is obtained from equations (1), (2) and (3) as follows:

$$f(u) = \frac{2m^m}{\sqrt{2\pi}\Gamma(m)} \int_0^\infty w^{2(m-1)} \exp\left(-\frac{u^2}{2w^2} - mw^2\right) dw \quad (4)$$

Figure 14 shows the calculated $f(u)$ curves for several values of m and the results of measurements. The close agreements of the theoretical and experimental results seen in the figure may verify the validity of the proposed model for sound-programme signals.

It seems worthy of note that this model can admit any desirable spectrum alteration without affecting the form of the model's probability density function $f(u)$.

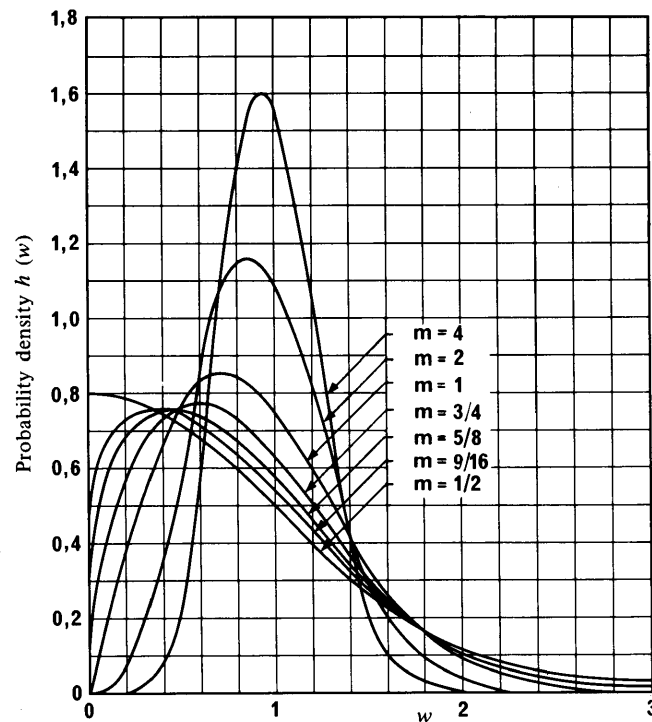


FIGURE 13 – The probability density function of the M-distribution

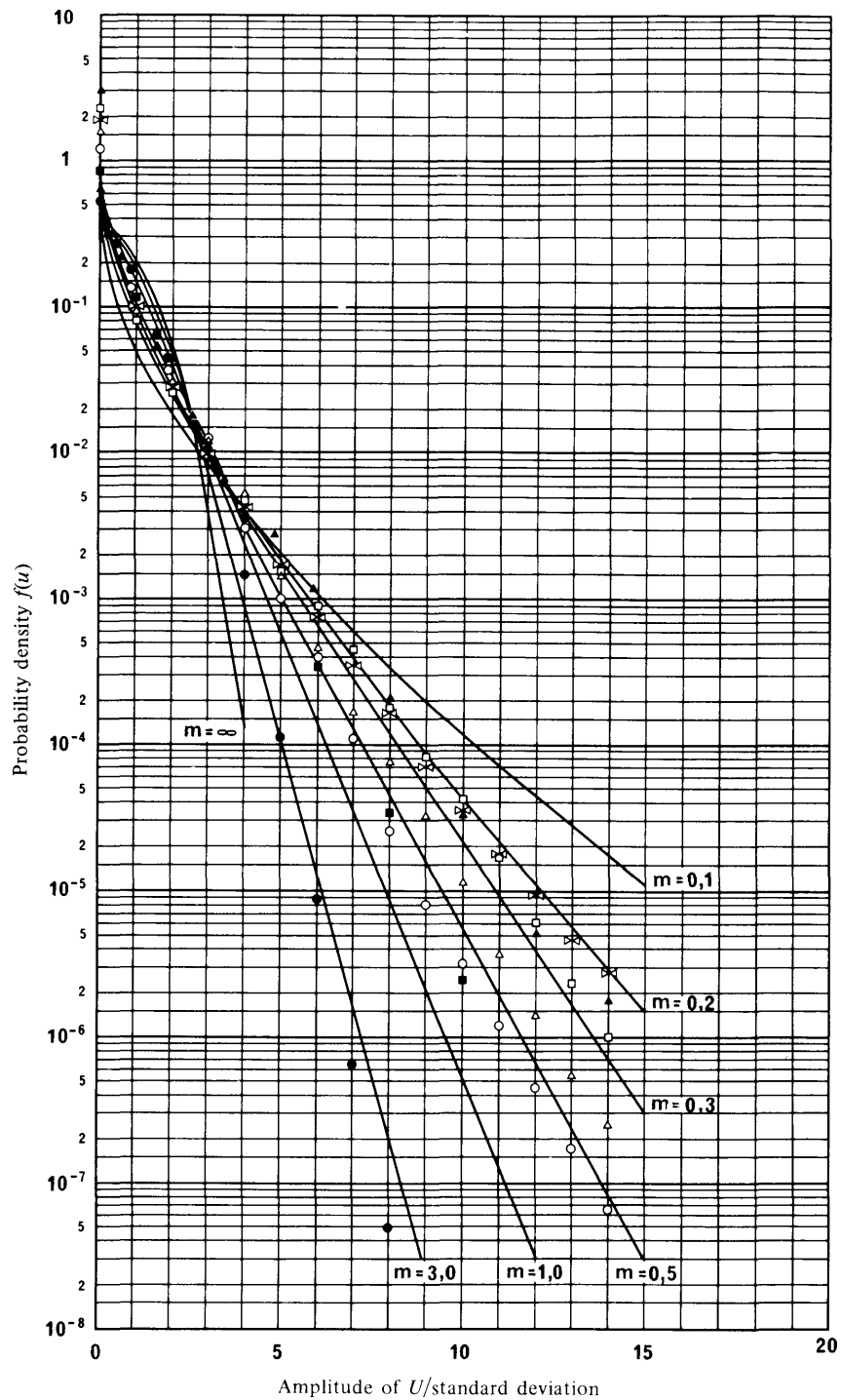


FIGURE 14 – The probability density functions of the $U(t)$ signal and sound-program signals

- Mozart, W.A.: symphony No. 29
- △ Beethoven, L.V.: symphony No. 2
- Beethoven, L.V.: symphony No. 9
- ◇ Tchaikovsky, P.I.: symphony No. 6
- Popular music (disk)
- ▲ Speech (positive) } measured by W. Weiss
- Speech (negative) }

3. A method of generating a model signal

A model signal using a $V(t)$ signal consisting of sine waves is here presented as an example.

3.1 The $V(t)$ signal with a constant r.m.s. value

To comply with the necessary property that the $V(t)$ signal be of the Gaussian process with a harmonious and consonant spectrum, the signal

$$V(t) = \sum_{i=1}^n C_i \cos(\omega_i t + \phi_i) \quad (5)$$

may be proposed, where C_i , ω_i and ϕ_i are constants, and the frequencies $f_i = \omega_i/2\pi$ should be determined so as to make irrational ratios which are very near to the simple integer ratios.

The coefficients C_i may be determined according to the spectrum of a sound-programme signal.

3.2 The $W(t)$ signal representing the fluctuating r.m.s. value

The $W(t)$ signal follows the M -distribution and the spectrum of the $W(t)$ may preferably be determined in accordance with the spectrum of a programme signal.

A simple method for generating the M -distributed signal may be depicted by the blocks B, C and D in Fig. 15.

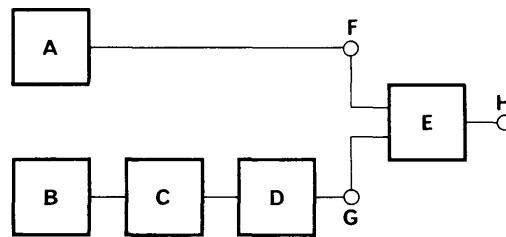


FIGURE 15 – A schematic block diagram for generating signals $U(t)$, $V(t)$, and $W(t)$

- A : sine wave generators and an adder of sine waves
- B : low frequency Gaussian noise generator
- C : full wave rectifier
- D : smoother
- E : multiplier
- F : $V(t)$ signal output
- G : $W(t)$ signal output
- H : $U(t)$ signal output

The full-wave rectifier C rectifies the output of the low-frequency Gaussian noise generator B, and a parallel CR smoothing circuit D smooths the output of the rectifier C. For the value of the capacitance $C = 0$, we obtain $m = 0.5$, and for $C = \infty$, we have $m = \infty$.

When the value of the capacitance C is set suitable for the detection of the envelope of the narrow band Gaussian noise, we have $m = 1.0$. The M -distribution with $m = 1.0$ represents the Rayleigh distribution.

The values of m other than these may be realized by choosing proper values of the capacitance C . Values of m less than 0.5 may be obtained with a modulating signal having an amplitude probability distribution giving larger amplitudes than are given by Gaussian noise at the lower probability levels.

3.3 The model signal $U(t)$

The $U(t)$ signal is obtained by multiplying $V(t)$ by $W(t)$ as shown in Fig. 15.

REPORT 820 *

**RELATIVE VALUES OF SOUND-PROGRAMME SIGNAL LEVELS
ESTABLISHED WITH THE VU METER AND WITH A
PEAK-PROGRAMME METER**

(Study Programme 17A/CMTT) **

(1978)

1. Introduction

This Report deals briefly with the different methods of sound-programme signal level control established using the vu meter in one case, and the peak-programme meter (ppm) in the other; and points out where the two methods give rise to differences in signal levels and operational problems as well as where misunderstandings of conventions could arise in an international sound-programme connection.

1.1 *The purpose of controlling sound-programme levels*

Sound-programme levels are controlled to obtain optimum use of a sound-programme transmission channel and to avoid interference to other channels in multiplex systems.

1.2 *Sound-programme level control*

The CCITT has recommended (Recommendation J.14) that at a point of zero-relative level (0 dBr), the sound-programme signals should be controlled by the broadcasting organization so that the peak level very rarely exceeds +9 dB, relative to the peak level reached by a sine wave of 0.775 volts r.m.s. across a 600 Ω resistive load at that point. (At the point of zero relative level this sine-wave level corresponds to 0 dBm0.)

2. Means of controlling sound-programme levels**2.1 *Methods in common use***

Two different methods are widely used to indicate relative signal levels in connection with the primary manual control of sound-programme signal levels.

Method A uses a vu (volume unit) meter [CCITT, 1968]. This meter basically gives an indication proportional to the average value of the programme signal.

The second, method B, uses a peak programme meter (ppm).

Note. — Information on these meters is given in Table I of Report 292.

2.2 *Response of the meters to sine-wave signals*

The vu meter in method A is usually connected to a relative level point in a transmission network so that a 0 dBm0 sine-wave test signal causes the meter to indicate at the reference mark on the scale (100% or 0 vu mark). It is usual practice to connect the vu meter at a +8 dBr point with an appropriate attenuator. At this point when programme signals cause the meter to indicate at the reference mark, the level on the circuit is +8 vu.

For an 0 dBm0 sine-wave test signal the meter (method B) is normally connected to a point in the transmission network so that the test signal level at that point causes the meter to indicate -9 dB with respect to that reference mark on the scale which corresponds to the peak of the programme.

2.3 *Response of the meters to sound-programme signals*

For real programme signals the reading of the two meters may be significantly different. However, for continuous complex signal waveforms with peaks occurring at intervals shorter than the integration time of the meters and having peak to average ratios of about 12 dB, both the vu meter and the peak-programme meter would show approximately the same deflection with respect to their peak programme reference marks, (e.g. they may each show a deflection 2 dB below their reference marks).

3. Setting-up levels**3.1 *Method "A" (vu meter) setting-up practice***

It is usual for test tones to be specified in terms of their level at a point of zero relative level, for example, -12 dBm0, 0 dBm0, etc.

* This Report should be brought to the attention of Study Group 10 and Study Groups IV and XV of the CCITT.

** Kyoto, 1978.

In the absence of limiters and compression devices a test signal of 0 dBm0 usually results in 40% modulation of a sound-broadcasting transmitter. In North America this corresponds to 31.6% modulation (in sound-programme transmission in North America this is commonly known as the “+8” level point). It should be noted that North America uses a test tone with an electrical value of 0 dBm which in relation to values given in CCITT Recommendation J.14, is a -8 dBm0 test tone.

3.2 *Method B (peak-programme meter) setting-up practice*

The practice normally adopted by the EBU and OIRT is to adjust the level so that maximum sound-programme level is 9 dB above the indicated level of a 0 dBm0 sine-wave signal.

4. **Interconnection of sound-programme circuits**

Provided that CCITT Recommendations J.14, N.21 and N.23 are observed, programmes derived by both monitoring methods may be directly interconnected.

In practice controlled sound-programme signals derived by the two monitoring methods may have different dynamic ranges and peak-level values at a point of zero relative level, and may not be compatible with CCITT Recommendation J.14.

These departures from the Recommendation, especially excessive peak levels, which occur with method A, can cause distortion and cross-talk into bearer systems. Low average levels, due to these control difficulties may result in poor signal-to-noise ratios at other points in the transmission system.

5. **Range of peak levels occurring in sound-programme signals**

5.1 *Evaluation of readings on ppm and vu meters*

Although circuits may be interconnected with a minimum of difficulty if connected as in § 4, there is a considerable difference in reading the ppm meter compared with the vu meter. For speech the ppm meter may indicate 10 to 15 dB higher than the vu meter, while for different types of music this will range from 4 to 10 dB. If both meters are read by the same operator this should be realized to avoid confusion.

It should be noted that the ppm gives a better control of signal voltages and it is evident that a move towards the ppm method would be highly beneficial [Schmid, 1977].

5.2 *Headroom or margin*

An important consideration in relation to the methods of monitoring and control is the range of voltage encompassed by a programme signal at a point of defined relative level. Report 491 gives results in Table II and in Figs. 4, 5 and 6 which relate to the voltage range encompassed at a point of zero-relative level controlled by method B. Further study would be required to find correspondingly detailed information about method A but it is believed that for a programme level of +8 vu, programme peaks frequently reach +18 dBm. No general conclusions are drawn as yet, but it is the intention of CMTT ultimately to define a conventional loading signal representative of sound-programme signals using information drawn from Report 491.

Tentatively it is suggested that at a point of zero-relative sound-programme level employing method B, programme signal peaks may extend to a level corresponding to about +15 dBm (level exceeded with probability of about 10^{-5}). A margin above this value is highly desirable to allow for misalignment, temperature, etc. A corresponding tentative figure for a programme level of +8 vu in method A is +21 dBm.

5.3 *Reduced headroom caused by pre-emphasis*

The tentative range given in § 5.2 relates, of course, to signals as supplied by the broadcaster. When pre-emphasis is used, or when frequency-sensitive/level-operated devices are employed, these characteristics will be modified. Some information is available in Report 491 about signals emphasized in accordance with CCITT Recommendation J.17 (1.5/800) employed in conjunction with method B; and, whilst it may be premature to draw conclusions, it is tentatively suggested that at a point of zero-relative level programme signal peaks may reach a level of +17 dBm (for probability of 10^{-5}). Again tentatively, it is suggested that peaks of pre-emphasized signals would (if unclipped), reach a level of +24 dBm at a programme level of +8 vu.

6. Conclusion

This subject requires further study and administrations are invited to provide contributions on this subject.

REFERENCES

CCITT [1968] Supplement 11 to Recommendation P., Vol. V, White Book, Mar del Plata.

SCHMID, H. [1977] Audio programme level, the vu meter, and the peak-programme meter. *IEEE Trans. Broadcasting*, Vol. BC-23, 1.

REPORT 493-3

COMPANDORS FOR SOUND PROGRAMME CIRCUITS *

(Question 17/CMTT)

(1970-1974-1978-1982)

1. Introduction

Documents provided by Canada and the United Kingdom together with some additional material from the Federal Republic of Germany, have been studied under this Study Programme. Paragraph 6 of this Report describes the subjective testing of two compandors – one of a new type being brought into use in Canada, the other conforming to § 1 of CCITT Recommendation J.31.

It is stressed that the results given only apply to these particular types of compandors.

2. The problem being studied

The maximum degree of noise reduction obtainable by the use of a compandor is achieved only in the absence of a signal. In the presence of a signal, the gain of the expander rises, and the amount of noise from the transmission circuit reaching the output of the system is therefore increased. The broadcasting authorities have been asked to provide information on the amount of such “expanded” noise that can be accepted when a compandor is used on a programme transmission circuit, so as to provide information relevant to the study of Study Programme 17B/CMTT (Kyoto, 1978). It should be considered as to what extent the load on the line is increased by the compressor, because a compandor reduces the noise by increasing the power of those portions of the programme which are of smaller energy.

3. Factors determining noise impairment

The subjective impairment produced by noise in a programme circuit cannot be expressed, as in a telephone system, by a readily definable quantity such as a percentage articulation score. The extent to which the listener’s enjoyment of the programme is disturbed by the presence of noise depends on a number of factors, few of which are capable of being precisely specified. In the case of a compandored circuit the situation is complicated by the fact that the “expanded” noise level fluctuates with programme level and may on that account be more obtrusive than a steady noise of the same average power.

The main factors which affect audibility of “expanded” noise may be classified as follows:

- the spectrum of the noise in relation to the spectrum of the programme material;
- any variation in companding action with frequency, e.g. whether the companding action is applied uniformly over the full frequency range of the system, or whether the frequency range is split into two or more bands, each provided with an independent compressor and expander; in the latter case, the number of bands and their location in the spectrum is relevant;
- the gain variation of the expander as a function of signal level, e.g. whether this gain – and hence the “expanded” noise – increases gradually or rapidly with signal level, and, if the latter, whether the transition from the minimum gain to the maximum gain takes place at low or at high levels;
- the dynamic characteristics of the expander, which determine the length of time for which any increase in the expander gain – and hence in the noise – persists after cessation of the signal which gave rise to it.

* For the definition of absolute power, relative power and noise levels, see Recommendation 574.

For additional clarity, the "expanded" noise should be divided into two categories:

- *Trailing noise*: objectionable noise occurring during a rapid transition from a high to a low level signal through an expander. The recovery time of the expander (to the new level) allows the transmission of a random sequence of noise bursts following each transient.
- *Cross-modulation noise*: a rapid decrease of a low-frequency, high-level signal transmitted simultaneously with a high-frequency, low-level signal through a compandored channel. The combined action of the compandor and the pre-distorter-restorer will cause the high-frequency, low-level signal to be masked by noise during the compandor recovery interval.

4. Measurement of "expanded" noise

The "expanded" noise is present only when the programme signal is present, and cannot, therefore, be measured directly. It is conceivable that after a particular operating condition has been established by subjective tests on the programme as representing, for example, "just perceptible" impairment, one or more test signals could be substituted for the programme and the resulting noise measured by appropriate filtering. However, apart from the obvious difficulty of devising suitable test signals, it should be noted that the information obtainable from such a test is not in a form directly applicable in practice. The test should be such as to give a direct answer to the questions:

- whether a given type of compandor is usable under given circumstances,
- which of several types of compandor will allow the greatest amount of noise in the transmission system taking account of the load on the line?

It is therefore suggested that no attempt should be made to measure "expanded" noise as such, but rather that tests should be aimed at determining, for each type of compandor, the maximum amount of noise that may exist on the transmission system without the noise heard in the presence of a programme exceeding the tolerable limit. If, at any time, a corresponding maximum noise determination were carried out with the compandor in operation but in the absence of programme, the difference between the two figures (in dB) should give a measure of "expanded" noise.

5. A practical method of measurement

A circuit is arranged so that the same programme material can be transmitted alternatively through either a path containing the compandor under test, or through a path without a compandor. Independently controllable noise can be applied to either path, that applied to the compandor being injected between the compressor and the expander.

With the noise power applied to the compandored path held constant, the noise power applied to the other path is varied until the listener's subjective assessment is the same for both paths. The difference between the noise powers applied to the two paths is the subjective reduction of noise by the compandor. This can be repeated for several values of the noise power applied to the compandored path.

If the same test is made in the absence of a programme (in which case the test can be made also with measuring sets), the difference between the noise powers applied to the two paths is the objective reduction of noise by the compandor. The objective reduction is in general somewhat higher than the subjective reduction depending on the compression characteristics used. The difference between the figures (in dB) is the above-mentioned measure of "expanded" noise.

Care must be taken that the programme material used for the tests is as noise-free as possible, so as to avoid false results. In particular, tape-recorded programmes may be troublesome.

Tests carried out by the Italian Administration using a compandor operating in the carrier frequency range, showed that this method of measurement is satisfactory. Similar compandors are used by the Federal Republic of Germany and by a number of other European Administrations.

6. Subjective tests performed on certain types of compandor

6.1 An earlier Document [CCIR, 1966-69] described the results of a subjective test by Canada on a type of compandor operating on the whole audio-frequency band for high-quality sound transmission.

A new type of dual-band 15 kHz compandor is being brought into use in Canada which has an improved performance and is described in [CCIR, 1974-78].

The essential features of the subjective test as described by Canada on the new type of compandor were:

- the path between compressor and expander was band limited to 8 kHz for the purpose of this test;
- the attack and recovery times of the compandor, as defined in CCITT Vol. III Fascicle III-2, Recommendation G.162, are 5 and 22.5 ms respectively, (20 and 75 ms respectively for the compandor as per [CCIR, 1966-69]);
- two different series of tests were conducted giving similar results:
 - (a) two panels of expert listeners equipped with high-quality headphones;
 - (b) expert listeners in an acoustically soft auditorium.

The results showed that the type of programme material used had a substantial effect on the “just detectable” impairment by expanded noise.

With a normal programme level, 19 to 25 dB more noise could be tolerated with most of the programme material used in the tests, which included Dixieland music and the male voice. (This is an 11 to 17 dB improvement over the compandor as per [CCIR, 1966-69].)

Less benefit (14 to 17 dB) is given for types of material for which, because of its masking effect, a higher noise level (in the absence of a compandor) is acceptable.

Further tests showed that up to six compandor links in tandem did not noticeably degrade the quality of programme material.

The programme-weighted signal-to-noise ratio which causes “just detectable” impairment for the more susceptible type of programme is in the range from 30 to 35 dB at the expander input. This is below the signal-to-noise ratio at which a transmission link would normally switch to an alternate facility in Canada.

Measurements of the spectral characteristics of programme signals fed through the dual band compandor showed that the compressor does alter the spectral character of the signal.

Depending on the programme signals produced, the higher audio frequencies could be transmitted 6 to 7 dB higher at compressed level as compared with a single band compressor. Caution must be taken in view of “head room” and pre-emphasis characteristics. In particular, the use of dual band compandors in conjunction with diplexed FM sub-carrier equipment which also employs signal pre-emphasis, should be treated with caution.

6.2 *Tests made on a system conforming to CCITT Recommendation J.31*

6.2.1 *Introduction*

This note describes subjective tests carried out on a system which meets CCITT Recommendation J.31, principally to grade any impairment of reproduced programme resulting from modulation of simulated bearer circuit noise by the action of the compandor incorporated in the equipment. The investigation was carried out in stringent listening conditions using the most critical programme material available. Tests were also conducted to study impairment due to modulation by a programme of interfering tones such as might result from carrier leaks in a frequency-division multiplex system.

For all the tests, the compandor and the pre- and de-emphasis in the equipment were in circuit. The 10 kHz and 14 kHz notch filters, provided to reduce the level of carrier leaks at the most likely frequencies, were switched out of circuit. The results of the tests are presented in this Report, but no conclusions are drawn.

6.2.2 *Preliminary tests*

The programme material used for the main tests was chosen by a few experienced observers in preliminary tests which used apparatus simulating the effects of programme modulation of background noise, a short passage of solo piano being selected as the most critical material available. The same passage was also found to constitute a stringent test for impairment due to programme-modulated tone at 6 kHz, and was, in fact, used for all the formal tests.

The preliminary tests also indicated that limits determined for monophony could also be safely applied to stereophony; the main tests were therefore carried out only monophonically and, as two complete sets of two-channel equipment were available, it was thus possible to arrange up to four modems in cascade.

6.2.3 *Test on the system*

6.2.3.1 *Test conditions*

For the present investigation, the sending and receiving equipments were connected back to back, and white noise, or a sinusoid simulating a carrier leak, was injected at various levels at the junction. The tests using injected noise were carried out for a single modem and for four modems in cascade (the levels

of injected noise then being such that the individual modems had the same output noise level); the test conditions being specified in terms of the noise level * at the audio output. The frequencies of the injected sinusoids were such as to produce tones of 2 kHz, ** 6 kHz, ** 10 kHz, or 14 kHz at the audio output, and the test conditions were specified by the level of tone at the output, measured with a selective receiver.

The subjective tests were carried out in a listening room — usually used for critical tests — having a volume of some 85 cubic metres and a mean reverberation time of about 0.3 second. The test programme was reproduced on a wide-range monitoring loudspeaker and the maximum indicated sound level, measured at the centre of the group of observers, was 83 dBA. ***

Most of the tests were carried out by six observers drawn from a group of about nine engineers experienced in assessing the quality of sound programmes. As a check, however, one set of tests was also carried out with six office staff, most of whom had little experience of critical listening. A number of the observers were around 40 years of age, and when investigating the effects of a simulated carrier leak producing a 14 kHz tone, it was at first difficult to get reliable results; this test was therefore repeated, substituting a group of observers aged 25 years or under.

6.2.3.2 Test procedure

The observers attended test sessions in groups of six. A brief period of introduction preceded the tests in each session. First, to familiarize the observers with the test material, the test item was played without added impairment; then, by way of example, the item was repeated with considerable impairment of the kind to be investigated in the test session. (Where two forms of impairment — for example, carrier leaks of different frequencies — were to be investigated in the same test session, examples of both were included in the introductory period.) Finally, before commencing the tests, the programme item was again presented without added impairment and the observers were asked to regard this presentation, including any unintended imperfections which might be present, as Grade 1 on the impairment scale to be used.

The tests which followed, covering a range of levels of noise or tone, were presented in arbitrary order. The observers were asked to grade the impairment of each test item according to the EBU 6-point scale given below (Report 405, Note 8).

<i>Grade</i>	<i>Impairment</i>
1	Imperceptible
2	Just perceptible
3	Definitely perceptible, but not disturbing
4	Somewhat objectionable
5	Definitely objectionable
6	Unusable (intolerable).

The audibility of interference at a single frequency depends very largely on the characteristics of the loudspeaker used and on the acoustic environment; it would therefore be difficult, except by a more comprehensive investigation than was possible in the present case, to obtain results of universal validity for programme modulated tones. In an attempt to circumvent this difficulty, each test item containing a programme modulated tone was followed by a further test in which the observer was asked to grade the impairment due to the steady tone in the absence of programme. The results of the two tests, presented together, enable the effect of the programme on its impairment to be separated from other factors which determine the subjective grade for a given absolute level of interference.

6.2.4 Results

Figure 1 gives the assessments of impairment due to simulated white noise on the bearer circuits. The mean subjective grade of impairment is plotted against the quiescent noise level at the output of the system. For each point plotted, the standard deviation of the mean is also indicated. Figure 1a shows the result, for one modem, of two separate tests, A and B, carried out by six experienced observers; while Fig. 1b shows the corresponding results obtained in a single test by six inexperienced observers. Fig. 1c gives the results from six experienced observers, obtained for four modems in cascade.

* For the tests, the ratio of maximum signal-to-weighted noise (CCIF 1949 weighting) was measured with a BBC peak programme meter. Where the measured signal-to-noise ratio is R dB, the figure for noise level given in this contribution is $-(R - 4)$ dBm0ps, and the noise axes on Figs. 1 and 2 are labelled accordingly.

** The manufacturer advises users of the equipment to avoid use of carrier channels which contain the less common carrier leaks resulting in 2 kHz or 6 kHz beat tones.

*** Measured with a sound level meter conforming to IEC Publication 123, with time-constant "slow".

The noise level for a single modem without simulated bearer circuit noise was about -71 dBm0ps, and this figure thus represented a limiting condition in Figs. 1a and 1b. The corresponding limiting condition for four modems in cascade, Fig. 1c, was about 6 dB higher, as expected.

The test results for experienced and inexperienced observers are in reasonable agreement, particularly for low grades of impairment. For example, both sets of results indicate that a quiescent noise level of about -68 dBm0ps is required to attain Grade 2, just perceptible impairment.

It must be emphasized that the gradings of impairment due to programme modulated noise, discussed above, were obtained with the most critical programme material available and in stringent listening conditions; in less demanding circumstances, the observers would undoubtedly have been more tolerant of the impairments produced. For instance, the impairment caused by programme modulation of bearer circuit noise will be concealed to some extent by the continuous random noise present in domestic "Hi-fi" equipment.

Comparing Figs. 1a and 1c, it will be seen that the gradings obtained for one modem and for four modems in cascade are approximately the same for a given value of quiescent noise level, the differences in mean grading being, in general, less than the standard deviation of the mean.

Figure 2 gives the assessment for tones at 2 kHz, 6 kHz, 10 kHz and 14 kHz. Here, the mean grade of impairment is plotted as a function of the quiescent level of tone. In Fig. 2a, for which the frequency of the tone was 2 kHz, the impairment gradings given during louder parts of programme differ very little from those during programme pauses; this indicated that the level of the 2 kHz tone, while being inevitably raised by the presence of the programme signal, was also partially masked by it. It is probable, however, that with alternative programme material, the masking effect could have been less and that, in these circumstances, the effect of programme modulation on 2 kHz tone would have been more marked.

The impairment due to carrier leaks giving tones of higher frequency was, however, found to be more obtrusive during louder parts of programme than during pauses. Fig. 2b and Fig. 2c indicate that, at subjective Grade 2, for example, the presence of the test programme lowered the tolerable quiescent level of the interfering tone by approximately 10 dB and 12 dB for frequencies of 6 kHz and 10 kHz, respectively.

