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INTERNATIONAL TELECOMMUNICATION UNION

CCIR

INTERNATIONAL
RADIO CONSULTATIVE
COMMITTEE

RECOMMENDATIONS AND REPORTS OF THE CCIR, 1986

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

XVIth PLENARY ASSEMBLY
DUBROVNIK, 1986

VOLUME XII

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



Geneva, 1986

CCIR

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

2. The objectives of the CCIR are in particular:

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

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

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



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

(Dubrovnik, 1986)

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.
VOLUMES 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-1	Land mobile service. Amateur service. Amateur-satellite service.
VOLUME VIII-2	Maritime mobile service.
VOLUME VIII-3	Mobile satellite services (aeronautical, land, maritime, mobile and radiodetermination). Aeronautical mobile service.
VOLUME IX-1	Fixed service using radio-relay systems.
✓ VOLUME X-1	Broadcasting service (sound).
✓ VOLUMES X/XI-2	Broadcasting-satellite service (sound and television).
✓ VOLUMES X/XI-3	Sound and television recording.
✓ 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 XVIth 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 1986 edition, unless otherwise noted; i.e., only the basic number is shown.

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

Volumes I to XIV, XVith Plenary Assembly, contain all the valid texts of the CCIR and succeed those of the XVth Plenary Assembly, Geneva, 1982.

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.

1.2 Recommendations

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48	X-1	367	II	478	VIII-1
80	X-1	368-370	V	479	II
106	III	371-373	VI	480	III
139	X-1	374-376	VII	481-484	IV-1
162	III	377, 378	I	485, 486	VII
182	I	380-393	IX-1	487-493	VIII-2
205	X-1	395-405	IX-1	494	VIII-1
215, 216	X-1	406	IV/IX-2	496	VIII-3
218, 219	VIII-2	407, 408	X/XI-3	497	IX-1
239	I	410-412	X-1	498	X-1
240	III	414, 415	X-1	500	XI-1
246	III	417	XI-1	501	X/XI-3
257	VIII-2	419	XI-1	502, 503	XII
265	X/XI-3	428	VIII-2	505	XII
266	XI-1	430, 431	XIII	508	I
268	IX-1	433	I	509, 510	II
270	IX-1	434, 435	VI	513-517	II
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290	IX-1	441	VIII-3	525-530	V
302	IX-1	443	I	531-534	VI
305, 306	IX-1	444	IX-1	535-538	VII
310, 311	V	446	IV-1	539	VIII-1
313	VI	450	X-1	540-542	VIII-2
314	II	452, 453	V	546-550	VIII-3
326	I	454-456	III	552, 553	VIII-3
328, 329	I	457, 458	VII	555-557	IX-1
331, 332	I	460	VII	558	IV/IX-2
335, 336	III	461	XIII	559-562	X-1
337	I	463	IX-1	564	X/XI-3
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362-364	II	475, 476	VIII-2	576-578	II

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1.2 Recommendations (cont.)

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581	V	609-611	II	643-644	X-1
582, 583	VII	612, 613	III	645	X-1 + XII
584	VIII-1	614	IV-1	646-647	X-1
589	VIII-2	615	IV/IX-2	648, 649	X/XI-3
591	VIII-3	616-620	V	650-652	X/XI-2
592-596	IX-1	621	VI	653-656	XI-1
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600	X/XI-2	625-631	VIII-2	658-661	XII
601	XI-1	632-633	VIII-3	662-666	XIII
602	X/XI-3	634-637	IX		
603-606	XII	638-641	X-1		

1.3 Reports

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32	X-1	322	VI (¹)	493	XII
109	III	324	I	496, 497	XII
111	III	327	III	499	VIII-1
122	XI-1	336	V	500-501	VIII-2
137	IX-1	338	V	509	VIII-3
176, 177	III	340	VI (¹)	516	X-1
181	I	342	VI	518	VII
183	III	345	III	519-522	I
184	I	347	III	524-526	I
195	III	349	III	528	I
197	III	354-357	III	530	I
200	III	358	VIII-1	533, 534	I
203	III	363, 364	VII	535, 536	II
204, 205	IV-1	371, 372	I	536-541	II
208	IV-1	374-376	IX-1	542	VIII-1
209	IV/IX-2	378-380	IX-1	543	II
212	IV-1	382	IV/IX-2	546	II
214	IV-1	383-385	IV-1	548	II
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238, 239	V	409	XI-1	574, 575	VI
249-251	VI	411, 412	XII	576-580	VII
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314	XII	488	XII	653-657	I

(¹) Published separately.

1.3 *Reports (cont.)*

Number	Volume	Number	Volume	Number	Volume
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670, 671	I	797-799	X-1	950	X/XI-3
672-685	II	800	X/XI-3	951-955	X/XI-2
687	II	801, 802	XI-1	956	XI-1
692-697	II	803	X/XI-3	958, 959	XI-1
699, 700	II	804, 805	XI-1	961, 962	XI-1
701-704	III	807-812	X/XI-2	963, 964	X/XI-3
706, 707	IV-1	814	X/XI-2	965-970	XII
710-713	IV-1	815-823	XII	972-979	I
714-724	V	826-842	I	980-988	II
725-729	VI	843-854	II	989-996	III
730-732	VII	857	III	997-1004	IV-1
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738	VII	867-875	IV-1	1007-1010	V
739-742	VIII-1	876, 877	IV/IX-2	1011-1015	VI
743, 744	VIII-2	879-880	V	1016, 1017	VII
747-749	VIII-2	882-885	V	1018-1025	VIII
751	VIII-3	886-895	VI	1026-1044	VIII-2
760-766	VIII-3	896-898	VII	1045-1051	VIII-3
768	VIII-3	899-906	VIII-1	1052-1057	IX-1
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774, 775	VIII-2	917-923	VIII-3	1073-1076	X/XI-2
778	VIII-1	925-929	VIII-3	1077-1089	XI-1
779-789	IX-1	930-934	IX-1	1090-1096	XII
790-793	IV/IX-2	936-942	IX-1		

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*

Number	Volume	Number	Volume	Number	Volume
4	VI	61	XIV-1	76	X-1
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15	I	63	VI	79-83	XIV-1
20	VIII-1	64	X-1	86, 87	XIV-1
23	XIII	66	XIII	88	I
24	XIV-1	71	I	89	XIII
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33	XIV-1	74	VI	96	XI-1
39	XIV-1				

1.5 *Opinions*

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14	IX-1	49	VIII-1	74	X-1
15	X-1	50	IX-1	75	XI-1
16	X/XI-3	51	X-1	77	XIV-1
22, 23	VI	56	IV-1	79-81	XIV-1
26-28	VII	59	X-1	82	VI
32	I	63	XIV-1	83	XI-1
35	I	64	I	84	XIV-1
38	XI-1	65	XIV-1	85	VI
40	XI-1	66	III	86	XIII
42	VIII-1	67-69	VI		

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1.6 Decisions

Number	Volume	Number	Volume	Number	Volume
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3-5	V	50	V	63	III
6	VI	51	X/XI-2	64	IV-1
9-11	VI	52	X-1	65	VII
18	XII	53, 54	I	66	XI-1
19	XIII	56	I	67, 68	XII
27	I	57	VI	69	VIII-1
32	VIII-3	58	XI-1	70	IV-1
42	XI-1	59	X/XI-3	71	VIII-3 + X-1
43	X/XI-2	60	XI-1	72	X-1 + XI-1

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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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.

1982-1986 *Chairman:* Y. ANGEL (France)
Vice-Chairman: W. G. SIMPSON (United Kingdom)

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

INTRODUCTION BY THE CHAIRMAN OF THE CMTT

1. CMTT activities from 1982 to 1986

1.1 During the study period 1982-1986, the CMTT held two meetings in Geneva:

- Interim Meeting from 29 August to 13 September 1983;
- Final Meeting from 21 October to 5 November 1985.

The volume of documentation amounted to 176 technical contributions.

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

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

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

<i>Working Party</i>	<i>Terms of Reference</i>	<i>Chairman</i>	
		<i>Interim Meeting</i>	<i>Final Meeting</i>
CMTT-AB	Q.13, 15, 21, 22	Mr. P. R. Wickliffe (United States of America)	Mr. J. McGrath (United States of America)
CMTT-AN	Q.14, 16, 23	Mr. J. M. Corbett (United Kingdom)	Mr. J. M. Corbett (United Kingdom)
CMTT-C	Q.17, 18, 19, 20	Mr. P. Wery (Canada)	Mr. P. Wery (Canada)

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

Interim Meeting

Mr. R. Gardiner (United Kingdom)
 Mr. S. Bonhoff (France)
 Mr. G. Duvic (France)
 Mr. E. Martinez de Aragon (Spain)

Final Meeting

Mr. R. J. Fevin (United Kingdom)
 Mr. C. Bremenson (France)
 Mr. Le Van Khiem (France)
 Mr. M. Jimenez-Rodriguez (Spain)

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

At these meetings, statements were made by the following:

— on television transmissions:

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

— on sound programme transmissions:

Mr. Steinke (OIRT), Chairman of Working Party 10-C;
 Mr. Wery (Canada), Chairman of Working Party CMTT-C.

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

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

1.5 In view of the limited facilities made available to the Study Groups, the CMTT set up a small organizational group to simplify the work of the Final Meeting consisting of its Chairman and Vice-Chairman, the Chairmen of the Working Parties, a representative of the Editorial Group and the CCIR Counsellors.

The setting up of this group made it possible to work with maximum efficiency in the limited time available.

1.6 The Special Rapporteurs of the CMTT for liaison with CCITT Study Groups during the study period 1982-1986 were as follows:

— CCITT Study Group IV — RA/TV Group, which became Working Party IV/6 in 1984:

Mr. R. Gardiner (United Kingdom) until the Interim Meeting,
 Mr. J. McNulty (United States of America) appointed at the Final Meeting.

— CCITT Study Group XV:

Mr. F. Hautsch (Germany (Federal Republic of)) until the Final Meeting,
 Dr. L. Stenger (Germany (Federal Republic of)) after the Final Meeting.

— CCITT Study Group XVIII:

Mr. P. Wery (Canada), throughout the period, his appointment having been renewed at the Final Meeting.

— CMBD:

Mr. S. Bonhoff (France), until the disbandment of this Study Group (VIIIth CCITT Plenary Assembly, Malaga-Torremolinos, 1984).

2. Results obtained

The Working Parties prepared a total of 152 documents during the two meetings, with the following results:

2.1 Texts submitted to the Plenary Assembly for approval

2.1.1 Recommendations

Of the 16 Recommendations appearing in Volume XII (Geneva, 1982) the CMTT proposed the modification of seven and the withdrawal of one.

It also proposed the adoption of the following four new Recommendations:

Section A

Recommendation 658 — Mixed analogue-and-digital transmission of analogue composite television signals.

Section C

Recommendation 659 – Digital transmission of sound programmes – General principles.

Recommendation 660 – Transmission of analogue high-quality sound-programme signals on mixed analogue-and-digital circuits using 384 kbit/s channels.

Section D

Recommendation 661 – Signals for the alignment of international sound-programme connections.

The CMTT also formally approved the draft texts of new Recommendations 642 and 645 prepared by CCIR Study Group 10. Since these texts relate to equipments and signals for long-distance sound-programme transmissions, the CMTT would like them to be published in CCIR Volume XII.

2.1.2 *Questions*

The CMTT has proposed the revision of the wording of six of the eleven existing Questions.

Previously, digital and mixed analogue-and-digital transmission of sound-programme signals has been covered by Questions 18/CMTT and 20/CMTT. Technological developments have caused the CMTT to consider it preferable to combine mixed transmissions with purely analogue transmissions, with consequent amendments to Questions 17/CMTT, 18/CMTT, 19/CMTT and 20/CMTT.

The advances made in recent years in multiplexing and high-definition television, have also made it necessary to update the relevant Questions 21/CMTT and 22/CMTT.

Finally, on the basis of contributions received, it is proposed to change Study Programme 21F/CMTT into a new Question 24/CMTT: Transmission of television signals using multiplexed analogue components (MAC).

2.1.3 *Opinions*

The CMTT has proposed the withdrawal of Opinion 41-2, on the grounds that the purpose of this text has been attained (preparation of Questions).

2.2 *Texts submitted to the Plenary Assembly*

2.2.1 *Reports*

The CMTT updated 19 of the 33 Reports in Volume XII (Geneva, 1982); mainly those appearing in Sections A and C. It also approved the withdrawal of Report 623, most of which now appears in Recommendation 562 (Volume X).

Finally, in the light of the contributions submitted, the CMTT adopted seven new Reports, on the following subjects:

Television

- Scrambling to ensure privacy (Report 1091);
- Bit-rate reduction for component-coded television signals (Report 1093);
- High-definition television transmission (Report 1092);
- Earth stations for outside broadcasts (Report 1090);
- Quality of transmission to a broadcasting satellite (Report 1094);
- Tests and measurements for transmitting MAC signals (Report 1096).

Sound programmes

- Transmission of signals of digital studio quality (Report 1095).

Although the Reports produced by the CMTT may seem to be relatively numerous, it should be noted that many technical standards, especially where digital transmissions are concerned, are still in a state of flux and cannot yet be presented as Recommendations.

2.2.2 Study Programmes

In addition to the revision of the Questions referred to in § 2.1.2, the CMTT reviewed the 40 Study Programmes appearing in Volume XII (Geneva, 1982). It amended 13 and withdrew four.

Nine new Study Programmes were adopted, one relating to picture transmissions:

- Transmitting earth stations for television broadcasts (SP 13G/CMTT);

and the others to sound-programme transmissions:

- Definition of a hypothetical reference connection of digital signals (SP 18J/CMTT);
- Characteristics of analogue or mixed analogue-and-digital circuits (SP 17G/CMTT);
- Interfaces between studios and digital circuits (SP 18K/CMTT);
- Digital transmission on one or two 64 kbit/s channels (SP 18G/CMTT);
- Digital transmissions of digital studio quality (SP 18F/CMTT);
- Analogue broadcasting-satellite feeder links (SP 17H/CMTT);
- Digital broadcasting-satellite feeder links (SP 18H/CMTT);
- Availability of digital connections (SP 20C/CMTT).

2.2.3 Decisions

During its meetings, IWP CMTT/1 reconsidered Decision 18 and updated it by further clarifying the responsibilities of various CCIR and CCITT Study Groups and the CMTT with respect to sound-programme and television transmissions. The new text appears in Volume XII as Decision 18-5.

After discussions in IWP CMTT/1, the Working Parties and the CMTT, two new Decisions were adopted by the CMTT:

Decision 67 – “Digital transmission of component-coded television signals”

This Decision established a new Interim Working Party (IWP CMTT/2) and its terms of reference are the same as its title. This IWP will have to cooperate closely with IWP 11/7, in order to ensure compatibility between studies. The IWP will be headed by:

Chairman: Dr. L. Stenger (Germany (Federal Republic of))

Vice-Chairmen: Dr. H. Murakami (Japan)
Mr. K. Davies (Canada)

Decision 68 – “Television and sound-programme signals in the broadband ISDN”

This Decision established a new Interim Working Party (IWP CMTT/3) to deal with problems of connection to the ISDN and will act in close collaboration with CCITT Study Group XVIII and IWP CMTT/2.

Its primary aim is to provide the Task Group on broadband aspects of ISDN set up by CCITT Study Group XVIII, with the information required by this Group concerning ISDN services which broadcasters may wish to use.

Since the Task Group had to submit its conclusions by June, 1986:

- the IWP was asked to begin its work as soon as possible after the Final Meeting;
- the IWP was authorized to submit the results of its work directly to Study Group XVIII, subject to their subsequent approval by the CMTT.

Mr. A. Brown (EBU) was appointed Chairman of IWP CMTT/3.

It was also decided that the Special Rapporteur of the CMTT for liaison with CCITT Study Group XVIII (Mr. P. Wery, Canada) should be a member of the IWP.

3. Acknowledgements

On behalf of the Vice-Chairman and himself, the Chairman wishes to express his gratitude to participating administrations and organizations for their contributions and to the delegates for their active work in an admirable spirit of cooperation, which has enabled the CMTT to achieve the substantial results described above.

He warmly thanks the Chairmen of Working Groups and Sub-Groups who, although often operating under difficult conditions, have skilfully guided the discussions which have produced the CMTT texts. He expresses his thanks to the Special Rapporteurs for having agreed to undertake their duties and for the reports they have prepared and submitted to the Final Meeting.

He also expresses his gratitude to Mr. D. Crampsey, who, in addition to his activities in a Working Group, has assumed the thankless task of Rapporteur for the Plenary Meetings of the CMTT, and congratulates him on the quality of his summary records.

Thanks were also due to Mr. A. W. Boyle and Mr. G. H. Grotelüschen, CCIR Counsellors, and to the members of the Editorial Group for their valuable assistance during meetings and in processing documents.

Finally, he extends his wholehearted thanks to Mr. R. C. Kirby, Director of the CCIR, and to all members of the staff and services of the CCIR, who had enabled the CMTT to carry out its work.

SECTION CMTT A: TELEVISION TRANSMISSION STANDARDS AND PERFORMANCE OBJECTIVES

Recommendations and Reports

RECOMMENDATION 567-2

TRANSMISSION PERFORMANCE OF TELEVISION CIRCUITS DESIGNED FOR USE IN INTERNATIONAL CONNECTIONS

(Question 13/CMTT)

(1978-1982-1986)

The CCIR,

CONSIDERING

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

UNANIMOUSLY RECOMMENDS

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

Introduction

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

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

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

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

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

The Recommendation contains five parts listed below:

Part A: Definitions of a connection and circuits

Part B: Definition of parameters

Part C: Measurement methods and test signals

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

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

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

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

A.1 Definitions

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

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

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

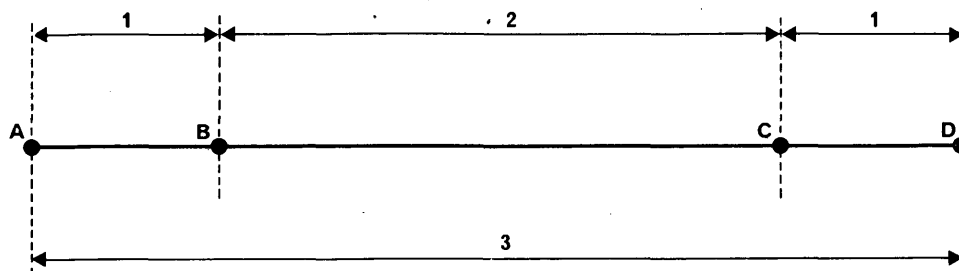


FIGURE 1 – An international television connection

- 1: Local television circuit
- 2: International television circuit
- 3: International television connection

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

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

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

The main features of the television terrestrial hypothetical reference circuit, which is an example of an international television circuit (BC in Fig. 1) and which may be of either radio or cable type, are:

- the overall length between video terminal points is 2500 km;
- two intermediate video points (M and M') divide the circuit into three sections of equal length;
- the three sections are lined up individually and then interconnected without any form of overall adjustment or correction;
- the circuit does not contain a standards converter or a synchronizing pulse regenerator, or an equipment for the insertion of signals in the line/field blanking interval.

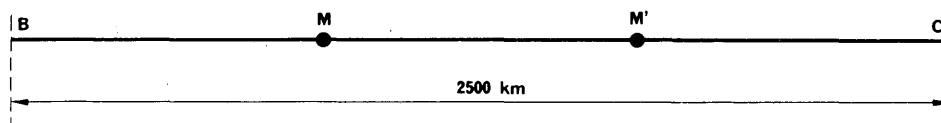


FIGURE 2a – Hypothetical reference circuit for television transmissions over a terrestrial system

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

A hypothetical reference circuit for a system in the fixed satellite service which may form part of an international television circuit (BC in Fig. 1) is defined as follows:

- it consists of one Earth station-satellite-Earth station system;
- it includes one pair of modulation and demodulation equipments for translation from the baseband to the radio-frequency carrier, and from the radio-frequency carrier to the baseband, respectively;
- it does not include a standards converter or a synchronizing-pulse regenerator, or equipment for the insertion of signals in the line/field blanking interval.

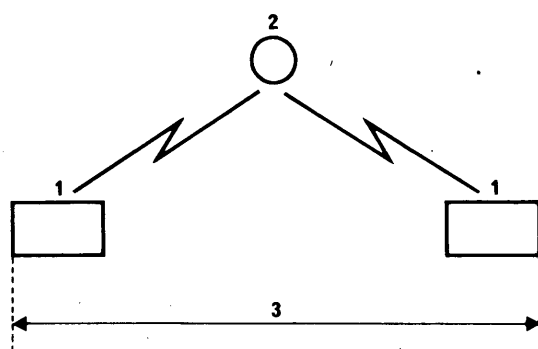


FIGURE 2b – *Hypothetical reference circuit for television transmissions over a system in the fixed satellite service*

- 1: Earth station
- 2: Satellite space station
- 3: Hypothetical reference circuit

PART B – DEFINITIONS OF PARAMETERS

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

B.1 Waveform terminology

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

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

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

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

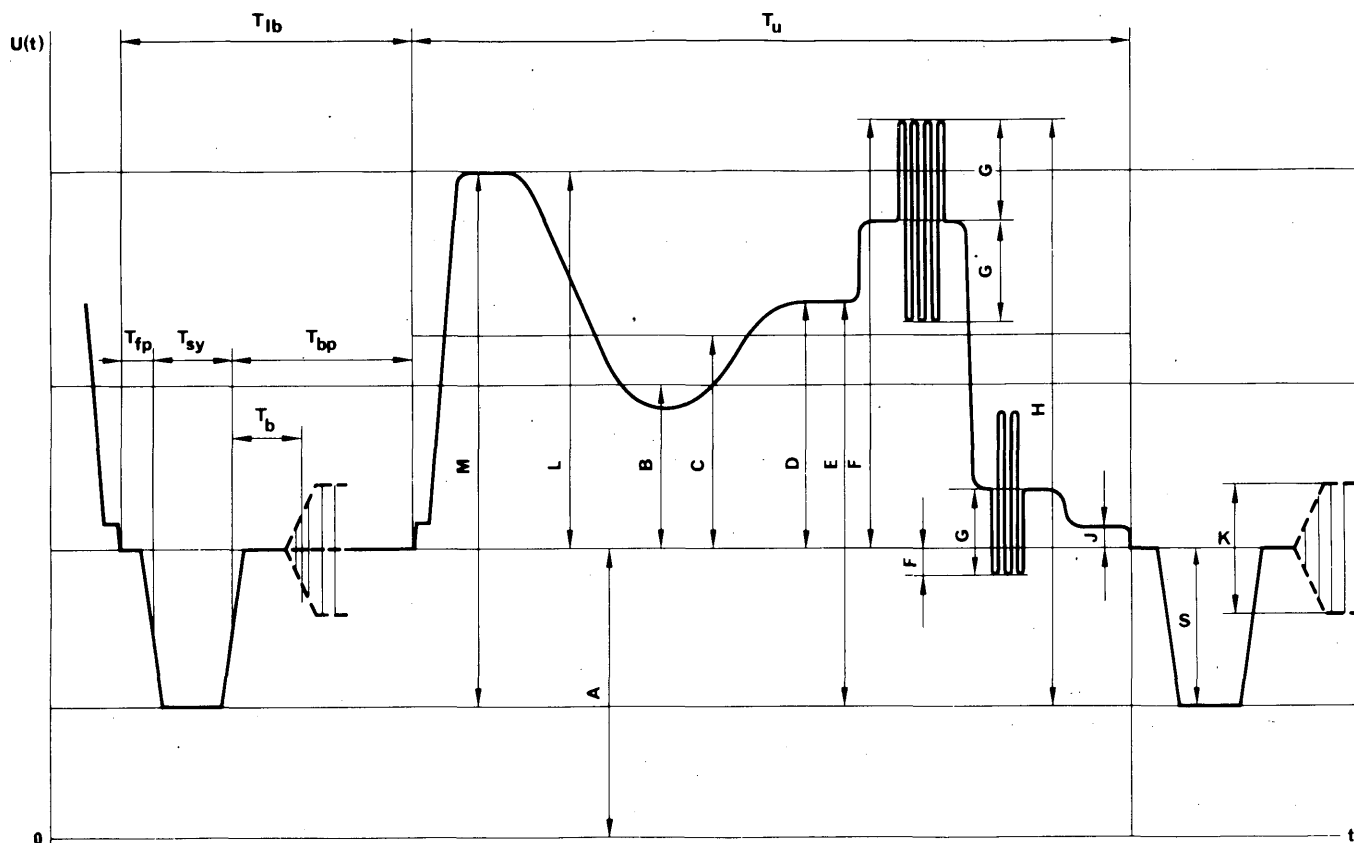


FIGURE 3 – One line of a composite colour video signal

B.2 Requirements at points of video interconnection

B.2.1 Nominal impedance (Z_0)

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

B.2.2 Return loss

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

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

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*

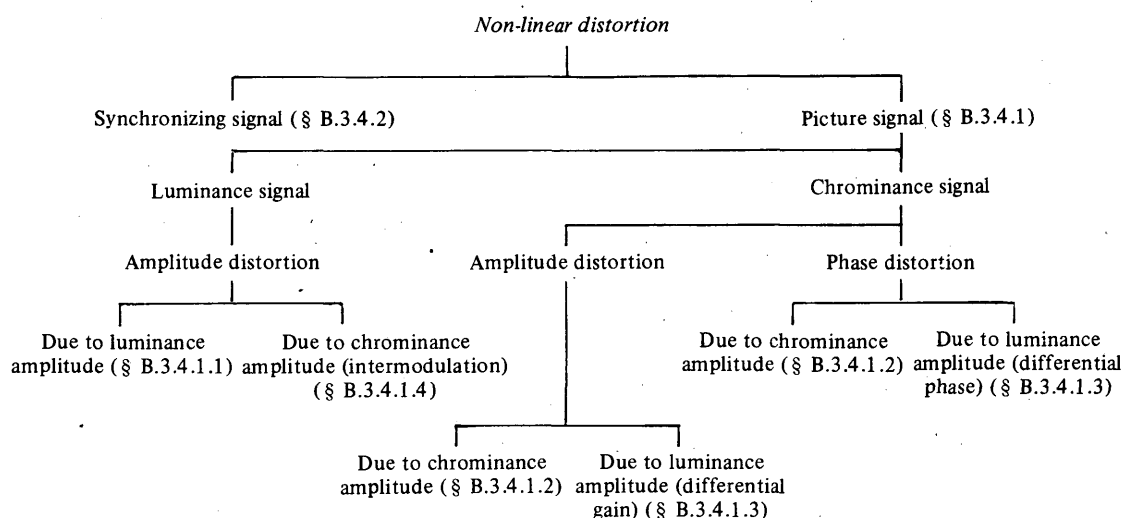
In a television circuit the transmission characteristic may not be completely linear. The extent of the non-linear distortion which is produced will depend primarily on:

- 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.

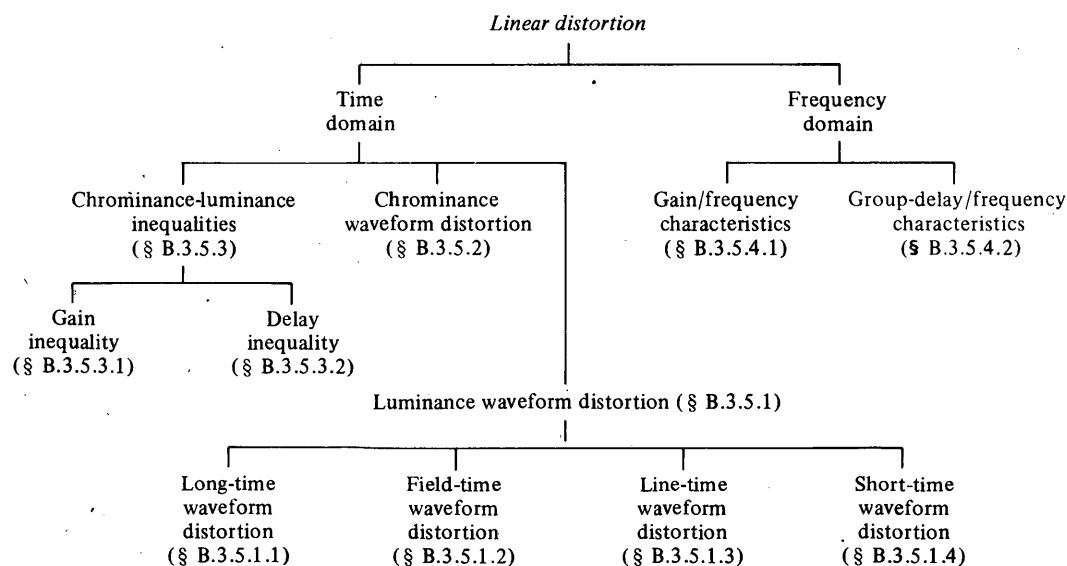
B.3.5 *Linear distortion*

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

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

C.3 Measurements of transmission performance

C.3.1 Insertion gain

The signal element used is *B3* for 625-line systems and *B2* or *B3* for 525-line systems. The amplitude *L* is measured between the centre of the bar (point *b₂* in Fig. 4) and blanking level (point *b₁* in Fig. 4). The resulting value of the received signal must remain inside the limits specified in Part D.

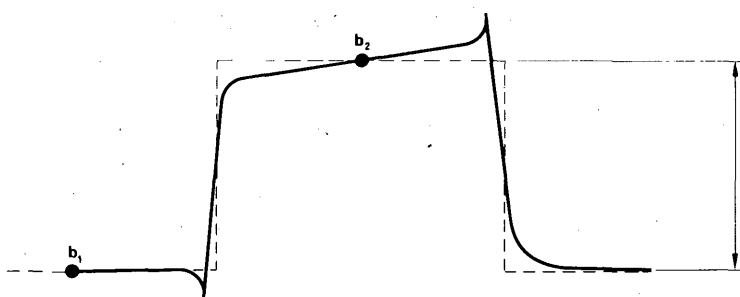


FIGURE 4 – Insertion gain measurement

C.3.2 Noise

C.3.2.1 Continuous random noise

Measuring equipment

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

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

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

Band limiting

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

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

Weighting

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

C.3.2.2 Low-frequency noise

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

C.3.2.3 Periodic noise

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

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

C.3.2.4 Impulsive noise

Impulsive noise voltages are measured by means of an oscilloscope.

C.3.3 Cross-talk from another television channel

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

Suitable combinations of elements *B*1, *B*2, *B*3 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 (*D*1) 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 (*G*2) 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 *G*2 (1 being the smallest, 3 the largest),

$k_i = \frac{2i - 1}{3}$ for 625-line signal *G*2

$k_i = 2^{i-2}$ 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 %/s.

C.3.5.1.2 *Field-time waveform distortion*

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

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

C.3.5.1.4 *Short-time waveform distortion*

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

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

C.3.5.2 *Chrominance waveform distortion*

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

C.3.5.3 *Chrominance-luminance inequalities*

C.3.5.3.1 *Gain inequality*

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

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

C.3.5.3.2 *Delay inequality*

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

C.3.5.4 *Steady state characteristics*

C.3.5.4.1 *Gain*

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

C.3.5.4.2 *Delay*

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

REFERENCES

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

CCIR Documents

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

ANNEX I TO PART C

TEST SIGNAL ELEMENTS

An indication of the signal elements required to carry out the tests mentioned in this Recommendation is given below in the form of figures. Preferred assemblies for insertion test signals are given in Recommendation 473. The preferred assemblies of elements for full-field measurements is the subject for further study. The reference designations used to describe these elements (e.g. signal *B* 1) 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 *B* 3.

Note 1. — The chrominance sub-carrier of test signal elements should be locked at the phase listed in Table I, where each phase angle is described with reference to the positive (*B*-*Y*) axis.

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

TABLE I

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

⁽¹⁾ Refer to Report 624 for system characteristics.

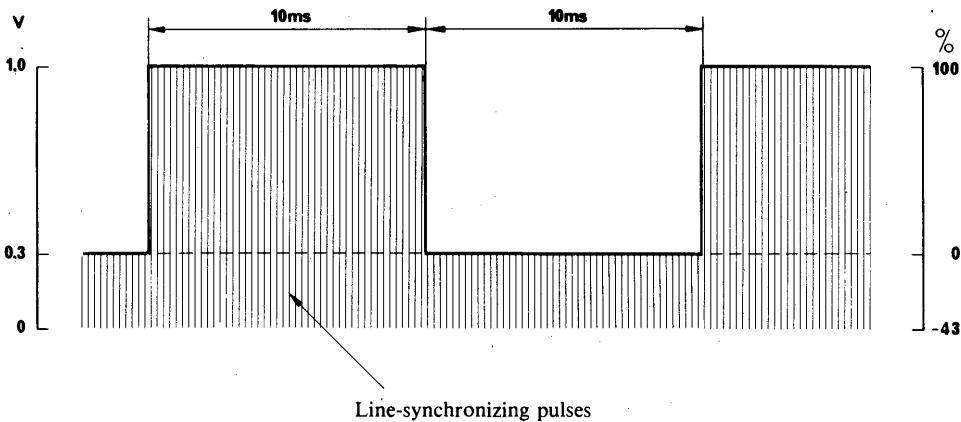


FIGURE 5 — Signal *A* for 625-line systems

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

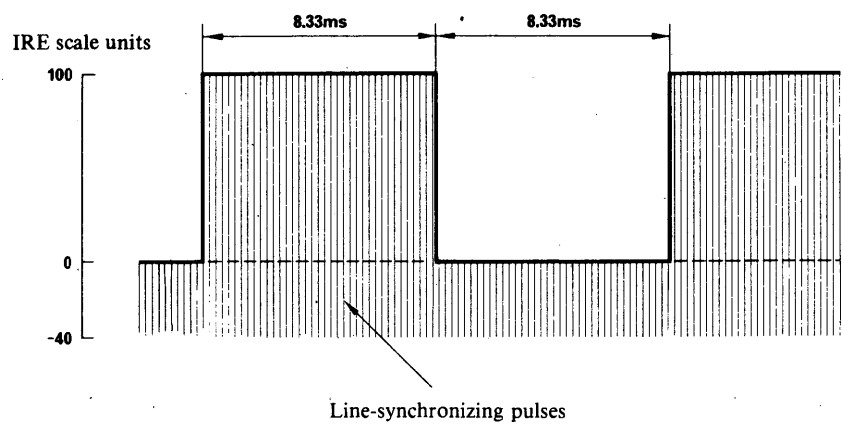


FIGURE 6a — *Signal A for 525-line systems*

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

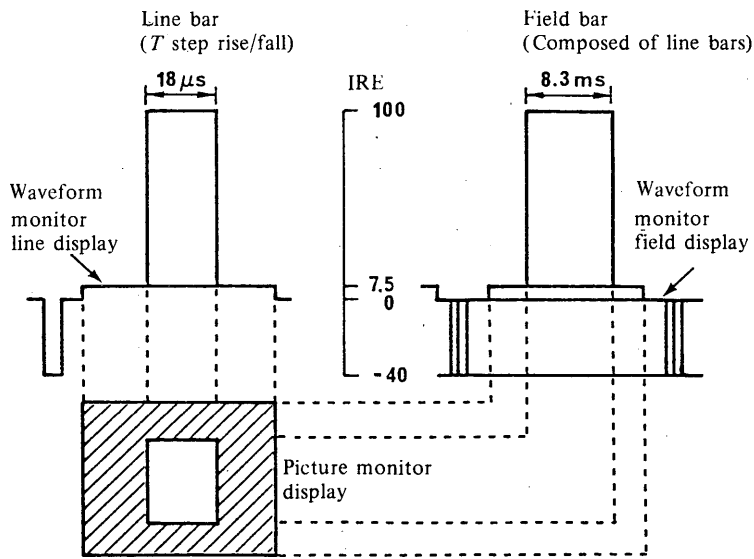


FIGURE 6b — *The window signal for 525-line systems*

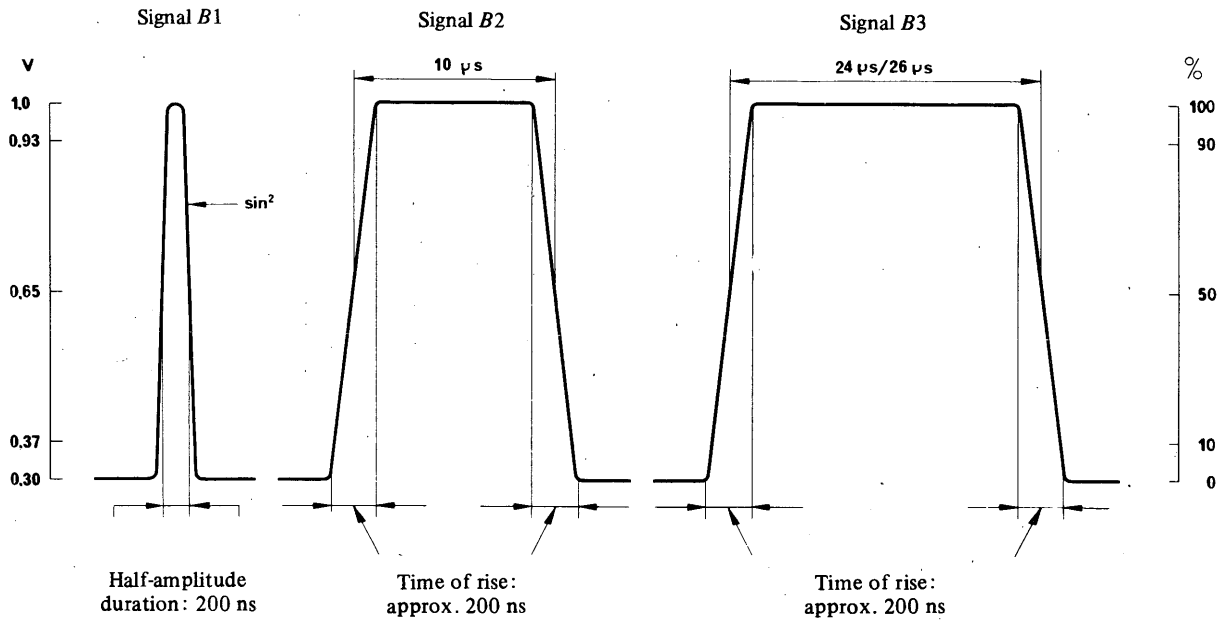


FIGURE 7 — Signal B for 625-line systems

Note 1. — In some OIRT countries, a half-amplitude duration of 160 ns is used for 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.

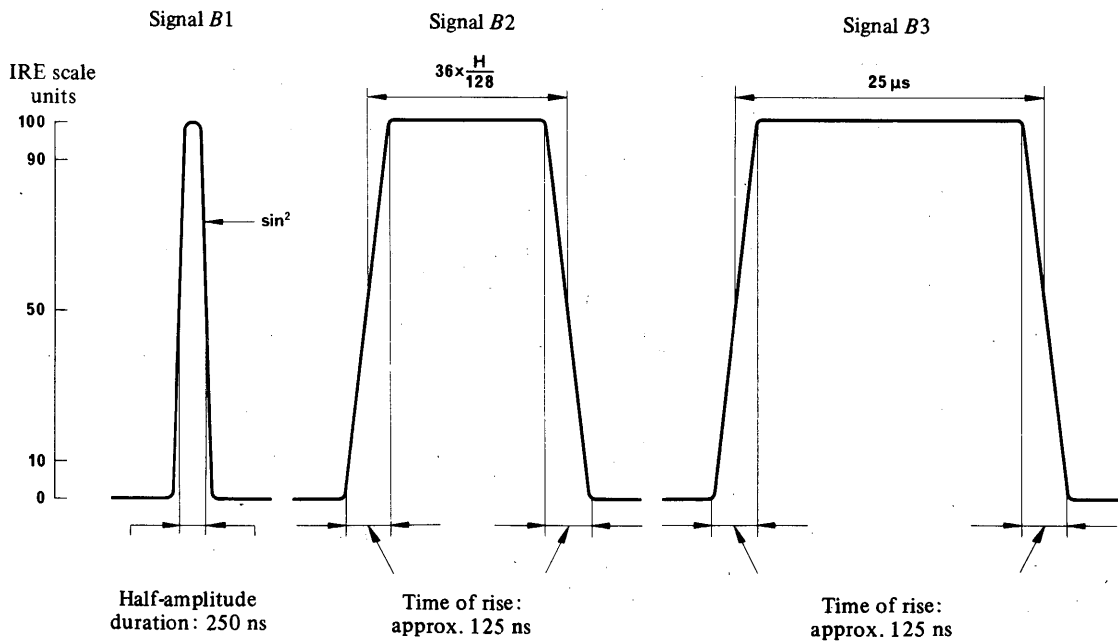


FIGURE 8 — Signal B for 525-line systems

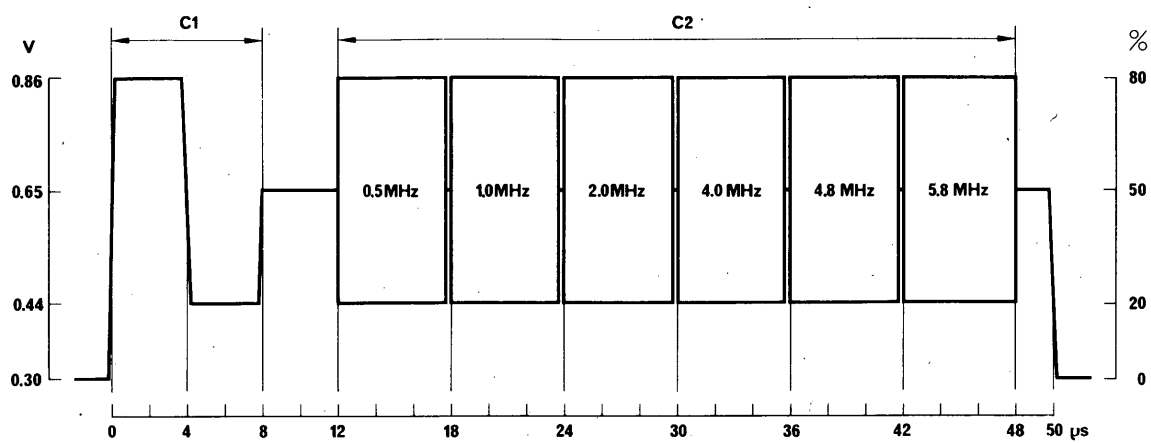


FIGURE 9 – Signal C for 625-line systems

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

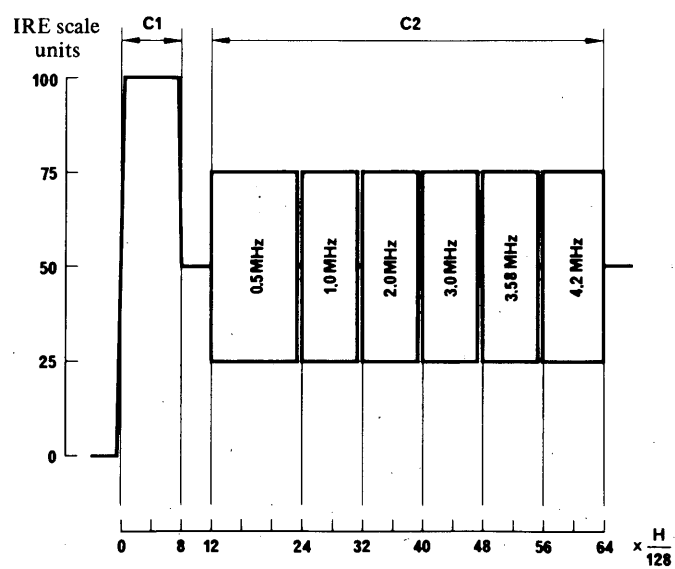


FIGURE 10 – Signal C for 525-line systems

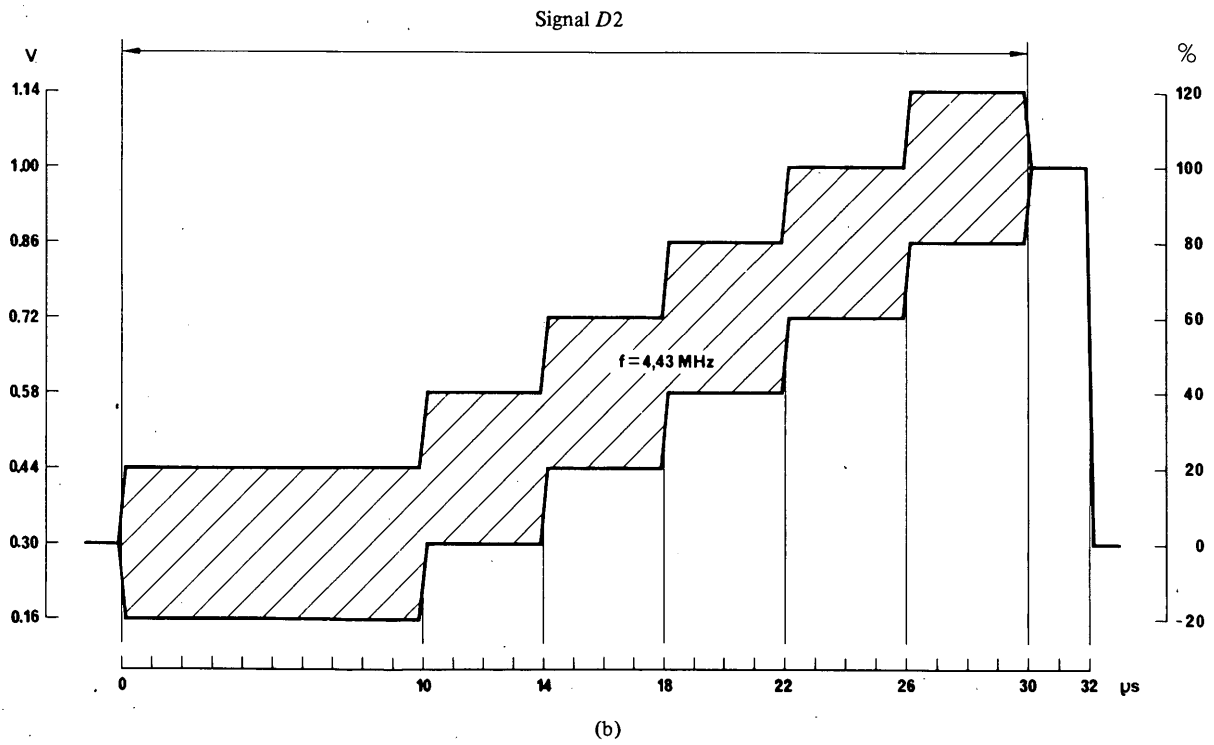
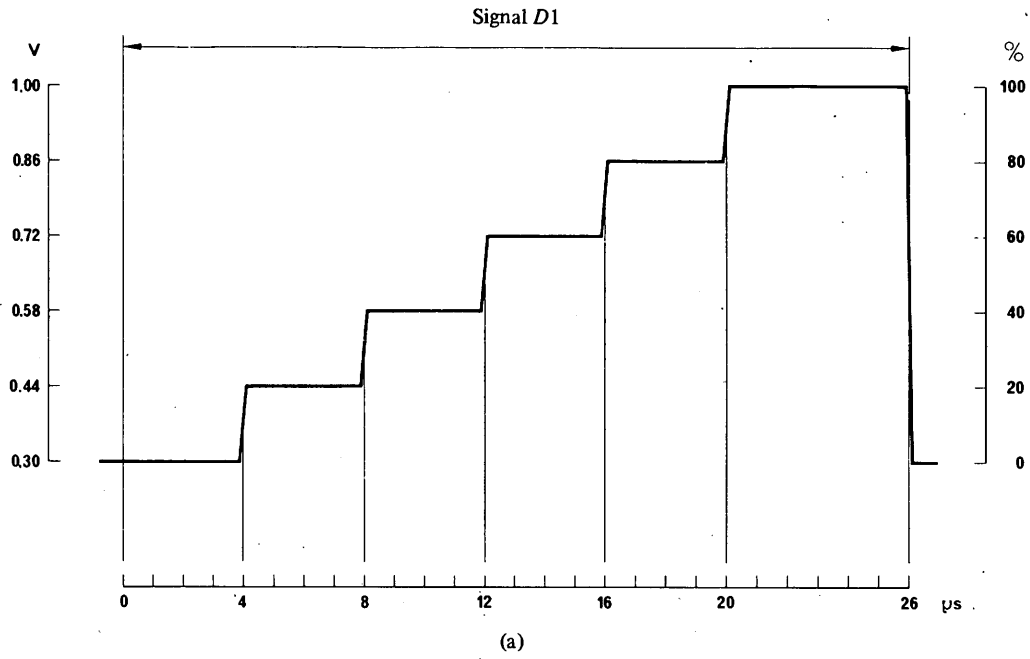


FIGURE 11 – Signal D for 625-line systems

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

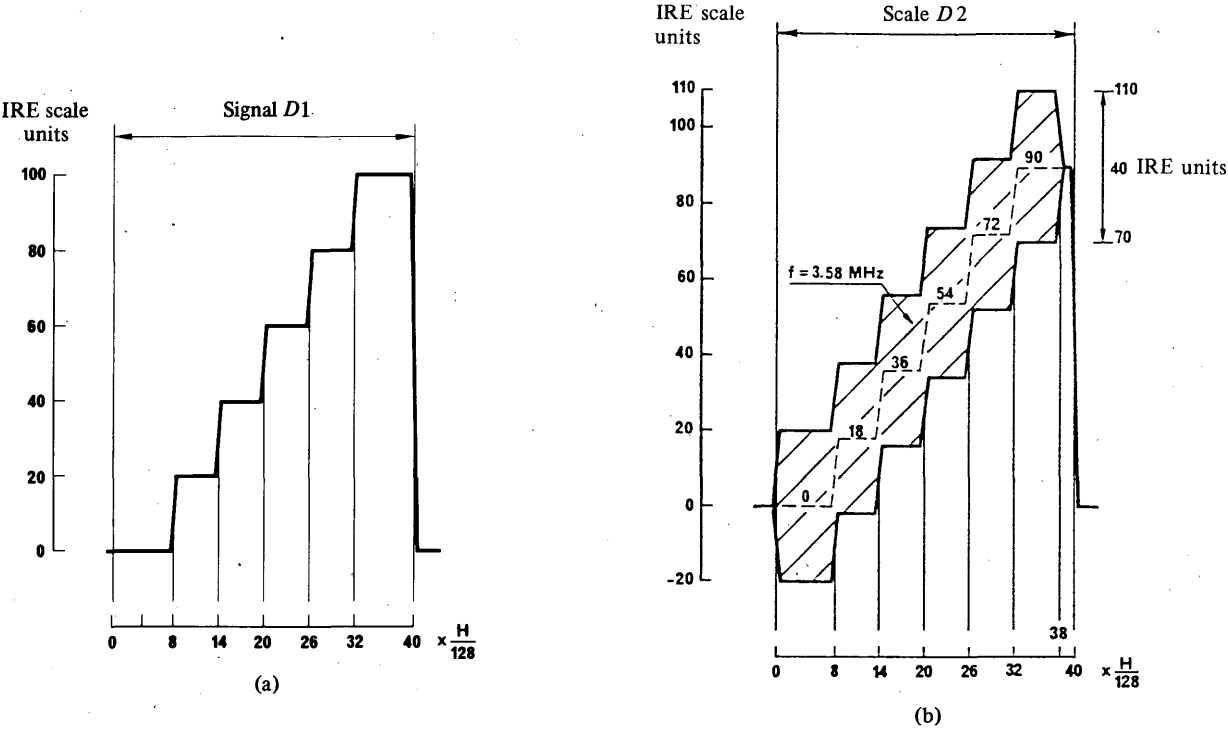


FIGURE 12 – Signal D for 525-line systems

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

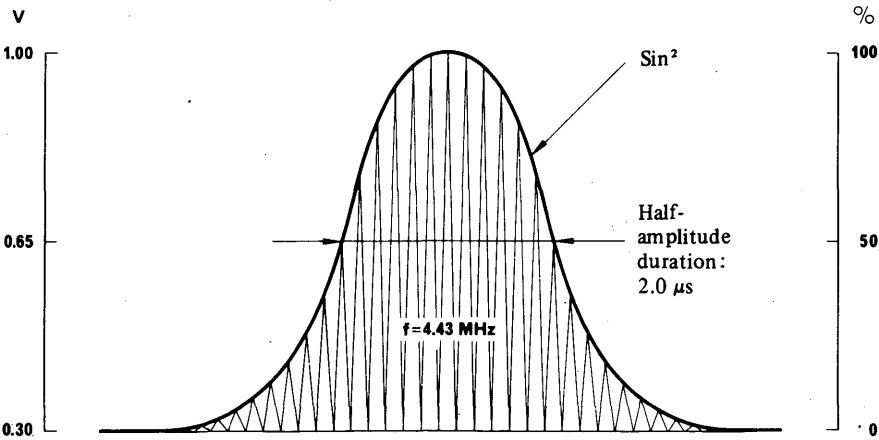


FIGURE 13 – Signal F for 625-line systems.

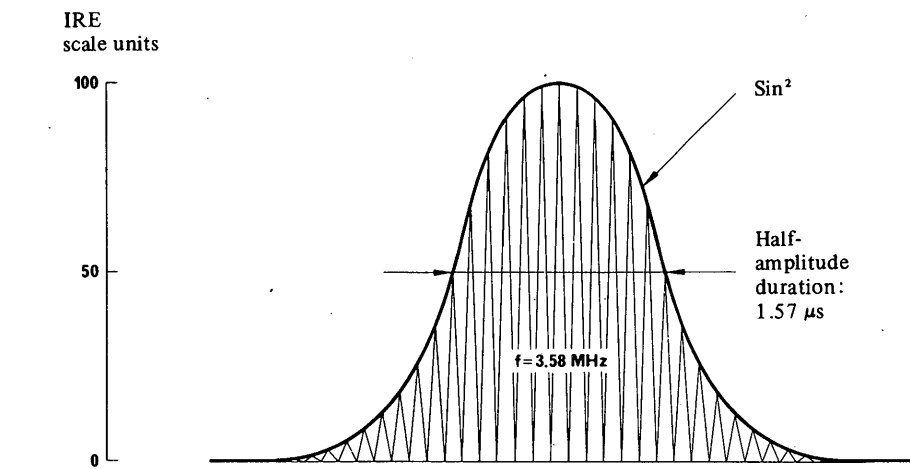


FIGURE 14 – Signal F for 525-line systems

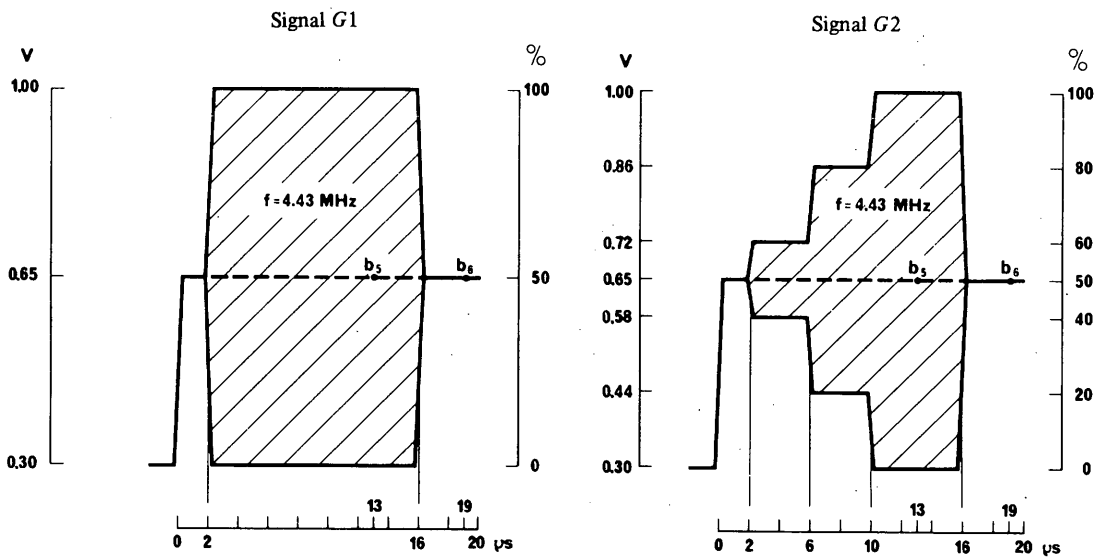


FIGURE 15 – Signal G for 625-line systems

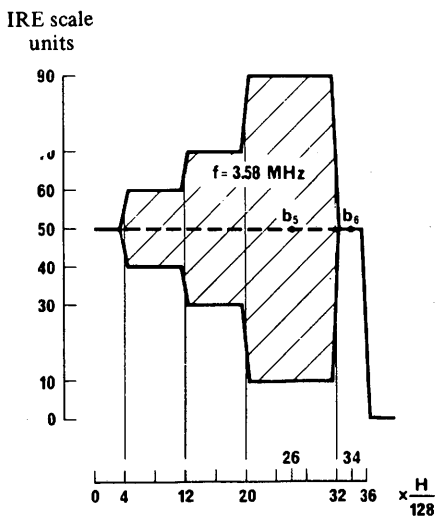


FIGURE 16 – Signal G for 525-line systems

ANNEX II TO PART C

DESIGN OF FILTERS USED FOR MEASUREMENTS

1. Low-pass filter for use in noise measurements

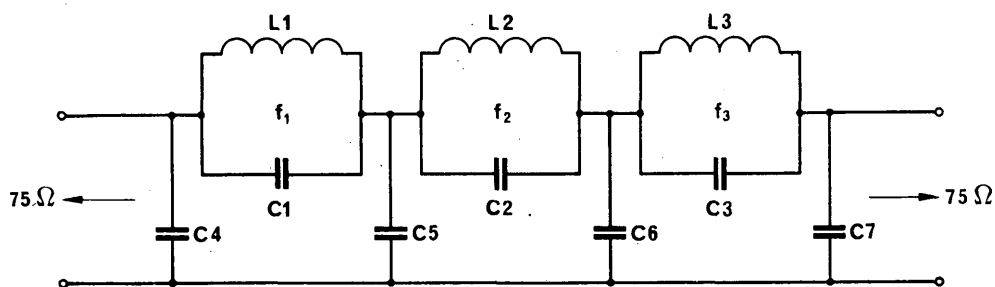


FIGURE 17 – Low-pass filter diagram

TABLE OF VALUES

Component	Multistandard value ($f_c = 5$ MHz)	Tolerance
C1	100	Note 2
C2	545	
C3	390	
C4	428	
C5	563	
C6	463	
C7	259	
L1	2.88	Note 3
L2	1.54	
L3	1.72	
f_1	9.408	
f_2	5.506	
f_3	6.145	

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

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

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.