Figure 2d shows the results for 14 kHz tone. It will be seen that these are less precise than those for the 6 kHz and 10 kHz tones, but are roughly similar in form and suggest a difference, at the Grade 2 level, of some 10 dB between the tolerable interfering tone levels with and without programme. For the tests for which the results are given in Fig. 2, no noise was added to the bearer circuit.

6.2.5 *Test on stereophonic sound-programme circuits*

Subjective quality assessments on the 15 kHz stereophonic sound-programme transmission equipments were carried out [CCIR, 1978-82], using the compandor characteristics specified in CCITT Recommendation J.31. Test results showed that the compandor causes fluctuation of sound image location when pianoforte solo or male speech is applied. It was found that very high accuracy is required for the characteristics of both compandors and transmission lines to suppress this fluctuation to an acceptable extent.

A new compandor characteristic which satisfies the requirements for both noise and sound image location is shown in [CCIR, 1978-82].

7. Further information

Further information on compandors is available in the documents of the CCITT which are related to Question 1 of Study Group XV of the CCITT.

Information regarding a general method proposed for the subjective assessment of compandor performance is given in Report 623.

8. Conclusion

It is concluded that the study can be best continued under the Question 17/CMTT.

Figure 1(a) is a line graph showing the relationship between Quiescent noise level (dBm0ps) on the x-axis and Mean grade on the y-axis. The x-axis ranges from -76 to -41, and the y-axis ranges from 1 to 6. A smooth curve is fitted to the data points, which include vertical error bars. The data points are as follows:

Quiescent noise level (dBm0ps)	Mean grade
-71	1.4
-69	1.6
-67	1.9
-66	2.8
-61	4.2
-56	4.4
-51	5.3

Figure 10(b) is a scatter plot with a fitted curve showing the relationship between Mean grade and Quiescent noise level (dBm0ps). The x-axis represents Quiescent noise level (dBm0ps) and ranges from -76 to -41. The y-axis represents Mean grade and ranges from 1 to 6. The data points, represented by open circles with vertical error bars, show a positive correlation. A solid curve is fitted to the data points.

Quiescent noise level (dBm0ps)	Mean grade
-71	1.9
-70	1.5
-69	1.6
-66	2.3
-61	3.4
-56	4.0
-51	5.4

Figure 10 is a line graph with error bars showing the relationship between Mean grade and Quiescent noise level (dBm0ps). The x-axis represents Quiescent noise level (dBm0ps) and ranges from -76 to -41. The y-axis represents Mean grade and ranges from 1 to 6. The data points show a positive correlation, with a smooth curve fitted to the data.

Quiescent noise level (dBm0ps)	Mean grade
-64.5	3.0
-63.5	3.5
-61.5	3.5
-58.5	4.2
-54.5	5.2
-50.5	5.8
-49.5	5.9

FIGURE 1 – Impairment versus quiescent noise level:
simulated white noise on bearer circuit

○ standard deviation of mean ($\pm$)

(a) one modem: experienced observers $\begin{cases} \circ \text{ Test A} \\ \bullet \text{ Test B} \end{cases}$

(b) one modem: inexperienced observers

(c) four modems in cascade: experienced observers

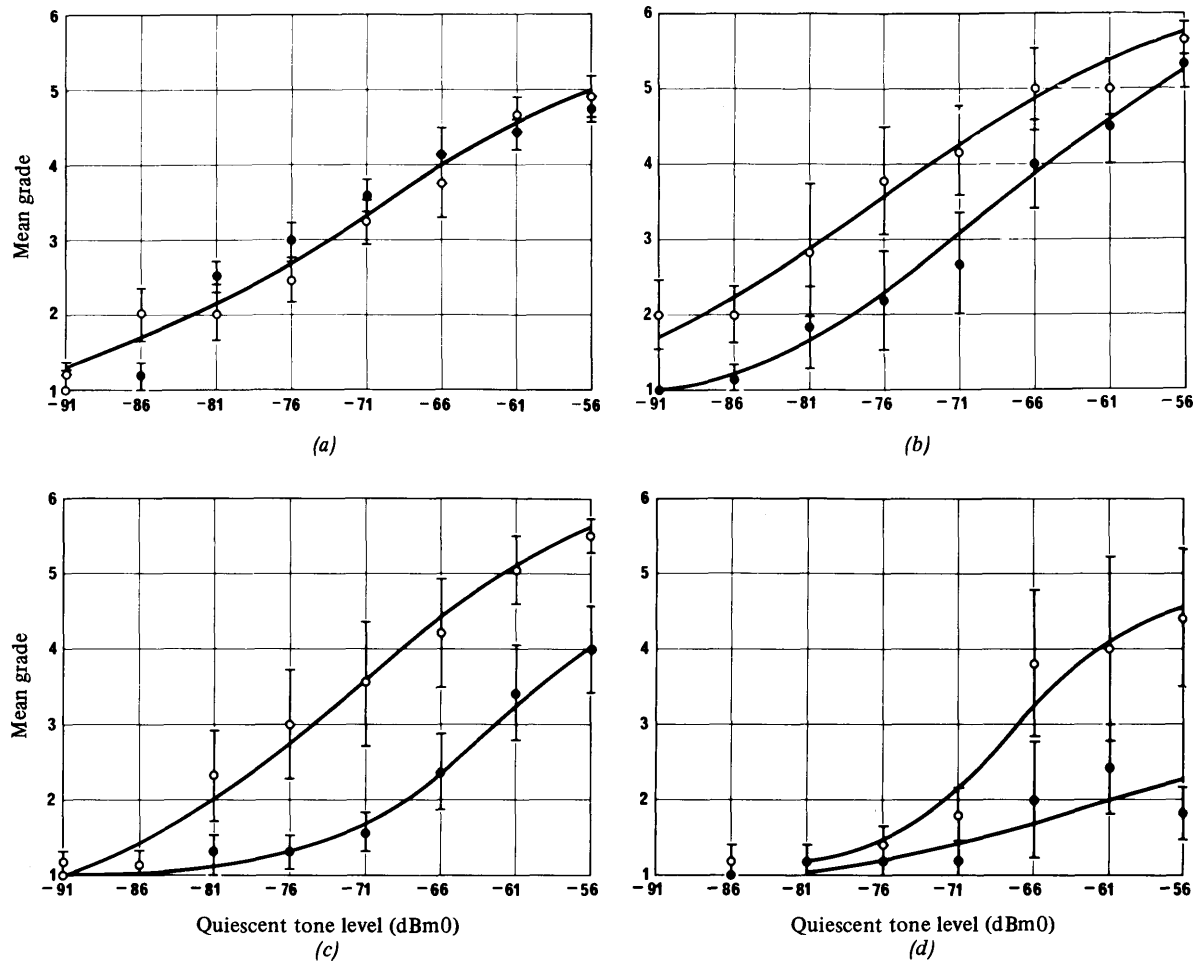


FIGURE 2 – Impairment versus quiescent tone level: simulated carrier leak

- standard deviation of mean ($\pm$)
- with programme
- without programme

Frequency of tone resulting from carrier leak: 2 kHz: Fig. 2 (a)
 6 kHz: Fig. 2 (b)
 10 kHz: Fig. 2 (c)
 14 kHz: Fig. 2 (d)

REFERENCES

CCIR Documents

- [1966-69]: CMTT/9 (Canada).
- [1974-78]: CMTT/70 (Canada).
- [1978-82]: CMTT/60 (Japan).

RECOMMENDATION 605

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS
SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT

(Study Programme 17D/CMTT)

(1982)

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:

$$D_n = D_3(n/3)^{1/h} \quad (1)$$

- for absolute levels:

$$D_n = D_3 + \frac{20}{h} \log(n/3) \quad (2)$$

- for logarithmic ratios:

$$D_n = D_3 - \frac{20}{h} \log(n/3) \quad (3)$$

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:

$$D_n = D_3 - \frac{20}{h} \log(L/l) \quad (4)$$

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.

TABLE I

<i>n</i>	Formula (1)			Formulas (2) and (3)		
	$\left(\frac{n}{3}\right)^{1/h}$			$\frac{20}{h} \log \left(\frac{n}{3}\right)$		
	<i>h</i> = 1	<i>h</i> = 3/2	<i>h</i> = 2	<i>h</i> = 1	<i>h</i> = 3/2	<i>h</i> = 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).

REPORT 821-1 *

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT **

(Study Programme 17D/CMTT)

(1978-1982)

Studies under Study Programme 17D/CMTT are ultimately concerned with providing a text for Recommendation 605 to give — for planning and maintenance purposes — a means of estimating the transmission performance of circuits, the configuration of which differs from that of the hypothetical reference circuit.

[CCIR, 1974-78a] gives some considerations concerning random noise. The limits which apply to a hypothetical reference circuit (hrc) (see for instance Recommendation 505, § 3.1.5) are given in the form of values not to be exceeded for more than a certain percentage of a month.

Estimated noise power levels expected in practice, are given in Recommendation 504, Annex II.

Paragraphs 2 and 3 of Recommendation 393, which concern the allowable noise power in the "line" (baseband) portion of an hrc for telephony over radio-relay systems, lay down principles for sub-dividing analogous limits; the limit for 20% of a month is sub-divided on a power basis and for smaller percentages on a time basis. This has been found to accord well with observed behaviour, and it is therefore proposed that the same principles be applied also to sound-programme circuits.

As regards the sub-division of the total noise power between multiplex and line equipment, two cases have to be distinguished:

- where a sound-programme channel is set up over a group link in a telephony multiplex system; here there will be an appreciable noise contribution from all higher modulation stages, and CCITT Recommendation G.223 will apply,
- a sound-programme channel provided above a radio-relay television channel by frequency modulating a sub-carrier which can then be directly fed into the baseband without further modulation: no sub-division has been fixed in Recommendation 402 or in Report 289.

Different sub-divisions of noise between modulation and line equipment could have some effect in margins to be allowed for fading.

Note. — When sub-dividing the noise tolerance recommended for the hrc between terminal and line equipment, great care should be taken to choose the two values as close as possible to those of real equipment, as any deviation inevitably will lead to more stringent requirements than those of existing Recommendations for sound-programme transmission.

For establishing maintenance limits, great accuracy is perhaps not required — compare the remarks in CCITT Recommendation G.226. It might be possible to find a method of calculation giving an approximate result which would be independent of the transmission technique involved. An example of a calculation was given in [CCIR, 1974-78a] but was felt by some administrations to go beyond the mandate of the question.

Paragraph 1 of Annex I summarizes [CCIR, 1974-78b] in which the measures considered by the OIRT to ensure good transmission quality for circuits longer than 2500 km are given. In § 2 of this Annex some methods (with examples) used in the OIRT for the summation of the numerical values of transmission parameters are given from [CCIR, 1974-78c]. These are noted here to assist further study.

Recommendation 605 is an answer to the request of Study Programme 17D/CMTT for analogue circuits.

Administrations are invited to confirm the recommended values for h as from their experience with actual circuits and to study those characteristics that are still outstanding in Table II of Recommendation 605.

* This Report should be communicated to CCITT Study Group IV.

** For the definition of absolute power, relative power and noise levels, see Recommendation 574.

REFERENCES

CCIR Documents

[1974-78]: a. CMTT/52 (L. M. Ericsson); b. CMTT/260 (OIRT); c. CMTT/259 (OIRT).

BIBLIOGRAPHY

[1978-82]: CMTT/233 (Japan).

ANNEX 1

1. Sound-programme transmission over circuits longer than 2500 km

In order to ensure high quality transmission of sound over circuits longer than 2500 km it is necessary to fulfil the following conditions:

1.1 International sound-transmission circuits over 2500 km should preferably contain no more than three low-frequency modulation sections.

1.2 The standards specified for amplitude frequency, non-linearity and amplitude amplitude distortions are then taken to be the same as for a 2500 km long hypothetical reference circuit; provided that the circuit does not contain audio sections of appreciable length.

1.3 The standards specified for weighted noise not exceeded for 80% of the time are determined according to:

$$P_l = P_{2500} + 10 \log \frac{l}{2500} \quad \text{dB} \quad (1)$$

where P_{2500} is the noise level in a channel of reference length, taking into account the noise introduced by the equipment forming the circuit, and l is the length of the circuit in km.

1.4 The standard specified for intelligible cross-talk on long distance circuits may be the same as for the reference circuit.

1.5 If the international sound circuit consists of more than three low-frequency modulation sections the statistical summation law should be applied when estimating the transmission performance of the entire circuit.

2. Statistical summation laws for the transmission parameters of sound-programme circuits

This Annex gives rules for the determination of the total values of the parameters of a sound circuit consisting of several modulation sections with different admissible values of their parameters.

Distinct summation laws should be applied to parameters with two-sided tolerances and to those with one-sided tolerances.

The proposed summation laws take into account both the probability and the statistical summation laws as well as the physical rules for adding up signals of various types.

2.1 Parameters with two-sided tolerances:

- amplitude/frequency characteristic;
- level variation;
- difference in gain between the A and B channels during stereophonic transmission;
- difference in phase between the A and B channels during stereophonic transmission.

Summation of the mean values of these parameters is linear, while for the admissible limit values the geometric (quadratic) summation law is applied:

$$m_s = m_1 + m_2 + \dots + m_n \quad (2)$$

$$a_s = \sqrt{a_1^2 + a_2^2 + \dots + a_n^2} \quad (3)$$

where

- m_i ($i = 1 \dots n$): admissible mean value of section i ,
- m_s : resulting mean value of the total sound-transmission circuit,
- a_i ($i = 1 \dots n$): admissible deviation from the mean value of section i with specified probability,
- a_s : resulting deviation of the total sound-transmission circuit with the same probability as for the individual sections.

Example of calculation of the amplitude/frequency characteristic:

section 1: (0 ± 1.0) dB
 section 2: (-1.0 ± 2.0) dB
 section 3: (0 ± 1.0) dB.

Summation of the mean values:

$$M = 0 - 1.0 + 0 = -1.0 \text{ dB.}$$

Summation of the deviations:

$$a_s = \sqrt{1.0^2 + 2.0^2 + 1.0^2} \text{ dB} = 2.45 \text{ dB}$$

Total tolerance: (-1.0 ± 2.45) dB.

In view of the fact that in most cases only the admissible limit values are known and not the mean values, the limit total values may be roughly determined by geometric summation of the individual limit values with Formula (3).

Example of calculation:

section 1: ± 1.0 dB
 section 2: $+1.0 \dots -3.0$ dB
 section 3: ± 1.0 dB

$$a_{s0} = \sqrt{1.0^2 + 1.0^2 + 1.0^2} \text{ dB} = +1.73 \text{ dB}$$

$$a_{sn} = \sqrt{(-1.0)^2 + (-3.0)^2 + (-1.0)^2} \text{ dB} = -3.32 \text{ dB}$$

Total tolerance: $+1.73 \dots -3.32$ dB.

A statistically more accurate method is linear summation of the mean values and geometric summation of admissible deviations from the mean value.

2.2 Parameters with one-sided tolerances:

- signal-to-noise ratio (see Note) (weighted and unweighted);
- cross-talk ratio between the A and B stereo channels;
- non-linearity distortions (harmonic-distortion factor and difference-tone factors).

For these parameters it is preferable to use geometric summation of limit tolerances.

For signal-to-noise (see Note) and cross-talk ratios, linear summation of powers is applied.

If the statistical summation law is applied, it is necessary to convert the ratio given in dB into the corresponding voltage in mV. These voltages are then introduced into the summation Formula (3) and the total ratio is calculated from the total voltage.

Example of calculation: weighted cross-talk ratio (see Note) referred to $+6$ dBm (600 Ω).

section 1: $60 \text{ dB} \cong -54 \text{ dBm} \cong 1.55 \text{ mV}$
 section 2: $58 \text{ dB} \cong -52 \text{ dBm} \cong 1.95 \text{ mV}$

$$a_s \approx \sqrt{1.55^2 + 1.95^2} \text{ mV} \approx 2.49 \text{ mV}$$

Total value: noise level $2.49 \text{ mV} \cong -49.9 \text{ dBm}$
 cross-talk ratio: 56 dB

For non-linearity distortions summation according to the amplitude and the phase should be used. However, all measurement methods concern solely the amplitudes. That is why before applying the geometric summation law, it is necessary to verify that all sections show an equal tendency to phase rotation. Should this prove impossible, it is preferable to use the $3/2$ summation law,

$$a_s = \sqrt[3]{a_1^{3/2} + a_2^{3/2} + \dots + a_n^{3/2}}^2 \quad (4)$$

and the results obtained will be more reliable.

Example of calculation of difference tone d_2 :

section 1: 0.2%
 section 2: 0.5%
 section 3: 0.2%

Geometric summation: $a_s = \sqrt{0.2^2 + 0.5^2 + 0.2^2} \% = 0.57 \%$

3/2 summation: $a_s = \sqrt[3]{(0.2 \sqrt{0.2} + 0.5 \sqrt{0.5} + 0.2 \sqrt{0.2})^2} \% = 0.65 \%$

Note. — These terms and parameters respectively, are not in accordance with the terminology of CCIR Recommendations.

RECOMMENDATION 474-1

MODULATION OF SIGNALS CARRIED BY SOUND-PROGRAMME CIRCUITS BY INTERFERING SIGNALS FROM POWER SUPPLY SOURCES

(Study Programme 17F/CMTT)

(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 (see Study Programme 17F/CMTT).

Note 3. — For design purposes this limit should be taken as applying to the hypothetical reference circuit of length 2500 km.

REPORT 647-2 *

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS **

(Question 18/CMTT)

(1974-1978-1982)

1. Introduction

This Report includes a description of digital network composition and gives details on several coding methods. Annex II contains information concerning subjective quality assessments for signals encoded with instantaneous companding systems.

2. Network composition

This section describes the composition of present and future international and/or national networks used for transmission of digital sound programme signals. The information is based on Report 647-1 of study period 1974-1978 and new contributions received during study period 1978-1982, which are quoted in the references. In particular, information from CCITT Study Group XVIII provides detailed aspects concerning the ISDN [CCIR, 1978-82a, b, c, d, e].

* This Report should be brought to the attention of CCITT Study Groups XV and XVIII.

** For the definition of absolute power, relative power and noise levels, see Recommendation 574.

2.1 Integrated services digital network

Digital sound programme signals are expected to be routed on digital bearer circuits which form part of the integrated services digital network (ISDN) with access at suitable digital interfaces [CCIR, 1978-82f, a]. In this case, CCITT Study Group XVIII is responsible for the definition of the multiplexing structure. It has been generally accepted that the overall bit rate required for a single digital sound programme channel (including coded samples, protection bits and bits for other purposes such as justification bits in the case of asynchronous access) should be an integer multiple of 64 kbit/s, so that the transmission capacity required is exactly that of several PCM telephone channels (e.g. 6 or 3 telephone channels) [CCIR, 1978-82g, h].

The adopted primary multiplex bit rates are 2048 kbit/s (CCITT Recommendation G.732) and 1544 kbit/s (CCITT Recommendation G.733). The evolution of the integrated services digital network [CCIR, 1978-82i, j, e] and additional digital hierarchy levels [CCIR, 1978-82f, k] introduce new considerations which are being studied by CCITT Study Group XVIII. Several aspects of these studies (e.g. transmission performance objectives) [CCIR, 1978-82j], should be considered in the CMTT.

CCITT Study Group XVIII drafted a revision to the definition of an ISDN, which is contained in Recommendation G.702. The revised definition for the ISDN now reads "A network evolved from the telephony IDN that provides end-to-end digital connectivity to support a wide range of services, including voice and non-voice services, to which users have access by a limited set of standard multipurpose customer interfaces".

As a first step in this direction a number of *information types* have been defined so that, e.g. ISDN access types and network models can be determined prior to detailed agreement on individual services. Of interest to CMTT are the information types designated as *u* information and defined as follows:

u_1 : digital information at bit rates greater than w up to the primary hierarchical level (1544/2048 kbit/s).

Note. — w information is composed of $n \times 64$ kbit/s with n in the order of 2 to 3.

u_2 : digital information at bit rates greater than u_1 up to the secondary hierarchical level (6312/8448 kbit/s).

u_3 : digital information at bit rates greater than u_2 up to bit rates of 140 Mbit/s.

These *u*-information types may be switched or non-switched. Digit sequence integrity is assumed to be required.

It is clear that sound programme signals fall under the *u* information types.

Furthermore it was agreed that every service/service category should be described by the following criteria:

- information type,
- information rate,
- traffic characteristics.

Study Group XVIII asks CMTT to consider submitting material regarding information types, bit rates and traffic characteristics as well as any other requirements/demands that sound programme services may have on the ISDN.

To perform the multiplexing of different digital signals, e.g. sound programme, telephone or other signals, originating in different locations, one administration [CCIR, 1978-82l] suggested the use of a special self-contained equipment providing "drop-insert" facilities conceptually similar to the external access equipment of CCITT Recommendation G.739.

2.2 Dedicated digital applications

In some particular cases, the digital sound-programme channels may be allocated by time division multiplex on circuits which are either part of the ISDN and dedicated to the use of broadcasting organizations [CCIR, 1978-82m, n, o] or part of separate dedicated networks [CCIR, 1978-82p, q, o].

The concept of "drop-insert" facilities given in § 2.1 is appropriate to the above application.

2.3 Network access

Examples of network access envisaged for sound programme signals are:

- a complete circuit at the primary level of the digital hierarchy is allocated to the transmission of sound signals from either a common source or several individual sources. The sound programme circuits are not individually routed via the ISDN (although the complete circuit may or may not be) and consequently, need not conform to the requirements of the Recommendations G.732 or G.733 of the CCITT;
- individual sound-programme circuits are routed over the ISDN and consequently must conform to the relevant CCITT Recommendations for primary PCM and digital multiplex equipment (in particular, the bit rate of these channels must be an integer of 64 kbit/s).

One administration is strongly in favour of a system having a simple frame structure which is compatible with that specified in Recommendation G.732 [CCIR, 1978-82r].

Some administrations consider that the same sound programme encoding frame (concept of sound frame) would be used for the different network access described above (concept of transmission frame) [CCIR, 1978-82s, n, t, u].

2.4 *Possible restrictions*

When a complete circuit at the primary level of the digital hierarchy is allocated to the transmission of sound programme signals, restrictions regarding the maximum number of consecutive zeros may exist at the 1544 kbit/s hierarchical level (CCITT Recommendation G.703). These restrictions could possibly be taken into account when defining the encoding law (character signals) for sound programme signals [CCIR, 1978-82b].

Furthermore, when circuits are leased at the full hierarchical bit rates no spare capacity is available for administrations to provide e.g. service digits for alarm and maintenance functions.

Other possible limitations should also be considered:

- digital speech interpolation (DSI) systems usually work at the primary hierarchical levels, i.e. sound programme signals may be affected rather than high bit rate video services. Since sound programme signals may have different characteristics from speech signals, for which DSI systems are optimized, they may affect the performance of DSI systems. On the other hand, DSI system characteristics may affect the performance of sound programme signals. It may be necessary to remove DSI systems when transmitting sound-programme signals or to route sound-programme signals such that they do not encounter DSI systems;
- three arrangements for interworking between digital systems based on different standards are presently considered by Study Group XVIII. These arrangements include multiplex system converters, A law to μ law converters and bit inversions and substitutions. Encoding law converters for speech signals would have to be removed when transmitting sound-programme signals. Multiplex system converters may affect the necessary time slot integrity for sound-programme signals. CMTT may therefore want to consider the possible implications of the interworking arrangements on the transmission of sound programme signals and advise Study Group XVIII accordingly.

2.5 *Aspects of multiplexing*

The technique of multiplexing high quality sound channels should be such that no bits are lost.

There appears to be four possible solutions:

2.5.1 All digital networks involved in the transmission are synchronized by the same master clock, in which case the sampling frequency of the sound channels is synchronized by the clock frequency of the multiplex. This solution is unlikely to be realized on a world-wide basis.

2.5.2 The primary digital multiplexes are synchronized by the tributary digital sound-programme signal. This may pose some problems of reliability and of possible conflict if, for example, two sound channels of different origins use the same multiplex.

2.5.3 The digital networks are plesiochronous but are controlled by clocks of very high stability.

The CCITT has recommended in Recommendation G.811 that the international digital links should be operated in a plesiochronous manner with reference clocks of very high accuracy (10^{-11}). This implies that the national networks are either fully synchronous or plesiochronous with the same accuracy.

It is pointed out that a slip rate of 1 slip in 70 days per plesiochronous interconnection is the resulting theoretical slip rate, taking into account clock accuracies according to Recommendation G.811 only. However, account should also be taken of practical network characteristics encountered under normal operating conditions even where synchronized national networks remain synchronized. As a guideline, Recommendation G.822 specifies a value of one controlled frame slip in 5 hours for a 64 kbit/s end-to-end connection (switched or permanent).

A digital sound-programme channel interconnection can be realized without any frequency adaptation if the terminal stations belong to the same synchronous network. Simple frequency adaptation by means of slip technique is permitted if the terminal equipment belongs to different synchronous networks with the clock-accuracy as specified in Recommendation G.811.

However, in the evolution towards this situation the terminal station will likely be operated at lower clock accuracies as specified by Recommendations G.732 and G.733 (5×10^{-5}). In this case the sound programme signal should be justified at all asynchronous interconnecting points, or the interconnection should be at analogue interfaces.

Another possibility is that timing information to the sound programme codecs (synchronous access) is provided by the primary digital multiplex equipment. In this case, no slips will occur between the sound programme codecs and the primary multiplex equipment.

It is noted that the above synchronous interconnection applies without any restriction as long as the telecommunication administrations are responsible for terminal equipment including encoding. In the event that encoding should become the responsibility of the broadcasting organizations, it is only applicable if their clocks are in synchronism with that in the network of the administration concerned or a synchronizing technique such as justification must be employed at the interconnection point.

2.5.4 A technique has been described [CCIR, 1978-82v] whereby synchronization of asynchronous sources may be accomplished without using justification. In this system, which is known as sample rate synchronization, a digital store is reset as necessary during a quiet audio passage, resulting in no audible disturbance.

3. Coding methods

Several different coding methods have been proposed for sound-programme transmission [CCIR, 1970-74a, b, c, and d; CCIR, 1974-78a, b, c, d, e, f, g and h; CCIR, 1978-82w, x, y, z, m, q, n, s, t, aa, ab, o, ac, ad, l, ae, af, u, ag]. Table I contains an updated survey of the most important parameters of some of these proposals.

The coding proposals use either instantaneous or near-instantaneous compression for bit-rate reduction. In these cases quantizing distortion is dependent on the sound-programme signal level. The distortion can be regarded as a programme-modulated noise (expanded noise) and can be measured by the same method proposed for analogue compandors bearing in mind the pre-emphasis characteristic [CCIR, 1978-82y].

Experience has shown that the quantizing noise has a uniform spectrum distribution. Therefore, the weighted noise values to be expected with sine wave tones can be calculated by the formula:

$$n_n = n_o - 20 \log 2^x + K_1 + K_2 - K_3$$

with the following explanations for the letters:

n_n : weighted noise level in dBq0ps;

n_o : overload level in dBm0s by a sinusoidal signal at the zero loss frequency of the pre-emphasis (2.1 kHz in the case of the pre-emphasis according to CCITT Recommendation J.17 with 6.5 dB loss at 0.8 kHz; 1.9 kHz in the case of the pre-emphasis according to Column B1). The difference between overload level and maximum programme-signal level of + 9dBm0s is the margin provided against overloading;

x : quantizing resolution in bit ($20 \log 2^x = x \cdot 6$ dB is the quantizing noise factor);

K_1 : takes into account the weighting of the noise. Two cases are considered:

- psophometric weighting (Report 496, Table II)
 - $K_1 = 8.5$ dB (15 kHz bandwidth)
 - $K_1 = 9.5$ dB (7 kHz bandwidth)
- psophometric weighting (Report 496, Table II) and de-emphasis
 - $K_1 = 4.5$ dB (J.17 de-emphasis with 6.5 dB gain at 0.8 kHz)
 - $K_1 = 3.0$ dB (column B1 de-emphasis)
 - $K_1 = 6.5$ dB (column B6)

K_2 : 5 dB quasi peak factor (Recommendation 505-1 (Kyoto, 1978), Note 1 to § 3.1.5 and Recommendation 503-1 (Kyoto, 1978), Note 1 to § 3.5);

K_3 : 1.8 dB improvement factor due to the r.m.s. ratio of a sinusoidal reference signal to a quantization noise signal with equal peak-to-peak values ($10 \log 12/8$).

The weighted noise levels calculated using the above mentioned formula considering the proposed range of sample word resolutions and the two proposed overload levels, restricted to 15 kHz bandwidth, are listed in Table II.