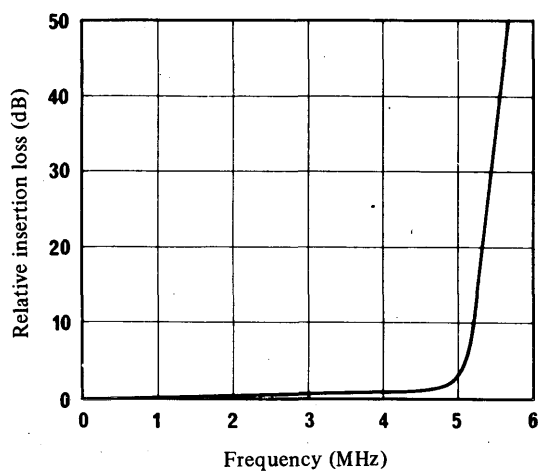


FIGURE 18 – Low-pass filter characteristic

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.

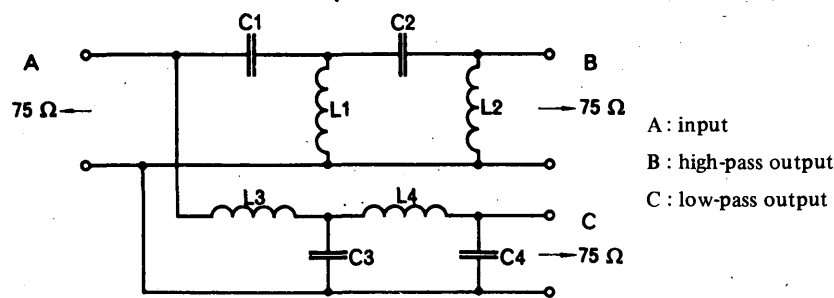


FIGURE 19 – Combined filter design diagram

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.

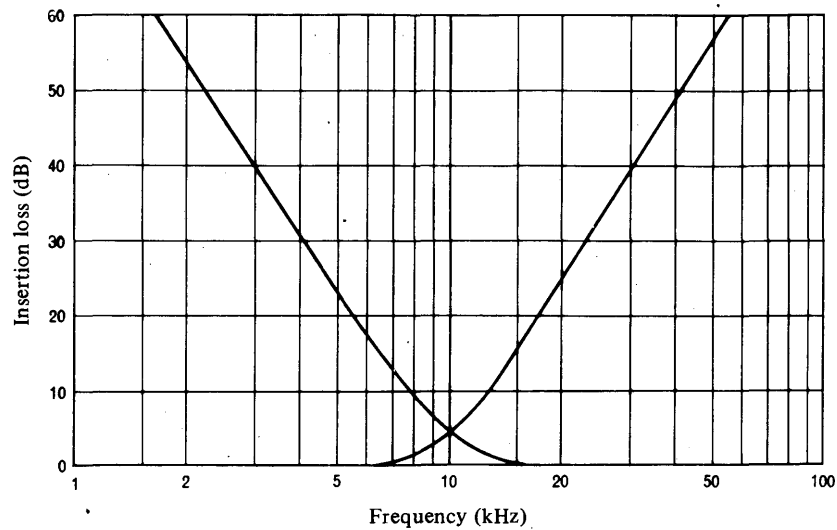


FIGURE 20 – Combined filter characteristic

3. Unified weighting network for random noise

3.1 Network configuration

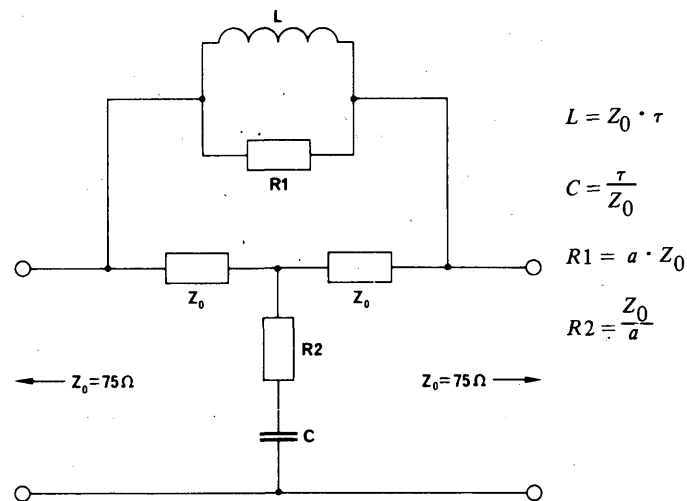


FIGURE 21 – Network diagram

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:

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

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

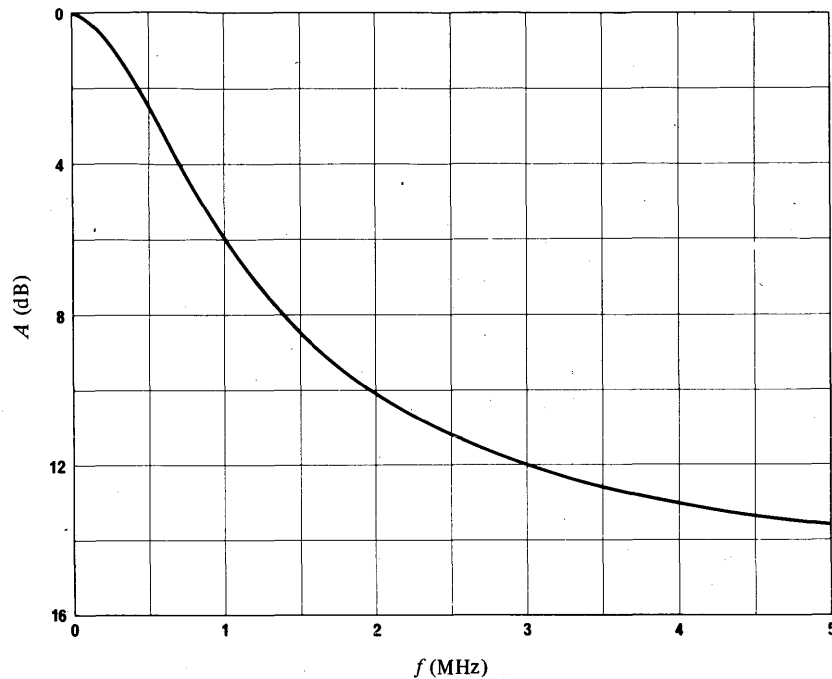


FIGURE 22 – Unified weighting characteristic

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

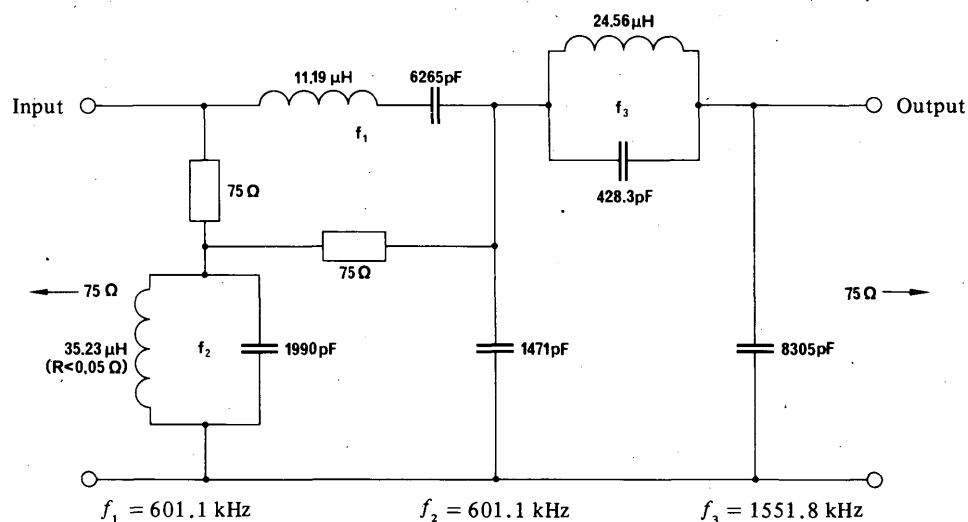


FIGURE 23 – Non-constant resistance network diagram

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.

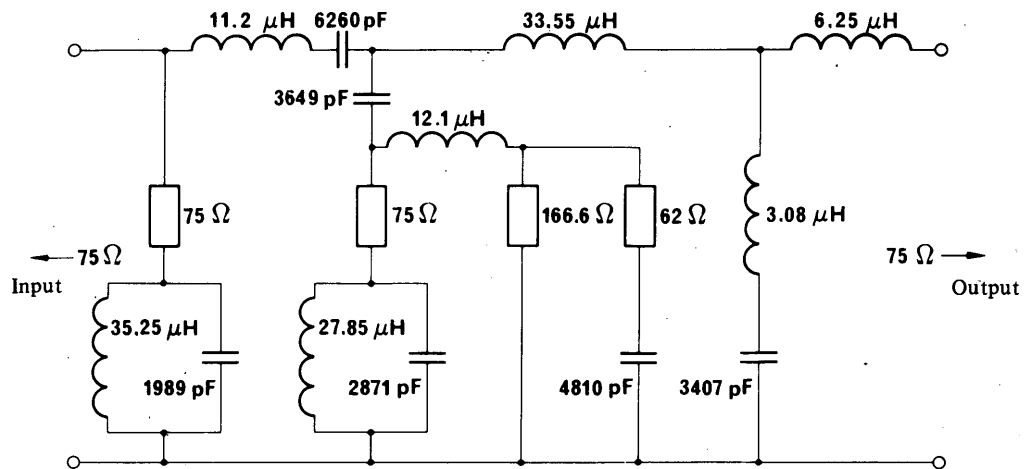
4.2 *Constant resistance form*

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.

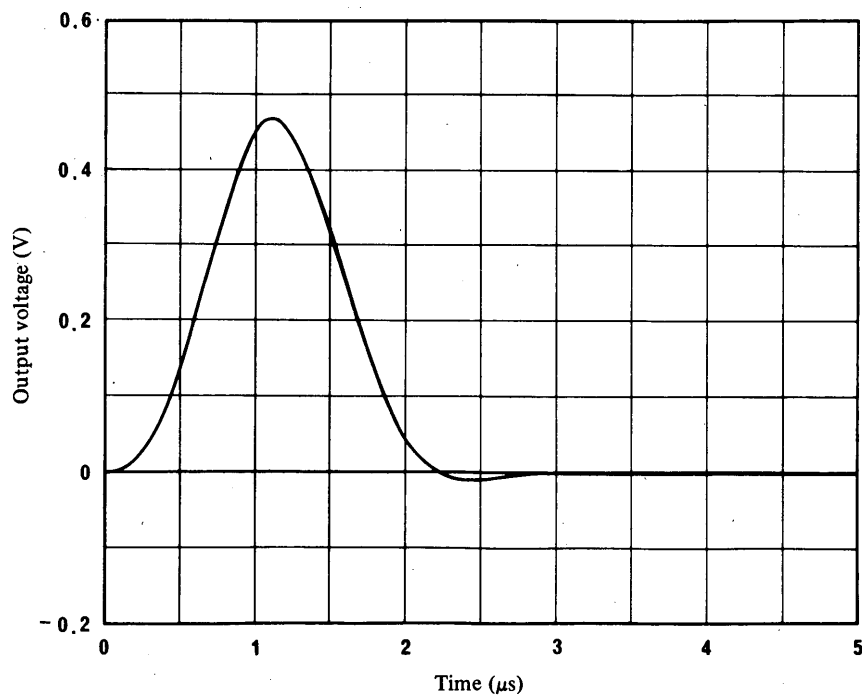
4.3 *Step response of staircase differentiating network*

FIGURE 25 – Transient response of the network

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

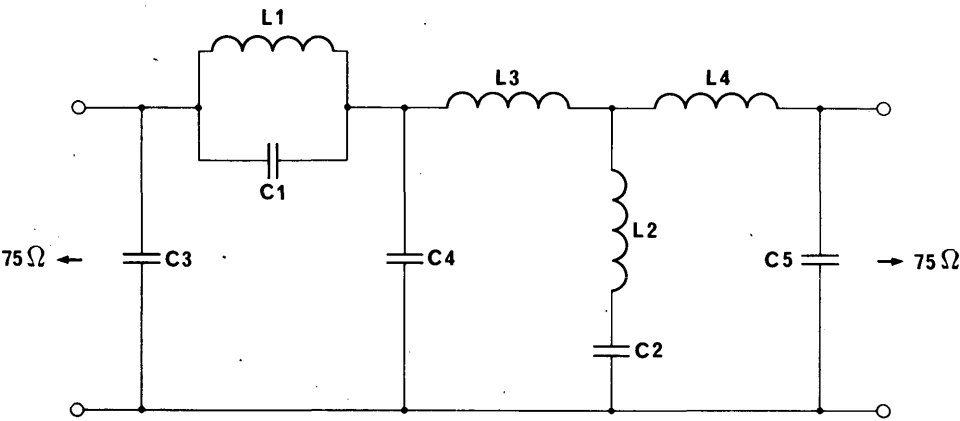


FIGURE 26 – Thomson filter diagram

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

2.1 Measurements which are different with insertion test signals

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

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

The amplitude *L* of the luminance bar is defined as the difference in level between the mid-point of the luminance bar (*b*₂ in Figs. 27 and 28) and a specified reference point (*b*₁ in Figs. 27 and 28).

Continuous random noise (§ C.3.2.1)

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

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

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

Luminance non-linearity (§ C.3.4.1.1)

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

Line-time waveform distortion (§ C.3.5.1.3)

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

The magnitude of the bar-top distortion is the maximum departure in the level in the interval between *b*₄ and *b*₃ (Figs. 27 and 28) expressed as a percentage of the bar amplitude.

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

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

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

after the half-amplitude point of the trailing edge of the luminance bar and the level at the reference point (*b*₁ in Figs. 27 and 28) expressed as a percentage of the luminance bar amplitude.

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

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

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

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

2.3 Measurements which are identical with insertion test signals

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

2.4 Measurements which are impracticable with insertion test signals

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

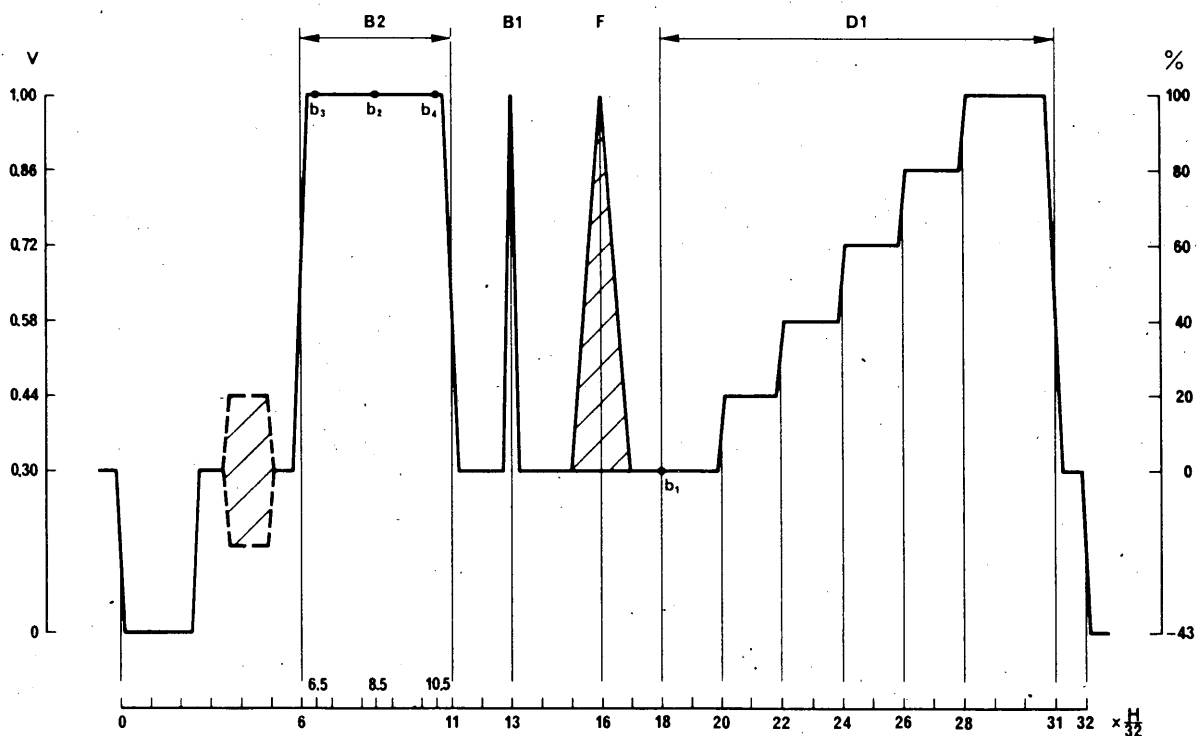


FIGURE 27 – Line 17 for 625-line systems

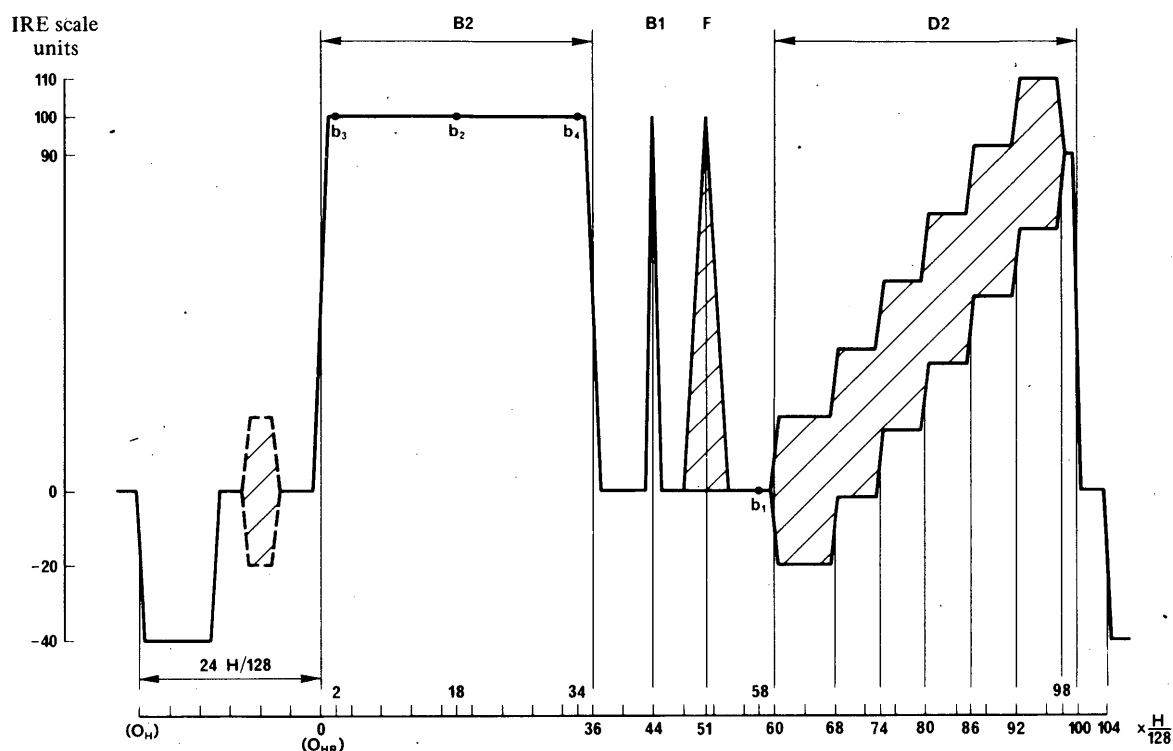


FIGURE 28 – Line 17/field 1 for 525-line systems

REFERENCES

CCIR Documents

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

BIBLIOGRAPHY

CCIR Documents

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

ANNEX IV TO PART C

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

1. Introduction

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

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

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

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

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

2. Routine-test method

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

2.1 $2T$ pulse response

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

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

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

2.2 $2T$ pulse/bar ratio

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

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

2.3 T pulse response

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

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

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

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

3. Acceptance-test method

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

Let the time series representing the filtered impulse response be:

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

and assume that this has already been normalized so that $B_0 = 1$; let the serial product of $B(rT)$ and the series $[\frac{1}{2}, 1, \frac{1}{2}]$ be

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

where:

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

then:

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

$$K1 \geq \left| \frac{C_r}{C_0} \right| \quad \text{for} \quad r \leq -8 \quad \text{and} \quad r \geq +8$$

and:

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

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

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

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

REFERENCES

- LEWIS, N. W. [1954] Waveform responses of television links. *Proc. IEE*, Vol. 101, Part III, 258-270.
 MACDIARMID, I. F. [1959] Waveform distortion in television links. *POOEJ*, Vol. 52, 108-114 and 188-195.
 SIOCOS, C. A. and CHOUINARD, G. [June, 1979] Subjective impairment units in relation with oscilloscope graticules for evaluating short-time linear waveform distortion of the luminance signal in 525-line television. *IEEE Trans. Broadcasting*, Vol. BC-25, 2, 63-71.

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

D.1 Introduction

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

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

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

D.2 Objectives and tolerances at points of video interconnection

D.2.1 Nominal impedance

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

D.2.2 Return loss

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

D.2.3 Non-useful d.c. component

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

D.2.4 Nominal signal amplitude

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

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

D.3 Transmission performance objectives and tolerances

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

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

D.3.1 *Insertion gain*

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

D.3.1.1 *Variations in the insertion gain*

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

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

D.3.2 *Noise*

D.3.2.1 *Continuous random noise*

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

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

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

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

D.3.2.2 *Low-frequency noise*

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

D.3.2.3 *Periodic noise*

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

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

D.3.2.4 *Impulsive noise*

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

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

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

D.3.4 *Non-linear distortion*

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

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

D.3.4.1 *Picture signal*

D.3.4.1.1 *Luminance signal*

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

D.3.4.1.2 *Chrominance signal*

For System M, the following figures apply:

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

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

Figures for other systems require further study.

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

Differential gain

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

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

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

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

Differential phase

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

For 3.58 MHz: x or y or $x + y$: 5°

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

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

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

The distortion should not exceed $\pm 3\%$ 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 $100 \pm 12\%$, corresponding to $K_{(P/B)} = 3\%$.

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

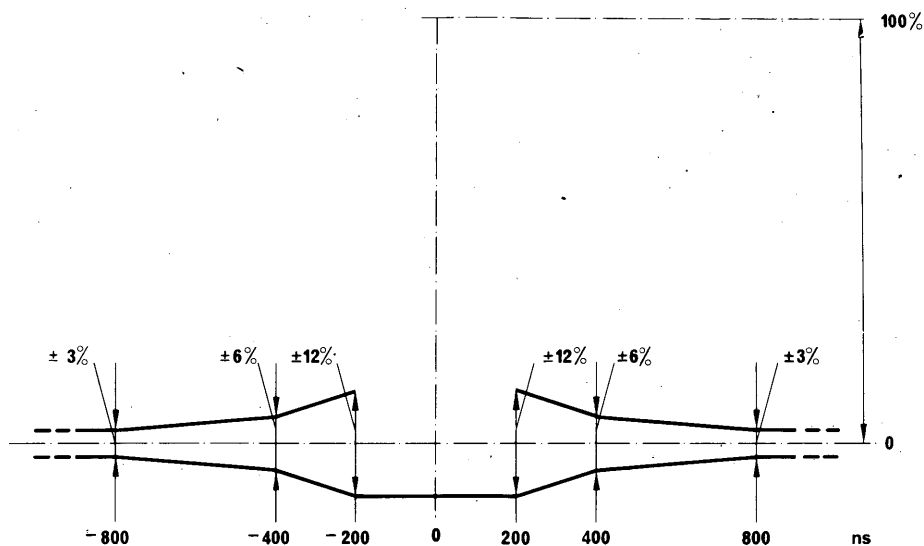


FIGURE 29a – Mask for response to test signal B1 (625-lines)
(Half-amplitude duration 200 ns)

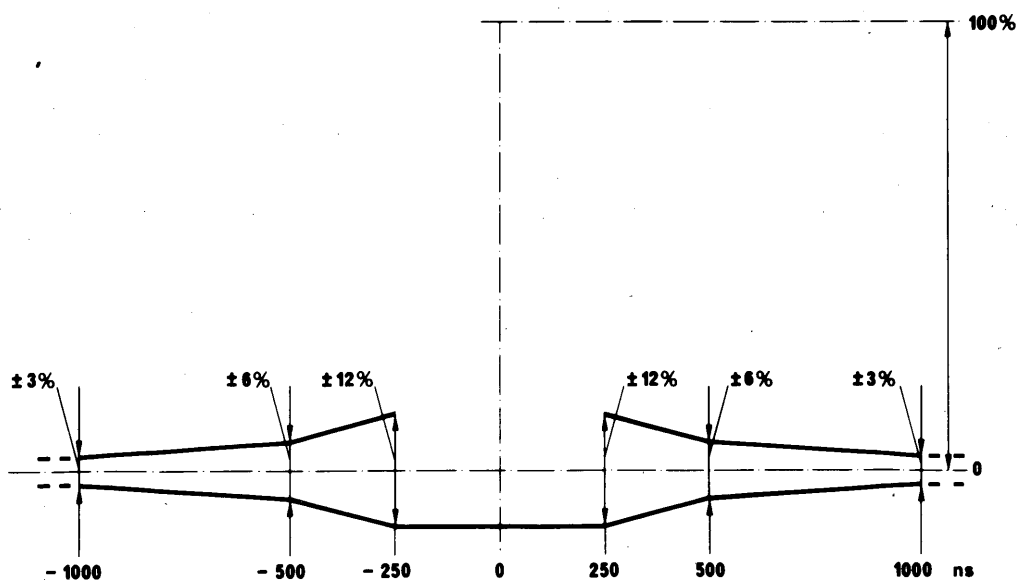


FIGURE 29b – Mask for response to test signal B1 (525-lines)
(Half-amplitude duration: 250 ns)

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

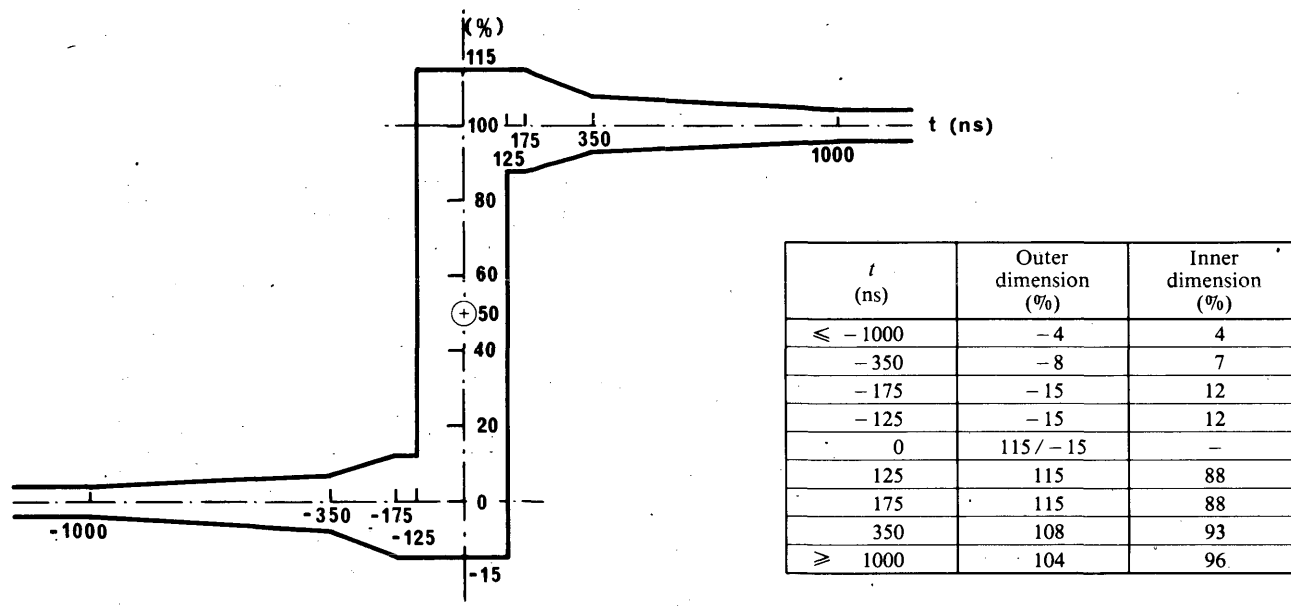


FIGURE 29c — Mask for response to B2 or B3 test signals (525-lines for Japan and Canada only)
(Time of rise: approx. 125 ns)

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

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

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

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

TABLE II

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

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

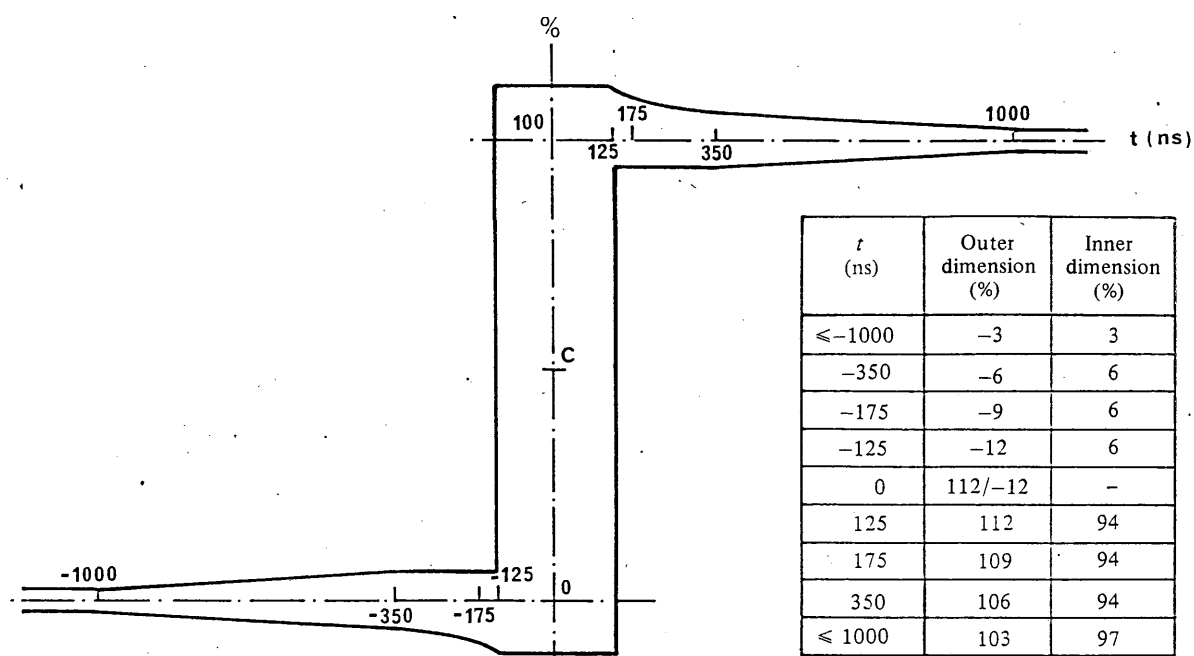


FIGURE 29d – Mask for response to *B2* or *B3* test signals
(525-lines for the United States of America only)
(Time of rise: approximately 125 ns)

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

D.3.5.2 Chrominance waveform distortion

Refer to § C.3.5.2.

D.3.5.3 Chrominance-luminance inequalities

D.3.5.3.1 Gain inequality

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

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

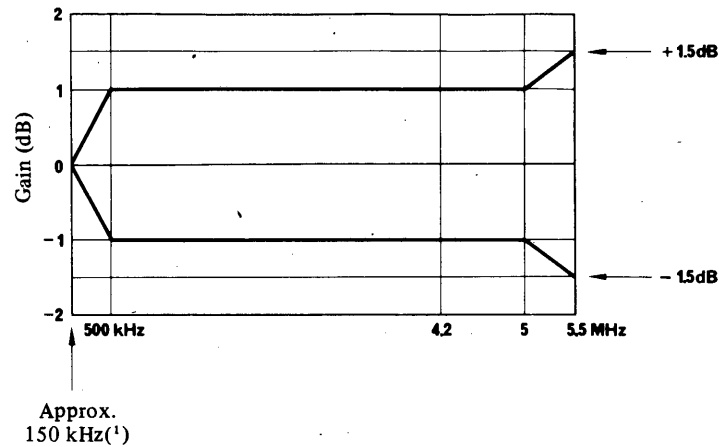


FIGURE 30 – Limits for gain/frequency characteristic

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

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

D.3.5.4.2 Delay

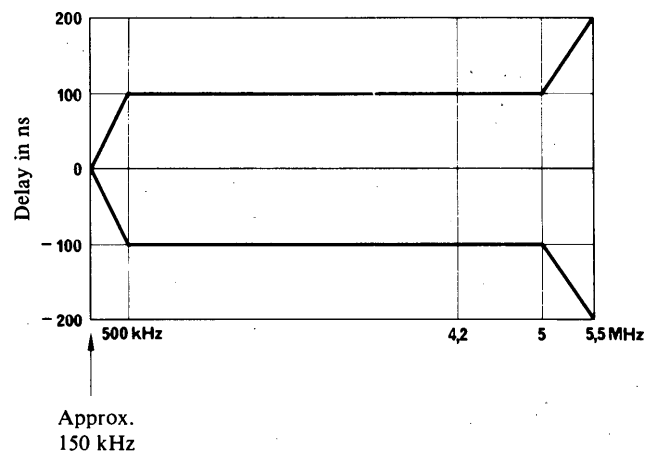


FIGURE 31 – Limits for delay/frequency response characteristic

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

REFERENCES

IEEE [1979] Video signal transmission measurement of linear waveform distortion. IEEE Standard 511-1979.

SIOCOS, C. A. and CHOUINARD, G. [June, 1979] Subjective impairment units in relation with oscilloscope gratitudes 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 III and IV provide only indications of the probable performance but for differential phase and gain, and luminance-chrominance gain inequality, such values may be considered accurate enough for practical applications. These values should be used with caution when studying the design of equipment because the laws of addition are not precisely known for every type of impairment.

E.2 Laws of addition

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

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

n : number of video-to-video sections,

L : length of the circuit,

l : 2500 km.

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

TABLE III

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

TABLE IV

§ of Part B	Characteristic	D_3 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	<i>Low-frequency noise</i>	dB	No law	—
3.2.3	<i>Periodic noise</i> Power-supply hum Single frequency	$\left\{ \begin{array}{l} \text{Noise} \\ \text{voltage} \end{array} \right.$	$\begin{array}{c} 2 \\ 2 \end{array}$	$\begin{array}{c} 2, 7 \\ 3 \end{array}$
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	$\begin{array}{c} \% \\ \% \\ \text{Degrees} \\ \% \\ \% \\ \text{Degrees} \end{array}$	$\begin{array}{c} 3/2 \\ 3/2 \\ 3/2 \\ 2 \\ 3/2 \text{ or } 2 \\ 3/2 \text{ or } 2 \end{array}$	$\begin{array}{c} — \\ — \\ — \\ — \\ 9 \\ 9 \end{array}$
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> Long-time waveform distortion: Field-time Line-time Short-time waveform distortion: Routine test method Acceptance test method	$\begin{array}{c} \% \\ \% \\ K_{(P/B)}, \% \\ K_{(2T)}, \% \\ \text{or } S, \% \end{array}$	$\begin{array}{c} 1 \\ 2 \\ 2 \\ 3/2 \\ 3/2 \end{array}$	$\begin{array}{c} \\ \\ 10 \\ 10 \\ 10 \end{array}$
3.5.3	<i>Chrominance-luminance inequalities</i> Gain inequality Delay inequality	$\begin{array}{c} \% \\ \text{ns} \end{array}$	$\begin{array}{c} 2 \\ 2 \end{array}$	$\begin{array}{c} 5 \\ 5 \end{array}$
3.5.4	<i>Steady state characteristics</i> Gain/frequency Delay frequency	$\begin{array}{c} \text{dB} \\ \mu\text{s} \end{array}$	$\begin{array}{c} 3/2 \\ 3/2 \end{array}$	$\begin{array}{c} 6 \\ 6 \end{array}$

⁽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 $h = 3/2$.

Note 10. — See Part C, Annex IV.

E.2.2 *Law pertaining to circuit configuration*

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

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

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

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

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

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

E.2.3 *Law pertaining to circuit length*

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

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

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

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

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

E.2.4 *Tables and 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 V for the difference value S .

– Addition of two distortions:

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

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

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

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

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

TABLE V

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

TABLE VI

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

REFERENCES

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

CCIR Documents

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

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

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

BIBLIOGRAPHY

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

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 1

§ 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 V		
2.4	Nominal signal amplitude	1 V		
3.1	Insertion gain	0 ± 0.25 dB	0 ± 0.3 dB	0 ± 0.5 dB
3.1.1	Insertion gain variation (1 s)	± 0.1 dB	± 0.2 dB	± 0.3 dB
3.1.1	Insertion gain variation (1 h)	± 0.25 dB	± 0.3 dB	± 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	± 50 ns	± 60 ns	± 90 ns
3.5.4.1	Gain/frequency characteristic (0.15 - 6 MHz)	± 0.5 dB	± 0.5 dB (4)	± 1.0 dB (4)
3.5.4.2	Delay/frequency characteristic (0.15 - 6 MHz)	± 50 ns	± 50 ns (4)	± 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.

REFERENCES

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

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.

* Strictly speaking the broadcasting path.

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.

REPORT 1090*

**USE OF TRANSPORTABLE TRANSMITTING EARTH STATIONS
FOR THE TRANSMISSION OF TELEVISION OUTSIDE
BROADCASTS OVER SATELLITES**

(Study Programme 13G/CMTT)

(1986)

1. Introduction

The establishment of national and international contribution links for television outside broadcasts using terrestrial circuits often involves significant problems with regard to circuit access, circuit reconfiguration and alignment. In difficult cases, substantial equipment and human resources are required and the viability of an outside broadcast from a site which is remote from normal points of signal origination, is often determined by the time required to establish the circuit.

Many of these problems can often be overcome by using transportable transmitting earth stations which may be quickly transported to the outside broadcast site and set up in an up-link configuration to access an available satellite transponder which is used by a satellite receive earth station in the fixed-satellite service.

The use of this concept introduces a need for standards on transmission performance objectives and the determination of the requirements for operation and command.

2. Transportable transmitting earth station basic characteristics

The transportable transmitting earth station should be compact, robust and easily set up and to reduce set-up difficulties, the antenna diameter should ideally be 2 to 3 m. This size suggests the earth station should operate in the 10 to 15 GHz band which is particularly well suited for this function.

The station should be equipped to accept standard video and audio signals as described in the various texts of the CCIR.

3. Specific needs for operation and command

3.1 Functions required

The initial assumption is that the transportable earth station may be operated from any point of the satellite up-link coverage area.

The power applied to the earth-station antenna should be adjustable to take account of its geographical location, the constraints imposed by the space sector operating characteristics, and to compensate for degradation due to rain attenuation.

To provide operational flexibility, the earth station should be equipped to access any of the available satellite transponders.

Although the main purpose of the transmitting earth station will be to transmit television signals (video and sound) to the satellite, a receiving capability is also required in order to:

- facilitate satellite acquisition;
- check the up link and satellite circuit by establishing a loop back to the transmit earth station;
- communicate with the receive earth station with a dedicated circuit either by terrestrial means or by satellite.

* This Report should be brought to the attention of Study Group 4.

3.2 Operating requirements

The station requires the following minimum measurement capabilities in order to verify:

- the frequency of transmission;
- the frequency deviation, including energy dispersal and modulation sense;
- the e.i.r.p.

To ensure correct operation, the station should have the capability of transmitting test signals (video and audio) to enable the receiving site to measure all necessary parameters (e.g. those described in Recommendation 567).

Deviation, frequency and antenna alignment should be adjustable as required by the satellite terminal control.

To maintain picture quality, the radiated power should be controllable by the satellite terminal control or the National Technical Control Centre (CNCT).

4. Overall performance objectives

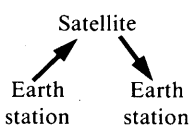
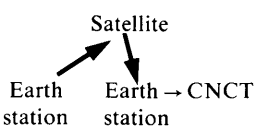
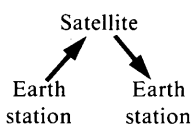
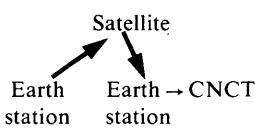
As a result of the relative simplicity of satellite circuitry, overall performance for most parameters is dictated by the earth-station equipment. With the exception of the noise parameter, experience to date indicates that performance is as good as, or in some cases better than, that outlined in Recommendation 567 for the video component and the various Recommendations for the sound component. In these cases, the deviation of the carrier has a direct impact on the results.

Noise, however, depends upon both antenna size and e.i.r.p. Recommendation 567 provides an objective for the weighted video signal-to-noise (S/N) ratio of 53 dB for the hypothetical reference circuit in the fixed-satellite service. Report 965 indicates that with the inclusion of a terrestrial link between the CNCT and earth stations, the S/N ratio is reduced to 51 dB. However, an S/N ratio of 51 dB is acceptable for the satellite circuit objective in the case of outside broadcast transmissions with a transportable transmitting earth station. This is based on the fact that the transmitting earth station will generally be located very close to the signal source and the performance of the receive end link to the CNCT will generally be much better than 53 dB.

Operations in France using satellites in the 10 to 15 GHz band, with a transmit antenna diameter size of 2.4 m, indicate that with clear sky conditions within an up-link coverage area as large as West Europe or Australia, the S/N ratio can be met with an e.i.r.p. of 73 dBW (see Table I). Further details can be found in [CCIR, 1982-86a and b].

This value of radiated power is typical for meeting a video S/N ratio of 51 dB. However, it may need to be modified depending on the noise contribution from the down link.

TABLE I

Satellite e.i.r.p. (dBW)	Trans- mitted e.i.r.p. (dBW)	S/N ratio (dB) received at large receiving antenna ($G/T = 39$ dB)		S/N ratio (dB) received at small receiving antenna ($G/T = 27$ dB)	
					
45 (limited coverage)	69	53	52	48	48
	69.7	53	52		
	75.5	57	55		
39 (global coverage)	76	55 to 58 ⁽¹⁾	54 to 57 ⁽¹⁾	49 to 50 ⁽¹⁾	49 to 50 ⁽¹⁾

⁽¹⁾ Values obtained over a long period, using several receiving stations in Europe.

5. Conclusion

This Report describes the basic requirements for the use of transportable transmitting earth stations for outside broadcast applications. Compromises between quality objectives and operational constraints which allow this procedure to produce acceptable results are proposed.

REFERENCES

CCIR Documents

[1982-86]: a. CMTT/178 (France); b. CMTT/179 (France).

RECOMMENDATION 568

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

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

(1978)

The CCIR,

CONSIDERING

- (a) that results of studies made by various administrations have shown the need for weighting the measured noise so that, in principle, a single value of signal-to-weighted noise ratio would result in the same subjective opinion value irrespective of the spectral distribution of the noise;
- (b) that such studies have also shown that the various television standards have different sensitivities to random noise, and consequently various noise weighting characteristics have been produced by many administrations;
- (c) that the acceptance of a single weighting characteristic for application to the hypothetical reference circuit for all television systems is necessarily based on a substantial compromise among the various noise weightings used by various administrations;
- (d) that it is furthermore necessary to measure random noise in a unified bandwidth in order to specify a single value of signal-to-weighted noise ratio as an objective for all television systems, so as to avoid any possible confusion when comparing results obtained at different points along an international television circuit;
- (e) that a general agreement has been obtained on a unified noise weighting network and on a single value of the signal-to-weighted noise ratio objective for application to the hypothetical reference circuit for all television systems,

UNANIMOUSLY RECOMMENDS

- 1. that the weighting characteristic and the weighting network shown in Annex II to part C of Recommendation 567 should be adopted for all international television circuits;
- 2. that the signal-to-weighted noise ratio should always refer to noise as measured in the band 0.01 to 5 MHz;
- 3. that a single value of signal-to-weighted noise ratio of 53 dB should be the objective for the 2500 km hypothetical reference circuit for 99% of any month.

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

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

ANNEX I

TREATMENT OF SPECIAL CASES

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

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

REFERENCES

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

CCIR Documents:

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

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

REPORT 637-3

SIGNAL-TO-NOISE RATIO IN TELEVISION

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

(Question 13/CMTT)

(1974-1978-1982-1986)

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	De-emphasis			Weighting			Upper frequency limit
	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,

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

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].

Figure 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.

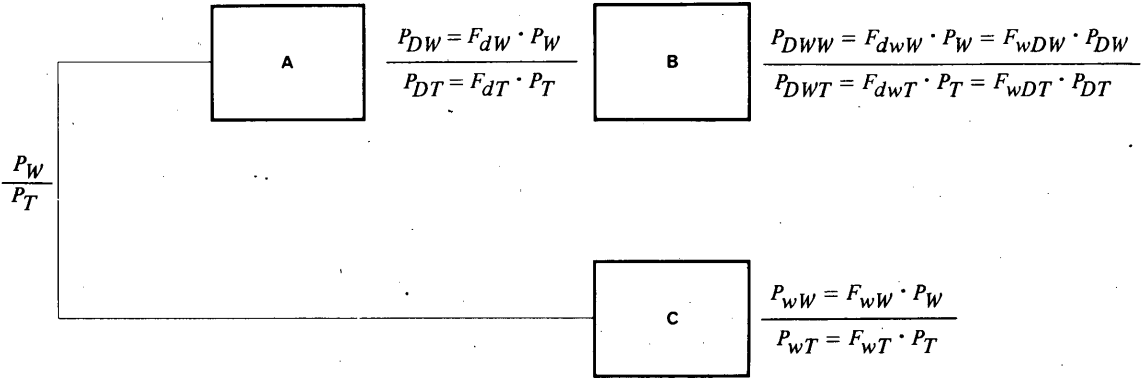


FIGURE 1 – Block diagram with conversion factors F

For definitions of terms see Table II

- A: de-emphasis network
- B: } weighting networks
- C: }

TABLE II

Measurement bandwidth (MHz)	Weighting	De-emphasis factors ⁽¹⁾ (dB)		Weighting factors (dB)				Weighting plus de-emphasis factors (dB)	
		(a) White noise F_{dw} ⁽²⁾	(b) Triangular noise F_{dT} ⁽²⁾	(c) White noise F_{ww}	(d) Triangular noise F_{wT}	(e) De-emphasised white noise F_{wDW} ⁽²⁾	(f) De-emphasised triangular noise F_{wDT} ⁽²⁾	(g) White noise F_{dwW} ⁽²⁾	(h) Triangular noise F_{dwT} ⁽²⁾
4.2 4.2 5.0	M ⁽³⁾ Unified Unified	525-lines							
		-0.4	2.9	6.2	10.3	2.7	9.9	2.3	12.8
		-0.4	2.9	6.8	11.5	2.7	10.9	2.3	13.8
5.0 5.0 5.0/6.0 ⁽³⁾	B, C, G, H I (luminance) I (chrominance)	625-lines							
		0.0	3.1	7.4	12.2	3.3	11.7	3.3	14.8
5.0 5.0 5.0/6.0 ⁽³⁾	B, C, G, H I (luminance) I (chrominance)	-2.1	2.0	8.5	16.3	3.1	14.3	1.0	16.3
		-2.1	2.0	6.5	12.3	2.3	10.9	0.2	12.9
		-2.1	2.0	4.8	0.8	9.4	1.3	7.3	3.3
5.0 6.0 6.0/5.0 ⁽³⁾	Unified D, K, L Unified	-2.1	2.0	7.4	12.2	3.0	11.2	0.9	13.2
		-1.6	2.3	9.3	17.7	3.5	15.8	1.9	18.1
		-1.6	2.3	8.2	14.6	3.2	13.3	1.6	15.6
6.0 6.0	Unified Unified	-1.6	2.3	8.0	12.8	3.3	12.0	1.7	14.3

⁽¹⁾ For 525-line systems, the de-emphasis constants are as described for system M in Table I. For 625-line systems, the de-emphasis constants are as described in Table I for other systems.

⁽²⁾ (a)+(e) = (g)
(b)+(f) = (h).

⁽³⁾ 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 (USSR).

[1974-78]: CMTT/64 (USSR).

ANNEX I

SYSTEM I (CHROMINANCE) WEIGHTING NETWORK CHARACTERISTICS

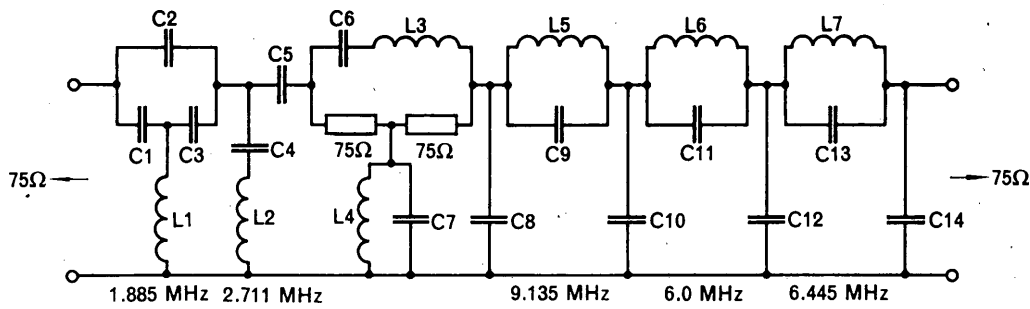


FIGURE 2 – Weighting network design diagram
(non-constant resistance)

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.

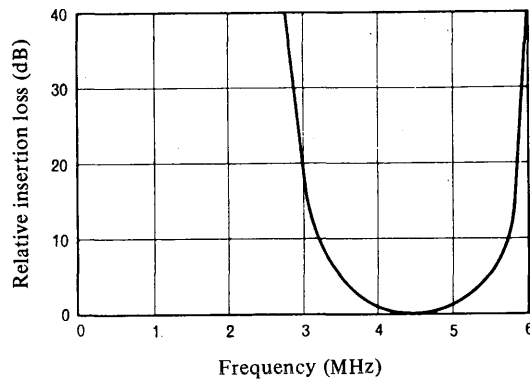


FIGURE 3 – Characteristics of the weighting network

REPORT 816-2

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

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

(1978-1982-1986)

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 and the United States of America, a variation on test signal element C, shown in Figs. 9 and 10 of Recommendation 567 has been developed. This consists of composite pulses (each based on the principle illustrated in Figs. 13 and 14 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 525-line and 625-line multiburst signals 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 these developments is contained in [CCIR, 1978-82a, 1982-86; 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.

REFERENCES

HOLDER, J. E. [April, 1977] A development of the multiburst test waveform. *Electron. Lett.*, Vol. 13, 9, 62.

CCIR Documents

[1974-78]: a. CMTT/37 (France); b. CMTT/237 (L. M. Ericsson); c. CMTT/270 (Italy); d. CMTT/273 (New Zealand); e. CMTT/69 (Canada).

[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 (USSR).

[1982-86]: CMTT/142 (United States of America).

REPORT 636-3*

LONG-TIME WAVEFORM DISTORTION IN LONG-DISTANCE TELEVISION CIRCUITS

(Study Programme 13B/CMTT)

(1974-1978-1982-1986)

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".

* This Report should be brought to the attention of Study Groups 4 and 9.

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 Macdiarmid, 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. Preservation of d.c. component in frequency modulators

In a pre-emphasized circuit, changes in the d.c. component are usually attenuated. Nevertheless, the dynamic capability of a pre-emphasized circuit can still be increased significantly by preserving the d.c. component.

In frequency modulation transmission circuits, both the voltage step and the oscillation which result from a.c. coupling cause the modulated signal to occupy a greater bandwidth.

If the d.c. component of the video signal is preserved in the frequency modulator in such a way that the centre frequency is set to the mid-point of a pre-emphasized peak white video signal, the available bandwidth is optimally utilized. This is important in satellite circuits which operate with narrow constraints of power and bandwidth [Australian Broadcasting Commission, 1983].

5. 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:

5.1 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.

5.2 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 § 5.6 for the linear component should be changed in direct proportion to the reduction of change in APL.

5.3 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).

5.4 Transient luminance non-linearity can be measured as in § 5.3 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).

5.5 Transient chrominance non-linearity can be measured by including the 3-level chrominance element *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.

5.6 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°.

5.7 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 a hypothetical reference circuit should be 40%.

REFERENCES

- AUSTRALIAN BROADCASTING COMMISSION [1983] Peak levels in a PAL signal after pre-emphasis to CCIR Recommendation 405-1. Engineering Development Report No. 127.
- COMBER, G. and MACDIARMID, I. F. [1972] Long-term response of a chain of a.c. coupled amplifiers. *Electron. and Lett.*, Vol. 8, 16, 379-380.

CCIR Documents

- [1970-74]: CMTT/181 (Germany (Federal Republic of)).
- [1978-82]: CMTT/113 (China (People's Republic of)).

BIBLIOGRAPHY

CCIR Documents

- [1970-74]: CMTT/5 (United Kingdom); CMTT/21 (USA); CMTT/40 (Germany (Federal Republic of)); CMTT/68 (Canada).
- [1974-78]: CMTT/19 (United Kingdom); CMTT/24 (OIRT).

REPORT 635-2

ALLOCATION OF TOLERANCES FOR COLOUR TELEVISION

(Study Programme 13C/CMTT)

(1974-1978-1986)

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.