TABLE I – Proposed coding methods

	Coding methods using different companding principles								Units	
	Near-instantaneous		Instantaneous							
	A1	A2	A3	B1	B2	B3	B4	B5		B6
Nominal bandwidth Pre/de-emphasis	0.04-15 ⁽¹⁾⁽²⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁴⁾		0.04-15 ⁽¹⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁴⁾	0.04-15 ⁽¹⁾ See Note ⁽⁵⁾	0.04-15 ⁽¹⁾ None	0.04-15 ⁽¹⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁴⁾	0.04-15 ⁽¹⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁴⁾	0.04-15 ⁽¹⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁴⁾	0.05-7 ⁽³⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁴⁾	kHz —
Overload point ⁽⁶⁾ Sampling frequency	+12 32		+12 32	+12 32	+12 32	+12 32	+12 32	+15 32	+15 16	dBm0s kHz
Companding law Bit rate reduction	5 ranges 14/10.1		5 ranges 14/10.1	13 segments 14/10	7 segments 13/11	7 segments 14/12	11 segments 14/11	11 segments 14/11	11 segments 14/11	— bits
Finest resolution and corresponding noise	14 —65		14 —65	14 —66	13 —55	14 —65	14 —65	14 —62	14 —60	bits/sample dBq0ps
Coarsest resolution at +9 dBm0s/ <i>f</i> ₀ * and corresponding noise	10 —41		10 —41	8 —30	10 —37	11 —47	9 —35	9 —32	9 —30	bits/sample dBq0ps
Resolution at +9 dBm0s/60 Hz and corresponding noise	12 —55		12 —55	10 —42	10 —37	12 —59	11 —47	12 —50	12 —48	bits/sample dBq0ps
Source coding	323		323 ⁽⁷⁾	320	352	384	352 ⁽⁸⁾	352 ⁽⁸⁾	176 ⁽⁸⁾	kbit/s
Error protection	12 13		11 ⁽⁷⁾	16	32	0	32	32	16	kbit/s
Framing and signalling	3 2		4 ⁽⁷⁾	0.66	0	0	0	0	0	kbit/s
Service bit rate	338		338	336.66	384	384	384	384	192	kbit/s
Transmission bit rate	~ 340** 384		340** 384	336.66** 384	384	384	384	384	192	kbit/s
Proposed by Doc. CMTT (1978-82)	France 35, 110, 93, 112, 277		UK 240	Italy 203 (1970-74) 271, 275	Japan 35, 51, 110, 231	USSR 35, 51, 110	Switzerland, Canada 110, 35, 276	Federal Republic of Germany, Netherlands, Denmark, Norway 6, 80, 81, 82, 110, 35, 251, 258, 297, 234	Federal Republic of Germany, Netherlands, 81, 82, 251, 258, 234	

* f_0 : zero loss frequency of pre-emphasis.

** Dedicated frame.

Notes to Table I:

- (1) Performance characteristics for analogue 15 kHz type sound-programme circuits are given in Recommendation 505 and the proposals in Table I are assumed to meet these requirements with at least three codecs in tandem. Performance characteristics for one codec are given in § 4.
- (2) These systems have also the possibility of nominal bandwidth of 0.05-7 kHz [CCIR, 1978-82u].
- (3) Performance characteristics for analogue 7 kHz type sound-programme circuits are given in Recommendation 503 and the proposals in Table I are assumed to meet these requirements with at least 3 codecs in tandem.
- (4) For convenience, the characteristic given in CCITT Recommendation J.17 has been reproduced in § 2.2.2 of Report 496.
- (5) The pre-emphasis used is: insertion loss = $10 \log \frac{8.5 + \left(\frac{f}{1900}\right)^2}{1 + \left(\frac{f}{650}\right)^2}$ (f in Hz with $f_0 = 1900$ Hz).
- (6) This is defined as the maximum r.m.s. level of sinusoidal signal which does not cause clipping; this value is independent of frequency if analogue peak limiter and pre-emphasis are removed and replaced by zero dB loss; with pre-emphasis the overload level is defined at the zero dB loss frequency (f_0).
- (7) These values are approximate. For more details see [CCIR, 1978-82o].
- (8) For sound-programme signals more than 36 dB below the overload point only 10 bits are needed for coding. In these cases the 11th bit can be used to transmit a 1-bit data signal.

TABLE II – Calculated weighted noise levels depending on quantizing resolution

Resolution		Weighted noise level in dBq0ps at 15 kHz bandwidth			
x (bits)	$20 \log 2^x$ (dB)	$n_o = +12$ dBm0s			$n_o = +15$ dBm0s
		Rec. J.17 de-emphasis with 6.5 dB gain at 0.8 kHz	B1 de-emphasis	Without de-emphasis	Rec. J.17 de-emphasis with 6.5 dB gain at 0.8 kHz
14	84	–65	–66	–61	–62
13	78	–59	–60	–55	–56
12	72	–53	–54	–49	–50
11	66	–47	–48	–43	–44
10	60	–41	–42	–37	–38
9	54	–35	–36	–31	–32
8	48	–29	–30	–25	–26

TABLE III – Comparison of programme-modulated noise in dBq0ps according to the proposed coding methods listed in Table I

Coding method	Sound-programme signal equivalent to a sinusoidal signal at f_0 zero loss frequency of pre-emphasis									

(1) μ law coding and no pre-emphasis are used in this system.

In Table III the calculated noise values are associated to the proposed coding methods listed in Table I to facilitate an objective comparison.

To aid comparison of various types of digital codecs, a proposal has been made [CCIR, 1970-74a] for a reference digital codec. The CMTT recognizes that other forms of reference may be possible, for example, the addition of noise to the unimpaired analogue signal, and suggests that a study of the preferred form of reference be carried out. This study should take account of the case where a number of codecs in tandem are used. [CCIR, 1970-74a] also indicates that a subjective impairment of 4.5 on the 5-point impairment scale given in Report 623 is likely to be met by 4 codecs, of the type described, in cascade. However, it is recognized that there may be difficulty in achieving this target figure and the results of tests by one Administration given in Annex II of this Report confirm this difficulty.

The CMTT asks Study Group 10 to consider whether a subjective impairment of 4.5 is appropriate for a hypothetical reference circuit such as that given in Recommendation 502, including digital sections.

Discussion of the documents [CCIR, 1970-74b, c and d] describing the proposed coding techniques which give little consideration to the hardware complexity involved (with the respective techniques), has indicated that it is important that further studies be made of the instrumental complexity necessary for each coding technique with regard to signal processing such as digital multiplexing (including error protection), studio processing and recording and direct broadcasting from satellites. Moreover, the EBU has proposed that further studies should be carried out for the choice of the best coding parameters for production, transmission and broadcasting and on the subject of transcoding between different systems [CCIR, 1978-82ah]. Some of these studies are properly the concern of Study Group 10, but, in the interest of progression towards the desirable objective of a single basic coding technique for all applications, the CMTT should give them due consideration.

It was recognized that the importance of such a consideration only exists if the proposed coding techniques meet specific quality objectives (e.g. corresponding to grade 4.5 on the CCIR 5-point impairment scale) yet to be agreed. In the case where two or more coding techniques meet the quality objectives, consideration of hardware complexity could be the determining factor in the choice of the coding standard to be adopted.

Concerning the sampling frequency, the EBU has concluded as a result of subjective tests that for high-quality sound signals a bandwidth of 15 kHz is sufficient [CCIR, 1978-82g]. Thus the choice of 32 kHz as the sampling frequency appears to be satisfactory [CCIR, 1970-74e, 1978-82h]. Since on this point all proposed coding methods listed in Table I (as far as 15 kHz bandwidth is concerned) are in agreement, 32 kHz can be unanimously recommended as the sampling frequency.

From Table I it can be seen that with only one exception all proposals are based on a finest resolution of 14 bits and, with two exceptions, on the use of pre-emphasis according to CCITT Recommendation J. 17 with 6.5 dB loss at 0.8 kHz. The question arises whether on these two points a single solution is possible for international cooperation [CCIR, 1978-82x].

As far as the overload level is concerned Table I shows that most of the proposals are in favour of +12 dBm0s; nevertheless, some administrations are strongly in favour of a value of +15 dBm0s because the level of +12 dBm0s would provide a margin of 3 dB only against overloading. At least 6 dB is suggested by these administrations. This value of +15 dBm0s takes into account:

- that in practice several sound-programme circuits occasionally have to be cascaded for international or especially for intercontinental links;
- that for maintenance purposes in CCITT Recommendation N.23 level deviations of ± 1 dB per circuit are tolerated;
- that different programme level meters are in use (see Report 820);
- that digital channels for sound-programme transmission are even more sensitive against overloading than those for analogue transmission;
- that the use of limiters either at the output of the broadcasting studios or at the input of sound-programme circuits is not covered by a Recommendation.

However a proposal for possible Recommendation has been made and is annexed to this Report (Annex III). Attention is drawn to potential difficulties which could arise with certain designs of limiters when a number of sound-programme circuits, each having such a limiter, are cascaded. It is also pointed out that only one limiter at the transmitting end would be necessary in any connection.

Further study is required and the administrations are asked for comments on the use and characteristics of limiters and on preferred overload point. Contributions are also invited on the subject as to whether limiters may be desirable if the overload point of 15 dBm0s is chosen.

A proposal has also been made [CCIR, 1978-82ai] that if no single companding system can be standardized, the possibility of using a 14 bit linearly coded system for international transmission should be considered.

4. Performance characteristics

The values specified should be met by a single codec with the input and output ports terminated in their nominal impedance and, if used under normal operating conditions, with pre-emphasis and de-emphasis networks in place, unless otherwise specified. Where applicable, performance characteristics for the hypothetical reference circuit (Recommendation 505) and laws of addition (Recommendation 605) were taken as a basis.

In general, it is expected that three codecs in tandem, interconnected at audio frequencies, will meet the performance characteristics specified for the hypothetical reference circuit.

4.1 *Monophonic programme transmission*

4.1.1 *Impedance of audio-frequency ports*

- To conform with Recommendation 505, § 2.2.

4.1.2 *Insertion loss*

- The nominal insertion loss should be 0 dB.
- The insertion loss variation should not exceed ± 0.3 dB.

Test signal level and frequency to be used are specified in Recommendation 505, § 3.1.2. However, the test signal frequency should not be a submultiple of the sampling frequency.

4.1.3 *Relative level*

- To conform with Recommendation 505, § 2.3 (see also CCITT Recommendation J.14).

4.1.4 *Adjustment of relationship between encoding law and audio level*

- Under study.

Note. — This parameter is used to specify the adjustment of encoder and decoder within a given tolerance.

4.1.5 *Gain/frequency response*

- The gain/frequency response referred to 0.8 kHz or 1 kHz should comply with the limits indicated:

0.04	to	0.125	kHz:	+0.25	to	–1.0	dB
0.125	to	10	kHz:	+0.25	to	–0.25	dB
10	to	14	kHz:	+0.25	to	–1.0	dB
14	to	15	kHz:	+0.25	to	–1.4	dB

Note. – A number of administrations felt that somewhat tighter limits, as proposed in [CCIR, 1978-82af], could be met without incurring a cost penalty.

4.1.6 *Group delay distortion*

- The difference between group delay at the given frequency and the minimum group delay should not exceed the following limits:

0.04	kHz:	18	ms
0.075	kHz:	8	ms
14	kHz:	3	ms
15	kHz:	4	ms

Note. – A number of administrations felt that somewhat tighter limits, as proposed in [CCIR, 1978-82af], could be met without incurring a cost penalty.

4.1.7 *Idle channel noise*

4.1.7.1 *Weighted noise*

- With the input and output ports of the channel terminated in the nominal impedance, the idle channel noise should not exceed -56 dBq0ps.

Note. – This value applies to codecs with pre-de-emphasis using 14 bits for the finest resolution. A formula to calculate idle channel noise is given in [CCIR, 1978-82ae]. The value to be applied to codecs which do not use 14 bits for the finest resolution or have no pre-de-emphasis is under study.

4.1.7.2 *Single frequency interference*

- The level of any single frequency (in particular the sampling frequency and its multiples), measured selectively, should not exceed $-X$ dBm0s.

Note. – The value for X is under study.

4.1.7.3 *Receiving equipment noise*

- Noise contributed by the receiving equipment alone should be less than $-X$ dBq0ps when its input is driven by a PCM signal corresponding to the lowest decoder output value number.

Note. – The value for X is under study.

4.1.8 *Discrimination against out-of-band input signals*

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

Note. – The values for X , Y and Z are under study to establish the desirable out-of-band characteristics of the attenuation required for a tolerable fold-over (alias – component) distortion. Attenuation at half the sampling frequency will probably need to be at least 80 dB for the combination of the input and output filters of a single codec.

4.1.9 *Spurious out-of-band signals at the channel output*

- With any sine-wave signal in the range 0.04 to 15 kHz at a level of 0 dBm0 applied to the input port of a channel, the level of spurious out-of-band image signals measured selectively at the output port should be lower than $-X$ dBm0s.

Note 1. – The spurious out-of-band signals should not give rise to unacceptable interference in equipment connected to the PCM channel. In particular, the recommended intelligible or unintelligible crosstalk in a connected FDM channel should not be exceeded as a consequence of the spurious out-of-band signals at the PCM channel output.

Note 2. – The value for X is under study to establish the desirable out-of-band characteristics of the attenuation required for a tolerable fold-over (alias – component) distortion. Attenuation at half the sampling frequency will probably need to be at least 80 dB for the combination of the input and output filters of a single codec.

4.1.10 Programme modulated noise

- Under study.

Note. — One administration [CCIR, 1978-82ac] proposed to use the test signal specified in Recommendation 505, § 3.1.5.2, i.e. +9 dBm0s/60 Hz. The reasons for this proposal are that:

- the test signal realistically simulates a sound-programme signal whose highest spectral energy density occurs within the low-frequency range in conformity with the conventional test signal defined in Recommendation 571;
- the suppression of the portion of noise in the range < 400 Hz cannot affect the test result due to the low weighting of this frequency range;
- the suppression loss of the 60 Hz test signal is not only determined by the high-pass filter but is also increased by approximately 24 dB by the weighting filter.

Other administrations expressed the view that a 60 Hz test signal would be attenuated by the pre-emphasis and that a more suitable test-signal frequency would be the 0 dB loss frequency of the pre-emphasis, e.g. 2.1 kHz for a pre-emphasis in accordance with CCITT Recommendation J.17 or 1.9 kHz for a pre-emphasis as used with encoding method B1 of Table I.

4.1.11 Non-linear distortion

4.1.11.1 Harmonic distortion

- Harmonic distortion factors measured with single-tone test signals at +9 dBm0s should not exceed the following limits:

TABLE IV

Test tone frequency (kHz)	Second harmonic and third harmonic, measured selectively (%)
0.04 to 0.125	<i>X</i>
0.125 to 7.5	<i>Y</i>

Note. — The values for *X* and *Y* are under study.

4.1.11.2 Spurious in-band signals

- With a sine-wave signal in the frequency range 0.04-15 kHz (excluding submultiples of 32 kHz) and at a level of 0 dBm0 applied to the input port of a channel, the output level at any frequency (excluding harmonics) other than the frequency of the applied signal, measured selectively in the frequency band 0.04 to 15 kHz should be less than $-X$ dBm0.

Note. — The value for *X* is under study. One administration proposed programme weighting for this measurement. Another administration suggested a tentative value of 50 for *X* [CCIR, 1978-82ag], pending further subjective evaluation.

Spurious in-band signals $S(f_d)$ can be produced by interference between a sound-programme signal $S(f)$ and the sampling frequency signal $S(f_0)$ according to the rule:

$$f_d = m f_0 - n f$$

where $m = 1, 2, 3 \dots$ and $n = 2, 3, 4 \dots$

Since results of subjective tests on this kind of distortion are not available a ratio of at least 40 dB between the test signal and any spurious signal is provisionally suggested. To facilitate the comparison of the results of measurements on a critical basis taking into account practical considerations it is proposed to use as test signals $S(f)$ 0 dBm0s/11 kHz and/or 0 dBm0s/13 kHz.

Then the spurious in-band signals are to be measured selectively at frequencies as follows:

TABLE V

f	$f_0 - 2f$	$f_0 - 3f$	$2f_0 - 4f$	$2f_0 - 5f$	
11	10	1	—	9	kHz
13	6	7	12	1	kHz

4.1.12 Variation of gain with input level

— Under study.

4.1.13 Crosstalk

The intelligible crosstalk in a multiplex equipment from other sound-programme circuits or from a telephone circuit into a sound-programme circuit should be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit. The sinusoidal test signal applied to the disturbing circuit should be in the frequency range of 0.04-15 kHz (excluding sub-multiples of the sampling frequency) and at a level of 0 dBm0s.

The crosstalk level should not exceed the following limits:

0.04 kHz:	X dBm0s
0.04 to 0.5 kHz:	oblique straight-line segment on linear-decibel and logarithmic-frequency scales
0.5 to 5 kHz:	Y dBm0s
5 to 15 kHz:	oblique straight-line segment on linear-decibel and logarithmic-frequency scales
15 kHz:	Z dBm0s

Note. — The values for X , Y and Z are under study. The following point requires particular attention: in the absence of any input signal, PCM codecs can cause crosstalk enhancement. This enhancement is most pronounced when the idle quantizer is biased at a decision level. In this case a low level crosstalk signal at the input to the encoder causes the zero crossings at the decoder to be determined by the crosstalk signal. In the worst case crosstalk in the order of magnitude of unweighted idle channel noise can be expected.

The use of an out-of-band “dither” signal at the encoder input reduces crosstalk enhancement.

4.2 Additional requirements for stereophonic programme transmission

4.2.1 Gain difference between channels A and B

The difference in gain between channels A and B should not exceed the following limits:

0.04 to 0.125 kHz:	0.7 dB
0.125 to 10 kHz:	0.4 dB
10 to 14 kHz:	0.7 dB
14 to 15 kHz:	1.4 dB

Note. — A number of administrations felt that somewhat tighter limits, as proposed in [CCIR, 1978-82af], could be met without incurring a cost penalty.

4.2.2 Phase difference between channels A and B

— The difference in phase between channels A and B should not exceed the following limits:

0.04 kHz:	14°
0.04 to 0.2 kHz:	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
0.2 to 4 kHz:	7°
4 to 14 kHz:	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
14 kHz:	14°
15 kHz:	19°

Note. — A number of administrations felt that somewhat tighter limits, as proposed in [CCIR, 1978-82af], could be met without incurring a cost penalty.

5. The connection of several codecs in tandem

During the introduction of digital transmission systems, digital sections of circuits will coexist with analogue sections. Therefore it is wise to study the impairments caused by the connection of several codecs in tandem. Simulation tests have been reported, and the results of these tests should be confirmed by practical experiments with several real codecs connected in tandem.

6. Subjective assessment of quality

Some administrations thinking that subjective assessments are important especially with several codecs in tandem, have presented the results of their tests as a confirmation that their proposed coding method produces a negligible impairment even with 3 or 4 codecs in tandem. The results of those tests are included in Annex II in a summarized form.

7. Bit rate

The overall bit rate used for the transmission of a sound signal contains an additional amount for protection against clicks caused by bit errors (see Report 648). The characteristics of bearer circuits (see Report 817) must be considered in the design of suitable error-protection techniques.

Some administrations considered that the overall bit rate of one single digital sound-programme signal (including sound-programme samples, protection bits and possibly, bits for other purposes) might be a multiple of 64 kbit/s, so that the transmission capacity required is exactly that of several PCM telephone channels (e.g. 5 telephone channels [CCIR, 1974-78f] or 6 or 3 telephone channels [CCIR, 1974-1978g]).

It must be noted that agreement has not been reached on whether the sampling frequency should be synchronized with the clock of the digital bearer.

ANNEX I

PRESENT STAGE OF THE DISCUSSION FOR A STANDARD FOR DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

Following the Interim Meeting of the CMTT (Geneva, 30 September-17 October 1980) at which a draft Recommendation for digital transmission of sound-programme signals was attached to draft Report 647-1 (MOD I) as Annex I, a number of contributions were received from administrations on this subject. An attempt was made at the Final Meeting to prepare an acceptable Recommendation taking account of the Annex I referred to, together with additional information as follows.

Japan in Document [CCIR, 1978-82aa] has suggested items for further study.

The Netherlands have proposed in [CCIR, 1978-82ab] that methods B5/B6 in Table I in this Report should be used.

The Federal Republic of Germany has given some results of subjective tests in [CCIR, 1978-82aj], and in [CCIR, 1978-82ad] has proposed the designation of two "cases" for digital transmission of sound-programme signals. Case 1 would indicate that analogue interfaces are involved, and case 2 would cover the situation when only digital interfaces are used. For case 1 the latter document proposes methods B5/B6 in the Table and it is confirmed that the overload point should be +15 dBm0s to avoid problems when analogue and digital circuits are connected in tandem.

The United Kingdom has proposed in [CCIR, 1978-82o] that the near-instantaneous coding method A1/A2 in the Table should be used, but they have also indicated that a uniformly quantized coded solution would be acceptable.

Italy has suggested in [CCIR, 1978-82ak] that the preferred overload level should be +12 dBm0s, and in [CCIR, 1978-82ae] has stated that the proposals in Annex I to draft Report 647-1 (MOD I) are not acceptable and that if uniformly quantized coding is adopted no emphasis should be employed.

Canada in [CCIR, 1978-82af] has supported method B4 in the Table and has explained how this might be employed on 1544 kbit/s circuits.

Switzerland has supported method B4 in the Table in Document [CCIR, 1978-82r].

Denmark has proposed in Document [CCIR, 1978-82ag] that method B5 in the Table should be used, but they have also indicated that a uniformly quantized interface might be considered failing an agreement for a single standard.

In addition the report of the Special Rapporteur CMTT/Study Group XVIII, Document [CCIR, 1978-82a], and Documents [CCIR, 1978-82b, c, e], from the CCITT, were considered, but it was felt that these documents were for information only as far as the proposed Recommendation was concerned. Similarly Document [CCIR, 1978-82a] was thought also to be for information in this respect. This document is a proposal to Study Group 10 from the European Broadcasting Union that for the emission of high quality sound signals in satellite broadcasting, either a linear 14-bit coding method, or a near-instantaneous companding method corresponding to column A in the Table should be used as appropriate. Also it was considered desirable that the Recommendation 606 for the sampling frequency should if possible be incorporated in the overall Recommendation.

After discussion on all of the above information, various proposals were made which include the systems listed in Table I of this Report. These are summarized below:

- (a) Method B4, with any other method in the Table by bilateral agreement.
- (b) Method A, with any other method in the Table by bilateral agreement.
- (c) Either method A or method B4 by arrangement between the administrations concerned, with any other method in the Table by bilateral agreement.
- (d) Uniform quantization coding method, 14 bits/sample, 32 000 (± 50 parts in 10^6) samples/second, without pre-emphasis (see Note), with any other method in the Table by bilateral agreement.
- (e) Any of the methods in the Table, particular cases being determined by the receiving country except by bilateral agreement.
- (f) As (e), but particular cases being determined by the sending country, except by bilateral agreement.
- (g) As (f), but with a uniformly quantized interface, 14 bits/sample at the receiving decoder input.
- (h) If none of the above proposed coding methods could be accepted a list of those parameters which could be agreed unanimously should be recommended.

Note. — Whether pre-emphasis should be used requires further study.

Particular attention was given to the first four proposals as these appeared to have most chance of being accepted as a Recommendation, but unfortunately it was not possible to reach unanimous agreement. The Recommendation 606 recommends a sampling frequency of 32 kHz.

The hope was expressed that consideration would be given to these proposals by administrations during the next Study Period. It was felt that all of the proposals were capable of providing a satisfactory performance, but with the aim of resolving the situation it was also thought that the following suggestions should be considered. These suggestions were made by various administrations and have not necessarily been agreed by the CMTT.

Several administrations have underlined the need in the future to consider these questions in a much more general context than just in the narrow field of transmission and to pay particular attention to the current studies in the use of digital techniques in the areas of production and sound broadcasting. Proposal (d) above seems in this respect to be the only solution presented so far, which could relatively easily be adapted to such an evolution.

One administration put forward the view that a variation of proposal (c) above should be considered, in which international sound-programme circuits occasionally leased to the broadcasters should, during the transition period before all-digital working, employ the coding method B4, whilst for completely dedicated circuits the coding method A should be used.

As far as proposal (d) is concerned several additional considerations have been put forward. These are, that:

- various systems described in Table I are being used nationally and a Recommendation conforming with this proposal for international transmission is unlikely to reduce the number of national systems;
- a system conforming with this proposal would not suffer from programme-modulated noise, and transcoding from such a system to any other system described in Table I is relatively simple and would result in negligible loss of quality, especially if no pre- and de-emphasis is used in the 14-bit uniformly quantized system. Post production processing would also be facilitated by such a system;
- if a system conforming with this proposal is used for international transmission then it is suggested that if a 14 bit uniformly quantized output is available then it would not normally be necessary to transcode up to that system from national systems;

- most requirements for international sound-programme exchanges will not involve more than two sound-programme channels, so in general the extra bit-rate needed for this proposal should not be very onerous;
- home audio equipment may in the future employ digital coding with 14 bits uniformly quantized, and it is also possible that a similar standard may be used for broadcasting. If this happens it seems desirable that transmission links should be capable of a compatible quality;
- a system conforming with this proposal may use one protection bit with each sample for simplicity. This would entail 480 kbit/s for each sound-programme channel. Four such channels could fit into 2048 kbit/s without encroaching on telephony time-slots 0 and 16 in the CCITT Recommendation G.732 frame. This could ease the situation as far as the ISDN is concerned. It may be possible to use signalling bits in telephony time-slot 16 as additional data bits in connection with the sound-programme channels;
- it is desirable that international sound-programme channels should take the place of an integer number of 64 kbit/s telephony channels. Companding methods which employ 384 ($= 6 \times 64$) kbit/s for each high quality sound-programme channel conform with this, but proposal (d) would probably employ 480 ($= 7\frac{1}{2} \times 64$) kbit/s which does not, although a stereo pair of channels would use 15×64 kbit/s. It should be noted that this proposal represents an increase of 25% in the bit rate;
- the proposal results in a bit rate (480 kbit/s per programme channel) which does not allow for an integer number of sound-programme channels and therefore is not compatible with the economic use of system capacity at the 1544 kbit/s digital hierarchy level;
- for interworking between 1544 kbit/s and 2048 kbit/s the situation appears to be as follows: if 384 kbit/s companded systems are employed then five sound-programme channels may be obtained in 2048 kbit/s. These five channels could be sent as four in one 1544 kbit/s multiplex and one in another 1544 kbit/s multiplex. If however proposal (d) is adopted with 480 kbit/s for each sound-programme channel only four such channels may be obtained in 2048 kbit/s. These four channels may commonly be two stereo pairs of channels, in which case it seems to be necessary to transfer two sound-programme channels to each of two 1544 kbit/s, to avoid splitting stereo signals between primary multiplexes;
- at present no administration had indicated that they are using or intending to use, a system conforming with this proposal. Thus every country would have to provide two systems, one for national and one for international use, and unless the international system extended to all parts of the country, transcoding between the national and international systems would be necessary. Thus cost penalties would be incurred. Additional complications may be caused if different pre- and de-emphasis is involved, and quality degradation is the likely result of such transcoding;
- consideration should also be given to the fact that for international programme exchanges, it may often be acceptable in the interests of economy to employ the uniform quantization principle with a sampling frequency of 16 kHz, thereby obtaining twice the number of channels in the same bit rate.

ANNEX II

SUBJECTIVE QUALITY ASSESSMENTS FOR SOUND-PROGRAMME SIGNALS ENCODED BY MEANS OF INSTANTANEOUS COMPANDING SYSTEMS

1. Introduction

Based on tests performed on their proposed coding method, some administrations (Italy, Japan and the Federal Republic of Germany) have presented the results of subjective assessments for sound-programme signals encoded by means of instantaneous companding laws. The tests were carried out on the basis of Recommendation 562 using the 5-grade rating scale. The results of these tests are not directly comparable because of some different test circumstances especially those concerning:

- the date and place of tests,
- the insertion of pre-/de-emphasis,
- the programme level adjustments,
- the particular programme excerpts,
- the listening conditions, and
- the listeners who took part in the tests.

Therefore the test results of the three Administrations should be considered separately.

2. Test results

2.1 *Italy* [CCIR, 1974-78i]

The tests were performed by the Radiotelevisione Italiana (RAI) in 1974. The results were taken as confirmation that the coding method proposed by Italy and detailed in column B1 in Table I of this Report produces a negligible impairment even with 4 codecs in tandem. For the tests, 7 programme excerpts were used:

- piano, Mozart,
- piano, Schubert,
- violin,
- trumpet,
- Glockenspiel,
- electronic signal (100 Hz),
- electronic signal (500 Hz).

The tests were performed both on programme-modulated noise and on granular distortion. For the 7 excerpts considered the average value of impairment due to programme-modulated noise is 4.5 for 1 codec and 4.3 for 4 codecs in tandem. If granular distortion is additionally weighted the equivalent values are 4.5 and 3.7.

2.2 *Japan* [CCIR, 1978-82x]

The tests were performed by the Nippon Telegraph and Telephone Public Corporation (NTT) in 1977. The results were also taken as confirmation that the coding method proposed by Japan and detailed in column B2 in Table I of this Report produces negligible impairment even with 3 codecs in tandem. Furthermore, the expected advantage due to pre-/de-emphasis (CCITT Recommendation J.17) could not be confirmed by a comparison test on one codec with and without pre-/de-emphasis. It was revealed that unacceptable impairments would be caused for sound-programmes which contain rather high-frequency components by the insertion of pre-/de-emphasis. The Administration of Japan decided therefore not to use any pre-/de-emphasis. In the tests were included the following 5 programme excerpts:

- pianoforte solo,
- male speech,
- symphony orchestra,
- drums,
- cymbals.

The test with cymbals was found as the most critical one if pre-/de-emphasis is used. With 3 codecs in tandem (all without pre-/de-emphasis) a very small degradation was found.

2.3 *Federal Republic of Germany* [CCIR, 1978-82aj]

The tests were performed by the Institut für Rundfunktechnik (IRT) in 1980. The tests were taken as a basis for the decision to use the coding method given in columns B5 and B6 in Table I of this Report instead of the coding method given in columns 4 and 5 in Table I of this Report published in Volume XII in 1978 (Kyoto). These tests do not refer to the coding method in column B1 of this Report. For the tests, 8 programme excerpts were used as follows:

- piano, Mozart,
- piano, Schubert,
- violin,
- trumpet,
- Glockenspiel,
- Frère Jacques,
- 100 Hz pulses,
- 500 Hz pulses.

For the proposed coding method (column B5/1982) the grade of impairment expressed by the average value of all weighted excerpts is 4.4 for 1 codec and 3.6 for 4 codecs in tandem. The equivalent values for the previous proposed coding method (column 4/1978) are 3.6 and 2.6 respectively and were regarded as unacceptable.

ANNEX III

PROPOSAL FOR A NEW RECOMMENDATION
LIMITERS FOR HIGH-QUALITY SOUND-PROGRAMME CIRCUITS

The CCIR,

CONSIDERING

- (a) that on high-quality analogue sound-programme circuits it is desirable to prevent distortion due to overloading, whilst keeping the signal-to-noise ratio as high as reasonably practical;
- (b) that on high-quality digital sound-programme circuits it is desirable to make the most efficient use of the available bits;
- (c) that if, in order to achieve this, pre-emphasis is employed at the sending end, then the use of a limiter to control the pre-emphasized signal is desirable to avoid over-modulation due to high-frequency peaks,

RECOMMENDS

that on high-quality (see Note 1) analogue or digital sound-programme circuits, employing pre-emphasis, a limiter should be employed at the interface between the broadcasting studio and the transmission circuit (see Note 2).

Note 1. — Consideration should be given as to whether this Recommendation should also cover medium and low quality circuits.

Note 2. — The action of a conventional non-frequency-selective limiter on a pre-emphasized signal may result in annoying level fluctuations of low and medium frequency components in the reproduced programme. One administration has suggested that a frequency-selective type of limiter is the best type to deal with this problem [CCIR, 1978-82am].

REFERENCES

CCIR Documents

- [1970-74]: a. CMTT/214 (EBU); b. CMTT/145 (United Kingdom); c. CMTT/190 (Germany (Federal Republic of)); d. CMTT/203 (Italy); e. CMTT/223 (CCITT Study Group XVIII).
- [1974-78]: a. CMTT/283 (France); b. CMTT/284 (France); c. CMTT/274 (EBU); d. CMTT/202 (United Kingdom); e. CMTT/88 (Italy); f. CMTT/282 (France); g. CMTT/75 (Germany (Federal Republic of)); h. CMTT/14 (Japan); i. CMTT/265 (Italy).
- [1978-82]: a. CMTT/285 (Report of Special Rapporteur to CCITT Study Group XVIII); b. CMTT/290 (CCITT Study Group XVIII); c. CMTT/291 (CCITT Study Group XV); d. CMTT/292 (CCITT Study Group XVIII); e. CMTT/294 (CCITT Study Group XVIII); f. CMTT/110 (CCITT Study Group XVIII); g. CMTT/10 (EBU); h. CMTT/35 (France); i. CMTT/108 (CCITT Study Group XVIII); j. CMTT/109 (CCITT Study Group XVIII); k. CMTT/241 (United Kingdom); l. CMTT/271 (Italy); m. CMTT/82 (Germany (Federal Republic of)); n. CMTT/92 (France); o. CMTT/240 (United Kingdom); p. CMTT/90 (France); q. CMTT/91 (France); r. CMTT/281 (Switzerland); s. CMTT/93 (France); t. CMTT/112 (France); u. CMTT/277 (France); v. CMTT/22 (United Kingdom); w. CMTT/6 (Norway); x. CMTT/51 (Japan); y. CMTT/80 (Germany (Federal Republic of)); z. CMTT/81 (Germany (Federal Republic of)); aa. CMTT/231 (Japan); ab. CMTT/234 (Netherlands); ac. CMTT/251 (Germany (Federal Republic of)); ad. CMTT/258 (Germany (Federal Republic of)); ae. CMTT/275 (Italy); af. CMTT/276 (Canada); ag. CMTT/297 (Denmark); ah. CMTT/247 (EBU); ai. CMTT/25 (United Kingdom); aj. CMTT/237 (Germany (Federal Republic of)); ak. CMTT/273 (Italy); al. CMTT/308 (CCIR Secretariat); am. CMTT/239 (United Kingdom).

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- HESSENMÜLLER, H. [1974] Die Übertragung von Rundfunkprogrammen in pulsecodemodulierter Form (The transmission of broadcast programmes in pulse-modulated form). *Rundfunktechn. Mitt.* Vol. 18, 6, 297-300.

CCIR Documents

- [1974-78]: CMTT/186 (Switzerland).
- [1978-82]: CMTT/232 (Japan); CMTT/272 (Italy).

RECOMMENDATION 606

**SAMPLING FREQUENCY TO BE USED FOR THE DIGITAL TRANSMISSION
OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS**

(Study Programme 18A/CMTT)

(1982)

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.

REPORT 648-2

**DIGITAL SOUND-PROGRAMME TRANSMISSION IMPAIRMENTS
AND METHODS OF PROTECTION AGAINST THEM ***

(Study Programmes 18A/CMTT and 18D/CMTT)

(1974-1978-1982)

1. Errors and loss of synchronization

The subjective effect of errors and slips (loss of synchronization) in the absence of properly designed mitigation or correction techniques, is an audible "click" of variable intensity, according to the weight of the erroneous bits, and of variable duration, according to the error burst or slip duration.

In theory an extensive series of lengthy subjective tests, on a number of programmes, is required to assess the error and slip rate limits which could be judged acceptable, e.g. corresponding to grade 4.5 on the CCIR 5-point impairment scale. (Such subjective tests should be carried out in accordance with Report 623.)

Whereas for high error and slip rates these tests are surely significant, for very low rates, the results will be affected by the fact that the listener's attention decreases with increasing test duration.

Moreover, it should be noted that even in the absence of transmission impairments the analogue studio apparatus might introduce some additional clicks.

Therefore it seems that even such exhaustive tests may give unreliable results. In accordance with the approach outlined in Report 642 which suggests a maximum tolerable rate of occurrence of short interruptions, it was suggested that for very low error and slip rates, the quality objective should be expressed in terms of rate of occurrence of clicks.

* For the definition of absolute power, relative power and noise levels, see Recommendation 574.

It seemed reasonable, from a conservative point of view, to assume that the occurrence of one click per hour, due to a single incorrect PCM sample, and of one click per day, due to a slip, which would produce several consecutive incorrect PCM samples, might be tolerable.

If we assume a 32×10^3 sample/s repetition rate of PCM sound samples, then a random error rate of about 10^{-9} would appear to be necessary on digital paths. However, all the data collected up to now on practical systems seem to exclude the possibility of ensuring a random error rate of 10^{-9} in any digital path.

The CCITT has, in its Recommendation G.821, proposed error performance objectives for 64 kbit/s connections in the ISDN used for voice traffic or as a "Bearer Channel" for data type services. The 64 kbit/s connection referred to is a subscriber-to-subscriber all-digital Hypothetical Reference Connection (HRX) of 25 000 km in length.

With regard to slip rate objectives the CCITT has, in its Recommendation G.822, proposed an end-to-end slip rate performance for telephone and non-telephone services based on a 64 kbit/s digital connection. The slip rate objectives are stated with reference to the Hypothetical Reference Connection (HRX) of 25 000 km in length.

Further information on error performance objectives and slip rate objectives is to be found in Report 817.

It would appear that error protection (correction or mitigation) techniques will probably be necessary for digital sound programme transmission.

In the case of slips, if the digital network does not offer a tolerable slip rate, some mitigation techniques, such as "muting" could be economically provided. The muting impairment could probably be less perceptible than that caused by a slip, unless the slip duration is longer than a few milliseconds. This matter requires further study, including considerations of the effect which a muting technique may have upon the design of equipment, with particular reference to reframing performance.

As far as errors are concerned, to optimize the error correction or mitigation strategy, subjective tests should be carried out to establish, for example:

- the perceptibility of errors according to the weight of the various digits in the sample word;
- whether it is better to leave single errors in a sample or to repeat the last error-free sample, as a function of the weight of various digits;
- whether it is better to repeat the last error-free sample or to mute, as a function of the number of samples repeated or muted.

In [CCIR, 1970-74a] an error mitigation technique is proposed to be used together with an A-law instantaneous compander. This technique provides the detection of errors in the five most significant bits of the 10-bit sample by parity checking and the consequent repetition of the last error-free sample for up to five consecutive sample periods, if necessary, before muting. With certain specific assumptions, detailed in the document and with the objective of not more than one click per hour, theoretical calculation shows that the maximum tolerable random error rate on a digital path is then of the order of 10^{-5} . Subjective tests have been carried out in this problem for BERs of 10^{-3} , 10^{-4} and 10^{-5} , and the results are given in the addendum 1 to [CCIR, 1978-82a].

The Document [CCIR, 1970-74b] discusses the effect of transmission errors in near-instantaneous companding and gives results of studies of protection based on error-concealment techniques. Further studies, reported in [CCIR, 1974-78a] suggest that error-correction techniques are desirable.

Table I indicates the error protection strategy in use for different systems.

TABLE 1 – Error protection strategies used in different coding systems

CCIR References	[CCIR, 1978-82b, c]	[CCIR, 1978-82c]	[CCIR, 1974-78b]	[CCIR, 1978-82d]	[CCIR, 1970-74a]
Coding method	14/10 bit near instantaneous companding with 5 ranges		13/11 bit instantaneous companding with 7 segments	14/11 bit instantaneous companding with 11 segments	14/10 bit instantaneous companding with 13 segments
Error control redundancy	a) 12 times 5 bits for 5 MSB of 192 samples b) 7 bits for 60 bits additional to 338 kbit/s c) 12.6 bit blockcode for 2 scale factors	a) 48 bits for 5 MSB of 192 samples (Reed Solomon code) b) 14.6 bits BCH code for 2 scale factors	1 bit (parity) for 3 MSB of 1 sample (Iwadare BI code of rate 3/4)	1 bit (parity) for 5 MSB of 1 sample (odd parity)	1 bit (parity) for 5 MSB of 2 samples (odd parity)
Net bit rate per channel	323 kbit/s		352 kbit/s		320 kbit/s
Bit rate for error protection per channel	53 kbit/s	12 kbit/s	32 kbit/s		16 kbit/s
Error detection	Syndrome calculation			Parity bit checking	Parity bit checking
Protection against random bit errors	Single bit errors in one protected block of code words are corrected		Single burst errors of up to 4 bit length followed by 51 error free bits are corrected; detection of 6 non-zero syndromes within 2 blocks of 13 syndromes each, causes the 13 samples of the former block to be muted	Repetition of the last error-free sample code word	Repetition of the last error-free sample code word
Protection against error bursts on consecutive samples	If no error correction is possible an error concealment is done by interpolation or extrapolation techniques	Correction for up to 3 error bursts each of 6 bits length, or one of 26 bit length, in one block		Muting, if more than 3 of 4 consecutive sample code words are detected as erroneous. Muting released at low level after 5 error-free samples	Muting, if more than 5 consecutive sample code words are detected as erroneous
Information on subjective quality assessment	Doc. CCETT/CTN/T3/79	[CCIR, 1978-82e]	[CCIR, 1978-82f]	[CCIR, 1978-82a]	[Tomati, 1980]

2. Jitter

An investigation of the effect of timing jitter on the quality of sound-programme signals is described in [CCIR, 1974-78c]. The jitter which causes impairment perceptible to 50% of experienced listeners was determined using critical programme material; results were obtained for both sinusoidal and random jitter. Figs. 1 and 2 give these results and also propose tolerance limits based on impairment being perceptible to only 5% of experienced listeners.

It is stressed that the proposed tolerance limits in Figs. 1 and 2 do not apply to a digital transmission system itself; they are concerned only with the decoded analogue signal after passing through a digital system. There is a need for information on jitter characteristics of digital transmission systems. It is also desirable to have means for objectively measuring jitter on the decoded analogue signals; CCIR Recommendation 409 (concerning the measurement of wow and flutter in recording systems) may help further studies on this aspect.

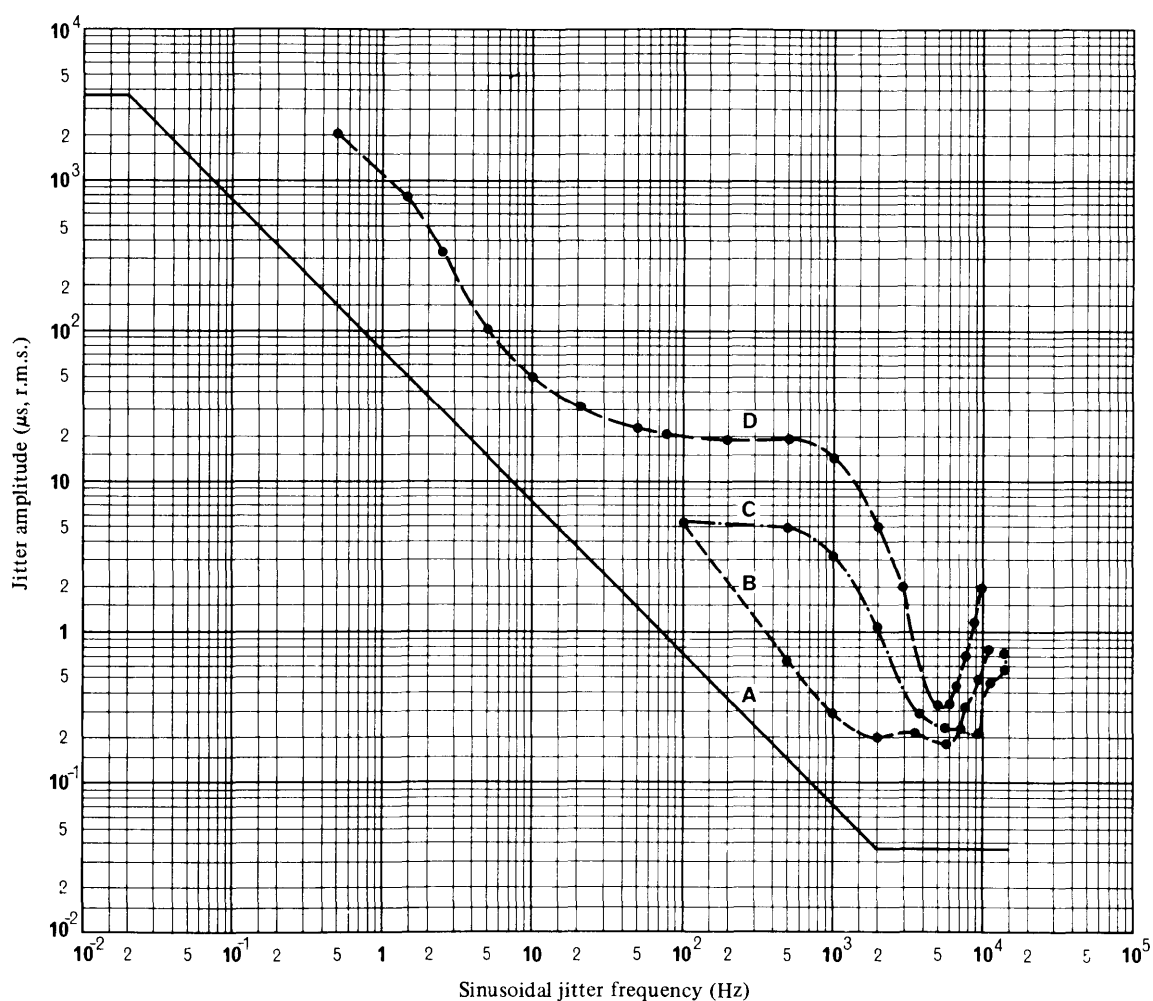


FIGURE 1 – Proposed maximum allowable r.m.s. amplitude of sinusoidal timing jitter in high-quality sound signals

Curves A : proposed tolerance limit

B : threshold, mean results, glockenspiel

C : threshold, mean results, piano

D : threshold, mean results, piano (derived from Stott and Axon [1955])

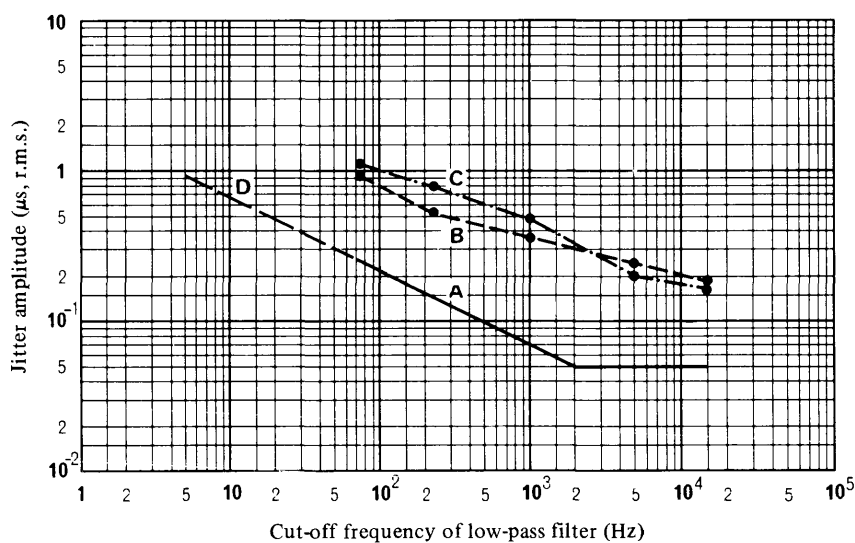


FIGURE 2 – Proposed maximum allowable r.m.s. amplitude of random timing jitter in high-quality sound signals

Curves A : Proposed tolerance limit

B : threshold, mean results, glockenspiel

C : threshold, mean results, piano

D : tentative tolerance limit for break-point in noise spectrum below 63 Hz

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STOTT, A. and AXON, P. E. [1955] The subjective discrimination of pitch and amplitude fluctuations in recording systems. *Proc. IEE*, Vol. 102B, 5.

TOMATI, L. [1980], Trasmissione numerica di segnali musicali in supervideo sui ponti radio televisivi (Digital transmission of high quality audio signals, inserted by means of a sound sub-carrier on television radio-links). *Elettronica e Telecomunicazioni*, 3.

CCIR Documents

[1970-74]: a. CMTT/204 (Italy); b. CMTT/145 (United Kingdom).

[1974-78]: a. CMTT/282 (France); b. CMTT/14 (Japan); c. CMTT/6 (United Kingdom).

[1978-82]: a. CMTT/82 (Add. 1) (Germany (Federal Republic of)); b. CMTT/92 (France); c. CMTT/93 (France); d. CMTT/81 (Germany (Federal Republic of)); e. CMTT/90 (France); f. CMTT/51 (Japan).

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SECTION CMTT D: METHODS OF OPERATION AND ASSESSMENT OF PERFORMANCE OF SOUND CHANNEL TRANSMISSIONS

Reports

REPORT 968

METHODS OF MEASURING THE BIT-ERROR DISTORTION OF DIGITAL CODED SOUND-PROGRAMME SIGNALS

(Study Programme 20A/CMTT)

(1982)

During the transmission of digital sound signals, bit-errors may occur along the transmission path. If there is a bit-error, the sample corresponding to the code group with the faulty bit will have an incorrect amplitude. So, the effect is the same as if there were an interference pulse superimposed on the audio signal. Where linear (uniform) coding is used, the size of the interference pulse depends simply on where the bit-error occurs in the code group. But when companding is used, the size also depends on the instantaneous value of the signal; and, in the case of near-instantaneous companding, on the scale factor. In these cases measurement of a parameter representative of the audible distortion is a problem. If we measure, for example, the errors occurring in a pseudo-random binary sequence, we are not taking into account the programme and therefore we may have an unrepresentative measurement.

One method [CCIR, 1978-82], devised by the EBU for research, to measure bit-error distortion in the presence of programme material on both transmission links and broadcasting paths is as follows. In summary, the audio signal is band-limited at the source; and, at the destination, the energy in a frequency window above the band limit is measured.

The method makes use of the fact that incorrect-amplitude samples, caused by bit-errors, all have the same maximum duration. They have a similar shape and the same zero crossings in their spectral power density function. Because of this * we need only measure the energies in a frequency window free of energy from the sound programme to obtain the relative total energies of the distortion pulses. The measurement is made in a window free of programme material above the band limit, but simultaneous with the programme, so that the effects of companding are part of the measurement. There are components of quantization noise in the measurement window, but (with an appreciable number of errors) they are very much smaller than the values to be measured.

In its studies, the EBU has used a low-pass filter at the sending end with a cut-off frequency of 12 kHz and a high-pass filter at the destination with a cut-off frequency of 13 kHz, so the measurement is made in the region 13-15 kHz. In fact the interference pulses are converted to damped oscillations by the receiving filters, but the measurement is not invalid because the damped pulses are of similar cyclic form.

Two measuring processes have been studied:

- measurement of the peak value of the interference pulse by a peak-detector;
- measurement of all interference pulses that exceed a certain threshold by a threshold-level detector.

By its nature, the method monitors the effectiveness of bit-error protection methods taking into account coding and transmission.

The EBU is currently assembling evidence on the relationship between subjective assessments of quality and objectively-obtained results using the method. Naturally, a consistent relationship between them must be shown for the method to be validated.

The method may still be valid for error bursts on the transmission-link where bit-dispersal (bit-shuffling) is used on the bearer medium. Experiments indicate that the method operates satisfactorily down to the limits of synchronization.

* The first zero crossing of the spectral power density function is at the sampling-frequency.

It should be possible to use the method with an unimpaired audio-bandwidth by using a measurement window of, say, 15-17 kHz with appropriate filters, where access to unfiltered D/A converter can be made available, and if there are no components due to possible dither of significant magnitude in the measurement window.

Administrations are requested to investigate the method and comment on its performance in practical cases.

REFERENCES

CCIR Documents

[1978-82]: CMTT/246 (EBU).

REPORT 642

INTERRUPTION TO SOUND-PROGRAMME SERVICES

(Study Programme 17C/CMTT)

(1974)

This subject requires considerable study and a possible approach is outlined below.

Since interruptions may be caused by many different phenomena such as circuit breaks, high noise levels and gross distortions, it may be that the most practical method of defining an interruption is by means of subjective assessment. It is suggested that an impairment which, if it were present continuously, would produce a subjective grade of 1 or 2 on the 5-point impairment scale (see Report 623), should be deemed to be an interruption according to a table of the form given below (Table I). This table is intended only as an example. Account is taken in the table both of the duration of the impairment, and of the number of such impairments in a given time.

TABLE I

Duration of impairment	Permitted rate of recurrence before condition is deemed to be an interruption
≤ 5 ms	1 per minute
> 5 ms but ≤ 0.5 s	1 per hour
> 0.5 s but ≤ 3 s	1 per day
> 3 s	Never

REPORT 822-1 *

TRANSMISSION OF SOUND-PROGRAMME SIGNALS OVER LONG DISTANCES **

Routine automatic testers for sound-programme circuits

(Question 19/CMTT)

(1978-1982)

Introduction

The general subject of measuring equipment for sound-programme circuits is dealt with by Study Group IV of the CCITT; and Recommendations O.31 and O.32 cover automatic measurements in a comprehensive way.

The availability of in-service testing methods for television circuits has directed attention to the development of analogous methods which could be used on sound-programme circuits. A suitable equipment has been designed by the Independent Broadcasting Authority (IBA) in the United Kingdom to perform a routine test of sound-programme circuits during the short interval which may occur between networked programmes. A similar equipment has also been developed in Italy by the Radiotelevisione Italiana (RAI).

The IBA experimental equipment which is described in part A below automatically records only amplitude/frequency response and noise level as it has been found by the IBA that these are the two parameters which cause most trouble on its sound-programme circuits. In this way the duration of test is kept to a minimum.

On the other hand, in addition to the amplitude/frequency response and weighted and unweighted noise, the RAI equipment permits the measurement also of harmonic distortion, as well as, in the case of stereophonic circuits, the total cross-talk and the non-linear cross-talk. This equipment is described in part B.

An automatic monitoring system for sound broadcasting channels without any interruption of the normal programme signal itself has been developed by the Federal Republic of Germany and is described below in part C. It has been in use in the Federal Republic of Germany for several years.

In the USSR, a complete system for the automatic monitoring of broadcast channel performance characteristics is under consideration, which monitors the whole transmission chain, including transmitters [CCIR, 1974-78].

A. AUTOMATIC MEASUREMENT EQUIPMENT PROPOSED BY THE IBA

1. General principles of operation

A remotely-controlled sender equipment disconnects the programme source and automatically generates a sequence of test signals. These signals are received at one or several receiving points on a sound programme network. A "gliding tone" signal is used. This is preferred to the use of a number of discrete frequencies because it has been observed that perturbations in the frequency responses of networks sometimes occur over a narrow range of frequencies. To allow the equipment to be used during short breaks between programmes it is essential that the test sequence should be completed very quickly; a duration of about 30 seconds has been achieved in practice.

The receivers have a high input impedance and are intended to bridge the programme path continuously. They respond to control tone signals from the sender which immediately precede the test sequence. The receivers detect the control code and activate a chart recorder which then records frequency response and noise level. This type of presentation of the measurements is preferred because it allows an immediate appreciation of the general performance of the network, — more readily for instance than from a column of figures.

2. The sender**2.1 The control code**

The control code is generated immediately before the glide tone. A sequence of three tones, each with a minimum duration of 20 cycles, is used. In present models the tones are at frequencies of 2.4, 2.6 and 2.8 kHz.

* This Report should be brought to the attention of CCITT Study Group IV.

** For the definition of absolute power, relative power and noise levels, see Recommendation 574.

2.2 *Gliding tone*

The test tone is swept from above 15 kHz passing through 0 Hz to approximately 0.1 kHz. The sweep is arranged to pass through 0 Hz to provide a further frequency calibration marker on the chart recording at the receiver. The total glide tone duration is 20 seconds. Two different rates of linear sweep are employed, from above 15 kHz to 0.5 kHz and below 0.5 kHz. This enables the frequency scale on the receiver chart to be expanded at the lower frequencies, thus allowing the chart to be read sufficiently accurately at those frequencies. The glide tone level is set at -10 dBm0s.

2.3 *Noise level*

At the end of the gliding tone sequence the code frequencies are again generated but in reverse order. This is to signal the beginning of the noise measurement period which lasts for about 5 seconds during the whole of which time the line at the sender end is terminated with a 600 ohms resistor. At the end of this period the line is automatically reconnected to the normal programme source.

2.4 *Emergency stop*

Because the equipment is designed to allow tests in intervals between programmes it is essential to allow for a situation where the equipment is energized when there is insufficient time available before the next programme transmission. The sender can be de-energized at any time after commencement of the test by pressing a reset button and this will immediately reconnect the network to the programme source.

3. **The receiver**

3.1 *Chart recorder*

When the first coded tone sequence is received, the chart recorder is switched on and runs at a paper speed of 5 mm per second. Thus, a complete test sequence will be drawn on about a 15 cm length of paper. The chart recorder is calibrated directly in dB. The centre of the scale represents -10 dBm0s and the paper is printed in ten large divisions. Each large division is sub-divided into five small divisions.

Full-scale deflection is set to plus and minus 10 dB so that each small division corresponds to 0.4 dB variation of the input signal.

3.2 *Frequency marker*

The recorder's stylus is deflected momentarily as the frequency sweep passes through 10, 5, 0.5 and 0.2 kHz. This allows the trace to be interpreted with linear interpolation between 0.5 kHz and 10 kHz, with extrapolation to 15 kHz; and with linear interpolation between 0.5 kHz and 0 kHz. (In practice the 0 kHz position can be clearly identified on the chart because the sweep passes through that frequency.)

3.3 *Noise test*

On receipt of the second coded tone sequence, an additional amplifier having a gain of 30 dB is switched into the signal path so that the dynamic range of the noise level display on the chart recorder is then -30 dBm0s to -50 dBm0s.

A Peak Programme Meter (PPM) type of circuit is used in the receiver (see Note). The time constants are modified in the circuit with the aim of giving, as closely as possible, the same indication of noise level from the chart recording as would have been obtained by an operator observing a standard ppm at the time of the test.

The total duration of the noise test is about 5 seconds. For the first 2½ seconds, "unweighted" noise level is recorded. For the remainder of the period a weighting filter is switched into the circuit so that "weighted" noise is recorded.

Note. — The principal characteristics of this type of instrument are shown in Table I of Report 292, and also in CCITT Recommendation J.15.

3.4 *Termination of test*

At the end of the test sequence the recorder is automatically switched off and the recording stylus de-energized.

B. AUTOMATIC MEASUREMENT EQUIPMENT PROPOSED BY THE RAI

1. Principles of operation

The measurements are undertaken at five different frequencies suitably chosen, in order to give the most significant results on various types of equipment, such as: cables, radio-links, circular broadcast transmitters.

The chosen frequencies are: 0.063 — 1 — 4.5 — 10 — 14.7 kHz.

The measurements made are: amplitude/frequency response, harmonic distortion, total cross-talk, non-linear cross-talk, weighted and unweighted noise.

Normally the test tones are produced at a level of -12 dBm (on 600 ohms) for the measurement of the amplitude/frequency response and are repeated at a level of $+9$ dBm for the measurements of distortion and cross-talk.

These levels may not be the most suitable in all cases, therefore it is possible to select, by means of a control placed on the front panel, a "measurement programme", i.e. a set of levels, one for each test tone. Some test tones may also be suppressed.

The measurement programmes may be pre-arranged by acting on internal diode matrixes.

A complete measurement cycle lasts about 130 s.

The equipment consists of a transmitting and a receiving unit. On both the transmitter and the receiver, there are indicators which identify the type of measurement being performed, and on the receiver a logarithmic instrument is used to show the result. The receiver has also an analogue output for a strip-chart recorder.

The measurement range is 40 dB; for each parameter this range is centred on the values that are generally found on a typical circuit.

The equipment is fully solid-state, internal switching devices included.

2. Description of the transmitting unit

In order to carry out the measurements, the transmitter is connected to the input of the sound circuit (one of the two channels in the case of a stereophonic circuit). When it receives a "start" command, the transmitter sends the sequence of frequencies as indicated in Fig. 1. The first group of 3 frequencies is the "start signal". When the receiver at the end of the circuit recognizes this signal, it prepares to make the measurements on the subsequent frequency groups.