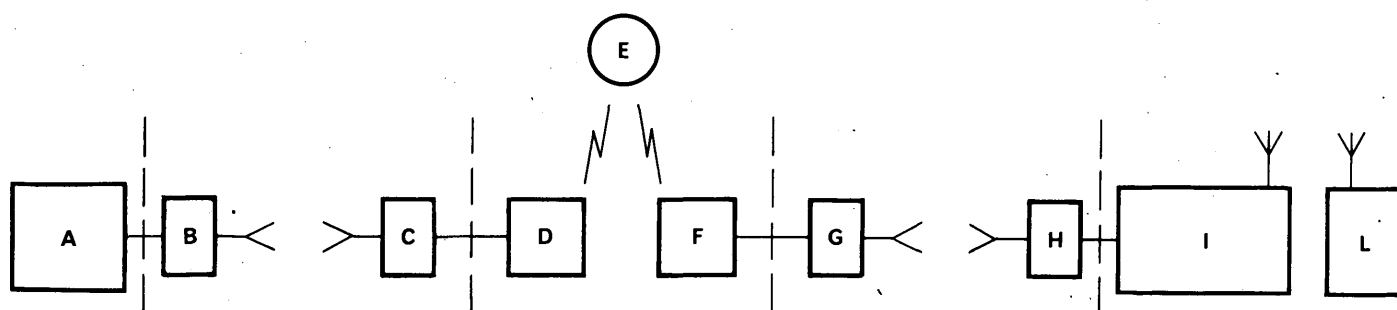


FIGURE 1 – Television system considered

- A: television studio
- B-C } 2500 km radio-relay links
- G-H }
- D: transmitting earth station
- E: satellite
- F: receiving earth station
- I: broadcasting transmitter
- L: domestic receiver

* 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, 1970-74b].

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, 1970-74c and d], 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.

* The adopted quality scale is the following:

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

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.

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.

In the People's Republic of Poland, formulae have been elaborated to calculate the overall differential gain and phase distortions of four-terminal television networks connected in cascade, based on the law of addition of instantaneous distortion values, at the defined luminance signal levels. The results of the research showed that the error obtained by the use of such a method is several times lower than the error given by calculations based on the method of addition of maximum distortion values.

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.

REFERENCES

- ALLNATT, J. W. [August, 1975] Opinion-rating model applied to television-transmission studies. *Proc. IEE*, **122**, 769-774.
- D'AMATO, P. [1974] Statistical properties of the distortion of a video chain. (In Italian). *Elettronica e Telecomunicazioni*, 233-242.
- LARI, M., MORGANTI, G. and SANTORO, G. [February, 1974] The statistical addition of distortion in transmission systems. *EBU Rev. Tech.*, **143**.
- PODEMSKI, A. [1973] Analysis of different methods in the addition of differential phase and gain of colour TV signals. (In Polish). *Rozprawy Elektrotechniczne*, **19**, 561-575.
- CCIR Documents*
- [1970-74]: a. CMTT/180 (Germany (Federal Republic of)); b. CMTT/58 (Italy); c. CMTT/56 (Italy); d. CMTT/57 (Italy).
- [1974-78]: CMTT/83 (USSR).

BIBLIOGRAPHY

- MACDIARMID, I. F. [1968] Performance requirements of links for colour television. IBC (London). IEE Conference Publication No. 46, 6, 8, 1 to 4.
- PODEMSKI, A. [1968] Summation of differential gain and phase distortions at the chrominance subcarrier frequency in circuit with static non-linearity. *Radiodif.-Télév.* (OIRT), **6**.
- PODEMSKI, A. [1974] Obliczanie wzmocnienia różnicowego połączonych kaskadowo urządzeń telewizyjnych (Calculation of differential gain in cascade-connected TV equipments). *Prace Instytutu Łączności*, **1**.
- PODEMSKI, A. [September, 1980] Addition des distorsions de phase et de gain différentielles des signaux de télévision en couleur. Document prepared for the conference "Dni Elektryki Polskiej we Francji" held by the Société des électriciens, des électroniciens et des radioélectriciens, Paris, France.

CCIR Documents

[1970-74]: CMTT/17 (USA); CMTT/63 (Italy); CMTT/64 (Italy).

REPORT 966-1 *

TRANSMISSION PERFORMANCE OBJECTIVES FOR TELEVISION STANDARDS CONVERTERS EMPLOYING FIELD STORES

(Study Programme 13F/CMTT)

(1982-1986)

In those circumstances where it is possible without affecting operation and performance, it is desirable to remove standards converters from a circuit to facilitate the overall assessment of its performance.

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 to determine what new parameters are required to provide a comprehensive set of performance specifications and what effects standards converters have on existing parameters described in Recommendation 567 in terms of the laws of addition defined for analogue circuits.

REFERENCES

CCIR Documents

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

ANNEX I

PRACTICAL PERFORMANCE OF TELEVISION STANDARDS CONVERTERS

1. In Japan

Television standards converters employing random access memories with the capacity to store three fields and which are capable of converting between 525/60 monochrome and 625/50 monochrome standards, and between NTSC, PAL and SECAM systems, are being operated at earth stations. Table I shows the specified major performance characteristics.

This is given as a reference for studying the relevant performance objectives for standards converters. It should be noted that the actual performances of these television standards converters are better than those given in Table I; also that some values differ from the design objectives for the hypothetical reference circuit given in Part D of Recommendation 567 due to the particular characteristics of standards converters.

It should also be noted that the measuring methods indicated in Recommendation 567 are not necessarily appropriate for these performance objectives.

2. In the United Kingdom

Four-field television standards converters, capable of converting between 625/50 and 525/60 monochrome standards, and from any one of the PAL, NTSC and SECAM systems to any other, are being operated in the United Kingdom and elsewhere. Table II shows the specified major performance characteristics.

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

TABLE I – Major performance characteristics of television standards converters used in Japan

Item		Specified values	Remarks
Signal-to-noise ratio	Continuous random noise	≥ 50 dB	Unweighted value
	Periodic noise (0 to 1 kHz)	≥ 50 dB	
	Periodic noise (1 kHz to f_c)	≥ 30 dB	
	Impulsive noise	≥ 30 dB	
Non-linear distortion	Colour saturation	Within $\pm 3\%$	Measured by using a vectorscope
	Phase error	Within $\pm 3^\circ$	
Linear distortion	Field-time	Within $\pm 1\%$	
	Line-time	Within $\pm 3\%$	
	Short-time (2 T pulse-bar)	Within $100^{+6\%}_{-10\%}$	
	Short-time (2 T pulse response)	Within 2%	
	Steady-state gain/frequency characteristics	See Fig. 1	
Resolution	Horizontal resolution	≥ 330 lines	280 lines in case of SECAM
	Vertical resolution	≥ 330 lines	

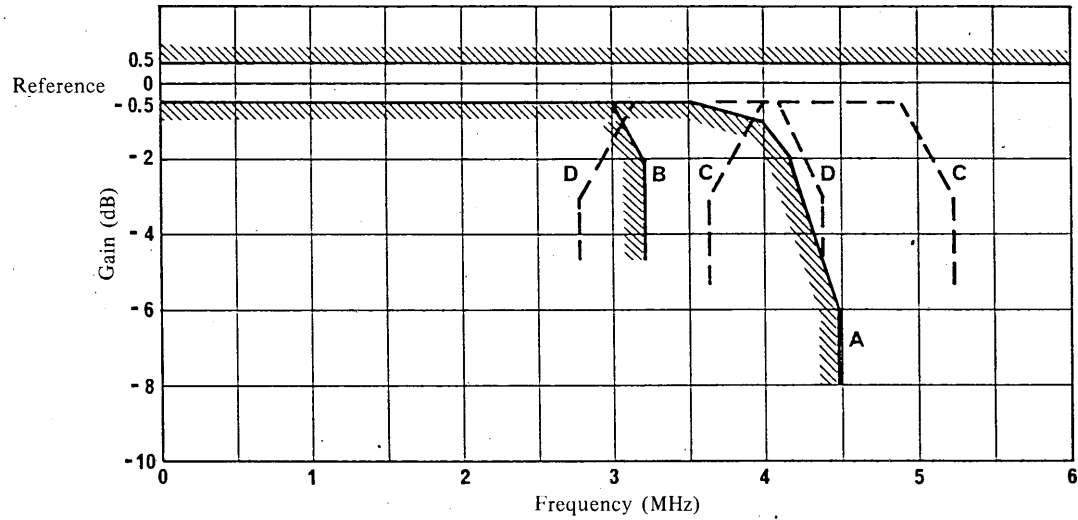


FIGURE 1 – Steady-state gain/frequency characteristics

 Luminance
 Chrominance

Conversion mode:

- A: PAL → NTSC, NTSC → NTSC, NTSC → PAL, NTSC → SECAM, PAL → SECAM, PAL → PAL
- B: SECAM → NTSC, SECAM → PAL, SECAM → SECAM
- C: SECAM → PAL, SECAM → SECAM, PAL → SECAM, PAL → PAL, NTSC → PAL, NTSC → SECAM
- D: SECAM → NTSC, PAL → NTSC, NTSC → NTSC

Note 1. – In the case where the input standard is NTSC or PAL, the characteristics depend on comb filter characteristics with the base clip circuit being excluded.

Note 2. – In the case where the output standard is SECAM, the characteristics are measured by excluding the rejection filter of 4.286 MHz.

TABLE II — Major performance characteristics of television standards converters used in the United Kingdom

Item		Specified values	Remarks
Signal-to-noise ratio	Unweighted 0 to 6 MHz Weighted	> 50 dB > 56 dB	} Total output noise figure
Non-linear distortion	Colour saturation Phase error	Within $\pm 2\%$ Within $\pm 2^\circ$	
Linear distortion	Field-time Line-time Short-time (2T pulse-bar) Short-time (2T pulse response) Steady-state gain/frequency characteristics	Within $\pm 1\%$ Within $\pm 3\%$ Within $100 \pm 5\%$ Within 2% See Fig. 2	

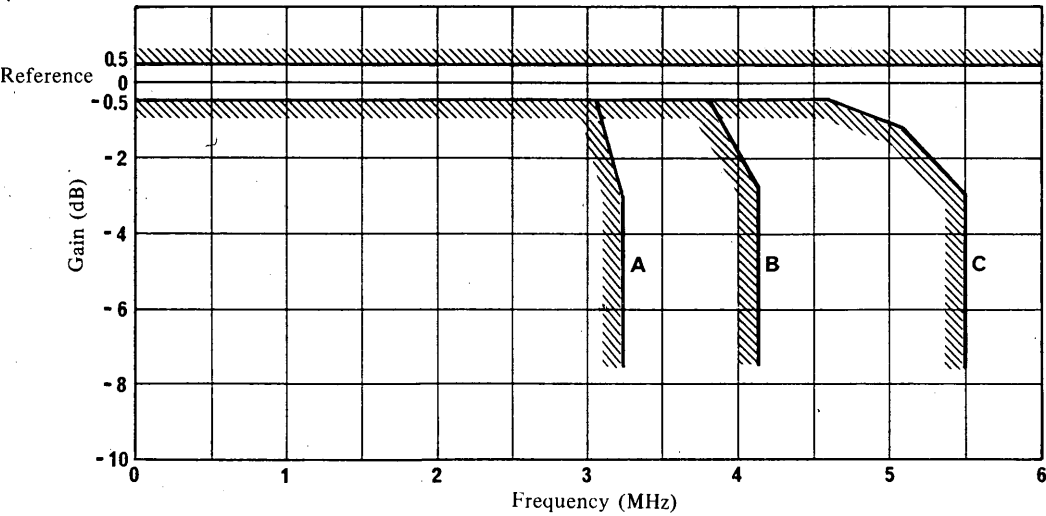


FIGURE 2 — Steady-state gain/frequency characteristics (luminance only)

Conversion mode:

- A: SECAM → PAL, SECAM → NTSC, SECAM → SECAM
- B: NTSC → NTSC, NTSC → PAL, NTSC → SECAM, PAL → NTSC
- C: PAL → PAL, PAL → SECAM

REPORT 1091

**PICTURE SIGNAL SCRAMBLING TECHNIQUES FOR ENSURING PRIVACY
IN ANALOGUE TELEVISION TRANSMISSIONS**

(Question 23/CMTT)

(1986)

1. Introduction

Under Question 23/CMTT the CCIR decided to study technical methods for ensuring privacy in long-distance international television transmission.

This Report is restricted to the description of processing techniques for scrambling the picture signal, excluding other important questions concerned with access control, which will have to be reported elsewhere.

2. Requirements and constraints

The criteria which should be taken into account when specifying the scrambling system are as follows.

2.1 *Intelligibility*

The scrambling process must ensure privacy of the picture signal. This means that the picture in its scrambled form is unintelligible, and is in a form that is unusable for broadcasting.

2.2 *Security*

The scrambling process must be such as to prevent any unauthorized person from descrambling the signal by analysing the waveform of the scrambled video signal (for example with electronic correlators).

2.3 *Quality*

The signal processing must not alter perceptibly the picture quality. In practice, it should be possible to assess the residual distortions by means of conventional measuring methods. If this is not the case, specific assessment methods will have to be defined.

Scrambling methods would probably be implemented with digital techniques to comply with the quality criteria. The following parameters will have to be specified:

- the number of quantization levels in the A/D and D/A converters;
- the sampling frequency;
- the input and output video filters;
- the other specific features of digital processing (linearity, residual clock, etc.).

2.4 *Compatibility of equipment*

The scrambling equipment must be able to process and restore the input signals, even when these signals are impaired by distortions, the limits of which will have to be specified.

The scrambling equipment must not suppress or disturb service signals or signals which are multiplexed with the picture signal such as:

- sync. pulses and colour synchronizing bursts;
- sound and data signals;
- test signals.

This requirement means that the scrambling process must be limited to the active part of the lines.

2.5 *Compatibility of the scrambled signal*

As far as possible, it should be as easy to handle a scrambled signal as a non-scrambled signal. This compatibility should apply especially to the following operations:

- transmission on terrestrial or satellite links;
- baseband switching;

- automatic equalization*;
- oscilloscope-based or automatic measuring*.

Compatibility with transmitting and switching equipments is ideally ensured if the fixed features of the signal are retained and if there is no significant change in the signal spectrum.

3. Scrambling techniques which could meet the criteria

- pseudo-random re-ordering of the lines;
- pseudo-random inversion of the line polarity;
- pseudo-random inversion of the horizontal scanning;
- pseudo-random delay of the lines with respect to sync. pulses (variable delay method);
- pseudo-random re-ordering of different parts of the line (line rotation method).

These techniques offer different levels of security. They may be more or less compatible and may have greater or lesser effect on quality.

4. Further information

[CCIR, 1982-86a] gives a brief description of the variable delay method and line rotation method applied to composite signals. It is stated that they meet the above criteria and thus enable the network to be operated in a conventional manner. The performance obtained is compatible with the design objectives and tolerances quoted in Recommendation 567.

A method using the line rotation technique applied to MAC signals is proposed in [CCIR, 1982-86b and c].

Pictures scrambled by the variable-delay method are very seriously impaired, but the subject matter remains intelligible. Pictures scrambled by the line rotation method are completely unintelligible. Pictures scrambled by the variable-delay method may be pirated (i.e. descrambled without authority) using relatively simple circuitry. Pictures scrambled by the line rotation method cannot be reliably pirated in real time with circuitry of a complexity that is practical today.

The line rotation method is tolerant of echoes and high-frequency distortions but is sensitive to line tilt and low-frequency distortions. Further information on this subject is given in [Knee, 1985].

REFERENCES

KNEE, M.J. [1985] DBS pay television: picture signal scrambling. Report BBC RD 1985/12. BBC Research Department, Engineering Division, London, UK.

CCIR Documents

[1982-86]: a. CMTT/191(Rev.1) (France); b. 10-11S/164 (EBU); c. 10-11S/182 (Germany (Federal Republic of), France).

* Correct functioning of these equipments requires that the insertion test signals (ITS) are located on the correct lines in the field-blanking interval. To meet this requirement, due to the scrambling and descrambling delays, it is necessary to delay the ITS. As a consequence, these signals will have to go through an A/D-D/A conversion process twice.

REPORT 1092

TRANSMISSION OF HIGH-DEFINITION TELEVISION SIGNALS

(Question 22/CMTT)

(1986)

1. Introduction

A study aimed at the establishment of a world-wide studio standard for high-definition television (HDTV) is being carried out in Study Group 11. The transmission of HDTV signals should be studied taking into account the conclusions derived by Study Group 11.

HDTV standards may be used in various media, and new telecommunication systems suitable for some applications of HDTV are rapidly being installed in some countries. Although such telecommunication services are not included in the CMTT terms of reference, it is desirable to exchange information with parties responsible in the CCITT and CCIR in order to establish well-aligned HDTV transmission systems.

2. Transmission signal format for HDTV

Several types of HDTV transmission signal format have been reported. These include: half-line offset PAL (HLO-PAL) (Report 801), time compression multiplexing (TCM), time compressed integration (TCI) and multiple sub-sampling encoding system (MUSE) [CCIR, 1982-86a]. The bandwidths of the signals used in these systems are shown in Table I.

In the TCM system the two colour-difference signals are compressed and in each line the luminance signal and one of the colour-difference signals are multiplexed. Each resulting signal is then transmitted line-sequentially. The proposed TCM signal bandwidth for a 1125-line system is 20 MHz [Tsuboi *et al.*, 1985].

There are several versions of TCI which depend on the choice of either a simultaneous or line-sequential arrangement of the colour-difference signals and on different compression ratios [Fujio *et al.*, 1982].

In the MUSE system, a bandwidth-reduced analogue signal is obtained by a 4-to-1 sub-sampling and a motion compensation technique which makes use of digital processing of the HDTV signal. This technique reduces the bandwidth of the 1125-line TCI signal to 8 MHz (see Report 1075).

In determining the signal format, it is necessary to consider the transmission quality level that is required to be maintained, as the required quality can vary according to the application for which the signal format is to be used. The major applications are:

- transmission of programme materials which may require further processing at a programme production centre;
- distribution of completed programmes;
- transmission of video signals by point-to-point transmission services; and
- lower capacity transmission for the case where urgent connections are required for news gathering.

As the use of digital transmission networks is increasing, the development of bit rate reduction coding systems for HDTV is becoming urgent. The bit rates of some possible formats are shown in Annex I. Ideally, the bit rates of such bit-rate reduced signals should coincide with the hierarchy levels used in digital transmission networks. Performance objectives in terms of such parameters as bit error ratios, jitter and slips based on subjective assessments of HDTV picture quality require further study.

It should be noted that a digitally sampled representation of HDTV studio signals, based on Recommendation 601, is being considered by Study Group 11.

3. Examples of HDTV signal transmission systems

Radio-relay, satellite and fibre-optic systems can be used for the transmission of HDTV signals. Examples of HDTV signal transmission systems are given in Table I.

TABLE I — *Examples of transmission systems for HDTV*

	Radio transmission		Fibre-optic transmission	
	Satellite feeder link	Terrestrial (FPU)	Subscriber loop	Trunk line
	Analogue	Analogue	Analogue	Digital
Transmission length (km)	—	32	2	2500
Carrier frequency/wavelength	14.17 GHz	38 GHz	0.89 μm	1.3 μm
Modulation scheme	FM	FM	Direct IM	PCM-IM
Transmission bandwidth/bit rate	27 MHz	110 MHz	20 MHz	400 Mbit/s
Sampling frequency	—	—	—	44.1 MHz
Coding method	—	—	—	8 bits/sample
Transmission signal format	MUSE	TCI	TCM	TCM
Baseband bandwidth (MHz)	8	20	20	20
Bandwidth after decoding (MHz)				
Y	20 ⁽¹⁾	20	20	20
C _r	7 ⁽¹⁾	7	5	5
C _b	7 ⁽¹⁾	5	5	5

⁽¹⁾ These figures apply to stationary parts of the picture.

For HDTV satellite broadcasting, it is possible to regard the system used for the feeder link, as the broadcast transmission system. The example shown in Table I uses a MUSE signal on the satellite feeder link with the intention of using the same signal for the transmission of HDTV signals (see Report 1075). For field pick-ups (FPU) using terrestrial radio links, a system using 38 GHz with an output power of 400 mW has been reported [CCIR, 1982-86b].

A fibre-optic HDTV transmission system, which makes use of the wide bandwidth and low loss properties of optical fibres, has been developed [Tsuboi *et al.*, 1985].

Digital transmission of HDTV signals with a 400 Mbit/s bit rate has been used for some transmission demonstrations in Japan [CCIR, 1982-86c].

REFERENCES

- FUJIO, T. *et al.* [June, 1982] High-definition television. NHK Tech. Monograph No. 32.
 TSUBOI, T., ASATANI, K. and MIKI, T. [January, 1985] Fiber-optic HDTV transmission technology. *SMPTE J.*, Vol. 94, 1, 4-10.

CCIR Documents

[1982-86]: a. 11/271(10-11S/135) (Japan); b. 11/20 (Japan); c. CMTT/151(11/265) (Japan).

ANNEX I

Figure 1 shows possible examples of formats for analogue component and composite signals. It also shows estimated bit rates for the digital transmission of HDTV signals. All figures have been derived for a system having 1125-line 60-fields/s, 2-to-1 interlace and 16-to-9 aspect ratio, which has been proposed in Study Group 11.

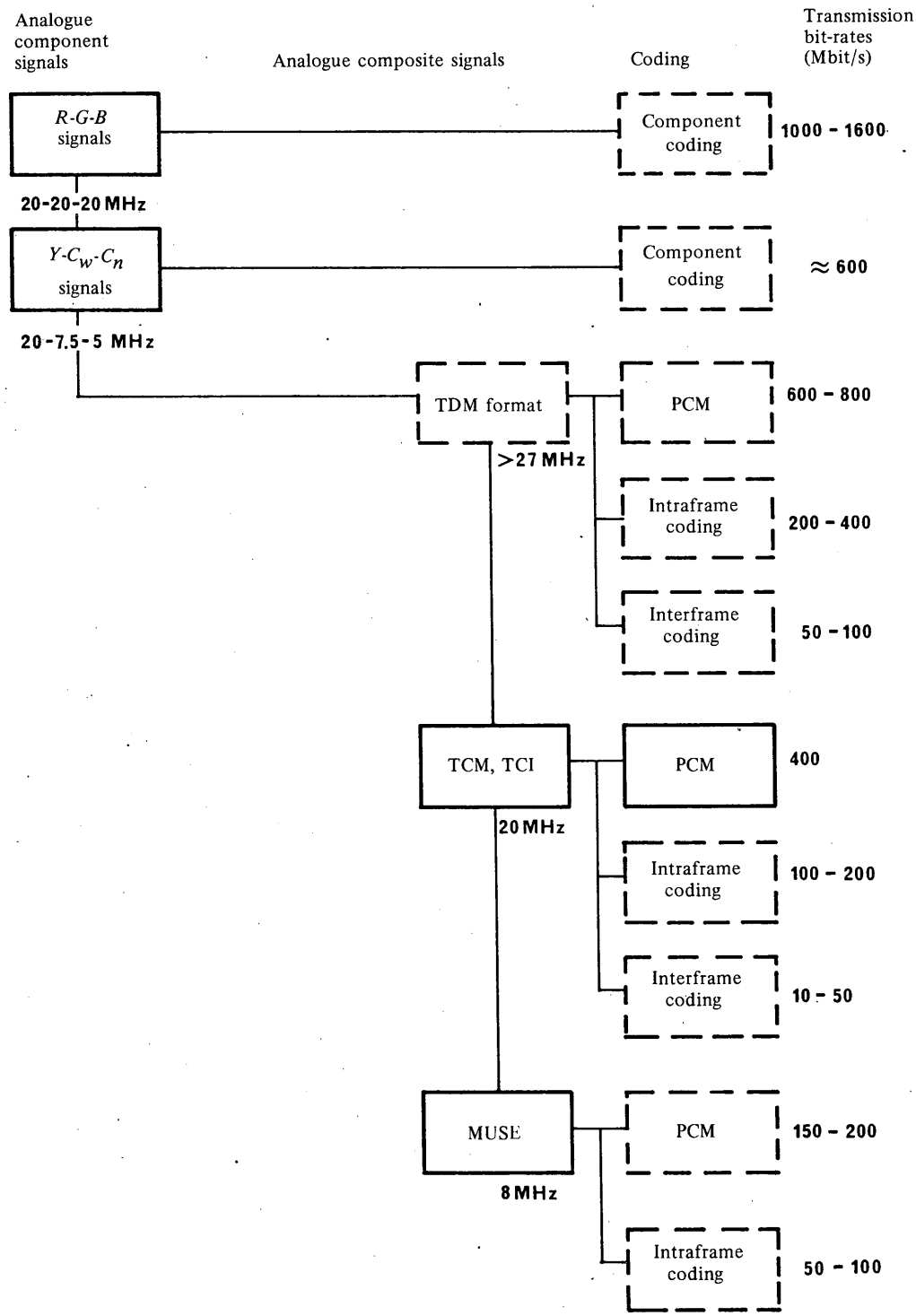


FIGURE 1 – Signal format and coding method for the 1125/60 HDTV system proposed in [CCIR, 1982-86a and b]

REFERENCES

CCIR Documents

[1982-86]: a. CMTT/157(11/268) (Japan); b. CMTT/158(11/269) (Japan).

RECOMMENDATION 603

**HYPOTHETICAL REFERENCE CHAIN FOR TELEVISION
TRANSMISSIONS OVER VERY LONG DISTANCES**

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

(1982)

The CCIR,

CONSIDERING

- (a) that, in some parts of the world, television transmission chains much longer than 2500 km are either existing or are under consideration;
- (b) that it is desirable to define one or more hypothetical reference chains corresponding to various typical connections;
- (c) that the concept of hypothetical reference chains of known composition would be very useful in progressing the studies under Study Programmes 13C/CMTT and 14A/11;
- (d) that Recommendation 567 recommends that the design objectives and tolerances for a terrestrial hypothetical reference circuit should be the same as those of a hypothetical reference circuit in the fixed-satellite service;
- (e) that, for television, inter-continental and very long distance transmission is normally by satellite,

UNANIMOUSLY RECOMMENDS

1. that it is only necessary to define one hypothetical reference chain which represents very long distance connections enabling any two points on the Earth's surface to be interconnected;
2. that the hypothetical reference chain should be defined as equivalent to five hypothetical reference circuits in cascade;
3. that the performance of the hypothetical reference chain be determined from the performance of the hypothetical reference circuit as defined in Recommendation 567 by the use of the methods recommended in Part E of that Recommendation.

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

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

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

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

BIBLIOGRAPHY

CCIR Documents

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

RECOMMENDATION 604-1 *

DIGITAL TELEVISION TRANSMISSION — GENERAL PRINCIPLES

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

(1982-1986)

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:
 - *contribution*: to carry signals to production centres where post-production processing may take place;
 - *distribution*: to carry television programmes when no further post-production processing is expected;
- (d) that it is highly desirable to facilitate the international exchange of television programmes by making digital television transmission circuits throughout the world transparent to television signals to the same or a higher degree than is possible with analogue circuits;
- (e) that when establishing a digital transmission circuit on a global basis, account should be taken of the different digital transmission hierarchies and 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.

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

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.

RECOMMENDATION 658 *

MIXED ANALOGUE-AND-DIGITAL TRANSMISSION OF ANALOGUE COMPOSITE TELEVISION SIGNALS

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

(1986)

The CCIR,

CONSIDERING

- (a) that, although television studios may progressively adopt operation based on separate component coding (e.g. according to Recommendation 601 for digital systems), existing analogue operation using composite signals will continue for a considerable time (see Note 1);
- (b) that a transitional period will occur during which analogue composite television signals will be transmitted through a circuit comprising analogue and digital sections in tandem;
- (c) that to facilitate the international transmission of television programmes during the transitional period, the methods of coding should preserve the quality of the colour television signals (e.g. NTSC, SECAM, PAL) and of the ancillary and additional signals (e.g. insertion test signals, teletext, etc.);
- (d) that the CCITT has recommended hierarchies of digital bit rates for the transmission of digital signals;
- (e) that while the use of bit-rate reduction techniques could be economically desirable, further studies are required before coding methods to comply with (c) above can be recommended,

UNANIMOUSLY RECOMMENDS

1. that when carrying a television signal presented in analogue composite form, preference should be given to all-analogue paths. However, in cases where mixed analogue-and-digital paths are unavoidable, the principles given in this Recommendation should be followed;
2. that the number of digital sections in a real circuit should be kept to the minimum;
3. that the hypothetical reference circuit for mixed analogue-and-digital transmission shall be equivalent to that defined in § A.1.2 of Recommendation 567, which applies to the case where the three sections all use analogue transmission. When one or more sections use digital transmission, the same structure should apply but modifications may be required for sections employing digital transmission (see Note 2). The signals at the input and output, and at the intermediate interconnection points of the hypothetical reference circuit, are in analogue form (see Note 3);
4. that the design objectives and tolerances specified for a hypothetical reference circuit in Recommendation 567 should also apply for mixed analogue-and-digital transmission (see Note 4);
5. that the overall quality of the hypothetical reference circuit, when considered against the objective and subjective criteria defined by Study Group 11, shall be no worse than the equivalent analogue system (see Note 5).