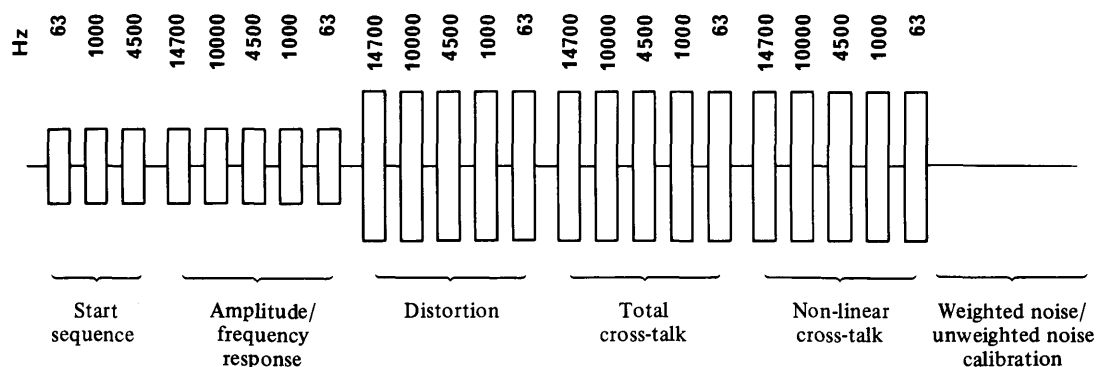


FIGURE 1

The test frequencies are derived from a 1 MHz crystal oscillator by means of a programmable logic divider. The transmitter logic section provides for the programming of the divider and for the generation of the appropriate necessary pauses.

The signals are made sinusoidal, with low distortion, by means of LC filters and are sent to an output amplifier which feeds the circuit under test.

The output level, $+9$ dBm nominally, is adjustable by 1 dB steps according to the characteristics of the circuit under test.

Moreover, an attenuator and an amplitude-correcting network can be inserted by the transmitter logic circuit, according to the measurements to be carried out and the "programme" chosen.

At the end of all the frequency groups, a pause is produced, during which the circuit is still occupied by the measuring system. During this time interval, the receiving unit carries out the measurement of noise. After that, the counter resets the bistable circuit and stops the transmitter.

3. Description of the test receiver

When a stereophonic circuit is to be tested, the receiver is normally connected to the channel which is fed by the transmitter, but, during cross-talk measurements, it is automatically switched to the other channel. Of course, for the complete test of a stereophonic circuit, two measurement cycles are needed. During the first cycle the transmitter is to be connected to channel L, while during the second, it is to be connected to channel R (or vice versa).

In order to carry out the measurement of harmonic distortion, the bandstop filter relevant to the frequency under examination is inserted. The cross-talk measurements are undertaken on the channel not fed by the transmitter. The total cross-talk is measured without including the bandstop filters, the non-linear cross-talk including these filters. When measuring the weighted noise, the psophometric filter is inserted.

Between the measurement of non-linear cross-talk and that of the noise, the equipment undertakes the measurement of the amplitude of an internal calibration generator which produces a staircase corresponding to 0, 10, 20, 30, 40 dB levels. This operation serves to check the receiver gain and the scale calibration of the chart-recorder.

The typical appearance of a chart-recording is illustrated in Fig. 2.

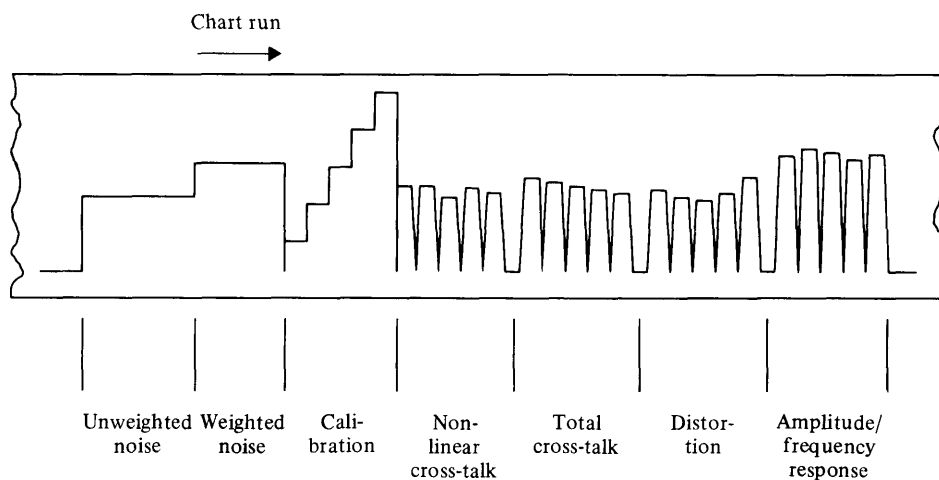


FIGURE 2

The frequencies are recognized by means of a number of selective detectors, which control the logic section.

The arrival of the first sequence of tones excites a "start signal identification circuit" which sets a bistable circuit in the state corresponding to the receiver operating condition and prepares it for the amplitude/frequency measurement.

The identification circuit has been designed so that it is insensitive to the programme signal.

The equipment stops at the end of the measuring cycle (after the measurement of noise) and also when the received test signal sequence is accidentally interrupted. In this way, the danger of the equipment remaining blocked in the middle of a measuring cycle is avoided.

C. AUTOMATIC MONITORING EQUIPMENT USED IN THE FEDERAL REPUBLIC OF GERMANY [CCIR, 1978-82]

1. Principles of operation

The programme signals themselves are used to monitor the transfer characteristics of the sound programme circuit, i.e. frequency response, non-linear effects and signal-to-noise ratio. The programme signals at the input to the circuit are analyzed for these parameters by means of three different filters, and the results digitized with a resolution of 0.25 dB. By using a low-level 15 kHz digitally modulated sub-carrier, the data on the input signal to the circuit are transmitted together with the programme signal. The receiver compares the transmitted data with simultaneous measurements on the programme signal at the output of the circuit.

2. The sender

The sender removes the programme signal in the range of 14.95 kHz to 15.05 kHz and inserts a carrier with -73 dBm0 at 15 kHz. The carrier is phase-modulated (2 PSK) with 25 bit/s and contains the information for synchronization of the downstream receivers, and data of the input programme signal to the circuit.

3. The receiver

The receiver decodes the data on the 15 kHz carrier and synchronizes the programme signal analyzer with the information about the input signals to the circuit. By means of a micro-computer, the differences in amplitude between the input and output programme signals of the circuit in each of the frequency bands defined by the filters, are compared with stored tolerance limits. If the tolerance limits are exceeded, alarms are given to take the transmission circuit out of service for fault localization and maintenance.

REFERENCES

CCIR Documents

[1974-78]: CMTT/109 (U.S.S.R.)

[1978-82]: CMTT/263 (Germany (Federal Republic of)).

REPORT 623

METHOD PROPOSED FOR THE SUBJECTIVE ASSESSMENT OF THE QUALITY OF SOUND IN BROADCASTING AND OF THE PERFORMANCE OF SOUND-PROGRAMME SYSTEMS

(Study Programme 19A/CMTT)

(1974)

1. Introduction

The purpose of subjective listening tests in sound broadcasting and reproduction is to assess by means of quality and impairment scales the degree of annoyance caused by any impairment to the wanted signal during the course of its transmission from the originating source to the listener. This implies that a programme sequence which is impaired after its passage through the system under test should be compared with the original sequence which preferably should be of "excellent" quality with "imperceptible" impairment.

Programme sequences used for testing should include silent intervals so that, in the absence of the wanted signal, the subjective assessment of the impairment caused by noise in the system is not excluded. On the other hand, the tests should exclude any assessment of defects, the audible effects of which might, in certain cases, not be objectionable and which might even give a subjective impression of improved quality. The programme sequences used for testing should, therefore, be free from any audible defects similar to those produced in the system under test but where this is impracticable, the consequent limitations on the validity of the results should be clearly indicated.

A system of lights should be used to indicate to the listener at all times the source (impaired or unimpaired) of the programme he is hearing.

The amount of data which needs to be collected depends upon such interrelated factors as the degree of statistical confidence which is needed in the answer, the standard deviation of the measurements, and the relative magnitude of the effect which it is required to detect. The following suggestions are intended as guide-lines to assist in formulating a considered experimental design.

2. Protection ratios

The assessment of receiver protection ratio, that is the wanted-to-unwanted signal ratio that results in a specified grade of impairment, requires a slightly different testing procedure. In this case, the unimpaired programme sequence used for comparison should be such that the sound quality reproduced by the receiver is appropriate to the broadcasting system for which the receiver is designed.

3. Selection of listening panel

Although in a normal listening audience there will be some expert listeners*, the proportion of them is likely to be so small that it is proper to concentrate the objective of laboratory tests on the opinions of non-experts, because the use of experts could lead to results which are much more critical than would be obtained with normal listeners. The choice of test listening conditions should be more critical than average but not unduly so. As tests with non-expert listeners tend to be lengthy, it is often desirable that a quick test should be carried out by experts. In this case, a smaller number of listeners can be used. However, it should be noted that in certain circumstances tests carried out with expert listeners may not be a satisfactory substitute for tests carried out by non-experts. In cases of doubt, the relationship between expert and non-expert opinion should be investigated.

The minimum number of non-expert listeners should normally be twenty whilst the minimum number of expert listeners should normally be ten. In all cases, the number and category of listeners and the duration of the tests should be stated. Whenever the system is intended for high-quality sound broadcasting or reproduction, expert listeners should be used exclusively.

4. Test procedure and duration

Because of the extreme unreliability of the long- or medium-term aural memory, the instantaneous comparison method should always be used.

Each process involves the repetition, four times consecutively, of the same programme sequence in the following order:

- original sequence;
- same sequence, impaired;
- original sequence (repeated);
- same sequence, impaired (repeated).

After these four repetitions, the listener should be allowed 15 to 20 s to make a judgment and to prepare for the next test.

Programme sequences should not last longer than 15 to 20 s.

The four repetitions of the same sequence should be separated by intervals of silence of about 0.5 to 1 s, the duration depending upon the type of programme. The instantaneous switching device should not introduce audible interference.

The programme sequences and impairments should be presented in random order subject to the condition that the same sequence should never be presented on two successive occasions with the same or different levels of impairment.

No session with any one listener should last for more than about 15 to 20 min without interruption. If the sessions must be consecutive, they should be separated by rest periods of roughly the same length.

5. Choice of programme sequences

Depending on the precise objective fixed and in particular on the category of the broadcasting or reproduction system tested, the following programme sequences should be used.

- either a representative selection of typical programme material,
- or, a selection of a few sequences picked deliberately for their highly critical behaviour with respect to the impairments introduced by the system being tested. For example, when assessing protection ratios, a suitably critical test sequence would be speech on the wanted programme impaired by “pop” music on the unwanted programme.

Whenever the system is intended to carry high-quality sound, the second type of programme sequence should be used.

In any event, the artistic or intellectual content of a programme sequence should be neither so attractive, nor so disagreeable, or wearisome, that the listener is distracted from his purpose.

6. Sound level

As it is a matter of establishing degrees of annoyance for certain impairments, the sound level during the tests is obviously of great importance as aural perception varies considerably with the sound level and any masking effect of room noise. The level should, therefore, be exactly specified.

The sound level, of the wanted signal or room noise, should be measured with a sound level meter with weighting *A* and the “slow” time constant standardized by the IEC (Publication 123). For measurement of the sound level of a programme sequence in the special conditions of the test and at a given position in the listening room, the sound level will be taken by definition as equal to the maximum value shown by the sound level meter during the sequence.

* The term “expert listeners” is considered to apply to listeners who have had recent extensive experience of assessing sound quality or impairment particularly of the type being studied in the subjective tests.

The sound level considered in defining exactly the conditions in which the tests have been carried out will be the mean of the sound levels measured at the various positions occupied by the listeners. The difference from this mean value for any position must be as small as possible. A value of ± 4 dB might be reasonable.

7. Listening conditions

Whenever the tests are conducted with loudspeakers, the organizer of the tests should give, for the benefit of other workers, all the essential information concerning the dimensions and the reverberation time of the listening room, the arrangement of listeners in the room and their distance from the loudspeaker or loudspeakers.

Generally speaking, an effort should be made to minimize the masking effect produced in the listening room by room noise, particularly when establishing tolerances for high-quality sound transmission.

Whatever mean level is fixed for the room noise, however, it should always be indicated and when it is manifestly likely to have a noticeable masking effect, the mean spectrum should also be indicated.

Furthermore, precautions should be taken to prevent as far as possible the listener(s) from being annoyed or distracted by certain features of the surroundings (temperature, light, moving objects or persons, etc.).

8. Grading scales

The following five-point scales which are identical to those used for the assessment of television pictures (see Recommendation 500) should be used for the assessment of quality and impairment. The nature and object of the tests will determine which of the two scales is the more appropriate.

Five-point scale	
Quality	Impairment
5 Excellent	5 Imperceptible
4 Good	4 Perceptible but not annoying
3 Fair	3 Slightly annoying
2 Poor	2 Annoying
1 Bad	1 Very annoying

For certain types of subjective tests it may be more convenient to use a comparison scale in which case the following seven-point scale should be used:

- 3 Much better
- 2 Better
- 1 Slightly better
- 0 The same
- −1 Slightly worse
- −2 Worse
- −3 Much worse.

9. Presentation of results

The results obtained by the use of expert listening panels should be presented separately from those provided by non-expert panels and details should be given of listening conditions, sound levels and any statistical methods used to analyze the test results should be described.

BIBLIOGRAPHY

CCIR Documents

[1970-1974]: CMTT/45 (France); CMTT/167 (10/202) (United Kingdom).

For information, see Report 820, Section CMTT C.

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SECTION CMTT E: TRANSMISSION OF SIGNALS WITH MULTIPLEXING OF VIDEO,
SOUND AND DATA AND SIGNALS OF NEW SYSTEMS*Recommendations and Reports*

REPORT 412-3

TRANSMISSION TIME DIFFERENCES BETWEEN THE SOUND
AND VISION COMPONENTS OF A TELEVISION SIGNAL

(Study Programme 21A/CMTT)

(1966-1970-1974-1978)

1. Introduction

In the early days of long-distance television transmission, when the video and sound were transmitted via different facilities, there occasionally existed a perceptible lack of simultaneity between sound and vision. Over the intervening years, the networks have been improved. Today, even though the sound and vision of a television programme may be transmitted via different facilities, the transmission velocities used, are such that little or no lack of simultaneity between sound and vision is experienced.

With the advent of satellite-communication systems, where the sound and vision signals may be transmitted via different media, difficulties may exist at the receiver location because of the lack of synchronization of sound and vision.

2. Conclusions based on observer reaction

Studies of observer reaction to non-simultaneous presentation of pictures and associated sound were reported in [CCIR, 1963-66a and b; CCIR, 1966-69]. General conclusions may be drawn from these experiments.

2.1 Qualitative conclusions

The observer is more sensitive to sound advances than to sound delays with respect to the correlated visible action.

Observer tolerance to sound/vision time differences varies according to the nature of the action. For example, the objectionable effects of the more easily correlated actions and sounds (such as the percussive sound produced by an object being struck) are detected at considerably shorter time differences than are lip motions of a speaker and the corresponding sounds. For scenes of people moving about, correlation between sound and action becomes more difficult for the observer to detect.

2.2 Quantitative conclusions

Although the conditions under which the experiments in Canada, the United States of America and the Federal Republic of Germany were conducted differed widely, the results were in broad agreement. In terms of these results, it is possible to derive some provisional figures for overall tolerance to the lack of simultaneity between the sound and the picture.

2.2.1 Sound delayed

For sound delayed with respect to vision, 140 ms will produce, approximately, a "just perceptible impairment" for 50% of the observers.

2.2.2 Sound advanced

For sound advanced with respect to vision, 70 ms will produce, approximately, a "just perceptible impairment" for 50% of the observers.

2.2.3 The above figures based upon the experiments in Canada and the United States of America are more stringent than necessary in some circumstances, but they allow for some evidence that sensitivity to sound/vision time differences might be greater for some languages than for English in which the first experiments were conducted.

The experiments in the Federal Republic of Germany [CCIR, 1966-69] have confirmed the above figures using a percussion test which constitutes the most critical case. Moreover, it was found that the tolerances depend very little on the different programme material used. Hence the conclusion may be drawn that the same holds true for other programme material and languages which were not included in the tests.

3. The problem of division of time differences among the links of a built-up connection

The figures cited in § 2.2 must be divided between acoustic links (sources to studio microphone, and television receiver to observer) and electrical links. The electrical links may in turn be subdivided between the broadcasting facilities and the telecommunications transmission networks. Very little information is available on which a practical division may be based, but this is being studied under Question 35/11.

As an example, in the United States of America, where a variety of programme material is transmitted, it has been found economically feasible on transmission networks to limit sound delay relative to the vision component of a television programme by not more than 50 ms, and sound advance by not more than 25 to 30 ms, at the receiving location. The advance or delay of sound of a television signal that may be introduced because of the difference in transmission velocities on network facilities, can be determined. The advance or delay figures of sound indicated are primarily concerned with the long-distance portion of the network and do not include the inter-city or other contributing sources to the time difference of sound and vision components of the television signal. Other electrical and acoustic links make up the remainder of the sound/vision time difference.

4. Responsibility for making necessary corrections of time differences (Study Programme 21A/CMTT)

If the sound and vision components of a television signal are transmitted along the same route, and by the same method, there will be no perceptible time difference between the two signals. Where such common routing is not feasible Administrations are urged to provide circuits resulting in time difference values as low as possible.

If, in spite of this, considerable time differences should occur, it should normally be the responsibility of the broadcasting authorities concerned to make the necessary corrections.

This arrangement seems to be most appropriate for the following reasons:

- as the two channels only have to coincide at the source and at the destination there is a greater flexibility for routing of channels;
- in general, broadcasting authorities will have various possibilities open to them to effect such corrections so that there is no need for the Administration and/or recognized private operating agencies to provide such equipment.

5. Methods of controlling time differences between sound and vision signals on long-distance connections

In a circuit of 2500 * km, the delays are as follows:

- for an individual modulation/demodulation equipment for high-quality sound transmission over carrier links (15 kHz): less than 3 ms;
- for modulation/demodulation equipment for sound transmission on a television circuit, e.g. systems given in Report 488: a few microseconds;
- where sections forming the circuit wholly comprise one of the following transmission media:
 - free space: 8.3 ms;
 - radio-relay links: 8.4 ms;
 - coaxial pair 2.6/9.5 mm (CCITT standard): 8.9 ms;
 - coaxial pair 1.2/4.4 mm (CCITT standard): 10.0 ms;
 - loaded lines:
 - 6.4 kHz bandwidth: 52.0 ms;
 - 10 kHz bandwidth: 40.0 ms;
 - 15 kHz bandwidth: 25.0 ms.

The time differences mentioned in § 2 will not be exceeded in practical applications if the following rules are observed:

- in the case of satellite transmissions, both the picture and the sound components should be transmitted via the same satellite;
- for terrestrial transmission, the television-sound circuit should not employ significant lengths, (e.g. not more than 100 km), of the following:
 - loaded circuits,
 - audio-frequency or carrier-frequency circuits, the basic delay of which is substantially increased by delay equalization.

* The length of 2500 km corresponds to the length of the hypothetical reference circuit for television given in Recommendation 567, and also to the length of the hypothetical reference circuit for sound-programme transmission given in Recommendation 502.

6. Further work

Contributions are invited from various Administrations based on experience and practice with regard to the following:

- the minimum time-difference which is generally considered practicable to achieve in long circuits;
- new experiments which introduce conditions beyond those previously used.

With regard to methods of controlling sound/vision time differences arising on the long-distance circuits, Administrations are invited to study § 5 above to determine its suitability as a basis for a new Recommendation.

REFERENCES

CCIR Documents

[1963-66]: a. CMTT/1 (Canada); b. CMTT/12 (United States of America).

[1966-69]: CMTT/55 (Federal Republic of Germany).

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.

REPORT 488-3 *

TRANSMISSION OF SOUND AND VISION SIGNALS BY TIME-DIVISION MULTIPLEX OR FREQUENCY-DIVISION MULTIPLEX **

(Study Programmes 21A/CMTT and 21B/CMTT)

(1970-1974-1978-1982)

1. Introduction

It is important to ensure that the difference in transmission time between the sound and picture signals comprising a television signal is kept to an acceptable minimum. One method of achieving this result is to combine these signals in a time-division or frequency division multiplex arrangement so that the combined signals can pass over a common circuit. The CMTT has been studying these techniques and the results which have been obtained so far are summarized below. ***

* This Report should be brought to the attention of Study Group 9.

** For the definitions of absolute power, relative power and noise levels, see Recommendation 574.

*** Other multiplex methods which can also provide acceptable differences of timing between the sound and picture signals, but which result in these signals being carried on two separate circuits or facilities within a common path, are being studied by CCIR Study Groups 4 and 9, and by CCITT Study Group XV. Examples of particular solutions are given in Recommendations 402, 403, Report 289 and other Reports of Study Group 9.

2. Time-division multiplex of sound and vision signals

A television signal requires a wide video bandwidth for satisfactory transmission by analogue means and this bandwidth is not fully utilized during the synchronizing and blanking periods. Therefore, it is practical for these periods to be used for the transmission of sound signals without requiring any increase in video bandwidth or making any significant extra demand on the video transmission system. It has been shown that practical systems of modulation which achieve the required standards of performance for the sound channels are feasible.

The United Kingdom has produced time-division multiplex equipment which is suitable for many 625- and 525-line systems [CCIR, 1970-74a]. Equipment of this type has been in service for some years in the United Kingdom and elsewhere [CCIR, 1974-78a and b]. The audio signal is sampled at twice line-frequency and the information is stored and coded into two groups of binary-coded pulses which, together with an additional marker pulse, are transmitted during the line synchronizing pulse period. A high quality music channel of about 14 kHz bandwidth is obtained and conforms to Recommendation 572. It operates satisfactorily under the following video circuit conditions:

- Video peak-to-peak picture to unweighted r.m.s. random noise ratio of 29 dB or better.
- Pulse lobes at the limiting values for the hrc. *
- Field-time distortion equivalent to that caused by a 1st order HP filter with a time constant of 15 ms.
- Signal/power-frequency periodic noise ratio of 11 dB. **
- RF loss such that the sine-squared pulse to bar ratio * is down to 72%.
- Long-term variations in the duration of the television line of +1.5 μ s.
- Instantaneous variation in the duration of the television line of 1 μ s.
- Chrominance/luminance delay inequality of 110 ns.

It is considered to be quite practical to equalize a long international television connection (Recommendation 603) to meet these conditions.

Recent tests [CCIR, 1974-78c] have shown that for half bandwidth satellite transponder operation the sound pulse amplitude should be reduced by 6 dB.

This system also conforms to Recommendation 505, except that the audio bandwidth is reduced from 15 kHz to 14 kHz.

Other information on this system is available [Shorter, 1969].

In the U.S.S.R., various studies have been made of time-division multiplex systems for the simultaneous transmission of sound and vision signals [Talyzin *et al.*, 1967; Sevalnev and Tsirlin, 1969; Kataev and Zubarev, 1965; Yaschenko and Zverev, 1964; Esin *et al.*, 1968]. In time-division multiplex systems it is possible to use methods of sound signal transmission other than PCM. On the Molniya and Orbita satellite systems of the U.S.S.R., sound signals are transmitted as a position modulation of pulses on the back-porch of the video line-blanking signal. Table I gives some details of two time-division multiplex systems for use with the SECAM colour system [CCIR, 1966-69].

The reference [CCIR, 1970-74b] describes an experimental system for analogue time-division multiplex transmission of one or more sound channels applicable to all video systems. Pulse-amplitude samples of the sound signals are time-compressed and inserted in vacant lines of the vertical blanking interval for transmission. At the receiver, the time-compressed signals are time-expanded to their original duration. The method is especially suited for transmission of many (for instance, 20) commentary channels of medium-quality performance.

The references [CCIR, 1970-74c] and [CCIR, 1974-78d] describe a system for digital time-division multiplex transmission of two high-quality 15 kHz sound channels according to Recommendation 505. The PCM encoded sound signals are placed on the back-porch of the line-blanking interval. Furthermore, in [CCIR, 1978-82a] a modification of the system is described in order to improve the mode of operation and to increase transmission reliability. The modified equipment is using the source coding according to Report 647, Table I, column B5 [CCIR, 1978-82b]. For insertion of two sound signals, the back-porch has to be extended by reducing the line sync. pulses. Each one of the 15 kHz sound channels can be replaced by two 7 kHz channels or three 5 kHz sound channels, according to Report 641 (Geneva, 1974), thus allowing for the transmission of up to four sound channels. For 625-line PAL systems, this system has been extensively and successfully tested on radio-relay and satellite links [CCIR, 1974-78e]. Modified equipment will be available for system B/G PAL. Adaptation of the equipment to standards NTSC and SECAM is under consideration [Dirks *et al.*, 1977].

* See Part D, § 3.5.1.4 of Recommendation 567.

** See Part B, § 3.2.3 of Recommendation 567.

The reference [CCIR, 1974-78f] describes an experimental system for transmission of four high-quality 15 kHz sound channels using the time-division multiplex technique according to Recommendation 505. The complete horizontal blanking interval is replaced by a binary NRZ-coded PCM-signal carrying the sound information as well as the information about line-synchronization and colour sub-carrier frequency. For other purposes each of the 15 kHz sound channels can be replaced by three 5 kHz sound channels according to Report 641 (Geneva, 1974), which allows the transmission of up to 12 sound channels supplementing the picture signal.

The reference [CCIR, 1970-74d] describes an experimental pulse-code modulation system for the 525-line system M. The audio signal, coded into two groups of 6 digit, ternary-coded pulses, modulates an elongated burst of chrominance sub-carrier situated in an extended video line-blanking interval. With the decreased duration of the line-synchronizing pulse, two high-quality sound channels having an upper frequency of about 14 kHz can be obtained [see also Fujio and Komoto, 1971].

The reference [CCIR, 1974-78g] describes a method for the transmission of two high-quality sound signals, or four medium-quality (commentary) sound signals, or eight telephony signals, or any suitable combination of these, by time-division multiplex with a video signal. Quaternary coding is employed, but half the above audio capacity may be obtained if necessary by using binary coding. The system is applicable to PAL, NTSC and SECAM.

Table I summarizes the information on the various systems described in § 2.

TABLE I – Summary of TDM systems

Country and recent CCIR Document	CMTT/6 (United Kingdom) 1970-74	CMTT/144 (U.S.S.R.) 1966-69		CMTT/12 (11/34) (Japan) 1970-74	CMTT/34 (Federal Republic of Germany) 1970-74	CMTT/197 (Federal Republic of Germany) 1970-74 CMTT/242 (Federal Republic of Germany) 1974-78 CMTT/79 (Federal Republic of Germany) 1978-82	CMTT/243 (Federal Republic of Germany) 1974-78	CMTT/187 (German Democratic Republic) 1974-78
Video System	625-line (PAL) 525-line (NTSC)	625-line (SECAM)		525-line (NTSC)	Adaptable to all systems. (Practical tests only with 626-line PAL)	Adaptable to all systems. (Practical tests only with 626-line PAL)	Adaptable to all systems. (Practical tests only with 626-line PAL)	Adaptable to all systems
Audio-modulation system	Pulse-code modulation	Pulse-width modulation		Pulse-code modulation (Ternary of colour burst)	Time compressed pulse amplitude modulation	Pulse-code modulation (Binary NRZ code)	Pulse-code modulation (Binary NRZ code)	Pulse-code modulation (Quaternary NRZ code)
Video-waveform modification	Increased duration of line-frequency equalizing pulses	(a) Increased deviation line-blanking (b) Line-sync. pulse displaced in time and its duration decreased (c) Decreased duration of front porch	(a) Increased deviation line-blanking (b) Line-sync. pulse displaced in time and its duration decreased	(a) Line-sync. pulse decreased in duration (b) Line-blanking inserted through field blanking (c) Colour burst extended	Unused lines of vertical blanking occupied by sound signal	Increased back porch of line blanking interval by reduced duration of line-sync. pulse	Replacing the complete line-blanking interval by a binary NRZ coded PMC signal	(a) Line-sync. pulse decreased in duration (b) Line-blanking inserted through field blanking (c) Colour burst modified
Audio bandwidth (kHz)	14	6	10-14	14	13	15 or 7 or 5	15 or 5	15, 7, 3.4
Number of audio channels	1	1	1	2	Equal to number of available lines in the vertical blanking	2 (15 kHz) or 4 (7 kHz) or 1 (15 kHz) + 2 (7 kHz) or 1 (15 kHz) + 3 (5 kHz) or 6 (5 kHz)	4 (15 kHz) or 2 (15 kHz) + 6 (5 kHz)	2 × 15 kHz or 4 × 7 kHz or 8 × 3,4 kHz
Audio signal-to-weighted-noise ratio (dB) ⁽¹⁾	53 dBQ0ps (with compandor) absolute noise level	60 (video S/N ratio: 41 dB)	53 (with compandor) (video S/N ratio: 55 dB)	52	39 (with compandor) (video S/N ratio: 52 dB)	68 (idle channel noise)	60 (with compandor)	63 dB } idle and 65 dB } channel noise (weighted according to Recommendation 468)
Audio S/N dependent upon video S/N	No	Yes	Yes	No	Yes	No	No	No

⁽¹⁾ The values given are not all on the same scale. Revised values should be contributed based on Recommendation 468 to enable valid comparisons to be made.

3. Frequency-division multiplex of sound and vision signals

Techniques of multiplexing sound and vision signals by frequency-division multiplex so that both signals are carried by the same video bearer circuit have been suggested. Such systems which are designed to be dependent on a particular transmission medium are the primary concern of the related CCIR and CCITT Study Groups. For instance, frequency-division multiplex techniques devised for radio relay systems, are described in Recommendations 402, 403 and Reports 289, 786. Systems which are designed to be independent of a particular transmission medium are the concern of the CMTT. It should be noted that unlike the time-division multiplex solutions proposed, the frequency-division multiplex techniques may require a bandwidth somewhat larger than the nominal video bandwidth of the standard concerned; also better linearity may be needed to avoid cross-talk problems. However, difficulties which may be associated with the presence of pulse trains in the video signals – sync. bottom clamping, monitoring, etc. – are eliminated.

Particular solutions using this technique have been described in [CCIR, 1974-78h] in which two methods currently being employed in the United States are mentioned. These use sub-carriers to diplex high-quality sound programmes along with the vision signal for the 525-line system.

Method 1 involves a sub-carrier at 5.5 MHz which is 4-phase-modulated by a digital signal with the capability of encoding 4 programme circuits of 15 kHz bandwidth. *

Method 2 involves two sub-carriers at 5.8 and 6.4 MHz each of which can be frequency modulated, providing 2 programme circuits of 15 kHz bandwidth. *

Neither of the two methods requires any change of video waveform to implement; nor do they measurably degrade performance or reliability of the vision signal for circuits of up to 6400 km.

In Japan, television broadcasting accompanied by two sound signals is offered to provide stereophonic or bilingual sound programmes. A television programme relay system for simultaneous transmission of picture signal and two sound signals [Matsumoto *et al.*, 1980] is operating satisfactorily on a commercial basis. The individual sound channel characteristics conform to performance objectives in Recommendation 504. Test results on transmission characteristics and subjective assessment were satisfactory [CCIR, 1978-82c].

Other solutions using this technique have been described in [CCIR, 1978-82d] in which a method currently being employed in France is mentioned. ** This uses sub-carrier at 7.5 MHz which is differentially phase modulated by a digital multiplex at 2048 kbit/s with the capability of encoding six programme circuits of 15 kHz bandwidth. For implementation this method requires no changes of video waveform.

Table II summarizes the information on the systems described in § 3.

4. TDM transmission of digital vision and sound signals

A technique for the transmission of digitally coded sound and vision signals by time-division multiplex (TDM) has been reported [CCIR, 1978-82f].

In the experimental system, two high-quality sound signals with 15 kHz bandwidth are sampled at twice television line-frequency and they are coded into 13 bit linear PCM form, which is then reduced to 11 bit non-linear PCM form by 7 segment companding. These bits are stored and transmitted after the DPCM coded vision signal for each line period. Thus, the time division multiplex frame period corresponds to the television line period.

Test results show that this system can well be used for stereophonic high-quality sound signal transmission.

5. Compatibility with sound programme signals on digital paths

The reference [CCIR, 1970-74e] draws attention to proposals which are being made in order to insert sound programmes on digital paths. If a Recommendation for such a system can be agreed, then it would be advantageous to adopt the same technique, with the same number of bits per sample and the same coding law, for the insertion of a sound programme in the video line-blanking intervals. In this case, the sampling frequency could be twice the appropriate line-repetition frequency, and may, if necessary, be synchronized with it. Since the most likely sampling frequency of a digital sound programme transmission system according to Report 647 will be a multiple of 8 kHz, a compatibility problem may arise. Direct digital-to-digital conversion may be used to solve this problem and contributions on this topic are invited.

* Other sub-carrier frequencies can also be used, but an analysis must be performed on the system intermodulation characteristics and impact on video S/N ratio.

** Similar systems will be employed in Norway [CCIR, 1978-82e], the Federal Republic of Germany and the Netherlands.

TABLE II – Summary of FDM systems

Country and recent CCIR document	CMTT/44 (United States) 1974-78		CMTT/53 (Japan) 1978-82	CMTT/90 (France) 1978-82	CMTT/6 (Norway) 1978-82
Video system	525-lines (NTSC)		525-lines (NTSC)	Black and white (SECAM, PAL)	625-lines (PAL)
Audio modulation system	Digital 4-PSK	FM	FM	Digital by 4-PSK Modulation	Digital QAM
Video waveform modification	None	None	None	None	None
Audio bandwidth (kHz)	15	15	10	15	15
Number of audio channels	4	2	2	6	5
Audio signal to weighted noise ratio (dB)	60	60	58	–	–
Quantizing noise for the finest resolution in digital systems in dBq0s (overload level)	– –	– –	– –	– 65 (12 dBm0s)	– 62 (15 dBm0s)
Audio S/N dependent on video S/N	No	No	No	No	No
Sub-carrier frequency	5.5 MHz(1)	5.8 MHz(1) 6.4 MHz(1)	6.5 MHz(1) 7.65 MHz(1)	7.5 MHz	7.232 MHz

(1) Other sub-carrier frequencies can be used, but in that case the system intermodulation effects must be analysed and taken into account in the video signal/noise ratio.

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[1970-74]: a. CMTT/6 (United Kingdom); b. CMTT/34 (Germany (Federal Republic of)); c. CMTT/197 (Germany (Federal Republic of)); d. CMTT/12 (11/34) (Japan); e. CMTT/212 (Italy).

[1974-78]: a. CMTT/23 (United Kingdom); b. CMTT/60 (EBU); c. CMTT/195 (United Kingdom); d. CMTT/242 (Germany (Federal Republic of)); e. CMTT/241 (Germany (Federal Republic of)); f. CMTT/243 (Germany (Federal Republic of)); g. CMTT/187 (German Democratic Republic); h. CMTT/44 (United States of America).

[1978-82]: a. CMTT/82 (Germany (Federal Republic of)); b. CMTT/79 (Germany (Federal Republic of)); c. CMTT/53 (Japan); d. CMTT/90 (France); e. CMTT/6 (Norway); f. CMTT/52 (Japan).

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REPORT 969

**MEASUREMENT METHODS AND TEST PROCEDURES FOR SPECIAL SIGNALS
TIME-MULTIPLEXED WITHIN THE FIELD BLANKING INTERVAL
OF AN ANALOGUE TELEVISION SIGNAL**

(Question 21/CMTT and Study Programme 21C/CMTT)

(1982)

1. Introduction

In conformity with Question 21/CMTT and Study Programme 21C/CMTT, studies are being carried out, inside the CMTT, to define how to multiplex within the field blanking interval of an analogue television signal some special signals devoted to the transmission of various information, such as: those relevant to the network management, those relevant to the transmission of sound programmes, and those devoted to "other types of information".

Certainly relevant to that last category is the information corresponding to the services that are being studied within Study Group 11, according to Question 29/11 and its associated Study Programmes.

Along these lines, the present Report is intended to collect the material dealing with the behaviour of international television circuits with respect to the transmission of such special signals. This material should belong to one of the three following sections:

- A. Identification of transmission impairments.
- B. Measurement methods and test procedures to determine quality of the special signals.
- C. Use of the measurement methods and test signals already defined for the analogue television signal as given in Recommendation 567, for the evaluation of the transmission quality of the special signals.

At the present time, contributions have only been received with respect to A and B, concerning a particular kind of special signal, which carries data and has the following main modulation characteristic:

- two level luminance modulation;
- compatibility within the maximum video signal bandwidth, obtained through an appropriate filtering of the modulated signal;
- time multiplexing within the active line periods of the field blanking interval.

2. Identification of transmission impairments (point A of the Introduction)

A contribution from the United Kingdom [CCIR, 1978-82a] describes measurements of teletext eye height [Sherry, 1976] which have been made at various points in a national transmission network in order to determine the factors contributing to eye closure and to estimate the resulting teletext service coverage in comparison with the UHF television service.

The eye height at the originating source of the teletext programme signal was a little better than 90% and the measurements included the eye height at the output of the terrestrial transmission network at each of 19 broadcast transmitter locations. It was found that the mean eye height at the output of the transmission network was 81.5% with a standard deviation of 7.1%. After further transmission through television transmitters and transposers, the eye height reduced to about 60%.

There is an indication therefore that the major impairment of the teletext signal occurs in the chain of television broadcast transmitters and transposers and that, a television programme distribution network, generally complying with Recommendation 567, and operated with normal maintenance directed towards the television parameters, will contribute only a small part of the total teletext signal impairment.

However, studies and field trials using satellite circuits carried out in France [CCIR, 1978-82b] indicate that truncation noise near to the FM reception threshold may impair the reception of data in the field blanking intervals of television signals, although the picture quality is still acceptable. In addition, when the frequency demodulator includes a phase-lock loop, special design precautions are necessary to obtain satisfactory data reception.

3. Measurement methods and test procedures to determine the quality of the special signals (point B of the Introduction)

Although data signals may be regarded as a particular case of a video signal, their special structure make them amenable to a more precise analogue characterization and to digital characterization.

3.1 Analogue characterization

The data signal may be split up in time to a series of slots within which the transmitted signal has a clearly defined waveform. By superimposing the waveform corresponding to the successive time slots on any display medium, it is possible to draw a diagram commonly known as an "eye" diagram, which is interpreted for the purposes of evaluating the performance of the data signal. This interpretation may be based on the use of a mask. An example of such a mask is given in [CCIR, 1978-82c].

3.2 Digital characterization

The data signal, unlike the picture signal, carries information which may be readily expressed since it is in digital form. Signal performance is therefore measurable by parameters directly linked to its information content, the transmitted information quantum being the bit.

Digital characterization depends on the definition of two parameters:

- the bit error-rate or ratio of the number of bits received erroneously to the number of bits received;
- the bit loss rate or ratio of the number of bits not received to the number of bits transmitted.

These two parameters are expressed in average values. It would be useful in some cases to be able also to determine the distribution of these errors and losses.

This characterization may easily be obtained through a test signal made up of a pseudo-random bit sequence broadcast in the normal framework of a data channel. This same test signal is ideal for plotting the eye diagram, since it comprises all the possible combinations of n consecutive bits, n being the degree of the polynomial used to generate the pseudo-random sequence. This signal is described with greater precision in [CCIR, 1978-82c].

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[1978-82]: a. CMTT/27 (United Kingdom); b. CMTT/265 (France); c. CMTT/38 (France).

REPORT 970 *

CHARACTERISTICS OF VIDEO SIGNALS**Limitation of out-of-band spectral components**

(Study Programme 21D/CMTT)

(1982)

The spectra of video signals in general extend beyond the nominal video bandwidth. This part of the spectrum may affect other signals transmitted on the same analogue transmission path above or below the television signal. Such signals are, e.g. frequency modulated sub-carriers for sound programme transmission, 2048 Mbit/s digital signals on radio-relay systems and telephony signals or other television signals on cable systems.

For the multiplexing of signals combining filters are used, the design of which depends on the out-of-band spectra of the signals. The design of these filters must take into account interference between the multiplexed signals on the one hand and undue distortion of the television signal on the other hand. As in general the responsibilities for the signal source and the transmission path are with different authorities, optimum design can hardly be achieved. Even a well designed transmission line might give unsatisfactory results, if new test signals, signals incorporated in the television signal to provide new services, or new processing methods of the video signal are introduced after installation of the equipment. It would be of great help to designers if a limitation of the out-of-band spectral components could be specified.

Practical tests with actual multiplex designs have shown that these are capable of handling actual television programme signals [CCIR, 1978-82]. It remains to ensure that signals applied for test purposes will not impose conditions which are unnecessarily onerous, in particular in respect of the characteristics of the conventional loading signal, as defined in Recommendation 570 and further treated in Report 643.

It is therefore proposed in [CCIR, 1978-82], that for any television signal which contains completely or partially electronically synthesized components and for any other inserted signal liable to be applied to the video input terminal of multiplex transmission equipment, the power in any 4 kHz band in the frequency range above 1.2 times of the nominal video bandwidth shall be at least 50 dB below that of a sine-wave with the same peak-to-peak amplitude as the nominal video signal (as defined in Recommendation 567, § B.2.4) at a point whose choice rests with the signal generating authority.

Further study of the implications of this limitation is required.

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REPORT 823

**PROCESSING OF SPECIAL SIGNALS INSERTED IN THE FIELD-BLANKING INTERVAL
OF A TELEVISION SIGNAL AT A POINT WHERE CONVERSION EITHER
OF TELEVISION STANDARD OR OF COLOUR SYSTEM OCCURS
ON AN INTERNATIONAL CIRCUIT**

(Study Programmes 14A/CMTT and 21C/CMTT)

(1978)

1. Introduction

Special signals can be inserted into the field-blanking interval of a television signal. These signals can be used for:

- supervision and measurement of various transmission characteristics,
- transmission of data concerning operation such as information instructions and remote control of equipment,
- transmission of sound and/or other information.

* This Report should be brought to the attention of Study Groups 4, 9 and 11 of the CCIR, Study Group XV of the CCITT and IEC Technical Committee No. 12.

In this regard, Report 314 states that “international signals should be inserted by the appropriate authority at the point of origin of the international circuit and should not be removed or replaced except by the authorities at the terminal point, or points of the international circuit”.

This statement is qualified by Recommendation 473 however, since the latter document recommends that international insertion test signals (ITS), (which may form only a part of the special signals referred to above), “should neither be removed nor replaced on the international circuit, except possibly at a point of conversion of either standard or colour system”.

It is obviously desirable that any converters in an international circuit should be as “transparent” as practicable to special signals transmitting data or other information. It is also desirable that such converters, which may well be located at an intermediate point in the circuit, e.g. at an earth station, should process the supervision and measurement signals inserted at the point of origin in such a way that these signals can be used for their intended purpose at points downstream of the converter. Unfortunately, standards converters employ processes, such as interpolation, which make it very difficult to transmit faithfully a standards converted version of an ITS.

2. Allocation of lines

Report 314 proposes the allocation of lines tabulated below:

Purpose	Standard			
	525 lines		625 lines	
	Line No.	Signal (see Rec. 473-2)	Line No.	Signal (see Rec. 473-2)
International ITS	17 of both fields	Figs. 5 and 6	17-18-330-331 ⁽¹⁾	Figs. 1, 2, 3, 4
International noise	—	— ⁽²⁾	22-335	—
Data signals	—	— ⁽³⁾	16-329	— ⁽³⁾
National ⁽⁴⁾	16 both ⁽⁵⁾ fields; 17-18-19-20-21 both fields	— — ⁽⁶⁾	19-20-21-332-333-334	—

⁽¹⁾ This should not prejudice their use for broadcasting signals such as teletext on a national basis.

⁽²⁾ It has been suggested that 525-line noise measurement might be possible using the bar element B₂ of the ITS on field 1.

⁽³⁾ In the EBU and the USA, experiments are being carried out with digital signals.

⁽⁴⁾ Must be removed if so requested by an authority at the receiving end of the international circuit.

⁽⁵⁾ Japan only.

⁽⁶⁾ Line 19 is reserved for Vertical Interval Reference signals in the USA.

Other special signals could be transmitted in the remaining lines of the field-blanking interval, when available.

3. Signals to be processed by an international converter

Reference to the above Table suggests that the minimum number of signals which it is desirable to process by an international converter, at present is as follows:

3.1 *For a 525-line input:* the ITS on line 17 of fields 1 and 2.

3.2 For a 625-line input: data on lines 16 (329), the ITS on lines 17 (330), the ITS on lines 18 (331), the noise on lines 22 (335).

4. Processing by a colour system converter (transcoder)

Colour system conversion (transcoding) differs from standards conversion in that no change is required in the line or field standards of the television signal being processed.

It seems reasonable to suggest that a colour system converter intended for inclusion in an international television circuit should transmit any international data signals with the minimum of distortion, possibly by arranging for them to by-pass the conversion process completely.

Possible methods of processing the international supervision and measurement signals include:

4.1 Deleting the signals completely at the converter. This is a simple but operationally unsatisfactory method.

4.2 Deleting the signals at the converter input and replacing them by newly generated signals inserted at the converter output. This is an operationally unsatisfactory method because information on any distortions occurring up-stream of the converter, and within the converter itself, is lost.

4.3 Arranging for the signals to by-pass the converter completely. This method gives no indication of any distortions occurring within the converter itself, but may nevertheless be simple to instrument, and an operationally satisfactory compromise.

4.4 Passing the signals through the same conversion process as the active picture lines. This method is unlikely to be completely satisfactory, firstly because any differential gain or phase distortion present on the incoming ITS is likely to be ignored by the converter when associated with an incoming SECAM or PAL * signal, and secondly because the frequency and/or phase of the sub-carrier component of an ITS which has been directly transcoded into PAL or SECAM, does not comply with the specification in the Recommendation 473, namely $4.43361875 \text{ MHz} \pm 10 \text{ Hz}$ and, in the case of PAL, locked at $60 \pm 5^\circ$ from the positive (B - Y) axis.

4.5 Passing the signals through as much as possible of the conversion process undergone by the active picture lines, but arranging, irrespective of the colour systems involved, for the decoded colour components of the converter input ITS to be recoded in accordance with Recommendation 473. This method appears to be the more satisfactory but it does involve some extra instrumentation in the converter, and may not give an entirely accurate indication of its performance. It also has the limitations, mentioned in § 4.4, with respect to differential gain and phase distortions on an ITS associated with an incoming SECAM or PAL signal.

4.6 Combining methods (§ 4.2 and 4.3), such that converter input, and locally generated output supervision and measurement signals, are transmitted downstream of the converter on an alternate basis at, say, half-field rate. This method requires extra instrumentation, (e.g. the measuring equipment needs to be able to distinguish the source of the particular signal being measured), but it could be operationally satisfactory although it gives no indication of converter performance.

4.7 Combining an automatic measuring device at the converter input with method (§ 4.2), so that distortions occurring downstream of the converter are measured on the reinserted supervision and measurement signals; whereas distortions occurring up-stream of the converter are assessed from data resulting from automatic measurement of the signals arriving at the converter input, this data being inserted, in, say, digital form, on another line in the field-blanking interval. This method also requires extra instrumentation and gives no indication of converter performance but could be considered operationally satisfactory.

This subject requires further study.

5. Processing by a standards converter

In general a conversion of colour system is required after a conversion of standard and, as a consequence, it is usual to associate a colour system converter (transcoder) with each standards converter.

The introduction of digital techniques has enabled standards converters to be designed with some capability for processing special signals inserted in the field-blanking interval. The duration of each line available for special signal insertion is very similar for all standards, but reference to Section 2 above, shows that not only is the structure of the ITS different between 525-line and 625-line standards, but that the format (i.e. the allocation of lines) is also different. In particular the present 525-line international allocation does not allow space for the insertion of a second line of ITS, data, or separate noise information which might be converted or bypassed from a 625-line input.

* Differential phase only in the PAL case.

Thus it would appear that the optimum processing arrangement for an ideal standards converter would be, in the case of 525-625 conversion, for the signal incoming on line 17 of field 1 to be transmitted, in some form, on line 17 of the converter output. Similarly for line 17 of field 2 to go out on 330. In the case of 625-525 conversion, lines 18 and 331 could be ignored by the converter, but lines 17 and 330 at its input could go out, in some form, on line 17 of fields 1 and 2 respectively.

Practical standards converters, however, employ processes, such as interpolation, which make it very difficult to convert faithfully an incoming test signal. This practical limitation, and the fact that the format of the converted ITS may not be compatible with the television standard with which it is associated, would seem to restrict the choice of practicable processing arrangements at this time to:

- deleting the signals at the converter (as in § 4.1),
- deleting and replacing by new signals added at the converter output (as in § 4.2),
- combining an automatic measuring device at the converter input with the method above (as in § 4.7).

Unfortunately none of the above arrangements allow information on the performance of the overall circuit (including standards converter) to be derived. This subject requires further study.

Special signals for testing standards conversion equipment have been proposed [EBU, 1975], but these, being of the full-field type, are not suitable for in-service testing.

REFERENCES

- EBU [1975] Operating procedure for use of the EBU test pattern generator with standards converters. *International Television Code of Practice*, Vol. 1, Supp. No. 4.
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QUESTIONS AND STUDY PROGRAMMES, RESOLUTIONS, OPINIONS AND DECISIONS

QUESTION 13/CMTT *

**STANDARDS FOR ANALOGUE SYSTEMS FOR THE TRANSMISSION
OF TELEVISION SIGNALS OVER LONG DISTANCES**

(1978)

The CCIR,

CONSIDERING

that all the information required by the CCIR and the CCITT relating to the requirements for the transmission of monochrome and colour television signals over long distances by analogue transmission systems, is not yet available,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monochrome or colour television signals over hypothetical reference circuits or chains entirely composed of analogue transmission systems:

what are the characteristics of the signal and of the circuit which must be considered, what are their recommended values, and what tolerances must be imposed to ensure satisfactory transmission?

Note. — See Recommendation 567, 568, 603 and Reports 637, 815, 816, 819 and 965.

STUDY PROGRAMME 13A-1/CMTT

**CHARACTERISTICS AND DESIGN OBJECTIVES
FOR INTERNATIONAL TELEVISION CIRCUITS**

(1978-1982)

The CCIR,

CONSIDERING

(a) that agreement has been reached on a large number of aspects of the problems which occur when transmitting television signals of various standards over international circuits, and that these agreements are recorded in Recommendation 567;

(b) that the need is recognized in Recommendation 567 that further study of a number of aspects is required;

(c) that new aspects of television transmission performance not recognized in Recommendation 567 may arise in the future,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the determination of uniform performance objectives for which there is no answer or unified acceptance in Recommendation 567;

2. the study of any aspects not investigated so far; examples of which are the correlation of design objective values for different parameters, the statistical variation of transmission performances with time, etc.

Note 1. — Reports 637, 815 and 816 contain information and references relevant to this Study Programme gathered during the periods 1974-1978 and 1978-1982.

Note 2. — Report 965 contains information relevant to this Study Programme gathered during the period 1978-1982.

Note 3. — See Recommendation 568 and Reports 816, 817 and 965.

* This Question replaces Question 2-1/CMTT (Geneva, 1974) and partly replaces Question 1-1/CMTT (Geneva, 1974).

STUDY PROGRAMME 13B/CMTT *

LONG-TIME WAVEFORM DISTORTION IN TELEVISION
CIRCUITS OVER LONG DISTANCES

(1978)

The CCIR,

CONSIDERING

- (a) that Recommendation 567 relates to the requirements for the transmission of television signals over long distances;
- (b) that long-time waveform distortion can be caused by the transient phenomena induced by a sudden change in the d.c. component of the picture signal, as indicated in § 3.5.1.1, Part B of Recommendation 567;
- (c) that such long-time waveform distortion can give rise to other distortions which can impair the quality of the received picture,

UNANIMOUSLY DECIDES that the following studies should be carried out:

for the transmission of monochrome or colour television signals over a hypothetical reference circuit (2500 km):

1. the maximum permissible magnitude, duration and shape of the long-time waveform distortion which can be set up by a sudden change or by periodic changes in the d.c. component of the picture signal;
2. the methods of calculation which can be used to find the permissible magnitudes for circuits which have more or fewer sections than the hypothetical reference circuit;
3. the other types of distortion which the long-time waveform distortion can produce;
4. the methods and best signals which may be employed to measure the long-time waveform distortion and other types of distortion in accordance with § 3.

Note 1. — Administrations are asked for information about the magnitude of long-time waveform distortion, about the kind and amount of non-linear distortion and about the use of clamping devices with relation to the number of equipment devices (e.g. amplifiers, modulation sections) connected in cascade.

Note 2. — Report 636 contains the information relevant to this Study Programme gathered during the periods 1970-1974, 1974-1978 and 1978-1982.

Note 3. — This Study Programme should be studied in close association with Study Programme 15F/CMTT.

STUDY PROGRAMME 13C-1/CMTT

ALLOCATION OF TOLERANCES FOR COLOUR TELEVISION

(1970-1978-1982)

The CCIR,

CONSIDERING

- (a) that the subjective quality of colour television pictures is affected by the objective performance of every component part of an overall television system from picture source up to and including the receiver;
- (b) that the relationship between objective parameters of television signals and subjective assessments of displayed picture quality is under study in Study Programme 3B/11;

* This Study Programme replaces Study Programme 1D-1/CMTT (Geneva, 1974).

(c) that the composition of overall television systems, as well as the specification of the corresponding overall quality levels, are under study in Study Programme 14A/11,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the allocation of the specified total tolerances among the component parts of an overall television system from the picture source up to and including the receiver, taking into account the statistical behaviour of departures from the nominal performance figures of the equipment. (In particular, for the CMTT, the performance of international television connections is of prime interest.)

Note. — Report 635 contains the information relevant to this Study Programme gathered during the periods 1970-1974 and 1974-1978.

STUDY PROGRAMME 13D-1/CMTT

TELEVISION REFERENCE CHAINS FOR TERRESTRIAL AND FIXED-SATELLITE LINKS

(1970-1974-1978-1982)

The CCIR,

CONSIDERING

- (a) that Recommendation 567 contains the definition and transmission performance requirements for a terrestrial hypothetical reference circuit of 2500 km divided into three sections of equal length by two intermediate video points; and for a fixed-satellite link comprising one Earth-satellite-Earth section;
- (b) that the characteristics of a fixed-satellite link may differ from those of a terrestrial link and are not directly related to the great-circle distance between the earth stations;
- (c) that fixed-satellite links may well be used to interconnect long terrestrial chains and that the resulting connections may include many intermediate video points;
- (d) that Part E of Recommendation 567 gives guidance for estimating the characteristics of a chain when the characteristics of the individual links are known;
- (e) that Recommendation 603 contains the definition for a hypothetical reference chain;
- (f) that Recommendation 603 notes that the performance of the hypothetical reference chain is less than satisfactory for some parameters;
- (g) that Opinion 38 applies to chains spanning countries with different television standards,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. determination of the objective performance of reference chains consistent with the desirable subjective quality of the overall television system, from picture-signal source to receiver, for the various types of links;
2. determination of the transmission performance to be recommended for the various types of hypothetical reference circuit, based on the desirable performance of reference chains;
3. the possible inclusion of standards converters in the reference chains.

Note. — See Recommendation 603.

STUDY PROGRAMME 13E-1/CMTT *

**NOISE MODULATION OF TELEVISION SIGNALS
ARISING FROM CARRIER PHASE JITTER**

Permissible levels

(1978-1982)

The CCIR,

CONSIDERING

- (a) that CCITT Study Group XV have asked for information relating to tolerable limits for unwanted modulation and jitter of signals arising from phase modulation upon the carrier supplies of analogue transmission systems;
- (b) that CMTT is competent to deal with the effects of noise in relation to television signals,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the tolerable levels of this particular form of noise modulation on television signals which arises from unwanted phase modulation of carrier supplies, taking into account the bandwidth and character of the noise.

Note 1. — In order to expedite these studies and to avoid unnecessary work, CCITT Study Group XV is asked to give such details as are known of the likely spectral distribution, etc., of the noise, or jitter, and also to give details of any proposed methods for specifying such noise or jitter upon the carrier supplies.

Note 2. — This Study Programme should be studied in close association with Study Programme 15G/CMTT.

Note 3. — The CCITT has indicated that information of interest to this Study Programme is contained in the CCITT texts, Annex 1 and Annex 2 to Question 11/XV and Recommendation G.229, which consider (provisional) requirements and measurement methods for phase jitter and sources of phase jitter.

STUDY PROGRAMME 13F/CMTT **

**THE PERFORMANCE OBJECTIVES FOR STANDARDS CONVERTERS
USED IN INTERNATIONAL TELEVISION TRANSMISSION**

(1982)

The CCIR,

CONSIDERING

- (a) that standards conversion between 625-line/50 Hz and 525-line/60 Hz standards for colour and mono-chrome signals is necessary for international television transmission between countries employing the different standards;
- (b) that standards conversion should be regarded as being of significance in international television transmission;
- (c) that the standards converter which is employed in an international television transmission may have a significant impact on the overall performance;
- (d) that, since standards conversion is likely to be included in international television transmission, it is necessary to study the relevant performance objectives for such standards conversion in CMTT;
- (e) that other aspects of performance, for example the ability of the standards converter to process moving pictures, are currently being considered by Study Group 11,

* This Study Programme should be brought to the attention of Study Groups 4 and 9 of CCIR and CCITT Study Group XV.

** This Study Programme should be studied in close coordination with Study Programmes 13A/CMTT, 14B/CMTT, 16A/CMTT and Question 15/CMTT. It should also be brought to the attention of Study Group 11.

UNANIMOUSLY DECIDES that the following studies should be carried out:

performance objectives for standards converters used in international television transmission.

Note. — Information relevant to this Study Programme is contained in Report 966.

QUESTION 14-1/CMTT *

STANDARDS FOR DIGITAL SYSTEMS FOR THE TRANSMISSION OF TELEVISION SIGNALS OVER LONG DISTANCES

(1978-1982)

The CCIR,

CONSIDERING

- (a) that the CCITT is studying the standards for future digital transmission systems to be used, *inter alia*, for the transmission of television signals;
- (b) that, in view of the development of methods for the digital processing of television signals, it is likely that these techniques will be widely used both for studio installations and direct broadcasting from terrestrial or satellite transmitters and for long-distance transmission;
- (c) that, to facilitate the international exchange of programmes and to rationalize the design of equipment, it would be desirable to standardize, as far as possible, the methods used for the digital coding of such signals for each application and to ensure the utmost compatibility among the methods used for the various applications,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of television signals over hypothetical reference circuits or chains either composed entirely of digital transmission systems or composed of a mixture of analogue and digital transmission systems:

1. what are the characteristics of the signals, of the coding methods, and of the various types of circuits which must be considered;
2. what are their recommended values, and what tolerances must be imposed to ensure satisfactory performance?

Note. — See Recommendation 604 and Report 817.

STUDY PROGRAMME 14A-1/CMTT **

DIGITAL TRANSMISSION OF TELEVISION SIGNALS

(1974-1978-1982)

The CCIR

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the methods to be recommended for the conversion of analogue television signals into digital coded form with a view to their transmission over long-distance links;
2. the methods to be recommended for the conversion of television signals already in one digital form into another digital form that is required for transmission over a different type of digital bearer, for example:
 - conversion of signals encoded in a digital form suitable for studio applications into another digital form suitable for long-distance transmission;
 - methods to ease the interconnection between signals coded for transmission over terrestrial systems and satellite systems;

* Cooperation with CCITT Study Group XVIII is essential to the study of this Question. Under Decision 18, the study of this Question should be carried out in coordination with studies concerning Question 25/11.