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

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

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

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

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

Note 5. — No specific coding parameters are defined because a practical coding method meeting the foregoing requirements for all colour television systems has not yet been described. However, administrations are reminded that they have the right to make bilateral agreements on coding parameters for a single colour television system, should this be required. If, however, such a circuit is to form part of an international television connection, it should meet the requirements specified in this Recommendation.

REPORT 646-3 *

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

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

(1974-1978-1982-1986)

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 Report.

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.

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

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.

Recommendation 658 for mixed analogue-and-digital transmission of analogue composite television signals has been approved. The discussions leading to this new Recommendation have drawn attention to the need for a number of new interrelated Recommendations dealing with the rapidly evolving developments in the requirements for the transmission of television signals. The situation in respect of the existing, proposed and required new Recommendations is summarized below.

Recommendation 604 is the first of the series of Recommendations which will lead the way into the use of all-digital techniques in the field of television transmission. Recommendation 604 requires expansion and/or the addition of further Recommendations which will fully cover the field of all-digital transmission.

Recommendation 658 deals with mixed analogue-and-digital transmission of analogue composite television signals and is intended to meet a relatively short-term requirement covering the transitional period in which analogue composite television signals must be transmitted on digital links forming part of a longer connection having analogue input and output ports. Most administrations consider that, although necessary at present, the need for this new Recommendation will diminish as time goes on.

Very rapid developments in the past few years have directed attention to the need for a new Recommendation or a group of Recommendations to deal with the transmission of multiplexed analogue component type signals. Question 24/CMTT should lead to a Recommendation on the analogue transmission of multiplexed analogue component type signals. Another Recommendation should deal with their transmission on mixed analogue and digital links. It is considered that the multiplexed analogue component Recommendations are urgently required and that they will grow in importance during the coming years, while the importance of new Recommendation 658 for mixed analogue-and-digital transmission of analogue composite signals, will correspondingly diminish.

The above-mentioned new and anticipated Recommendations pertain to interrelated functions, namely the transmission of analogue composite or component television signals over mixed analogue-and-digital circuits. Further, economic considerations dictate that, if possible, common transmission facilities should be used for these purposes. Therefore, it is imperative that the Recommendations be prepared in a manner to ensure the compatibility of the common transmission facility with all these signals.

The anticipated use of encryption to ensure privacy in long-distance transmission (see Question 23/CMTT) may require modifications or supplements to existing and projected Recommendations dealing with digital coding, because scrambling of the analogue signal prior to digital coding may comprehensively modify the signal statistics at the digital coder.

The requirements for interrelated Recommendations, the need for which has been identified, are summarized in Table I.

TABLE I — *Requirements for Recommendations on television signal transmission*

Recommendation	Topic	Requirements
567	Analogue transmission of analogue composite signals	No change
604	Digital transmission of digital component signals	New Recommendations with details
658	Mixed analogue-and-digital transmission of analogue composite television signals	More details
Anticipated	Analogue transmission of multiplexed analogue component signals	Contributions for new Recommendation
Anticipated	Mixed analogue-and-digital transmission of multiplexed analogue component signals	Contributions for new Recommendation
Anticipated	Digital coding of encrypted analogue video signals	Contributions for new Recommendation

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 Recommendations 604 and 658 on mixed analogue-and-digital transmission of analogue composite television signals. 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 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.

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 25H/11. The characteristics of the digitally encoded television signals are under study in Study Programmes 25G/11 and 25J/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.

Problems may arise when a signal of a particular television standard is transmitted via a digital codec optimized for a different television standard. Similarly, problems may arise when a scrambled signal is transmitted via a digital codec. These problems are discussed in [CCIR, 1982-86a].

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. In the case of composite signals, it might be an advantage if the codec was able to transmit sound in sync. (SIS) (see [CCIR, 1982-86a]).

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.

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. For the former case, there may be certain difficulties as mentioned in [CCIR, 1982-86a]. Japan states in [CCIR, 1982-86b] that when a bit rate reduction method is applied to video signals, signals originated in digital form such as teletext may require special treatment and may need to be transmitted separately.

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.

[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].

[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.

[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.

[CCIR, 1982-86c] gives the conclusion of a study carried out in France on a suitable organization of a digital multiplex for transmission on the third level of the CCITT digital hierarchy. This study concluded that 27 Mbit/s could be the bit rate for the coding of a video signal complying with Recommendation 601. Sound and auxiliary data were multiplexed by means of the same packet mode multiplex as systems using multiplexed analogue components. Bit rates at the third level of the CCITT digital hierarchy were obtained by the addition of tributaries at lower bit rates within the digital hierarchy (1.544 Mbit/s or 2.048 Mbit/s).

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.

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].

Several administrations have an interest in the use of bit rates in the region of 70 Mbit/s for television transmission. Such a level is not part of the CCITT hierarchy as at present defined, but the CMTT has requested that arrangements should be made by the CCITT to ensure that customer access at a level between the present third and fourth hierarchical levels is feasible and economic, because of its potential importance for television. The choice of the new access level should be the subject of cooperation between the CCITT and the CMTT, to ensure that the level is optimized for broadband video signals as well as for other ISDN services. Account should also be taken of the need to exchange digital television signals between regions having different digital hierarchies. CCITT Recommendation G.802 makes recommendations for international interworking between networks using different digital hierarchies. These matters have been drawn to the attention of CCITT Study Group XVIII in accordance with the footnote to Annex I.

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 [CCIR, 1974-78a] and France [CCIR, 1982-86d] state that the bit rate of 34 368 kbit/s is a desirable objective for use in the European digital hierarchy.

[CCIR, 1982-86e] states that not all the bit rates currently favoured for the transmission of digital television signals appear to be compatible with those terrestrial systems which are based on 2048 kbit/s, and none with those based on 1544 kbit/s, also none of the bit rates are directly compatible with communication satellite systems as they are operated at this time. The situation could be eased, for regional satellites in Europe, Australasia and parts of Africa at any rate, by increasing the maximum possible bit rate through the 40 MHz transponders from 60 Mbit/s to about 70 Mbit/s, and that through the 80 MHz transponders from 120 Mbit/s to the CCITT fourth order level of 139 264 kbit/s. There are some indications that these levels may be achievable in the future with existing plant. In order to assist in the implementation of Recommendation 604, administrations should be urged to investigate incompatibility problems with a view to eliminating, or at any rate minimizing, the requirement for recoding at earth-stations, with its consequent penalties of increased cost, inefficient use of bandwidth and loss of operational flexibility, together with the possible added hazard of increased impairment to the television signal.

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. It is important that such constraints should be minimized and that the possibilities offered for television are not unduly restricted by the requirements of other services such as telephony. A contribution is given in the following paragraph, but further information on these subjects is needed urgently by the CMTT as discussed in [CCIR, 1978-82g].

[CCIR, 1982-86f] describes a multiplexer which combines two signals, each having a bit rate of about 70 Mbit/s, to produce a signal at the fourth order hierarchical level, namely 139 264 kbit/s. Each 70 Mbit/s tributary may contain either one digital television signal or two signals of 34 368 kbit/s. The frame alignment signal and the net bit rate of the two tributaries are adjusted to the requirements of both telephony and television. According to the same principle, it may be possible for digital television signals to be adapted to other hierarchical bit rates. The insertion of a frame alignment signal at the interface between the television studio and the transmission system would considerably facilitate the transmission of television signals in the integrated services digital network (ISDN).

[CCIR, 1982-86g] describes a practical system for transmitting a composite PAL vision signal, two associated high-quality sound signals and other high-quality sound signals and telephone channels into 68.736 Mbit/s (2×34.368 Mbit/s). Two such signals can be combined into 139.264 Mbit/s by a modified commercial multiplexer designed for multiplexing four signals at 34.368 Mbit/s into 139.264 Mbit/s.

[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 25G/11.

[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. [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.

Studies under Study Programme 25H/11 on the requirements for pre- and post-filtering the signals for the 4:2:2 coding standard of Recommendation 601 are reported in Report 962. The filter specifications resulting from these studies are given in that Report and in a new Annex to Recommendation 601.

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 [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.

A study on PCM codecs using 8 bits per sample [CCIR, 1982-86h; Devereux, 1982] draws attention to the importance of the performance of both the basic coder/decoder and the associated circuitry in determining the performance of a cascaded chain of codecs. Of particular importance are:

- pre- and post-filters;
- blanking-level stabilization;
- quantization distortion, particularly on signals with high rate-of-rise (or fall);
- timing jitter on clock pulses;
- line-time non-linearity.

In a well-designed codec where these factors have been taken into account, quantization noise is likely to be the dominant impairment, and this is shown to be about 2 dB worse than the theoretical value in the codecs tested. A cascaded connection of 8 codecs is shown to increase the quantization noise by approximately 9 dB, as theory would predict.

Experimental results concerning the cascading of NTSC PCM codecs are given in [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 cascaded codecs is less than 6. For more than 8 cascaded codecs, 9 bits quantizing is considered desirable. Also, in [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.

These are important factors in establishing performance objectives for hypothetical reference circuits and chains as discussed in § 6 of this Report.

A practical system for the digital transmission of two composite video signals over an optical fibre link is described in [CCIR, 1982-86i]. Each signal was sampled at a frequency of 13.9264 MHz, one tenth of the fourth level of the CCITT bit rate hierarchy, and converted into PCM form with 8 bits/sample.

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 [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, 1978a; 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: [CCIR, 1974-78c] describes a technique for sampling the PAL signal at a frequency precisely equal to $2 f_{sc}$. Rossi [1978] proposes a similar method for NTSC where again the sampling frequency is $2 f_{sc}$, the samples being derived from original $4 f_{sc}$ samples. A phase shift of 180° every alternate line is necessary in order to permit the removal of alias components. This technique is used in the system described in [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.

[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]. [CCIR, 1978-82p] 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.

[Sawada and Kotera, 1978a] describes a 32 Mbits composite intrafield DPCM coding system for NTSC, which employs two-dimensional prediction and dual word length coding (entropy coding). A 28 M/bytes intrafield DPCM coding system for electronic news-gathering applications is described in [Rossi, 1980].

It is claimed in [CCIR, 1978-82q] that 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 using intrafield coding. In such a case, interfield coding and/or interframe coding techniques will be effective alternative means. These techniques are also very effective for attaining better picture quality at a bit rate greater than 32 Mbit/s. In [CCIR, 1978-82r] 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. An experimental interfield NTSC codec, with $2 f_{sc}$ sampling frequency, sub-sampled from $4 f_{sc}$ sampling frequency, is described in [Yoshino and Ohya, 1980]. By using 5 bits/sample, and removing the blanking intervals, a data rate of 32 Mbit/s can be achieved.

[Hatori *et al.*, 1979] describes 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. [Koga *et al.*, 1979] also 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.

An experimental interframe NTSC codec, with $3 f_{sc}$ sampling frequency and movement compensated prediction, is described in [Ninomiya and Ohtsuka, 1982]. It is suggested that with such a codec, a colour television signal, together with audio and error correction data, could be transmitted at 16 Mbit/s.

Subjective test results of the effects of up to 5 NTSC DPCM codecs connected in cascade are given in [Murakami *et al.*, 1982]. Various moving picture test sequences were used which were typical of broadcast television scenes and included still pictures, scenes with violent movement, scenes with abrupt changes (cuts) and scenes with smooth changes (camera panning and zooming). The test sequences also covered a range of levels of contrast and average brightness, and differed in the amount of detail and degree of colour saturation. It was concluded that cascading DPCM codecs of this type is not advisable, although the connection of 2 codecs in cascade may be acceptable. However, [CCIR, 1982-86g] describes a bit-rate reduction system that was designed to be used in cascade. It uses sub-Nyquist sampling and three-dimensional DPCM to reduce the bit rate of a composite PAL signal to 53.2 Mbit/s. No change in impairment could be seen between pictures processed through one codec and pictures processed through three codecs.

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.

The encoding parameters for the 4:2:2 digital studio standard belonging to an "extensible family of compatible digital coding standards" were adopted in Recommendation 601. The parameters for the other members are still to be defined. It is stated, however, that sampling frequencies related to the members should be easily convertible from one to the other. Additional information is contained in Report 962.

In [CCIR, 1982-86j] an attempt is made to derive, from the 4:2:2 standard, sets of lower sampling frequencies suitable for other applications, e.g. electronic news gathering, transmission, etc., which take into account the requirement mentioned above.

On the other hand, the higher levels of the digital transmission hierarchies which have been defined by the CCITT, are suitable for digital transmission of television signals. It is shown in [CCIR, 1982-86j] how various source coding procedures and bit-rate reduction techniques can be used to adapt the various members of the extensible family of compatible digital coding standards, mentioned in Recommendation 601, to suitable levels of the transmission hierarchy based on 2048 kbit/s. In the interests of progressing the work of the CMTT, Study Group 11 should provide comments on the respective advantages of these different options.

Two separate applications are foreseen for the transmission of digital video signals:

- *contribution*: to carry signals to production centres where post-production processing may take place;
- *distribution*: to carry television programmes when no further post-production processing is expected.

Some general aspects on the choice of coding parameters for contribution connections at 140 Mbit/s, where post-production processing is required, are given in [CCIR, 1982-86k]. In order to maintain the processing capabilities of 4:2:2 signals the full spatial and temporal resolution should be retained.

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.

A 34 Mbit/s transmission system using these three methods but using initial sampling parameters different from those specified in Recommendation 601, is described in [Sabatier *et al.*, 1982].

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]. A phase-alternating sub-Nyquist sampling method is described in [Yoshino and Ohya, 1980].

A spatial temporal sub-sampling method is described in [CCIR, 1982-86l]. The impact of the sub-sampling process on picture quality can be minimized by the use of an appropriate sampling pattern coupled with corresponding pre-filtering and post-filtering operations.

For the special application of TV programme distribution in possible future subscriber networks based on optical fibre, a coding method is described in [CCIR, 1982-86m]. Assuming that the bit rate is not a cost-sensitive factor in the subscriber area, a net bit rate of 135 Mbit/s was chosen using the sampling frequencies of the 4:1:1 level in order to allow simple and inexpensive terminal equipment.

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 component 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.

[Wengenroth, 1982] describes a system for coding a television signal at a bit rate of 70 Mbit/s using the DPCM method (folded quantization with 5 bits/sample).

A DPCM method with an adaptive quantizer which uses variable word length coding and transmission of the residual length in the horizontal blanking interval, is described in [Zühlke, 1981; CCIR, 1982-86n]. The codec, based on these principles, operates at a bit rate of 34 Mbit/s. [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 [Pirsch, 1981].

[Murakami *et al.*, 1983] describes a codec, adaptable to the extensible family of compatible digital coding standards mentioned in Recommendations 601 and 604, which is applicable to both 525-line and 625-line systems. This codec applies motion-compensated interframe, interfield and intrafield adaptive prediction techniques. It can be used to transmit electronic news gathering programmes at 15 Mbit/s and television programmes at 30 Mbit/s. Subjective evaluation experiments were performed using test moving pictures which included still-to-violent motion, camera panning or zooming and scene changes. For these tests, analogue NTSC and PAL signals were decoded to component form. The component output of the bit-rate reduction codec recoded to NTSC and PAL was compared with the unprocessed component signal recoded to NTSC and PAL. The results show that the codec can attain satisfactory picture quality for practically all of the pictures in actual TV programmes, including still to violently moving scenes in both 525-line and 625-line TV systems [Murakami *et al.*, 1984; Matsumoto *et al.*, 1985]. As indicated in [Ninomiya and Ohtsuka, 1982], motion-compensated interframe coding appears particularly promising for component coding.

In [CCIR, 1982-86m] reference is made to a system allowing digital TV transmission via digital channels of the third hierarchical level. For bit-rate reduction, an adaptive interframe/intrafield DPCM method is used based on the 3 : 1 : 0 level. In order to adapt to a given channel bit rate ranging from 32 to 44 Mbit/s, the system operates with variable code word lengths.

A form of hybrid DPCM coding scheme, in which both PCM and DPCM samples are transmitted, is described in [Wilson and Carmen, 1982]. The coded component signals, sampled at 14 : 7 : 7 MHz, were processed to produce an output at 140 Mbit/s. During subjective tests, viewers judged the picture quality of the signal reduced to a bit rate of 140 Mbit/s to be the same as that of the original signal sampled at 14 : 7 : 7 MHz.

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-component 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]. In one experimental adaptive group coding system [Epanechnikov *et al.*, 1982], the luminance component was averaged over a group of 4×2 elements. The mean value was coded in 7 bits, 8 bits (1 bit per element) were used to transmit the sign of the deviation of the actual element luminance from this mean value and 5 bits to transmit the mean deviation modulus. The resulting bit rate of the luminance signal was 25 Mbit/s. 5 Mbit/s was reserved for the transmission of the chrominance information, and the colour-difference components were averaged over a group of 4×4 elements on two frames.

A proposal for transmission at 140 Mbit/s is described in [CCIR, 1982-86o]. By the use of folded quantization and a form of hybrid coding degradation of picture quality due to overload effects can be significantly reduced.

Entropy coding has been applied to the NTSC signal in the separate-component 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 [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, 1978-82t]. 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 in quality. Such a technique could further the evolution from mixed analogue-and-digital transmission to all-digital separate-components transmission.

[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 $4 f_{sc}$ affords the best algorithms, a close second choice is $3 f_{sc}$ phase alternating line encoding (PALE); but straight PCM encoding at $3 f_{sc}$ is a poor third choice.

However, it is claimed in [Clarke, 1981] that all performance limitations with composite PAL analogue to digital *YUV* converters 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 $3 f_{sc}$ to $4 f_{sc}$ is possible. It is further claimed that a completely reversible $3 f_{sc} \rightarrow 4 f_{sc} \rightarrow 3 f_{sc}$ 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.

[CCIR, 1982-86a] discusses some basic requirements and corresponding measuring methods to evaluate the performance of composite PAL codecs. It particularly addresses the problems of measuring the performance of analogue-to-digital and digital-to-analogue converters and methods of using conventional test signals.

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]. [CCIR, 1978-82u] 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.

[Kobayashi, 1977; CCIR, 1978-82v] indicate 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. [CCIR, 1982-86p] shows a method for classifying picture quality impairments by using psychological factor analysis, in cases where the classification procedure cannot take place *a priori*. [Hishiyama and Inoue, 1984] presents classification results on picture quality degradation factors associated with interframe coding and the physical measures to specify them.

It is very important to use moving picture sequences for the evaluation of codecs using interfield or interframe coding. Moving picture test sequences in composite NTSC form for this purpose are described in [Murakami *et al.*, 1981].

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].

Comparison among coding algorithms should be carried out by taking into account of the trade-off relation between the bit-rate reduction effect and the corresponding picture quality degradation arising from the type of signal processing employed. [Inoue and Hishiyama, 1984] gives the investigation results of signal processing employed in interframe coding algorithms.

The use of zone plate test patterns for studying the performance of digital coding equipment when using such techniques as spatial and temporal filtering, sub-sampling, interpolation etc., has been described in [Weston, 1982]. This type of test signal is most useful in the design and optimization of coding equipment and by revealing and distinguishing between certain types of impairment, it aids the selection of critical test material using real pictures for use in subjective assessments.

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 slips.

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 the cases of 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, 1974-78] 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, 1974-78], 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. Experimental results concerning the cascading of NTSC DPCM codecs are given in [Murakami *et al.*, 1982]. A summary of the results 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 (= 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?

REFERENCES

- BOYER, R. [1978] Un système de transmission numérique. Deuxième partie: Réalisation de codeurs et décodeurs MICD pour un système à 52 Meb/s. *Radiodif.-Télév.*, 51.
- CLARKE, C. K. P. [February, 1981] Digital decoding of PAL and NTSC signals using field delay comb filters and line-lock sampling. *Television Technology in the 80's*, SMPTE, Scarsdale, NY 10583. 15th Annual SMPTE Television Conference, San Francisco, CA, USA.
- DEVEREUX, V. G. [1982] Tests on eight video pcm codecs in tandem handling composite PAL and monochrome video signals. Report BBC RD 1982/19. BBC Research Department, Engineering Division, London, UK.
- EPANECHNIKOV, V. Y., SARDYKO, S. V. and TSUKKERMAN, I. I. [1982] Kodirovaniye tsvetnykh TV izobrazhenii tsifrovym potokom 30 Mbit/s (Coding of colour TV pictures at 30 Mbit/s). *Tekhnika kino i Televideniya*, 2, 37-40.
- HATORI, Y., MURAKAMI, H. and YAMAMOTO, H. [June, 1979] 30 Mbit/s codec for NTSC-CTV by interfield and intrafield adaptive prediction. IEEE International Conference on Communications (ICC '79), Boston, MA, USA.
- HATORI, Y. and YAMAMOTO, H. [April, 1979] Predictive coding for NTSC composite colour television signals based on comb-filter integration method. *Trans. Inst. Electron. Comm. Engrs. Japan*, Vol. Eb2-4.
- HEITMANN, J. [1976] Ein systemkompatibles digitales farbfemsehsignal (System-compatible digital codec for colour television signals). *Rundfunktechn. Mitt.*, Vol. 20, 6, 236-241.
- HISHIYAMA, K. and INOUE, M. [September, 1984] Physical measures for interframe coded picture quality. *Rev. Elec. Comm. Labs.*, Vol. 32, 5.
- INOUE, M. and HISHIYAMA, K. [September, 1984] Trade-off between information suppression effect and picture quality degradation with interframe coding. *Rev. Elec. Comm. Lab.*, Vol. 32, 5.
- KOBAYASHI, Y. [1977] Picture quality evaluation method for digitally encoded video signals (in Japanese). *Trans. Inst. Electron. Comm. Engrs. Japan*, Vol. J60-B, 5.
- KOGA, T., IJIMA, Y., IINUMA, K. and ISHIGURO, T. [1979] Statistical analysis of NETEC-22H system performance. IEEE International Conference on Communications (ICC '79) Boston, MA, USA.
- KRETZ, F., BOUDEVILLE, J. L. and SALLIO, P. [1978] Qualité subjective en télévision numérique. Deuxième partie: Critères objectifs de qualité et exemple d'optimisation d'un codage numérique. *Radiodif.-Télév.*, 52.
- MATSUMOTO, S., MURAKAMI, H., HATORI, Y. and YAMAMOTO, H. [1985] 15/30 Mbit/s motion-compensated interframe, interfield and intrafield adaptive prediction coding – component coding based on CCIR Recommendation 604 (in Japanese). *J. Inst. Telev. Engrs. (Japan)*, Vol. 39, 10.
- MESSERSCHMID, U. [1969] Bandbreitenreduktion bei der Fernsehübertragung durch Verringerung der Abtastfrequenz (Bandwidth reduction for television transmission by reducing the sampling frequency). *NTZ*, Vol. 22, 9, 515-521.
- MOTSCH, R. [September, 1977] Adaptive bit-rate reduction by Hadamard transform Proc. Institution of Electronic and Radio Engineers, Conference No. 37, London.
- MURAKAMI, H., HATORI, Y. and YAMAMOTO, H. [August, 1981] Subjective evaluation of interfield-intrafield adaptive prediction coding. Technical Note No. 160. KDD Research and Development Laboratories, Japan.
- MURAKAMI, H., HATORI, Y. and YAMAMOTO, H. [October, 1982] Subjective evaluation on interfield-intrafield adaptive prediction coding. *SMPTE J.*, Vol. 91, 10, 912-918.
- MURAKAMI, H., MATSUMOTO, S., HATORI, Y., YAMAMOTO, H., OSHIMA, T. and SANO, Y. [19-23 September, 1983] A 15 Mbit/s universal codec for TV signals using a median adaptive prediction coding method. Conf. Record, VIII-A-17. IEEE Sixth International Conference on Digital Satellite Communications, Phoenix, AZ, USA.
- MURAKAMI, H., MATSUMOTO, S., HATORI, Y. and YAMAMOTO, H. [1984] A 15/30 Mbit/s universal digital TV codec and its performance. International Symposium on Image Processing and its Applications.
- NINOMIYA, Y. and OHTSUKA, Y. [1982] A motion-compensated interframe coding scheme for television pictures. *IEEE Trans. Comm.*, Vol. COM-30, 1, 201-211.

- PIRSCH, P. [July, 1981] Design of DPCM quantizers for video signals using subjective tests. *IEEE Trans. Comm.*, Vol. COM-29, 7, 990-1000.
- RANQUET, A. [1978] Un système de transmission numérique. Troisième partie. Système expérimental de transmission de télévision numérique: modulation et multiplexage. *Radiodif.-Télév.*, 53.
- ROSSI, J. P. [September, 1978] Sub-Nyquist sampled PCM NTSC colour TV signal derived from four times the colour subcarrier sampled signal. International Broadcasting Symposium, London, UK.
- ROSSI, J. P. [September, 1980] A 28 Mbit/s NTSC color television system for ENG applications. International Broadcasting Convention Proceedings, IEE Conf. Publ. No. 191.
- SABATIER, J. [1977] Un système de transmission numérique. Première partie: Les paramètres du codage à réduction de débit. *Radiodif.-Télév.* 50.
- SABATIER, J., AGUILLE, G., JOLIVET, J. C., LE BLOCH, H. and RIOU, A. [1982] Système de codage pour la transmission à 34 Mbit/s d'un programme de télévision en couleur. *Commutation et transmission*, 1, 69-82.
- SARDYKO, S. V., TSUKKERMANN, I. I. [1977] Gruppovoe kodirovanie TV izobrazhenii (Group coding of TV pictures). *Tekhnika kino i televideniya*, 9.
- SAWADA, K. and KOTERA, H. [October, 1978a] 32 Mbit/s transmission of NTSC colour TV signals by composite DPCM coding. *IEEE Trans. Comm.*, Vol. COM-26, 10, 1432-1439.
- SAWADA, K. and KOTERA, H. [April, 1978b] A 32 Mbit/s component separation DPCM coding system for NTSC color TV. *IEEE Trans. Comm.*, Vol. COM-26, 4, 458-465.
- TSUKKERMANN, I. I. [1980] Izbytochnost televizionnykh soobshchenii i problema effektivnogo kodirovaniya v tsifrovom televidenii (Redundancy of TV information and the problem of effective coding in digital television). *Tekhnika sredstv svyazi* — seriya tekhnika televideniya, 6.
- TSUKKERMANN, I. I. et al. [1981] *Tsifrovoe kodirovanie televizionnykh izobrazhenii* (Digital coding of TV pictures). Radio i svyaz, Moscow, USSR.
- WENGENROTH, G. [1982] Die Codierung von Farbfernseh- und Bildfernsprechsignalen in einem digitalen optischen Teilnehmeranschlussnetz (Coding of colour television and visual telephone signals in a digital fibre-optic cable distribution network). *NTZ-Archiv.*, Vol. 4, 4.
- WESTON, M. [October, 1982] The zone plate: its principles and applications. *EBU Rev. Tech.* 195, 3-11.
- WILSON, E. [January, 1981] Bit rate reduction to 140 Mbit/s from a television component standard requiring 216-218 Mbit/s. IBA E&D Interim Report 111/81.
- WILSON, E. J. and CARMEN, P. R. [March, 1982] Bit-rate reduction for 140 Mbit/s links. *IBA Tech. Rev.*, 16.
- YOSHINO, T. and OHYA, A. [1980] Digital frame memory for still picture television receivers — PASS encoding system and applications. *SMPTE J.*, Vol. 89, 4, 257-262.
- ZÜHLKE, W. [1981] DPCM mit variabler Wortlänge für die Fernsehbildcodierung (DPCM with variable word length for television picture coding). *Nachrichtentechn. Elektronik*, Vol. 31, 4, 143-145.