** Results arrived at in response to this Study Programme, should be brought to the attention of Study Groups 4, 9 and 11, and transmitted to CCITT Study Group XVIII.

3. the studies detailed in § 1 and 2 should take account of:
- the concept of an extensible family of compatible digital coding standards as studied under Study Programme 25G/11;
- and include the study of:
- sampling parameters;
 - methods for the reduction of redundancy; *
 - methods of protection against digital transmission errors;
4. the definition of hypothetical reference chains and circuits either composed entirely of digital transmission systems or composed of a mixture of analogue and digital transmission systems;
5. design objectives and tolerances applicable for the transmission of television signals over hypothetical reference chains or circuits having digital sections.
- Note.* — See Recommendation 604 and Reports 646, 819, 823, and 967.

STUDY PROGRAMME 14B/CMTT **

QUALITY PARAMETERS FOR TELEVISION TRANSMISSION SYSTEMS USING DIGITAL OR ANALOGUE-AND-DIGITAL MODULATION

(1978)

The CCIR,

CONSIDERING

- (a) that for the use of the concept of the hypothetical reference circuit and for the checking of real systems, it is essential to define objective quality parameters for transmission systems and associated measurement methods;
- (b) that for analogue transmission systems, Recommendation 567 describes measurement methods used in determining various transmission parameters;
- (c) that in a digital transmission system, impairments may be caused either by the coding equipment or by the transmission circuits,

UNANIMOUSLY DECIDES that the following studies should be carried out:

- 1. the objective parameters to be applied in characterizing the quality of an analogue-to-digital coding system or of a digital-to-digital converter;
 - 2. the parameters to be applied in specifying the quality of an all-digital television transmission system.
- Note.* — See Recommendation 604.

STUDY PROGRAMME 14C/CMTT

LAWS OF ADDITION FOR IMPAIRMENTS ASSOCIATED WITH ALL-DIGITAL AND MIXED ANALOGUE-AND-DIGITAL TRANSMISSION OF TELEVISION SIGNALS

(1978)

The CCIR,

CONSIDERING

- (a) that requirements for the transmission of television programme signals by digital means is under study;

* Studies under this Study Programme should be carried out in close collaboration with Study Group 11.

** This Study Programme should be studied in close coordination with Study Programme 16A/CMTT.

- (b) that hypothetical reference circuits and chains, which include digital as well as analogue sections are under study;
- (c) that circuits and chains of different lengths are of interest;
- (d) that the laws of addition are not known for circuits and chains containing digital sections as well as analogue sections;
- (e) that the concept of hypothetical reference circuits and chains has no meaning unless appropriate laws of addition can be determined,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the laws of addition considered appropriate for the parameters which describe the television circuit performance when all sections of the circuit are digital; *
2. the laws of addition considered appropriate for the parameters which describe the television circuit performance when the circuit contains some analogue and some digital sections.

Note. — See Recommendation 604.

STUDY PROGRAMME 14D/CMTT **

PERFORMANCE OF LONG-DISTANCE DIGITAL CIRCUITS CARRYING BROADCASTING TELEVISION SIGNALS

(1982)

The CCIR,

CONSIDERING

- (a) that performance objectives for an international 64 kbit/s hypothetical reference connection have been drawn up (CCITT Recommendations G.821 (errors) and G.822 (slip));
- (b) that as recognized [CCIR, 1978-82], the method of specifying error-performance objectives in these Recommendations is not satisfactory for the more stringent requirements of broadcasting television signals (see for example Reports 646, 639 and 967);
- (c) that these broadcasting signals have bit rates greater than 64 kbit/s and may not encounter all the parts of the network which was considered when the CCITT Recommendation was drawn up;
- (d) that it is intended that broadcasting signals shall use parts of the international long-distance digital network interchangeably with telephony and other traffic;
- (e) that appropriate standards may therefore be defined for international connections which may be used for digital transmission of broadcasting signals,

UNANIMOUSLY DECIDES that the following studies should be carried out to determine:

1. the impairments to television signal quality caused by errors, slip and jitter that may be introduced by a long-distance digital circuit, operating both normally and under fault conditions;
2. the limits to be placed upon the magnitude and/or rate of occurrence (as appropriate) of these degradations to avoid significant impairments to the television signal quality;

* This includes the case where the digital coding format of the signal may be changed between sections.

** This Study Programme should be studied in close coordination with Study Programmes 14A/CMTT, 14B/CMTT, 14C/CMTT, 16A/CMTT and Question 9 of CCITT Study Group XVIII.

3. the effect of such limits on the planning of the international long-distance digital network which may convey broadcast television signals.

REFERENCES

CCIR Documents

[1978-82]: CMTT/12 (EBU).

STUDY PROGRAMME 14E/CMTT *

**SERVICE REQUIREMENTS FOR LONG-DISTANCE DIGITAL TELEVISION
CONNECTIONS ON THE INTEGRATED SERVICES DIGITAL NETWORK (ISDN)**

(1982)

The CCIR,

CONSIDERING

- (a) that CCITT considers that the ISDN may be used for the transmission of digital information arising from a wide range of services including the broadcast television service;
- (b) that CCITT is thus requesting advice from CMTT on possible service requirements concerning the transmission of signals of the broadcast television service,

UNANIMOUSLY DECIDES that studies should be carried out to determine:

the service requirements to be satisfied for the transmission of signals of the broadcast television service on the ISDN and especially:

- the traffic characteristics,
- the opportunity to envisage switching on a call by call basis,
- the possibility to tolerate call set-up delays.

QUESTION 15-1/CMTT

**METHODS OF MEASUREMENT, TEST SIGNALS AND OPERATIONAL
CHARACTERISTICS OF ANALOGUE TELEVISION TRANSMISSION SYSTEMS**

(1978-1982)

The CCIR,

CONSIDERING

that all the information required by the CCIR and the CCITT relating to the requirements for the transmission of monochrome and colour television signals over long distances by analogue transmission systems is not yet available,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monochrome or colour television signals over an international television connection entirely composed of analogue transmission systems:

what methods of measurement, test signals and operational characteristics can be recommended for checking the transmission performance?

Note. — See Recommendation 569 and Report 818.

* The results of this Study Programme should be brought to the attention of CCITT Study Group XVIII.

STUDY PROGRAMME 15A/CMTT *

**INSERTION OF TEST SIGNALS IN THE FIELD-BLANKING INTERVAL
OF AN ANALOGUE TELEVISION SIGNAL**

(1978)

The CCIR,

CONSIDERING

- (a) that Recommendation 473 recommends the insertion of certain signals in the field-blanking interval of an analogue television signal for the measurement, monitoring and control of television circuits;
- (b) that the nature of such test signals and their mode of operation should continue to meet as closely as possible the measurement, monitoring and control requirements of international television transmissions;
- (c) that it may be necessary also to use other special signals for network supervision and control,

UNANIMOUSLY DECIDES that the following studies should be carried out:

- 1. whether these signals can be inserted and removed without detriment to the quality of the television picture itself;
- 2. the purpose for which such signals should be used internationally;
- 3. the points at which these signals should be inserted in the international television connection and, possibly, removed again;
- 4. the provisions to be made to avoid confusion between signals for national and international use;
- 5. the forms of special signal to be recommended for international use;
- 6. the position of these signals in the field-blanking interval;
- 7. the best system of encoding for network supervision and control signals.

Note 1. — Information relevant to this Study Programme is contained in Reports 314 and 819 and Recommendation 473.

Note 2. — The study of this Study Programme should be co-ordinated with the study of Study Programme 21C/CMTT.

STUDY PROGRAMME 15B-1/CMTT

AUTOMATIC SWITCHING SYSTEMS FOR TELEVISION CIRCUITS

(1974-1978-1982)

The CCIR,

CONSIDERING

- (a) that the performance definitions and design objectives for television hypothetical reference circuits are given in Recommendation 567;
- (b) that availability of a television circuit is considered in Report 639;
- (c) that television circuit performance in normal operation is being studied by CCITT Study Group IV under Question 20/IV;
- (d) that the performance and availability of a television circuit may be improved by the use of automatic switching systems;
- (e) that the operating of automatic switching systems may introduce impairments to the television signal;
- (f) that the specific characteristics of the switching systems for cable, radio relay and satellite circuits are being studied by the CCIR and CCITT Study Groups concerned,

* This Study Programme partially replaces Study Programme 1C-1/CMTT (Geneva, 1974).

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the criteria which should be used for initiating the automatic switching of television circuits;
2. the restrictions to be placed on the impairments introduced by the switching from a normal channel to a stand-by channel.

Note 1. — The term “automatic switching” is defined as designating an operation carried out automatically and consisting of, for example:

- passing from a normal channel to a stand-by channel;
- changing the route used to link one point with another.

Note 2. — Information relevant to this Study Programme is contained in Report 818.

STUDY PROGRAMME 15C-1/CMTT

AVAILABILITY OF AN INTERNATIONAL TELEVISION CIRCUIT OR CHAIN

(1974-1978-1982)

The CCIR,

CONSIDERING

- (a) that the overall availability and reliability objectives for the telecommunication network are under study (Question 2/CMBD of Joint Study Group CMBD);
- (b) that the availability objectives of transmission systems are under study in several Study Groups (e.g. Question 24/4 and Question 5/9 of the CCIR, and Questions 32/XV and 10/XVIII of the CCITT);
- (c) that the preferred availability objectives for transmission of a television signal (including associated sound programme channels) may be different from those for transmission of other signals, such as telephony,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. type and magnitude of faults which are considered to be an interruption of television signals;
2. analysis of availability data gathered from actual television transmission circuits and chains;
3. the preferred availability objective in an international television circuit or chain.

Note 1. — Information relevant to this Study Programme is contained in Report 639.

Note 2. — These studies should be coordinated with work done under Study Programme 16C/CMTT to ensure compatibility of objectives.

STUDY PROGRAMME 15D/CMTT *

AUTOMATIC MEASUREMENT AND MONITORING ON TELEVISION CHAINS

(1978)

The CCIR,

CONSIDERING

- (a) that the automation of national and international television networks is one form of their evolution;
- (b) that it would be desirable to unify automatic measuring and monitoring methods, which are used in international transmissions and in other sections of global television systems;

* This Study Programme replaces Question 7/CMTT (Geneva, 1974) and Study Programme 7A-1/CMTT (Geneva, 1974).

(c) that such unification can facilitate co-ordination between the operating services of the different telecommunications organizations concerned, and improve the overall performance of the systems in service,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the use that can be made of test signal elements * for automatic measuring, monitoring and operation of networks;
2. the methods of analysis and processing that are appropriate for automatic monitoring and operation of networks;
3. the form in which the results should be expressed in order to facilitate the transmission or interchange of measurement and monitoring information;
4. the methods which permit the comparison of the results of measurement data with the appropriate specifications;
5. the extent to which it is possible to unify the measuring and monitoring methods used in television networks with those used in other sections of a global television system (in this respect see Question 15/11);
6. the way in which the monitoring and measuring data can be used for the automatic control and optimization of television chain characteristics;
7. the additional characteristics which should be specified for test and other special signals with a view to solving these problems.

Note. — Information relevant to this Study Programme is contained in Report 411 and Recommendation 569.

STUDY PROGRAMME 15E-1/CMTT

STANDARD TEST SIGNAL FOR CONVENTIONAL LOADING OF A TELEVISION CHANNEL

(1978-1982)

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 some test signals which may be used for the television circuit can cause substantial disturbance in a sound programme channel carried on a common path and that this might in some circumstances occur at times when the programme channel is in service;
- (e) that Recommendation 570 proposes a composite colour-bar signal;
- (f) that such a colour-bar signal might not be available in case of monochrome transmission;
- (g) that CCITT Study Group XV, in connection with the study of the television transmission on 60 MHz systems has suggested a test signal on the basis of available information of actual television signal statistics,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the characteristics of a television test signal for conventional loading of channels designed for analogue monochrome transmission and digital colour and monochrome transmission when measuring, calculating, or specifying the noise in other circuits carried in a common path with the television circuit;

* The use of such test signal elements is referred to in Recommendation 473.

2. the statistical distribution of typical television signal loadings;
 3. the importance of the rise time of the luminance component of the recommended colour-bar signal.
- Note.* — Information relevant to this Study Programme is contained in Report 643.

STUDY PROGRAMME 15F/CMTT *

**LONG-TIME WAVEFORM DISTORTION IN TELEVISION
CIRCUITS OVER LONG DISTANCES**

Methods of measurement and test signals

(1978)

The CCIR,

CONSIDERING

- (a) that Recommendation 567 relates to the requirements for the transmission of television signals over long distances;
- (b) that long-time waveform distortion can be caused by the transient phenomena induced by a sudden change in the d.c. component of the picture signal, as indicated in § 3.5.1.1, Part B of Recommendation 567;
- (c) that such long-time waveform distortion can give rise to other distortions which can impair the quality of the received picture,

UNANIMOUSLY DECIDES that the following studies should be carried out to determine:

for the transmission of monochrome or colour television signals over a hypothetical reference circuit:

the methods and test signals which may be employed to measure the long-time waveform distortion and the other distortions which may be caused by long-time waveform distortion.

Note 1. — Administrations are asked for information about the magnitude of long-time waveform distortions, about the kind and amount of non-linear distortion and about the use of clamping devices with relation to the number of equipment devices (e.g. amplifiers, modulation sections) connected in cascade.

Note 2. — Report 636 contains the information relevant to this Study Programme gathered during the periods 1970-1974, 1974-1978 and 1978-1982.

Note 3. — This Study Programme should be studied in close association with Study Programme 13B/CMTT.

STUDY PROGRAMME 15G-1/CMTT **

**NOISE MODULATION OF TELEVISION SIGNALS
ARISING FROM CARRIER PHASE JITTER**

Methods of measurement

(1978-1982)

The CCIR,

CONSIDERING

- (a) that CCITT Study Group XV asked for information relating to tolerable limits for unwanted modulation and jitter of signals arising from phase modulation upon the carrier supplies of analogue transmission systems;
- (b) that the CMTT is competent to deal with the effects of noise in relation to television signals,

* This Study Programme replaces Study Programme 1D-1/CMTT (Geneva, 1974).

** This Study Programme should be brought to the attention of Study Groups 4 and 9 of CCIR and CCITT Study Group XV.

UNANIMOUSLY DECIDES that the following studies should be carried out:

the methods and test signals which may be employed to measure the noise resulting from unwanted phase modulation on carrier supplies.

Note 1. — In order to expedite these studies and to avoid unnecessary work, CCITT Study Group XV is asked to give such details as are known of proposed methods of measurement of the noise or jitter resulting from unwanted phase modulation of carrier supplies.

Note 2. — This Study Programme should be studied in close association with Study Programme 13E/CMTT.

Note 3. — The CCITT has indicated that information of interest to this Study Programme is contained in the CCITT texts, Annex 1 and Annex 2 to Question 11/XV and Recommendation G.229 which considers (provisional) requirements and measurement methods for phase jitter and the sources of phase jitter.

QUESTION 16-1/CMTT

METHODS OF MEASUREMENT, TEST SIGNALS AND OPERATIONAL CHARACTERISTICS OF DIGITAL TELEVISION TRANSMISSION SYSTEMS

(1978-1982)

The CCIR,

CONSIDERING

that all the information required by the CCIR and the CCITT, relating to the requirements for the transmission of monochrome and colour television signals over long distances by digital transmission systems is not yet available,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monochrome or colour television signals over an international television connection either composed entirely of digital transmission systems or composed of a mixture of analogue and digital transmission systems:

what methods of measurement, test signals and operational characteristics can be recommended for checking the transmission performance?

STUDY PROGRAMME 16A/CMTT *

MEASUREMENT METHODS FOR TELEVISION TRANSMISSION SYSTEMS USING DIGITAL OR ANALOGUE AND DIGITAL MODULATION

(1978)

The CCIR,

CONSIDERING

(a) that for the use of the concept of the hypothetical reference circuit and for the checking of real systems, it is essential to define objective quality parameters for transmission systems and associated measurement methods;

(b) that for analogue transmission systems, Recommendation 567 describes measurement methods used in determining various transmission parameters;

(c) that in digital transmission systems, impairments may be caused either by the coding equipments or by the transmission circuits,

* This Study Programme should be studied in close co-ordination with Study Programme 14B/CMTT.

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. measurement methods to be used in checking the parameters defined in § 1 and 2 of Study Programme 14B/CMTT for digital television transmission systems;
2. measurement methods to be used in checking the parameters for mixed analogue and digital television transmission systems.

Note. — See Report 819.

STUDY PROGRAMME 16B/CMTT

APPLICATION OF INSERTION TEST SIGNALS TO DIGITAL AND MIXED ANALOGUE AND DIGITAL TELEVISION CIRCUITS

(1978)

The CCIR,

CONSIDERING

- (a) that Study Programme 15A/CMTT asks for the study of special test signals which can be used for monitoring and checking the performance of the circuits during the transmission of programmes;
- (b) that Recommendation 473 specifies, for analogue systems, special forms of insertion test signals for the determination of various transmission parameters;
- (c) that it is desirable to have available similar facilities of continuous supervision of digital and mixed analogue/digital systems;
- (d) that broadcasting organizations and Administrations have already introduced, and are using insertion test signals,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. feasibility of the application of the existing insertion test signals in transmission chains comprising digital and analogue sections;
2. reduction in accuracy of the evaluation of the insertion test signals at the receiving end by distortions originating in digital sections;
3. methods to be used to transmit through the digital sections the information corresponding to the insertion test signals;
4. the insertion test signals needed to evaluate the performance of digital sections.

Note. — See Report 819.

STUDY PROGRAMME 16C/CMTT

AVAILABILITY OBJECTIVES APPLICABLE TO TELEVISION TRANSMISSION SYSTEMS USING DIGITAL MODULATION

(1982)

The CCIR,

CONSIDERING

- (a) that the faults which are considered as interruptions of television signals and the availability objectives to be recommended for an international television circuit or chain are under study (Study Programme 15C/CMTT);
- (b) that availability objectives for digital transmission systems are under consideration by several Study Groups (e.g., CCIR Study Groups 4 and 9, CCITT Study Group XVIII, CMBD etc.);

- (c) that the performance of digital television systems in the event of transmission incidents is likely to differ from that of analogue systems;
- (d) that the availability objectives recommended may have considerable implications for the selection of methods for the digital encoding and multiplexing of signals;
- (e) that it would be undesirable to have incompatible availability objectives between digital and analogue television transmission systems,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. characteristics of faults liable to cause an interruption to a digital television transmission;
2. supplements that may be required, with reference to digital modulation systems, to the definitions and objectives established under Study Programme 15C/CMTT;
3. implications of the methods selected for the digital encoding and multiplexing of television signals for the availability of a television transmission system using digital modulation.

Note. — These studies should be co-ordinated with work done under Study Programme 15C/CMTT, to ensure compatibility of objectives.

QUESTION 17/CMTT *

STANDARDS FOR ANALOGUE SYSTEMS FOR THE TRANSMISSION OF SOUND-PROGRAMME SIGNALS OVER LONG DISTANCES

(1970–1978)

The CCIR,

CONSIDERING

that all the information required by the CCIR and the CCITT relating to the requirements for the transmission of monophonic and stereophonic sound-programme signals over long distances by analogue transmission systems is not yet available,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monophonic and stereophonic sound-programme signals over hypothetical reference circuits or chains entirely composed of analogue transmission systems,

what are the characteristics of the signal and of the circuit which must be considered, what are their recommended values, and what tolerances must be imposed to ensure satisfactory transmission?

Note 1. — At present the performance characteristics of certain types of sound-programme circuits are laid down in Recommendations 503, 504 and 505. For more information see Report 496.

Note 2. — See also Recommendation 502 and Reports 491, 820.

STUDY PROGRAMME 17C/CMTT **

TRANSMISSION OF SOUND-PROGRAMME SIGNALS OVER COMMUNICATION-SATELLITE LINKS

(1970)

The CCIR,

CONSIDERING

that the use of communication-satellite links for the transmission of sound-programme signals, and related auxiliary signals (whether accompanying a television signal or not) has become common practice,

* This Question replaces Question 5-2/CMTT (Geneva, 1974).

** Formerly Study Programme 5G/CMTT (Geneva, 1974).

UNANIMOUSLY DECIDES that studies should be carried out to determine:

1. the general characteristics of such sound-programme circuits and, particularly, how these may differ from the characteristics specified for sound-programme circuits provided by other means;
2. the general characteristics of circuits provided for the transmission of related auxiliary signals; such as commentary, cue, control and engineering service circuits.

Note. — See Report 498-2 (Kyoto, 1978) and Report 642.

STUDY PROGRAMME 17D/CMTT *

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT

(1974)

The CCIR,

CONSIDERING

- (a) that Recommendation 502 gives performance objectives for a hypothetical reference circuit consisting of three sections and having a length of 2500 km;
- (b) that Recommendation 567, Part E shows laws of addition for nearly all the parameters to be measured on a television circuit;
- (c) that CCITT Study Group IV has asked the CMTT whether it will be possible to give laws of addition also to a certain number of sound-programme circuit sections in order to have some guidance for the relevant parameters when setting up and maintaining complex sound-programme connections;
- (d) that the laws of addition will also be of interest for Study Group 9 for incorporation into Report 375,

UNANIMOUSLY DECIDES that the following study should be carried out:

1. the laws of addition considered appropriate for the parameters defined for sound-programme circuit performance;
2. in particular, the utility of Part E of Recommendation 567 as a guidance for some parameters, e.g. insertion gain, random noise, crosstalk, linear distortion, non-linear distortion.

Note. — See Recommendation 605 and Report 821.

STUDY PROGRAMME 17F/CMTT **

NOISE IN SOUND-PROGRAMME CIRCUITS FROM THE POWER SUPPLY

(1974)

The CCIR,

CONSIDERING

- (a) that Recommendation 474 gives limits for unwanted modulation of sound-programme signals by low order harmonics of the public power supply frequency;
- (b) that some units used to supply power to equipment carrying sound-programme signals may produce unwanted components at higher frequencies than normal power supply sources;

* Formerly Study Programme 5K/CMTT (Geneva, 1974).

** Formerly Study Programme 5L/CMTT (Geneva, 1974).

(c) that these unwanted components may cause the sound-programme signals to be modulated by them,

UNANIMOUSLY DECIDES that the following study should be carried out:

the highest permissible level for the most intense unwanted side component, due to modulation of a sound-programme signal caused by interfering signals from power supply sources having frequencies higher than the usual low-frequency mains supplies.

QUESTION 18-1/CMTT *

**STANDARDS FOR DIGITAL SYSTEMS FOR THE TRANSMISSION
OF SOUND-PROGRAMME SIGNALS
OVER LONG DISTANCES**

(1978-1982)

The CCIR,

CONSIDERING

(a) that the CCITT is studying the standards for future digital transmission systems to be used, *inter alia*, for the transmission of sound-programme signals;

(b) that, in view of the development of methods for the digital processing of sound signals, it is likely that these techniques will be widely used both for studio installations and direct broadcasting from terrestrial or satellite transmitters and for long-distance transmission;

(c) that, to facilitate the international exchange of programmes and to rationalize the design of equipment, it would be desirable to standardize, as far as possible, the methods used for the digital coding of such signals for each application and to ensure the utmost compatibility among the methods used for the various applications;

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monophonic and stereophonic sound-programme signals over hypothetical reference circuits or chains, either composed entirely of digital transmission systems or composed of a mixture of analogue and digital transmission systems:

what are the characteristics of the signals, the coding methods and the circuits which must be considered, what are their recommended values, and what tolerances must be imposed to ensure satisfactory transmission?

Note. — See Reports 647, 817 and 819.

STUDY PROGRAMME 18A-1/CMTT **

**STANDARDS FOR THE DIGITAL TRANSMISSION
OF SOUND-PROGRAMME SIGNALS**

(1978-1982)

The CCIR

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the methods to be recommended for the conversion of analogue sound-programme signals into digital coded form with a view to their transmission over long-distance links;

* Cooperation with CCITT Study Group XVIII is essential to the study of this Question. Under Decision 18, the study of this Question should be carried out in coordination with studies concerning Question 51/10.

** Results arrived at in response to this Study Programme should be brought to the attention of Study Groups 4, 9, 10 and 11, and transmitted to CCITT Study Group XVIII.

2. the methods to be recommended for the conversion of sound-programme signals already in digital form, into a form suitable for transmission over long-distance links (e.g. conversion of signals encoded in a digital form suitable for studio applications * into another digital form suitable for long-distance transmission);
 3. the methods to be recommended for the conversion of sound-programme signals already in digital form and encoded for transmission, using one particular encoding law into digital signals conforming to another encoding law. Studies on this point should take into consideration:
 - analogue and digital interconnection;
 - effects on transmission performance;
 - composition of international sound-programme connections determining the necessary number of conversion processes in tandem;
 - the use of pre-/de-emphasis;
 - maintenance aspects;
 - economical considerations;
 - drop and insert facilities on the international sound-programme connection;
 - possibility of bi-directional sound-programme transmission;
 4. the studies detailed in § 1 to § 3 should take into account:
 - the work being carried out in CCIR Study Groups 10 and 11 and CCITT Study Groups IV and XVIII;
 - the corresponding requirements defined by CCIR Study Group 10 for the processing and the quality of such signals;
 - particular constraints in relation to digital transmission impairments (e.g. errors, jitter, slips);
 5. the definition of hypothetical reference chains and circuits either entirely composed of digital transmission systems or composed of a mixture of analogue and digital transmission systems;
 6. design objectives and tolerances applicable for the transmission of sound-programme signals over hypothetical reference chains or circuits having digital sections.
- Note.* — See Recommendation 606 and Report 648.

STUDY PROGRAMME 18B/CMTT

STANDARDS FOR THE USE OF A SIGNALLING CHANNEL TIME-MULTIPLEXED WITH DIGITAL SOUND SIGNALS

(1978)

The CCIR,

CONSIDERING

- (a) that in principle it is possible to insert a signalling channel, time-multiplexed with digital sound signals, in digital transmission paths;
- (b) that such a signalling channel might be used for various purposes, for example for:
 - signals associated with the programme (source identification, the timing and the destinations of the programme);
 - supervision and control signals (type of coding, location of transcoder, automatic switching);
 - technical information on the signal (sound level, stereo phase, noise level and signal spectrum at the source),

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. whether these signals can be time-multiplexed with, and demultiplexed from, the digital sound bit stream, without increasing digital transmission impairments (e.g. error-rate, slips);

* This aspect is being studied by Study Group 10.

2. the format of signals to be recommended for international use and the minimum repetition rate required for certain types of information;
3. transmission capacity, type of insertion (synchronous or asynchronous) and multiplexing frame characteristics of the signalling channel.

STUDY PROGRAMME 18C/CMTT *

**TRANSMISSION OF TIME-DIVISION-MULTIPLEXED SOUND-PROGRAMME
SIGNALS ON A DIGITAL TRANSMISSION PATH**

(1982)

The CCIR,

CONSIDERING

- (a) that the CCITT is studying the characteristics to be recommended for digital transmission equipment capable of being used for signals in the sound-broadcasting service;
- (b) that the CMTT is studying the standards to be recommended for digital systems transmitting sound programme signals;
- (c) that, when recommending these standards, the CMTT should take account of multiplexing methods;
- (d) that the CMTT should also take account of certain operational considerations,

UNANIMOUSLY DECIDES that the following studies should be carried out:

determine the information relating to the transmission of time-division-multiplexed sound-programme signals on a digital transmission path which should be passed to the CCITT to be taken into account by that body when preparing Recommendations concerning equipment characteristics.

Note. — In order to carry out these studies, the CMTT must receive the relevant information from the CCITT on the characteristics of digital paths available for sound-programme transmission.

STUDY PROGRAMME 18D/CMTT *

**PERFORMANCE OBJECTIVES FOR INTERNATIONAL
DIGITAL TRANSMISSION LINKS CARRYING SOUND-PROGRAMME SIGNALS**

(1982)

The CCIR,

CONSIDERING

- (a) that the quality of sound-programme signals carried by international digital links will be affected by:
 - the digital sound coding method;
 - the error control strategy;
 - the performance of the international digital network (bit error rate, error distribution, slip rate and jitter amplitude);
- (b) that the influence of digital circuit transmission performance on sound-programme signal impairment has yet to be determined,

UNANIMOUSLY DECIDES that the following studies should be carried out to determine:

1. the impairments to sound-programme signal quality caused by errors, slip and jitter that may be introduced by an international long-distance digital circuit, operating both normally and under fault conditions;

* This Study Programme should be brought to the attention of CCITT Study Groups XV and XVIII.

2. the limits to be placed upon the magnitude and/or rate of occurrence (as appropriate) of these degradations to avoid significant impairments to the sound-programme signal quality;
3. the effect of such limits on the planning of the international long-distance digital network which may convey sound-programme signals.

Note 1. — In order to carry out these studies, the CMTT must receive the relevant information from the CCITT on the characteristics of integrated digital networks and of integrated services digital networks.

Note 2. — See Report 648.

STUDY PROGRAMME 18E/CMTT *

SPECIFIC REQUIREMENTS FOR SOUND-PROGRAMME SERVICES WHEN TRANSMITTED OVER THE INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

(1982)

The CCIR,

CONSIDERING

- (a) that the CCITT is studying the network characteristics to be recommended for the transmission of wideband services over the ISDN;
- (b) that the CMTT is studying the standards to be recommended for digital systems transmitting sound-programme signals over the ISDN;
- (c) that, when recommending these standards, the CMTT should take account of the characteristics of and the facilities offered by, the evolving ISDN;
- (d) that the CMTT should also take account of certain operational considerations,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. determine the information relating to the transmission of time-division-multiplexed sound-programme signals on a digital transmission path, which should be passed to the CCITT to be taken into account by that body when preparing Recommendations concerning network characteristics;
2. determination of the service requirements of sound-programme circuits routed over the ISDN:
 - for example, if the connections are required to be established by switching on a demand basis, to determine the bit rates at which switching is required,
 - alternatively, to determine if a delay can be tolerated which would allow connections to be established on a semi-permanent basis; and if so, to determine the extent of the delay which can be tolerated in establishing a call.

Note. — In order to carry out these studies, the CMTT must receive the relevant information from the CCITT on the characteristics of ISDN.

QUESTION 19-1/CMTT

METHODS OF MEASUREMENT, TEST SIGNALS AND OPERATIONAL CHARACTERISTICS OF ANALOGUE SOUND-PROGRAMME TRANSMISSION SYSTEMS

(1978-1982)

The CCIR,

CONSIDERING

that all the information required by the CCIR and the CCITT relating to the requirements for the transmission of monophonic and stereophonic sound-programme signals over long distances by analogue transmission systems is not yet available,

* This Study Programme should be brought to the attention of CCITT Study Group XVIII.

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monophonic and stereophonic sound-programme signals over an international sound-programme connection entirely composed of analogue transmission systems:

what methods of measurement, test signals, and operational characteristics can be recommended for checking the transmission performance?

Note. — See Report 822.

STUDY PROGRAMME 19A/CMTT *

**METHOD FOR THE SUBJECTIVE ASSESSMENT OF
IMPAIRMENT DUE TO THE CHARACTERISTICS OF SOUND-PROGRAMME
TRANSMISSION SYSTEMS**

(1974)

The CCIR,

CONSIDERING

(a) that with the appearance of certain new phenomena due to the evolution of transmission techniques there are insufficient known references to be able to assess directly the actual quality of transmission from the measurements obtained;

(b) that in some cases it is therefore essential to carry out listening tests — which obviously have to be statistically based — to supplement the objective measurements obtained with the usual measuring devices, so that valid conclusions can be drawn from the combined data;

(c) that the conclusions which may be drawn from listening tests depend largely on the actual conditions in which the tests are carried out,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the methods to be recommended for the conduct of listening tests to arrive at a subjective assessment of the quality or impairment of a transmission;
2. the conditions to be met and the mode of presentation to be recommended to ensure comparability of the results of tests carried out by different organizations;
3. the use of the methods referred to in § 1 and 2 during international transmissions (Note 1).

Note 1. — Study under § 3 is to be carried out in cooperation with Study Group 10. (Study Programme 50C/10 refers.)

Note 2. — See Report 496, § 2 and Report 623.

STUDY PROGRAMME 19B/CMTT **

**TYPE OF FAULTS WHICH MUST BE CONSIDERED IN AN INTERNATIONAL
SOUND-PROGRAMME CIRCUIT OR CHAIN**

(1974)

The CCIR,

CONSIDERING

(a) that the definition of the overall reliability and quality of a sound-programme circuit or chain is under study (Question 2/CMBD of Joint Study Group CMBD);

* Formerly Study Programme 5H/CMTT (Geneva, 1974).

** Formerly Study Programme 5J/CMTT (Geneva, 1974).

- (b) that Report 642 specifies what constitutes an interruption in sound-programme services,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the type and magnitude of faults that will cause an interruption of sound-programme signals.

STUDY PROGRAMME 19C/CMTT

NOISE MODULATION OF SOUND-PROGRAMME SIGNALS ARISING FROM CARRIER JITTER

(1978)

The CCIR,

CONSIDERING

- (a) that CCITT Study Group XV has asked for information relating to tolerable limits for unwanted modulation and jitter of signals arising from noise modulation upon the carrier supplies of analogue transmission systems;

- (b) that CMTT is competent to deal with the effects of noise in relation to sound-programme,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the tolerable levels of noise modulation of sound-programme signals arising from noise on carrier supplies taking into account bandwidth and character of the noise.

Note 1. — To expedite these studies and to avoid unnecessary work, CCITT Study Group XV is asked to give such details as are known of the likely spectral distribution etc. of the noise, or jitter, and also to give details of any proposed methods for specifying or measuring such noise or jitter upon the carrier supplies.

Note 2. — This Study Programme should be brought to the attention of Study Groups 4 and 9 of the CCIR and CCITT Study Group XV.

STUDY PROGRAMME 19D/CMTT

EXAMINATION OF METHODS USED FOR SETTING-UP AND MONITORING OF SIGNAL LEVELS ON SOUND-PROGRAMME CIRCUITS, AND DERIVATION OF CHARACTERISTICS FOR A CONVENTIONAL LOADING SIGNAL REPRESENTING SOUND-PROGRAMME SIGNALS

(1978)

The CCIR,

CONSIDERING

- (a) that it is desirable to have universally understood methods for the setting up and monitoring of signal levels on sound-programme circuits, and compatible information for the design of systems suitable for sound-programme transmission;

- (b) that the characteristics of the instrument used for the purposes of level control of sound-programme signals can influence the signal characteristics which are encountered upon a circuit;

- (c) that a variety of instruments are listed in Report 292 and in CCITT Recommendation J.15 as being used for monitoring sound-programme transmissions;

- (d) that Reports 491 and 497 already give some information regarding the measured characteristics of sound-programme signals,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the significant differences between different methods of setting up and monitoring programme signal levels on international circuits, and the effects upon the signals carried;

2. the minimizing of level differences arising from monitoring methods;
3. the characteristics which should be assumed for sound-programme signals for the purposes of system design;
4. the ways in which adequate amplitude range and margins against overload should be determined (Note 1).

Note 1. — CCITT Study Group XV is studying the “usable power reserve” in relation to carrier sound-programme circuits under Question 1/XV, but is awaiting further information from CMTT on the characteristics of sound-programme signals.

Note 2. — See Recommendation 571 and Report 497-3.

QUESTION 20-1/CMTT *

METHODS OF MEASUREMENT, TEST SIGNALS AND OPERATIONAL CHARACTERISTICS OF DIGITAL SOUND-PROGRAMME TRANSMISSION SYSTEMS

(1978-1982)

The CCIR,

CONSIDERING

that all the information required by the CCIR and the CCITT relating to the requirements for the transmission of monophonic and stereophonic sound-programme signals over long distances by digital transmission systems is not yet available,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of monophonic and stereophonic sound-programme signals over an international sound-programme connection, either entirely composed of digital transmission systems or composed of a mixture of analogue and digital transmission systems:

what methods of measurement, test signals and operational characteristics can be recommended for checking the transmission performance?

Note. — The attention of the CMTT is drawn to CCITT Recommendation G.704, “Maintenance of digital networks”.

STUDY PROGRAMME 20A/CMTT **

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

Methods of monitoring and measurement

(1978)

The CCIR

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the simplest and most effective technique for monitoring digitally coded, sound-programme signals for long-distance transmission;
2. the methods of measurement and test signals to be recommended for checking performance over long-distance transmission links when digital modulation is used.

Note. — See Report 968.

* This Question should be brought to the attention of CCITT Study Group XVIII.

** This Study Programme partly replaces Question 10-1/CMTT (Geneva, 1974) and Study Programme 10A-1/CMTT (Geneva, 1974).

STUDY PROGRAMME 20B-1/CMTT

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

Operational characteristics

(1978-1982)

The CCIR

UNANIMOUSLY DECIDES that the following studies should be carried out:

the operational characteristics of digital sound-programme circuits that should be maintained to ensure satisfactory performance.

 QUESTION 21-1/CMTT *

STANDARDS FOR TRANSMISSION OF SIGNALS WITH MULTIPLEXING
OF VIDEO, SOUND AND OTHER TYPES OF SIGNAL USED
FOR THE TELEVISION SERVICE

(1978-1982)

The CCIR,

CONSIDERING

- (a) that various operational constraints, particularly those stemming from television service requirements, make it necessary to associate the video signal with other signals such as sound and data signals;
- (b) that some of these requirements can be met through systems of multiplexing in the analogue television signal video baseband;
- (c) that other multiplexing systems have been developed for this purpose and are designed to be independent of the transmission medium;
- (d) that certain systems, on the other hand, are closely dependent on the transmission medium and in that case the specification should be studied by the relevant Study Group; **
- (e) that certain systems may associate the various constituent signals in analogue, digital, or combined analogue/digital form;
- (f) that standards must be defined for these new systems on a basis of cooperation between CCIR and CCITT Study Groups, taking into account their respective terms of reference,

UNANIMOUSLY DECIDES that the following question should be studied:

what standards would be applicable to systems of multiplexing the video signal with associated signals such as sound and data signals:

1. if the television signal is analogue in form:
 - when the multiplexed signal can be transmitted within the video bandwidth of the television standard concerned;
 - when the multiplexed signal is such, that an extended video bandwidth is required provided the system is independent of a particular transmission medium;
2. if the television signal and the associated signals are digital in form?

Note. — See Report 969.

* This Question should be brought to the attention of Study Group 11.

** An example of such systems is provided by the simultaneous transmission of sound and picture signals over radio-relay systems (Study Programme 3A/9).

STUDY PROGRAMME 21A/CMTT *

TRANSMISSION TIME DIFFERENCES BETWEEN THE SOUND
AND VISION COMPONENTS OF A TELEVISION SIGNAL

(1978)

The CCIR,

CONSIDERING

- (a) that the time of transmission of television signals over a terrestrial or a communication-satellite system is generally not negligible;
- (b) that, if the sound and picture signals are transmitted by different methods or routes, there may be an error in timing between the sound and picture signals at the receiving end;
- (c) that Study Group 11 are studying the acceptable overall tolerances of synchronization of the sound signal with respect to the picture signal,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the methods which may be used to ensure that the differences in transmission time are minimized, and the minimum value that can be generally realized (see Note 1);

2. types of testing methods which should be recommended for the measurement of such time differences.

Note 1. — The allocation of the tolerances between the transmission system and other parts of the overall television system are being studied under Question 35/11.

Note 2. — See Reports 412 and 488.

STUDY PROGRAMME 21B/CMTT **

TRANSMISSION OF SOUND AND PICTURE SIGNALS
BY TIME-DIVISION MULTIPLEX OR FREQUENCY-DIVISION MULTIPLEX

(1978)

The CCIR,

CONSIDERING

- (a) that if the sound and picture components of a television signal are transmitted over different routes or by different methods there may be an error of timing between the signals at the receiving end;
- (b) that one method of minimizing this problem would be to transmit both signals over the same circuit by multiplexing;
- (c) that the study of systems of frequency multiplexing is associated with Question 3/9 and Study Programme 47A/10 and certain information related to these systems is contained in Recommendation 402 and Report 289;
- (d) that multiplexing systems can be designed which are independent of the transmission medium,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. techniques which can be used to enable a sound signal to be transmitted in time, or frequency-division multiplex with the picture signal;
2. standards of performance that can be obtained by the use of these techniques;
3. special problems or requirements that might be introduced if these techniques were to be adopted for the international transmission of sound and picture signals.

Note. — See Report 488 and Recommendation 572.

* This Study Programme replaces Question 4-1/CMTT (Geneva, 1974).

** This Study Programme replaces Study Programme 4B/CMTT (Geneva, 1974).

STUDY PROGRAMME 21C-1/CMTT *

**TIME-DIVISION MULTIPLEXING OF SPECIAL SIGNALS
IN THE FIELD-BLANKING INTERVALS OF AN ANALOGUE TELEVISION SIGNAL**

(1978-1982)

The CCIR,

CONSIDERING

- (a) that it is current practice in a number of countries to transmit special signals by time-division multiplex in the field-blanking interval of an analogue television signal;
- (b) that such signals can be used for various purposes, such as:
- the operational management of national and international networks,
 - transmission of information associated with the programme,
 - transmission of sound,
 - transmission of other information,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. whether special signals can be inserted in, and removed from, the field-blanking interval of the television signal, without detriment to the quality of the television picture itself;
2. the purpose for which such signals should be used internationally;
3. the points at which these signals should be inserted in the international television connection and, possibly, removed again;
4. the provisions to be made to avoid confusion between signals for national and international use;
5. the position of these signals in the field-blanking interval;
6. the best system of encoding the signals under consideration;
7. methods of measurement and test procedures that can be recommended for checking the impairments of special signals resulting from transmission over television circuits.

Note. — See Reports 823 and 969.

STUDY PROGRAMME 21D-1/CMTT

CHARACTERISTICS OF VIDEO SIGNALS

(1978-1982)

The CCIR,

CONSIDERING

- (a) that nominal video bandwidths have been assigned to television circuits in various transmission systems;
- (b) that multiplex transmission over a common path depends on the absence of significant signal levels at frequencies outside the band required for satisfactory transmission of television signals;
- (c) that knowledge of the actual video signal spectrum is required for the design of sampling circuits in digital encoders;
- (d) that protection from unwanted frequencies needs to be economically provided,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the characteristics to be expected of a video signal source feeding the input terminals of a transmission circuit: in particular, the spectral energy distribution of a typical and worst-case video signal, both inside and outside the nominally transmitted band (Note 3);

* This Study Programme should be brought to the attention of Study Group 11.

2. the levels of spectral components, outside the nominal bandwidth, which can reasonably be accepted by various types of television transmission systems without special precautions (Note 4);

3. recommendations that could be made regarding the worst-case signal to be permitted at video interconnecting points.

Note 1. — The designer of television transmission equipment (whether for use on radio-relay or cable systems) needs some indication of what protection it is reasonable to provide and which will directly cater for the majority of cases. Certain signal sources (for instance some test signal generators) can produce amounts of out-of-band energy well in excess of that of a normal television programme signal. It seems uneconomic to design all transmission equipment for exceptional cases; these can probably be dealt with better at the origin. If specific interface conditions can be agreed, there could be full flexibility for connecting any signal source to any circuit.

Note 2. — See Report 970.

Note 3. — Study Group 11 is invited to contribute to this aspect of the Study Programme.

Note 4. — Study Groups 4 and 9 of CCIR and Study Group XV of the CCITT are invited to contribute to this aspect of the Study Programme.

STUDY PROGRAMME 21E/CMTT *

DIGITAL TRANSMISSION OF SIGNALS COMPRISING A TIME-DIVISION MULTIPLEX TELEVISION PROGRAMME

(1978)

The CCIR

UNANIMOUSLY DECIDES that the following studies should be carried out:

methods of multiplexing the various signals making up a television programme, in their digital forms (vision signals, test signals, sound signals, data signals, etc.).

QUESTION 22/CMTT **

HIGH-DEFINITION TELEVISION

(1978)

The CCIR,

CONSIDERING

- (a) that present systems for television are not ideal in respect to picture quality; for example, sharpness and reality;
- (b) that transmission of wideband video signals in bands for terrestrial and broadcasting-satellite systems, as well as over cable systems, will be possible;
- (c) that the rapid progress in the development of displays will permit the use of large-screen, high-definition television displays for domestic reception;
- (d) that Study Group 11 is studying the standards which should be recommended for high definition television systems intended for broadcasting to the general public under Question 27/11;
- (e) that problems of long-distance transmission are involved in the choice of standards for high-definition television systems intended for broadcasting to the general public;
- (f) that the existing CCIR Recommendations do not seem to provide a complete solution to these problems,

* This Study Programme partly replaces Study Programme 10A-1/CMTT (Geneva, 1974).

** This Question replaces Question 12/CMTT (Geneva, 1974) and Study Programme 12A/CMTT (Geneva, 1974).

UNANIMOUSLY DECIDES that the following question should be studied:

1. what characteristics of telecommunication systems are to be envisaged for the long-distance transmission of high-definition television systems forming the study of Question 27/11;
2. what is the value of these characteristics and the relevant tolerances allowed to ensure satisfactory transmission quality;
3. what are the measuring and monitoring methods to be recommended?

QUESTION 23/CMTT

TECHNICAL METHODS FOR ENSURING PRIVACY IN LONG-DISTANCE INTERNATIONAL TELEVISION TRANSMISSION

(1982)

The CCIR,

CONSIDERING

- (a) that radio signals are by their very nature likely to be received by a large number of unidentified receivers and that in the case of international television transmission using telecommunication satellites, stations to which the information is not addressed may receive and interpret the signals;
- (b) that the number of stations capable of receiving such signals is steadily increasing;
- (c) that undesired access to the transmitted signal is facilitated when similar technical characteristics are used for broadcasting as those used in transmission,

UNANIMOUSLY DECIDES that the following question should be studied:

what technical methods can be used for ensuring privacy in long-distance international television transmission using radiocommunication techniques?

Note. — Both in the Study Group and in the Plenary Assembly (see minutes of Eleventh Plenary Session of the XVth Plenary Assembly) some administrations questioned the conformity of this Question with the terms of reference of the CCIR and expressed their concern about the difficulties involved.

OPINION 41-2

BANDWIDTHS OF SOUND-PROGRAMME CIRCUITS

(1970-1974-1978)

The CCIR,

CONSIDERING

that the views of the CMTT were sought by CCITT Study Group IV in [CCIR, 1966-69] on two proposed classifications by bandwidth of programme circuits,

IS UNANIMOUSLY OF THE OPINION

1. that the function of the CMTT is to draw up overall performance specifications for programme circuits of various nominal bandwidths;
2. that the nominal bandwidth of a programme circuit should be defined as the band of frequencies over which the performance is specified;

3. that a classification of programme circuits is necessary;
4. that the numbers of nominal bandwidths should be restricted as far as possible in order to limit the amount of work involved in arriving at specifications;
5. the nominal bandwidths should be chosen taking due account of practicable methods of providing the circuits; it is recognized that while from the circuit user's point of view it does not seem necessary to take account of the methods of providing circuits when choosing nominal bandwidths for classes of circuits, the economics of long-distance circuit provision will be strongly influenced by the relationship between the bandwidth of the programme circuit and the frequency allocation used for multi-channel telephone systems;
6. that the chosen nominal bandwidths should be restricted to the following:
 - that corresponding to the highest quality (40 Hz to 15 kHz) dealt with in Recommendation 505 and Report 496;
 - a narrow bandwidth of about 100 Hz to 5 kHz, for which the nearest corresponding specifications are contained in Recommendation 503 and CCITT Recommendation J.33 (6.4 kHz limit);
 - an intermediate bandwidth, for example, of 10 kHz. Recommendation 504 specifies a circuit which would be in this category.

Note. — This paragraph takes account of the views of the broadcasting authorities.

7. that consideration of this subject should continue with the co-operation of the broadcasting authorities.

Note. — This Opinion should be brought to the attention of Study Group 10.

REFERENCES

CCIR Documents

[1966-69]: CMTT/53 (CCITT)

DECISION 18-3 *

**DIGITAL SYSTEMS FOR THE TRANSMISSION OF
SOUND-PROGRAMME AND TELEVISION SIGNALS**

(1974-1976-1978-1981)

The CMTT,

CONSIDERING

- (a) that CCITT Study Group XVIII is studying the problems associated with the establishment of an integrated digital network for the transmission of all kinds of signals, including those for sound-programme and television;
- (b) that there is an urgent need to ensure that the standards adopted for the integrated digital network shall be compatible with the requirements necessary for the transmission of sound-programme and television signals;
- (c) that CCIR Study Groups 10 and 11 are studying the use of digital techniques in programme production and emission;
- (d) that the CMTT is studying the performance requirements applicable to long-distance digital transmission circuits for sound-programmes and television;
- (e) that the CMTT has to consider the transition period during which technological changes will take place;
- (f) that there is a need to co-ordinate the studies undertaken by Study Groups 10 and 11 and the CMTT, with those in progress in CCITT Study Group XV and XVIII,

DECIDES

1. that an Interim Working Party (CMTT/1) be established with the following terms of reference:
 - 1.1 to examine the results of studies on digital techniques effected in Study Groups 10 and 11 and the CMTT and to make any proposals necessary to ensure maximum compatibility between the recommendations of the Study Groups;
 - 1.2 to co-operate with CCITT Study Group XV and XVIII to ensure that the needs for the transmission of sound-programme and television signals are satisfied by the integrated digital network;
2. that the Chairman of the Interim Working Party shall be a representative of the Administration of the United Kingdom;
3. that the Interim Working Party shall be composed of the Chairmen and Vice-Chairmen of the CCIR and CCITT Study Groups concerned, as well as of representatives nominated by the following Administrations: Germany (Federal Republic of), Australia, Canada, Denmark, United States of America, Finland, France, India, Italy, Japan, Norway, Netherlands, United Kingdom, Sweden, Switzerland, U.S.S.R. and Yugoslavia;
4. that, as far as possible, the work of the Interim Working Party shall be carried out by correspondence;
5. that coordination of digital studies in the CCIR shall be carried out in accordance with Annex I;
6. that the specification of performance for digital transmission circuits in general shall be in accordance with the principle that the first objective for systems employing digital or digital and analogue methods should be that the overall performance is not worse than the equivalent analogue system. Later in the development of digital systems it may be possible to improve standards of performance.

* This Decision should be brought to the attention of CCIR Study Groups 10 and 11 and CCITT Study Groups XV and XVIII.

ANNEX I

COORDINATION OF DIGITAL STUDIES IN STUDY GROUPS 10, 11 AND CMTT

1. Organization of work

To provide effective coordination of studies of Study Groups 10, 11 and the CMTT on digital techniques, the results of the work of these Study Groups will be examined by the Interim Working Party before transmission to the CCITT Study Group XVIII. If necessary, it will make proposals to ensure compatibility between the results of the Study Groups. Between Study Group meetings this work will continue by correspondence. The results of the study by correspondence will be transmitted to the Study Groups prior to their meeting. The timing of the work of the Interim Working Party must take account of meetings of CCITT Study Groups XV and XVIII as well as those of the CCIR Study Groups.

The Interim Working Party considers that the work of coordination would be expedited if the meetings of Study Groups 10, 11 and the CMTT could be held simultaneously and separate from meetings of the other CCIR Study Groups.

2. Coordination of studies

The Interim Working Party proposes that particular parts of studies concerning digital techniques should receive emphasis by Study Groups in the manner given in the Table below.

TABLE I

*Distribution of the emphasis on studies concerning
digital techniques among certain Study Groups of the CCITT and the CCIR,
the CMTT, and the Interim Working Party CMTT/1*

1. Specification of PCM systems and hierarchy of binary bit rates ensuring compatibility between the needs of various services making use of the digital transmission networks, (CCITT Study Group XVIII).
2. * Coding methods, including redundancy reduction for foreseen application of sound broadcasting and television, and standards to be adopted for:
 - 2.1 studio installations including recording { sound: (S.G. 10)
television: (S.G. 11);
 - 2.2 terrestrial radio-relay links {
 - 2.3 cable transmission links { (CMTT);
 - 2.4 satellite transmission links {
 - 2.5 broadcasting from terrestrial transmitters and from satellites { sound: (S.G. 10)
television: (S.G. 11).
3. * Study of ways of protecting against digital transmission impairments (e.g. errors, jitter, slip) for:

	<i>Studio</i>	<i>Transmission link</i>
3.1 sound programmes	(S.G. 10)	{ (CMTT).
3.2 television	(S.G. 11)	
4. * Compatibility of the relevant methods for:
 - 4.1 studio installations, including recording; **
 - 4.2 terrestrial radio-relay links;
 - 4.3 cable transmission links;
 - 4.4 satellite transmission links;
 - 4.5 broadcasting from terrestrial transmitters and from satellites between themselves and with the other services envisaged under § 1 (Interim Working Party).

* Close cooperation between the CMTT Interim Working Party and Study Group XVIII is essential for these studies.

** Aspects of compatibility relating to the exchange of recordings are matters for Study Groups 10 and 11.

5. Study of transmission methods over various transmission media, including supervision, for:
 - 5.1 studio installation, including recording { sound (S.G. 10)
television (S.G. 11);
 - 5.2 terrestrial radio-relay links (S.G. 9);
 - 5.3 cable transmission links (S.G. XV);
 - 5.4 satellite transmission links (S.G. 4);
 - 5.5 broadcasting from terrestrial transmitters and from satellites { sound (S.G. 10)
television (S.G. 11).
6. PCM multiplex equipment for sound and television programmes, including insertion of programme signals into telephony digital paths (Study Group XV and XVIII).
7. Methods of code conversion of programme signals in digital form, for example, conversion of signals encoded in a form suitable for studio application into a form suitable for long-distance transmission:
 - 7.1 sound programmes {
 - 7.2 television { (CMTT).
8. Definitions of hypothetical reference circuits and chains for sound programme and television circuits (CMTT).
9. Transmission requirements to be satisfied by the hypothetical reference circuits and chains defined in § 8 (CMTT).
10. Definitions of hypothetical reference circuits and reference connections for the integrated digital network (Study Group XVIII).
11. Measuring methods for performance and test signals for sound programmes and television;
 - 11.1 within each part of the system for:
 - 11.1.1 studio installations, including recording { sound (S.G. 10)
television (S.G. 11);
 - 11.1.2 terrestrial radio-relay links (S.G. 9);
 - 11.1.3 cable transmission links (S.G. XV);
 - 11.1.4 satellite transmission links (S.G. 4);
 - 11.1.5 broadcasting from terrestrial transmitters and from satellites { sound (S.G. 10)
television (S.G. 11);
 - 11.2 between the input ports and output ports of the transmission chain of § 11.1.2, 11.1.3 and 11.1.4 (CMTT).
12. Study of subjective quality of signals encoded and transmitted in digital form, or which have been processed (e.g. chroma-key/colour-matte or other special effects) using signals transmitted in digital form for:
 - 12.1 sound programmes (S.G. 10);
 - 12.2 television (S.G. 11).
13. Problems of switching with a view to:
 - 13.1 switching and mixing in the studio (cutting from one camera to another cross-mixing, inlay and other special effects and equivalent operations in sound broadcasting) { sound (S.G. 10)
television (S.G. 11).
 - 13.2 switching in the transmission network (programme switching or otherwise) (CMTT).

3. Digital sound programme

During the Study Period 1978-1982 the work of Study Group 10, the CMTT and CCITT Study Group XVIII was reviewed.

It was noted that in the CMTT the following activities took place.

3.1 An attempt to get agreement on a single solution for a Recommendation for digital transmission of sound-programme signals.

Following the Interim Meeting of the CMTT (30 September - 17 October 1980, Geneva) at which a draft Recommendation for digital transmission of sound-programme signals was attached to Report 647 as Annex I, a number of contributions were received from administrations on this subject. An attempt was made to prepare an acceptable Recommendation. The different methods as listed in Table I of Report 647 were considered and some of them were put for adoption as a single solution. It was not possible to get unanimous support to any of the methods. Finally a method with uniform quantization, 14 bits/sample was suggested as a compromise but also for that method no common agreement could be reached. The revised text of Annex I of Report 647 summarizes the different methods which were considered. Some additional information which could possibly help for future progress on this subject is given in that Annex and the attention of administrations is drawn to that text for further action.

3.2 Furthermore CMTT did receive considerable information of CCITT SG XVIII referring to new services, network aspects etc. as regards to the future digital transmission network. Besides some amendment in the CMTT documentation, a new Study Programme: 18E/CMTT is proposed to study the specific requirements of sound-programme circuits transmitted over the integrated service digital network (ISDN).

In Study Group 10 a draft Report has been prepared on the subject of standard digital audio interface for studios. Another draft Report contains a proposal for a draft new Recommendation on the sampling frequency for digital sound signals in broadcasting studios.

Study Group 10 is of the opinion that a large consensus was reached on the use of a common sampling frequency of 48 kHz for studio applications, including recording of sound on audio tape recorders and video tape recorders.

Study Group 10 therefore decided to prepare a proposal for a draft new Recommendation and it is hoped that, in view of the present trend in the industry this proposal could be unanimously accepted during the next interim meetings of Study Group 10, in 1983.

4. Digital television

During the Study Period 1978-82 the work of Study Group 11, the CMTT and CCITT Study Group XVIII was reviewed. It was noted that several contributions had been received concerning practical digital television transmission systems and methods of protection against bit errors. Transmission impairments were deemed to be a sufficiently important topic to justify a separate Report and accordingly a new Report 967 was generated as a companion to the main Report on digital television transmission (Report 646).

During the period further information was received from CCITT Study Group XVIII concerning hierarchical bit rates and digital circuit performance characteristics and this has accordingly been incorporated into Report 817. Clarification, however, is still required on the precise customer-access interface requirement both in regard to leased circuits and to the short-term use of circuits established through the ISDN. In this context CCITT Study Group XVIII should note that the CMTT is considering the use of bit rates at the 32.064, 34.368, 44.736 and 139.264 Mbit/s hierarchical levels for international exchanges. Partial use of the latter type of circuit at 68.736 Mbit/s could also be of interest in order to achieve a balance between the cost of bit-rate reduction equipment for component-coded signals and the cost of transmission.

In Study Group 11 a Recommendation (601) on digital coding standards was adopted. The principal items to be recommended include:

- the standards should use component coding based on the luminance and the two colour-difference signals, or the three primary signals R.G.B.;
- the standards should allow the evolution of an extensible family of compatible digital coding standards;
- the member of the family recommended for use in interfaces between main digital studio equipment and for international programme exchange is the 4:2:2 member in which the sampling frequency is 13.5 MHz for the luminance signal and 6.75 MHz for each of the two colour-difference signals. The same sampling frequencies apply to both the 525-line and the 625-line system. Sampling structures are orthogonal, and line, field and picture repetitive in both systems;
- uniformly-quantized PCM, using 8 bits for each component signal, should be the recommended form of coding.

The standards for the lower members of the family will be a main subject in the next study period.

Report 629 and Report 962 were revised to include new technical information, in particular, new explanatory and supporting information to the new Recommendation.

Revision was also made of Question 25/11 and its Study Programmes in order to update and improve the texts in accordance with the adoption of the new Recommendation.

As a result of the adoption by Study Group 11 of Recommendation 601 a corresponding Recommendation (604) was generated by the CMTT on the general principles to be followed for the all-digital transmission of component-coded signals. A proposal for a similar Recommendation for mixed analogue-and-digital transmission was considered but was not adopted because of the fundamentally divergent opinions that are expressed in Annex II of Report 646.

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