CCIR Documents

- [1970-74]: a. CMTT/174 (Germany (Federal Republic of)); b. CMTT/241(Rev.1) (USA).
- [1974-78]: a. CMTT/247 (EBU); b. CMTT/218 (USA); c. CMTT/17 (United Kingdom); d. CMTT/12 (Japan); e. CMTT/232 (France); f. CMTT/55 (Germany (Federal Republic of)); g. CMTT/11 (Japan); h. CMTT/191 (Japan); i. CMTT/228 (Germany (Federal Republic of)); j. CMTT/233 (France); k. CMTT/216 (USA); l. CMTT/45(Rev.1) (USA); m. CMTT/205 (United Kingdom).
- [1978-82]: a. CMTT/18 (United Kingdom); b. CMTT/267 (France); c. CMTT/52 (Japan); d. CMTT/85 (Germany (Federal Republic of)); e. CMTT/86 (Germany (Federal Republic of)); f. CMTT/286 (EBU); g. CMTT/13 (EBU); h. CMTT/296 (Canada); i. CMTT/7 (EBU); j. CMTT/14 (USA); k. CMTT/17 (United Kingdom); l. CMTT/123 (11/128, USSR); m. CMTT/8 (EBU); n. CMTT/59 (Japan); o. CMTT/230 (Japan); p. CMTT/45 (Japan); q. CMTT/47 (Japan); r. CMTT/229 (Japan); s. CMTT/261 (Germany (Federal Republic of)); t. CMTT/31 (United Kingdom); u. CMTT/236 (Japan); v. CMTT/49 (Japan).
- [1982-86]: a. CMTT/177(Rev.1) (Denmark); b. CMTT/146 (Japan); c. CMTT/184 (France); d. CMTT/176 (France); e. CMTT/39 (United Kingdom); f. CMTT/16 (Germany (Federal Republic of)); g. CMTT/171 (United Kingdom); h. CMTT/36 (United Kingdom); i. CMTT/40 (United Kingdom); j. CMTT/15 (Germany (Federal Republic of)); k. CMTT/216 (Germany (Federal Republic of)); l. CMTT/218 (Germany (Federal Republic of)); m. CMTT/203 (Germany (Federal Republic of)); n. CMTT/51 (German Democratic Republic); o. CMTT/217 (Germany (Federal Republic of)); p. CMTT/31 (Japan).

BIBLIOGRAPHY

- GUIONNET, J. and SALLIO, P. [1978] Système expérimental de télévision numérique à 52 Mbit/s. Évaluation subjective de l'effet des erreurs de transmission. *Radiodif.-Télév.*, Vol. 4/5, 54, 1-7.

ANNEX I*

MATTERS RELATING TO DIGITAL TELEVISION TRANSMISSION
THAT THE CMTT WISHES TO DRAW TO THE
ATTENTION OF THE CCITT

In the present Report, the CMTT summarizes its ideas about the digital coding and multiplexing arrangements for transmitting digital television signals.

In choosing suitable bit rates, account has to be taken of the requirements of contribution connections which carry the signals to production centres where the signal may undergo various forms of processing, as well as the requirements of distribution connections after which no processing is done.

[CCIR, 1982-86] explains some of the differences, and it indicates that the use of the 140 Mbit/s hierarchical level is expected to satisfy all the requirements of both distribution and contribution connections. However, the use of such a high bit rate may prove to be unduly expensive, and it would present difficulties for satellite transmission. The CMTT is therefore considering the use of the third hierarchical level. So far, it is expected that such a bit rate will satisfy the needs of distribution, but there are doubts about its suitability for contribution connections. For this reason, Report 646 draws attention to the desirability of having access to a bit rate somewhere between the existing third and fourth hierarchical levels. The choice of the exact bit rate must be a matter for collaboration between the CMTT and the CCITT, and must take account of the needs of television as well as practical constraints. Consideration must also be given to the need for interworking between regions having different hierarchies.

The CMTT also hopes that, when being offered a nominal bit rate for television, there are not any significant restrictions on the bit rate that can actually be used for the television signal or on the way the television data is formatted.

REFERENCES

CCIR Documents

[1982-86]: CMTT/176 (France).

REPORT 1093

**DIGITAL TRANSMISSION OF COMPONENT-CODED TELEVISION SIGNALS
AND BIT-RATE REDUCTION TECHNIQUES**

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

(1986)

1. Introduction

The scope of this Report is to address the issues related to digital transmission of component television signals and the specifications of bit-rate reduction methods based on Recommendations 601 and 604.

The general principles of digital transmission of television signals are given in Recommendation 604, and it is noted that two different applications need to be considered:

- *contribution*: to carry signals to production centres where post-production processing may take place;
- *distribution*: to carry television programmes when no further post-production processing is expected.

The scope of this Report covers both applications.

In order that the CMTT can produce the necessary Recommendations and Reports on the coding methods to be used for both applications, an Interim Working Party (IWP CMTT/2) has been set up to consider the factors relating to the transmission of signals in digital form, and IWP 11/7 will continue to study the coding methods in respect to picture quality and processing capability. Both IWPs will collaborate closely on these aspects to ensure compatibility between the results of the studies.

* This Annex together with Annex I to Report 967, has been forwarded by the CMTT to CCITT Study Group XVIII for information and possible action.

2. Digital bit rates for transmission

The issues related to bit rates for transmission of component television signals are the subject of studies in IWP CMTT/2. The following points should be taken into consideration:

- interworking between different levels in the CCITT hierarchies;
- compatibility between different bit rates;
- existing practices for transmission over satellite circuits;
- the third hierarchical level may be the most appropriate for distribution.

3. Performance objectives

The performance objectives are governed by the requirements of the intended application (contribution, distribution) and the available bit rate for transmission.

Although further progress in bit-rate reduction techniques can be expected in the future, it seems unlikely that the quality of the pictures, coded according to the 4 : 2 : 2 standard, can be maintained at rates around the CCITT third-order hierarchical levels. At these rates, the use of sophisticated coding algorithms gives a picture quality which is better than that produced by composite encoding-decoding equipment currently in use, although inferior to that of the 4 : 2 : 2 standard.

Quality and performance objectives will be studied in IWP CMTT/2, in close collaboration with IWP 11/7. Examples of performance objectives are given in [CCIR, 1982-86a, b and c]. The performance objective given as an example in [CCIR, 1982-86a] states that the subjective quality of the pictures transmitted by the coding system to be recommended should be equivalent to, or better than, that obtained when pictures are transmitted over a typical analogue circuit whose performance conforms to the requirements for a hypothetical reference circuit as specified in Recommendation 567.

4. System specification

Table I indicates the basic elements of a specification, the details of which are for further study.

TABLE I – *Basic elements of specification*

Format of input/output video signals	The 4 : 2 : 2 member specified by Recommendation 601 (either of 525-line and 625-line system) taking into account Recommendation 656
Bit rates for transmission	To be studied by IWP CMTT/2
Pre-processing blanking signal: – element sub-sampling/ line sub-sampling; – pre/post filtering. Coding algorithm	To be studied by IWP 11/7 and IWP CMTT/2
Channel error protection	Methods of error control should be provided
Sound channel	At least two monophonic sound channels should be provided
Digital-frame configuration	To be studied by IWP CMTT/2

A practical example of a coding system is given in Fig. 1, and a description of the coding system and the system specification are described in [CCIR, 1982-86a; Murakami *et al.*, 1983; Matsumoto *et al.*, 1985].

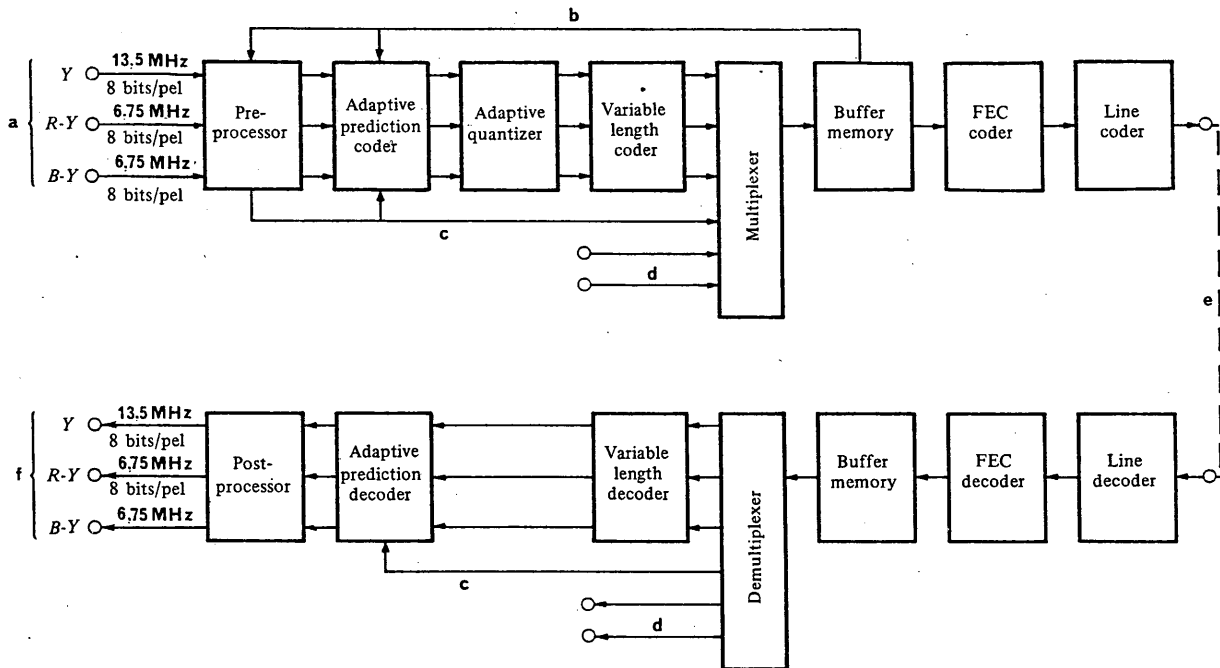


FIGURE 1 – Block diagram of the codec of [CCIR, 1982-86a]

- a : input video signals
 b : mode control
 c : amount of displacement
 d : sound signals
 e : transmission channel (30, 32, 34 and 45 Mbit/s)
 f : output video signals
 FEC: forward error correction

REFERENCES

- MATSUMOTO, H., MURAKAMI, H., HATORI, Y. and YAMAMOTO, H. [1985] 15/30 Mbit/s motion-compensated inter-frame, interfield and intrafield adaptive prediction coding – component coding based on CCIR Recommendation 604 (in Japanese). *J. Inst. Telev. Engrs. (Japan)*, Vol. 39, 10.
- MURAKAMI, H., MATSUMOTO, S., HATORI, Y., YAMAMOTO, H., OSHIMA, T. and SANO, Y. [19-23 September, 1983] A 15 Mbit/s universal codec for TV signals using a median adaptive prediction coding method. Conf. Record, VII-A-17. IEEE Sixth International Conference on Digital Satellite Communications, Phoenix, AZ, USA.

CCIR Documents

[1982-86]: a. CMTT/148 (Japan); b. CMTT/176 (France); c. CMTT/203 (Germany (Federal Republic of)).

BIBLIOGRAPHY

- BULEY, H. and STENGER, L. [April, 1985] Inter/intraframe coding of colour TV signals for transmission at the third level of digital hierarchy. *Proc. IEEE*, Special Issue on Visual Communications Systems, Vol. 73, 4.
- WILSON, E. J. and CARMEN, P. R. [March, 1982] Bit-rate reduction for 140 Mbit/s links. *IBA Tech. Rev.*, Vol. 16, 70-82.

REPORT 817-2*

**INFORMATION RECEIVED FROM CCITT STUDY GROUPS XV AND XVIII
REGARDING DIGITAL TRANSMISSION SYSTEMS**

(Questions 14/CMTT and 18/CMTT)

(1978-1982-1986)

1. Introduction

A list of CCITT Study Groups and Questions under study is given in Volume I of the CCITT Red Book. It should be noted that a number of questions relating to digital equipment (multiplex, line systems, etc.) which were previously studied in CCITT Study Group XVIII have now been assigned to CCITT Study Group XV.

There are several points of particular importance which CCITT Study Group XVIII has brought to the CMTT's attention. They are listed below.

1.1 The CMTT may wish to give guidance to the newly established Task Group on broadband aspects of ISDN on the required capacity for broadband channels transmitting digital television, and as a result has set up Interim Working Party CMTT/3.

1.2 Recommendation G.821 on error performance was considerably revised and previous CMTT references to it should be reviewed for content change.

1.3 Recommendation G.824 on the control of jitter and wander is due to be significantly revised in the near future. Any new reference to Recommendation G.824 should bear this in mind.

1.4 The CMTT may wish to examine Recommendation 502 on hypothetical reference circuits for sound-programme transmission for its compatibility with CCITT Recommendation G.801. The CMTT may also wish to advise CCITT Study Group XVIII to include, in Recommendation G.801, transmission models for digital television.

1.5 The CMTT may wish to advise Study Group XVIII on which bit rates are preferred for digital television transmission. This may include a level between 3 and 4, a level above this for high-definition television and other yet unidentified hierarchy levels.

1.6 In response to a CCITT Study Group XVIII proposal, the CMTT has drafted a new Study Programme to study broadcast-related applications for the 64 kbit/s wideband speech coding algorithm.

1.7 Recommendation G.700 shows the framework of the CCITT Study Group XVIII Series G.700, G.800 and G.900 Recommendations.

1.8 Recommendation I.110 shows the framework of the CCITT Study Group XVIII Series I Recommendations.

1.9 The CMTT should continue the special rapporteurship with CCITT Study Group XV and Study Group XVIII.

The CMTT has noted that ISDN connections are in general full duplex, that is they offer the same capabilities in both directions. However, the CMTT may require unidirectional connections or bidirectional connections with different capabilities in each direction.

2. Digital transmission models

Digital transmission network models, defined in CCITT Recommendation G.801, are hypothetical entities of a defined length and composition for use in the study of digital transmission impairments (e.g. bit errors, jitter and wander, transmission delay, availability, slip, etc.). The diversity of possible network situations requires that individual models can only represent a small portion of typical real entities. However, a limited number of such models (e.g. 2 or 3) together may be sufficiently representative to provide a useful tool upon which studies may be based.

* This Report should be brought to the attention of CCITT Study Groups XV and XVIII.

The network models, where applicable, should take account of the following features:

- physically reflect the length of the overall connection with some indication of frequency of occurrence,
- identify boundaries between switching and transmission elements,
- give no indication of the means of implementing transmission between switching elements (e.g. metallic, optical, radio media, satellite, etc.),
- describe in detail the user/network access arrangement in the local portion (i.e. customer to local exchange),
- take account of all possible usages or be independent of them,
- reflect the use of additional digital processing elements required in particular network configurations (e.g. A-μ converters, digital pads, transcoders, etc.).

2.1 Hypothetical reference connection

A digital hypothetical reference connection (HRX) is a model in which studies relating to overall performance may be conducted, thereby facilitating the formulation of standards and objectives. It is noted that the CCITT have defined HRXs that are based principally on telephony applications.

2.2 Hypothetical reference digital link

To facilitate the study of digital transmission impairments (e.g. bit errors, jitter and wander, slip, transmission delay) it is necessary to define network models comprising a combination of different types of transmission elements (e.g. transmission systems, multiplexers, demultiplexers, digital pads, transcoders). Such a model is defined as a hypothetical reference digital link (HRDL). The exact length and composition in respect of the number, type and disposition of equipments will depend on the digital impairment under study. In addition, the HRDL can be regarded as a constituent element of an HRX, thus permitting the apportionment of overall performance objectives to a shorter model. A length of 2500 km is considered as a suitable distance of an HRDL.

2.3 Hypothetical reference digital section

To accommodate the performance specification of transmission systems (i.e. digital line and radio systems) it is necessary to introduce a hypothetical reference digital section (HRDS). This model is homogeneous in that it does not include other digital equipment such as multiplexers/demultiplexers. This entity can form a constituent element of an HRDL.

Examples of HRX, HRDL and HRDS are contained in CCITT Recommendation G.801.

3. Digital hierarchies

It is noted from Recommendation G.702 that the CCITT has recommended two bit rate hierarchies.

TABLE I – Digital hierarchies

Digital hierarchy level	Hierarchical bit rates (kbit/s) for networks with the digital hierarchy based on a first level bit rate of		
	1544 kbit/s		2048 kbit/s
	64		64
1	1544		2 048
2	6312		8 448
3	32 064	44 736	34 368
4	97 728		139 264

4. Transmission performance of digital networks

4.1 Error performance

The CCITT has given in its Recommendation G.821, performance objectives for 64 kbit/s connections in the ISDN. These connections may be for voice traffic or as a "bearer channel" for data type services. The 64 kbit/s circuit-switched connection referred to, is an all-digital hypothetical reference connection (HRX) of 27 500 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 . The objectives are specified in Table II.

TABLE II — Error performance objectives for international ISDN connections

Performance classification	Objective
(a) (Degraded minutes) (Notes 1, 2)	Fewer than 10% of one-minute intervals to have a bit error ratio worse than 1×10^{-6}
(b) (Severely errored seconds) (Note 1)	Fewer than 0.2% of one-second intervals to have a bit error ratio worse than 1×10^{-3}
(c) (Errored seconds) (Note 1)	Fewer than 8% of one-second intervals to have any errors (equivalent to 92% error-free seconds)

Note 1. — The terms "degraded minutes", "severely errored seconds" and "errored seconds" are used as a convenient and concise performance objective "identifier". Their usage is not intended to imply the acceptability, or otherwise, of this level of performance.

Note 2. — The one-minute intervals mentioned in Table II and in the notes are derived by removing unavailable time and severely errored seconds from the total time and then consecutively grouping the remaining seconds into blocks of 60. The basic one-second intervals are derived from a fixed time pattern.

Definitions of available and unavailable time are given in Annex A of CCITT Recommendation G.821 as follows:

"A period of unavailable time begins when the bit error ratio (BER) in each second is worse than $1 \cdot 10^{-3}$ for a period of ten consecutive seconds. These ten seconds are considered to be unavailable time. The period of unavailable time terminates when the BER in each second is better than $1 \cdot 10^{-3}$ for a period of ten consecutive seconds. These ten seconds are considered to be available time. Excessive BER is only one of the factors contributing towards the total unavailable time."

Definitions concerning availability can be found in Recommendation G.106.

The performance objectives represent a compromise between a desire to meet service needs and a need to realize transmission systems taking into account economic and technical constraints. The performance objectives, although expressed to suit the needs of different services are intended to represent a single level of transmission quality.

The performance objective for degraded minutes as stated, is based on an averaging period of 1 min. This averaging period and the exclusion of errors occurring within severely errored seconds which occur during this one minute period (see Table II, Note 2), may allow connections with frequent burst errors to meet this particular part of the overall objective, but such events will be controlled to a certain extent by the severely errored seconds objective. However, there is some doubt as to whether the objectives are adequate for proper operation of real-time video services with relatively long holding times, and this is the subject of further study.

Work in CCITT Study Group XVIII is now concentrating on determining error performance objectives for higher bit rates.

4.2 *Slip rate objectives*

In its Recommendation G.822 the CCITT has proposed an end-to-end slip rate performance for telephony and non-telephony services based on a 64 kbit/s digital connection. The slip rate objectives are stated with reference to the hypothetical reference connection (HRX) of 27 500 km in length.

It is assumed that International Exchanges (ISC) 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, 1978-82a].

In a practical international end-to-end connection containing both international and national portions, the slip rate may significantly exceed the value computed from n plesiochronous interexchange links due to various design, environmental and operational conditions in international and national sections. These include:

- configuration of the international digital network,
- national timing control arrangements,
- wander due to extreme temperature variations,
- operational performance characteristics of various types of switches and transmission links (including diurnal variations of satellite facilities),
- temporary disturbances on transmission and synchronization links (network arrangements, protection switching, human errors, etc.).

4.3 *Jitter requirements*

CCITT Recommendation G.823 provides jitter guidelines and limits to be applied in the planning of networks and in the design of equipment which are based on the 2048 kbit/s hierarchy. Recommendation G.824 gives corresponding information for the 1544 kbit/s hierarchy.

CCITT Study Group XVIII states that 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.803, 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*

The CCITT defines ISDN as follows:

“A network evolved from the telephony ISDN 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.”

CCITT Study Group XVIII, in its coordinating role on ISDN matters, has established a new series of Recommendations, Series I, pertaining to user/network interfaces, network related aspects, architectural issues, etc. of the ISDN. Recommendations on essential ISDN characteristics in a single volume assists the user as well as the provider of the ISDN [CCIR, 1982-86a].

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

6.2 *ISDN services*

The CCITT has issued the following Recommendations that deal with ISDN services:

Recommendation I.210 describes the principles of telecommunication services supported by an ISDN. Two general classes of services are defined, bearer services and tele-services.

Recommendation I.211 specifies bearer services, that is services which provide the capability for the transmission of signals between user/network interfaces and involving only low layer attributes.

Recommendation I.212 specifies tele-services, that is services which provide the complete capability for communication between users of specified user terminals and involving low and high layer attributes.

Both bearer and tele-services are characterized by attributes and their corresponding values.

Sound programme and television signal transmission could thus be provided as both a bearer or a tele-service, depending on the attributes requested by a customer.

For services less than 2048 kbit/s, definitions for service attributes and values are contained in Recommendation I.130. CCITT Study Group XVIII is working on other Recommendations for ISDN services at bit rates greater than 2048 kbit/s.

6.3 *Available bit rate of the access to the ISDN*

In the case of access to the ISDN for broadband services, bit rates up to the primary hierarchy level are specified in the I.200 series of Recommendations.

The CCITT Study Group XVIII Task Group on broadband aspects of ISDN has tabulated [CCIR, 1982-86b] various proposals for H-channel capacities together with the main applications. The range 30 to 70 Mbit/s is being considered for the capacity of the basic broadband channel. In this range, three bit rates seem more appropriate for transmitting over existing systems of the various hierarchies, namely 30 to 34 Mbit/s, around 45 Mbit/s and 60 to 70 Mbit/s.

6.4 *ISDN user-network interface and channel types*

The CCITT has issued three new Recommendations to specify the ISDN user-network interface. Recommendation I.410 discusses general aspects and principles relating to Recommendations on ISDN user-network interfaces. Recommendation I.411 provides reference configurations for ISDN user-network interfaces. Recommendation I.412 defines limited sets of channel types and interface structures for ISDN user-network physical interfaces.

These Recommendations provide for bidirectional transmission at H0 (384 kbit/s), H11 (1536 kbit/s) and H12 (1920 kbit/s) channel signals or combinations of B (64 kbit/s) and H0 channel signals. All H channels (H0, H11, H12) appear at the ISDN primary user-network interface by means of the "primary rate user-network interface", that is 1544 kbit/s or 2048 kbit/s.

7. **Non-ISDN services**

7.1 *Guidelines for transport of service signals*

In general, with regard to bit rates available for transport of service signals, the following guidelines apply:

A.1 For the hierarchies based on 1544 kbit/s primary rate, the principle has been established that some bits in the frame should be reserved, in particular to perform end-to-end quality control of the digital paths when several digital sections in tandem are involved. An example of implementation of this principle is the 1544 kbit/s rate, in which some F-bits (see Recommendation G.704) are reserved for this purpose. This principle does not necessarily imply that there are basic restrictions on the offering of the full hierarchical bit rate. For example, at 6312 kbit/s, there is no fundamental restriction on the use of full capacity of the digital path, however, the principles above may still need to be observed.

A.2 In case of networks using 2048 kbit/s based hierarchy there is in principle no basic restriction on the use of full capacity of the digital path. However, it is recognized that compatibility with recommended frame structures at the various levels of the 2 Mbit/s hierarchy (e.g. the use of the same frame alignment pattern) could be a preferred solution since it offers the following advantages:

- use of the same framing devices for switched and non-switched applications;
- end-to-end quality control performed in a unique way by the network when the maintenance entity that terminates the service (e.g. the encoding device) does not belong to the network;
- possibility of performing additional network management functions that could be required, depending on the applications.

The preference for compatibility with recommended frame structures could be reconsidered for the applications where significant restrictions on the efficient use of the digital path capability can be identified.

7.2 *Equipment interface for sound-programme signals*

The characteristics of primary PCM multiplex equipment operating at 2048 kbit/s and offering digital access at 384 kbit/s and/or synchronous digital access at 64 kbit/s are described in Recommendation G.735. Recommendation G.737 gives the characteristics of equipment (external to PCM muldaxes) providing the same digital access.

For establishing sound-programme connections according to Recommendations J.41 and J.42, the 384 kbit/s channel described in Recommendations G.735 and G.737 would be used.

REFERENCES

CCIR Documents

[1978-82]: a. CMTT/292 (Study Group XVIII, CCITT); b. CMTT/294 (Study Group XVIII, CCITT).

[1982-86]: a. CMTT/228 (CMTT Special Rapporteur to CCITT Study Group XVIII); b. CMTT/183 (CCITT).

REPORT 967-1 *

DIGITAL TELEVISION: TRANSMISSION IMPAIRMENTS AND METHODS OF PROTECTION

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

(1982-1986)

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. System design for digital transmission of television signals

The digital transmission of television signals has some features that are different to analogue transmission. For digital transmission, the following points should be considered:

- the insertion, switching and routing of the television signals should not cause bit errors or slips;
- methods of inserting and extracting television signals in a transmission system should be economic;
- route diversity, protection channels and stand-by equipment may have to be used to ensure sufficiently high transmission reliability;
- the maximum acceptable bit error ratio should be less for television than for telephony;
- error-correction techniques will be useful when transmitting television signals after bit-rate reduction.

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

3. 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, resulting in severe impairment. Provided the demultiplexing can be achieved, the phase-locked loops usually involved in the demultiplexing process, can give a large degree of jitter reduction.

High frequency jitter on the recovered clock is reduced by the phase-locked oscillator used for clock recovery. But, as this is not the case for the data signal, timing error arises between the recovered clock and the data signal. This increases the probability of a symbol error and means that, even if the r.m.s. value of the jitter level is the same, the effect differs with jitter frequency. In general, the effect is small for low frequencies and large for high frequencies which implies that the jitter performance specification should be more stringent for high frequencies [CCIR, 1982-86a].

Residual jitter may still be present on analogue signals 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 which comprise 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. 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.

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

4. Transmission errors

4.1 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. The CMTT wishes CCITT Study Group XVIII to note the views expressed in Annex I to this Report on error ratios.

The CCITT has so far proposed performance specifications for 64 kbit/s voice traffic connections or data-type bearer channels and the figures are quoted in CCITT Recommendation G.821. [CCIR, 1978-82a] suggests that for 34 Mbit/s channels, it can be assumed that bit error ratios do not exceed $1 \cdot 10^{-4}$ for a significant time. It is hoped that the CCITT will regard an error ratio of worse than $1 \cdot 10^{-4}$ as indicating a condition of complete failure [CCIR, 1982-86b]. This is consistent with [CCIR, 1978-82b] which discusses the general difficulty of specifying error performance objectives and suggests that a long-term mean bit-error ratio of $1 \cdot 10^{-6}$ is an appropriate objective for a high bit rate, maximal length, hypothetical reference connection.

Experiments described in [CCIR, 1982-86c] have shown that service was interrupted, as judged subjectively, when the bit-error ratio exceeded $1 \cdot 10^{-3}$. Furthermore, full scale measurements showed that this level is reached for a C/N ratio measured in a 30 MHz band and equal to:

- 16.5 dB if 2 coded 34 Mbit/s programmes are transmitted simultaneously over the same radio-relay channel using 8-phase modulation (8-PSK);
- 7.5 dB if a single 34 Mbit/s programme is transmitted using minimum shift keying (MSK) modulation.

For purposes of comparison, the interruption in service occurs at around 11 dB for frequency modulation analogue transmission.

In a prolonged field trial described in [CCIR, 1982-86d], it was found that the effect of fading of an 11 GHz radio link was a sudden rise in error ratio, barely affecting television signals until complete failure occurred due to loss of demultiplexer synchronization. Measurements of degraded minutes and severely-errored seconds were compared with records of losses of synchronization but showed no correlation. For an error ratio of $1 \cdot 10^{-3}$, a much shorter averaging interval is needed, perhaps of the order of 500 μ s.

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, 1978-82c].

The parameter "mean bit-error ratio" is not in itself sufficient to specify the error-performance requirement for broadcast quality 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 could be within a mean bit-error ratio tolerance, and yet not be capable of correction by an error-protection system which 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.

4.2 *Circuit availability*

Recordings carried out on a real link using a frequency of 4 GHz [CCIR, 1982-86c] have shown that the unavailability rate varies throughout the day and is greater at night than in the day-time. The majority of breaks last less than 2 s, and on average last between 0.3 and 0.4 s.

A prolonged field trial using an 11 GHz radio link [CCIR, 1982-86d] has shown that the unavailability rate in the United Kingdom is highest at night during the winter, but during the summer there are bursts of unavailability during day-time.

4.3 *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 $1 \cdot 10^{-9}$ assuming that error events are essentially random in nature. Such an error ratio would be acceptable for the most susceptible coding method [CCIR, 1978-82c].

The provision of relatively simple error-correcting codecs could, however, reduce the $1 \cdot 10^{-9}$ error criterion for vision signals to about $1 \cdot 10^{-6}$ with an excess transmission bit rate not exceeding 10%. It is also possible that the criterion may be further relaxed to about $1 \cdot 10^{-4}$ 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 $1 \cdot 10^{-4}$ to $1 \cdot 10^{-9}$. [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, 1978-82a] 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 $1 \cdot 10^{-6}$ to $1 \cdot 10^{-8}$, depending on the system. [CCIR, 1978-82d] 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 $3 \cdot 10^{-7}$ and $1 \cdot 10^{-5}$ 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, 1978-82e] describes an interfield and intrafield adaptive prediction method, for 30 Mbit/s, which has an inherent permissible error-ratio tolerance of about $1 \cdot 10^{-8}$.

4.4 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.

[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. [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.

[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 $1 \cdot 10^{-4}$ is $1 \cdot 10^{-8}$. The burst correcting capability is 26 bits.

[CCIR, 1978-82e] 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 $1 \cdot 10^{-5}$ can be reduced to $1 \cdot 10^{-11}$. 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 $1 \cdot 10^{-4}$.

[CCIR, 1982-86e] describes the performance of a self-orthogonal convolution (8, 7) double error-correcting code. The scheme has been implemented for a 32 Mbit/s DPCM codec and showed that a bit error ratio of $2 \cdot 10^{-4}$ can be reduced to an effective ratio of $1 \cdot 10^{-7}$.

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.

[Murakami *et al.*, 1983; Matsumoto *et al.*, 1985] have described a method of protecting against channel errors. This method employs periodic refreshing and error concealment as well as a BCH (255, 239) error-correction arrangement. These measures can reduce the number of occasions that picture degradation occurs to no more than once every two days, with a channel error ratio of $9 \cdot 10^{-6}$.

[CCIR, 1982-86d] describes a Reed-Solomon (63, 59) error-correcting code, interleaved six ways. This code can correct, in a block of 2268 bits, any single burst of length not exceeding 67 bits, or certain combinations of shorter bursts.

[CCIR, 1982-86f] describes a hybrid DPCM system. This system has a reduced sensitivity to transmission errors which is comparable to PCM representation.

5. 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:

5.1 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.

5.2 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.

REFERENCES

- BBC [1974a] BBC Report 1974/41.
 BBC [1974b] BBC Report 1974/40.
 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 Télévision Conference, San Francisco, CA, USA, 80-88.
 GUIONNET, J. and SALLIO, P. [1978] Système expérimental de télévision numérique à 52 Meb/s. Evaluation subjective de l'effet des erreurs de transmission. *Radiodif.-Télév.*, Vol. 4/5, **54**, 1-7.
 MATSUMOTO, S., MURAKAMI, H., HATORI, Y. and YAMAMOTO, H. [1985] 15/30 Mbit/s motion-compensated inter-frame, interfield and intrafield adaptive prediction coding — component coding based on CCIR Recommendation 604 (in Japanese). *J. Inst. Telev. Eng. (Japan)*, Vol. 39, **10**.
 MURAKAMI, H., MATSUMOTO, S., HATORI, Y. and YAMAMOTO, H. [1983] A 15/30 Mbit/s universal digital TV codec and its performance. International Symposium on Image Processing and its Applications.
 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/16 (Germany (Federal Republic of)); b. CMTT/107 (Study Group XVIII, CCITT); c. CMTT/12 (EBU); d. CMTT/268 (France); e. CMTT/47 (Japan).
 [1982-86]: a. CMTT/21 (Japan); b. CMTT/137 (EBU); c. CMTT/49 (France); d. CMTT/171 (United Kingdom); e. CMTT/20 (Japan); f. CMTT/217 (Germany (Federal Republic of)).

ANNEX I*

MATTERS RELATING TO DIGITAL TELEVISION TRANSMISSION THAT THE CMTT WISHES TO DRAW TO THE ATTENTION OF THE CCITT

In its Report 967, the CMTT summarizes its ideas about transmission impairments and methods of protection, including error ratio.

Regarding the permissible error ratio, Report 967 draws attention to the wish of the CMTT not to be forced to work with an error ratio of worse than $1 \cdot 10^{-4}$ as indicating a condition of complete failure. In principle, error-correction techniques can be used at the television decoding terminals, but the complexity of correcting for an error ratio of worse than $1 \cdot 10^{-4}$ is considerable. This will not only make the terminal equipment expensive, but the error-correction arrangement will probably demand an unduly high proportion of the total available bit rate. This would be a significant disadvantage particularly in the case of a television signal which has already been reduced to a relatively low bit rate.

* This Annex together with Annex I to Report 646 has been forwarded by the CMTT to CCITT Study Group XVIII for information and possible action.

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

Recommendations and Reports

RECOMMENDATION 473-4*

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

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

(1970-1974-1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that it is already current practice in a number of countries to use insertion test signals in the field-blanking interval of monochrome and colour television signals;
- (b) that such signals can be used for the measurement of performance and the monitoring, control and correction of characteristics of international transmission circuits;
- (c) that Report 314 proposes that certain specific lines in each field be allocated for the insertion of special test signals for international transmissions;
- (d) that traffic demands may make it necessary to perform all operational measurements by means of insertion test signals, to an accuracy approaching that of conventional out-of-service measuring methods,

UNANIMOUSLY RECOMMENDS

1. that for the international transmission of television signals, insertion test signals in accordance with Annex I (625-line systems) and Annex II (525-line systems) may be inserted at the origin of the circuit;

Note. — As an interim measure some administrations may decide to omit some of the waveforms described. Where waveforms are omitted:

- waveforms other than those described should not be inserted;
- care must be taken to ensure that the line-time luminance components of equivalent lines in each field (e.g. 17 and 330 for 625-line systems) are similar.

2. that these signals should neither be removed nor replaced on the international circuit, except possibly at a point of conversion of either standard or colour system.

ANNEX I

625-LINE SYSTEMS

1. Introduction

For the international transmission of 625-line television signals, Recommendation 472 and Report 314 propose the use of lines 17 (330) and 18 (331) for insertion test signals.

This Annex describes a comprehensive arrangement of insertion test signals (see Note) to which the following general considerations apply:

- it is assumed that the line duration H is divided into 32 equal time periods. This division defines the characteristic instants;
- the time periods shall not differ from each other by more than ± 40 ns;
- the characteristic instants are referred to the mid-amplitude point of the leading edge of the synchronizing pulse. The half-amplitude points of the luminance and chrominance transitions and the peaks of the pulses occur at the characteristic instants;

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

- the actual characteristic instants of any luminance waveform shall not differ by more than 250 ns from their nominal positions;
- except in the case of the 20T composite pulse, the actual characteristic instants of any chrominance waveform shall not differ by more than 500 ns from their nominal positions;
- the colour burst is present in the line-blanking period only in colour transmission;
- in the case of PAL transmissions, the chrominance sub-carrier of the insertion signals is locked at $60 \pm 5^\circ$ 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.

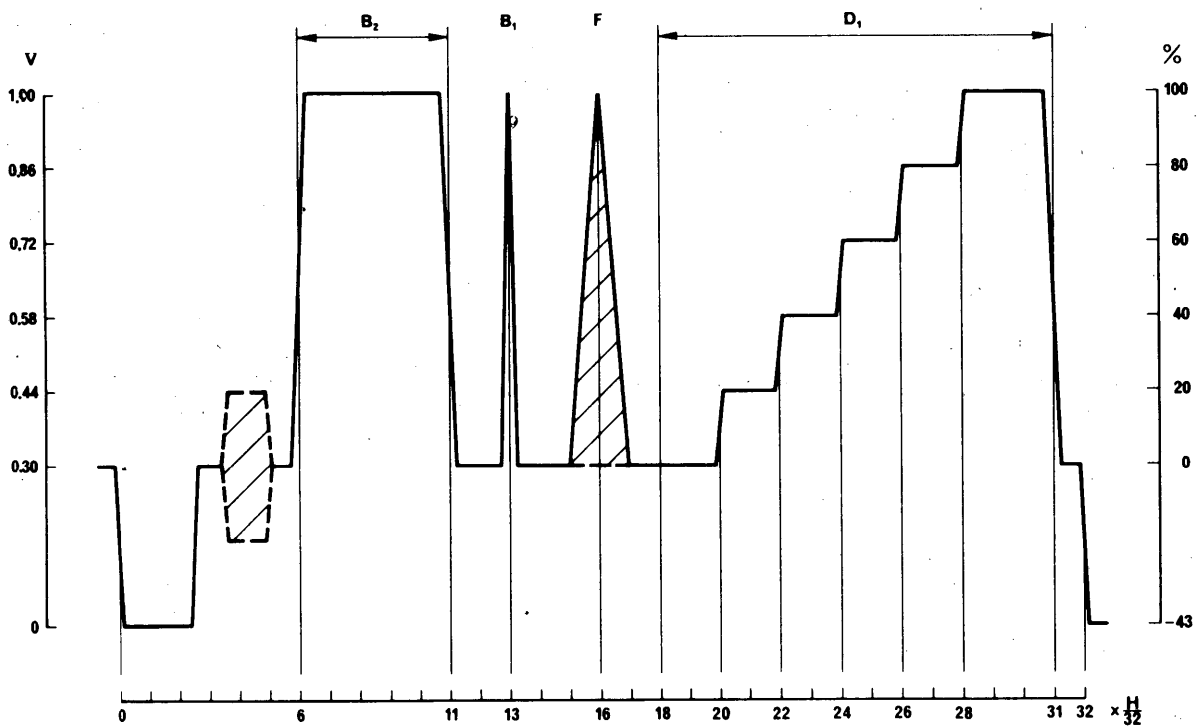


FIGURE 1 – Line 17

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 (see Table I);

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.

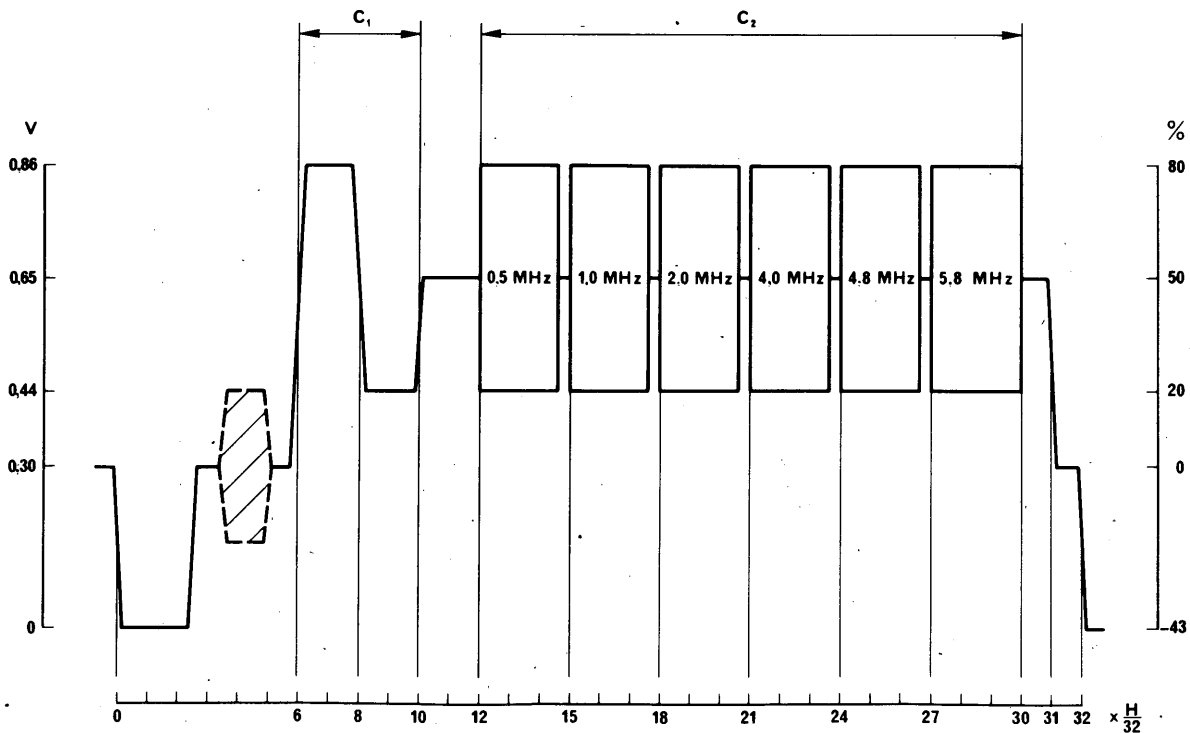


FIGURE 2 - Line 18

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 μ s approximately.

5. Particulars of signals inserted in line 331 (Fig. 4)

5.1 Luminance pedestal

- position of transitions: $6H/32$, $31H/32$;
- amplitude measured from blanking level: within $\pm 1\%$ 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 transactions: 1 μ 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: $\leq 2^\circ$.

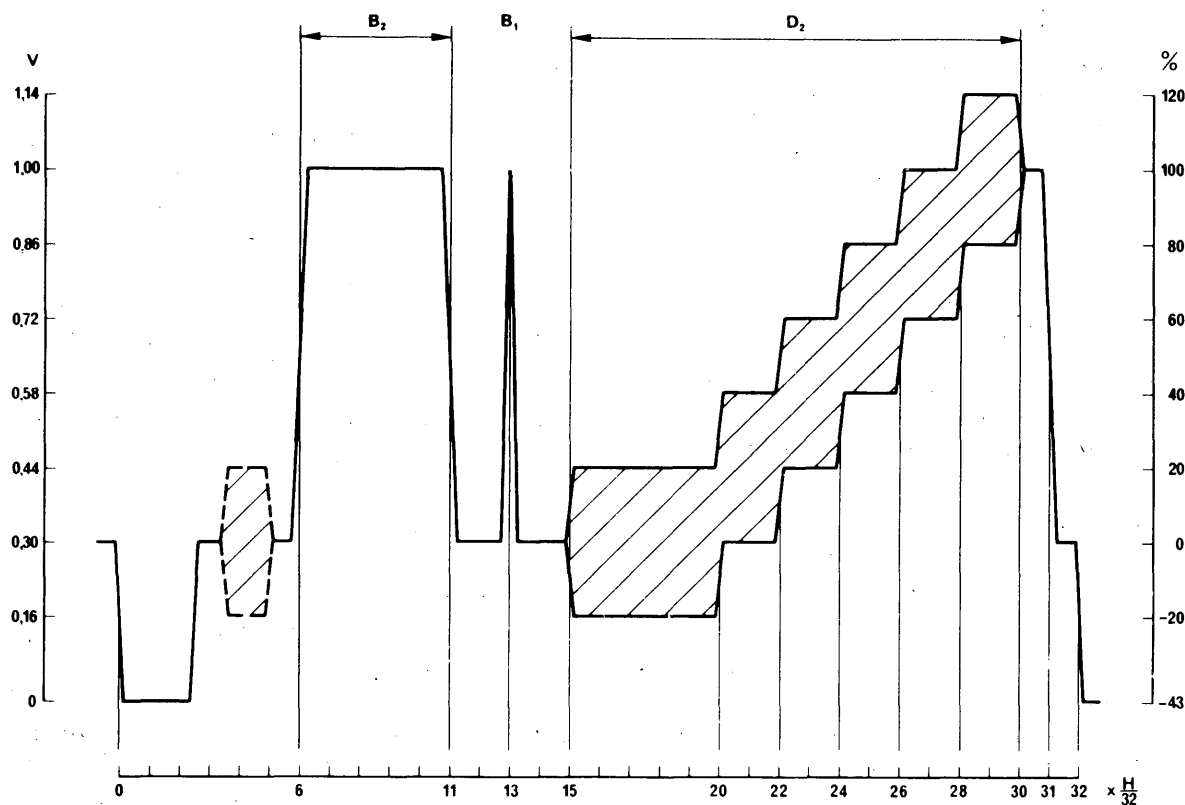


FIGURE 3 - Line 330

5.3 Superimposed three-level chrominance signal (G_2)

This signal may be used as an alternative to the superimposed chrominance bar signal defined above:

- position of transitions: $7H/32$, $9H/32$, $11H/32$ and $14H/32$;
- peak-to-peak amplitudes:
 - 1st section: within $\pm 1\%$ 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 μ 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 μ s approximately;
- phase difference between the sub-carrier superimposed on the staircase in line 330 and the sub-carrier superimposed on the line 331: $\leq 2^\circ$.

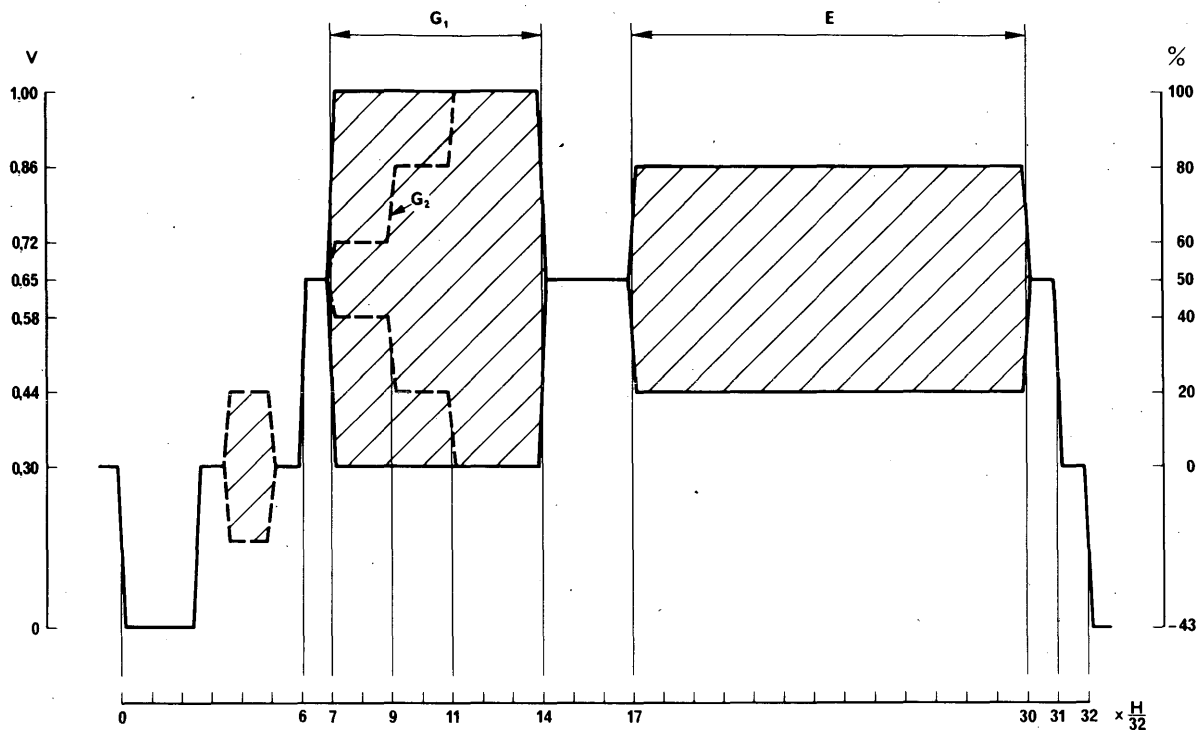


FIGURE 4 - Line 331

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_2 and C_1	18
Line-time waveform distortion	B_2	17 and 330
Short-time waveform distortion:		
— step response	B_2	17 and 330
— pulse response	B_1	17 and 330
Chrominance-luminance gain inequality	B_2 and G_1 or G_2	17 and 330, 331
Chrominance-luminance delay inequality	B_2 and F	17
	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 (see Note 1);
- the signals defined in this Annex apply to both monochrome and colour television transmission, as shown in Figs. 5 and 6. For monochrome transmission, some simplifications of the test signal by the omission of one or more of its components may be desirable. Such simplified signals are shown in Figs. 7 and 8;
- the line duration H is divided into 128 equal parts, and the position and duration of test signals are determined in $H/128$. This division defines the characteristic instants;
- the characteristic instants are referred to the half-amplitude points of the leading edge of the luminance bar signal (B_2) in Figs. 5 or 7 and the reference bar signal (C_1) in Figs. 6 or 8, which are inserted in fields 1 and 2 respectively, (O_{HR}). The half-amplitude point of the luminance and chrominance transitions and peak of the pulses occur at the characteristic instants;
- positioning of the reference point (O_{HR}) shall not exceed $24H/128 \pm 125$ ns relative to the mid-amplitude point of the leading edge of the horizontal synchronizing pulse (O_H);
- the systematic offset in the defined characteristic instants of any luminance and chrominance waveforms shall not differ by more than ± 150 ns (see Note 2) and ± 300 ns (see Note 2) respectively, from the nominal points;
- the random error in the defined characteristic instants for both luminance and chrominance waveforms shall not exceed ± 25 ns from a fixed position which lies within the above systematic offset;
- the colour burst is present in the line-blanking period only in colour transmission;
- the frequency of the colour sub-carrier is 3.579 545 MHz for system M/NTSC, and 3.575 611 49 MHz for system M/PAL, ± 10 Hz.

Note 1. — The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

Note 2. — Reduction in these tolerances is a matter for further study.

2. Particulars of signals inserted in line 17 of field 1 (Figs. 5 or 7) (see Note 1 of § 1)

2.1 Luminance bar (reference white level) (B_2)

- position of transitions: $0H/128$ (O_{HR}) and $36H/128$; duration of bar $36H/128$;
- bar amplitude: 100 ± 0.5 IRE units (see Note);
- rise and fall times of transitions (integrated sine-squared shape): 125 ± 5 ns;
- overshoot and undershoot: $\leq 1\%$;
- 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) (see Note 1 of § 1)*

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 $0.4 \mu\text{s}$, nor longer than $2.0 \mu\text{s}$ 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 (see Note 1 of § 1)

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.

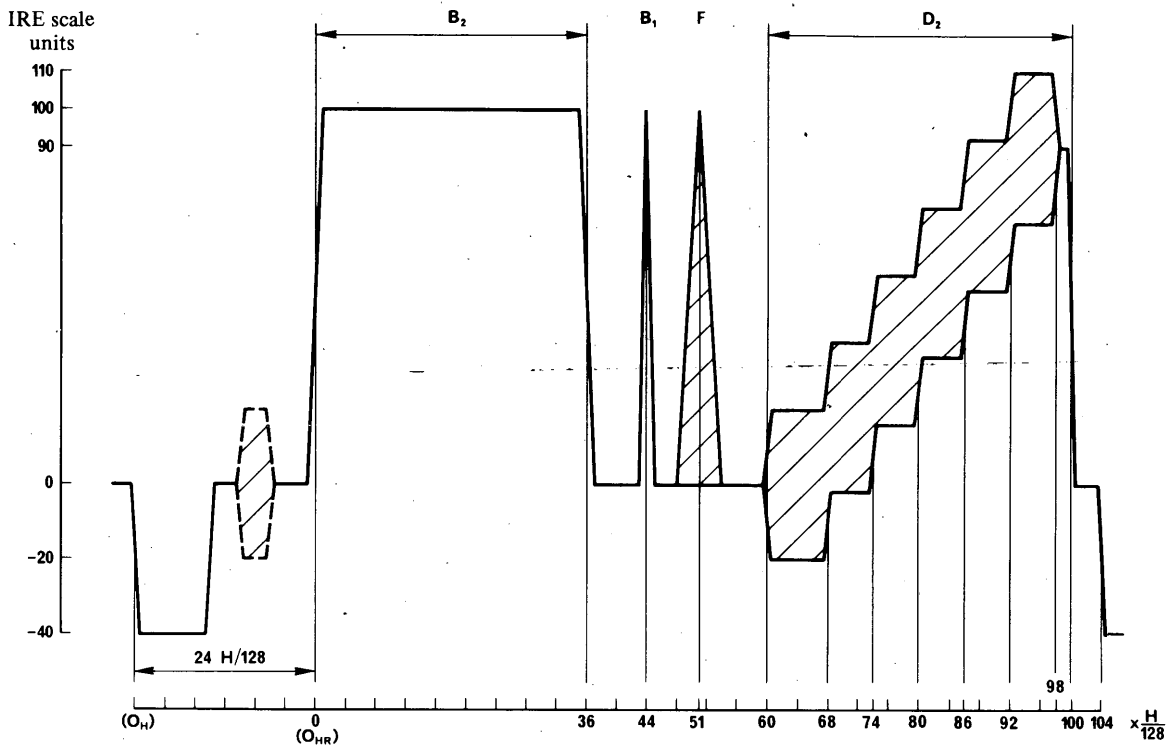


FIGURE 5 – Line 17 of field 1

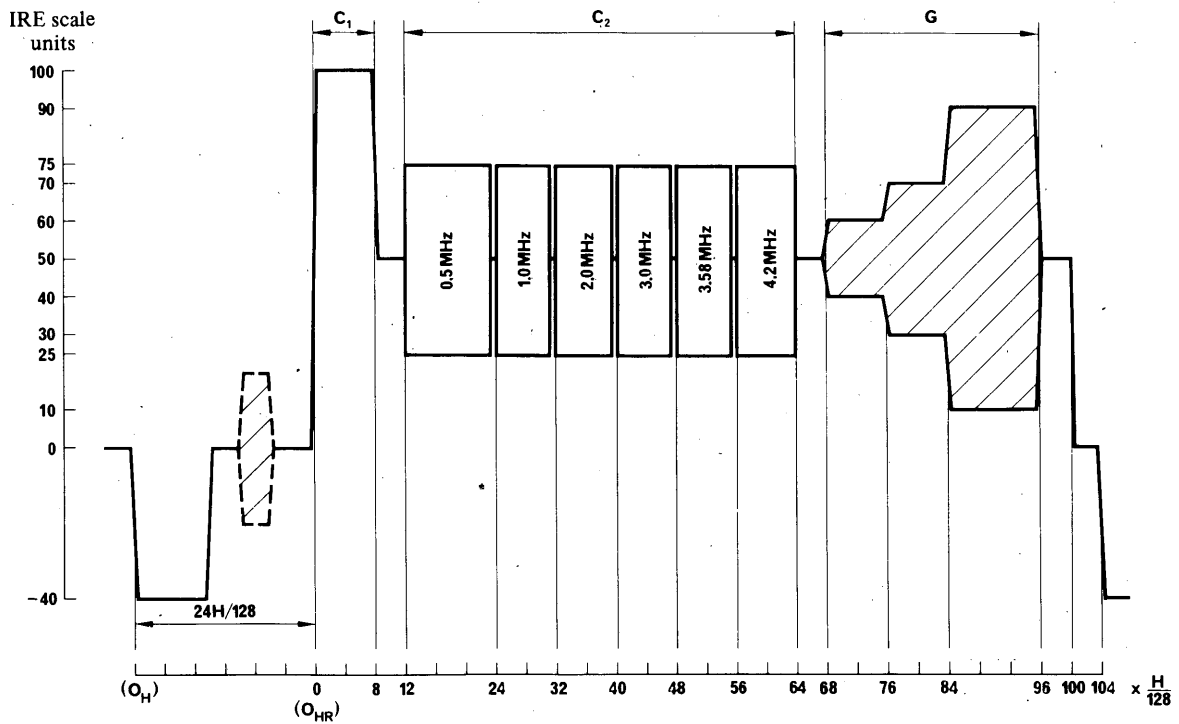


FIGURE 6 – Line 17 of field 2

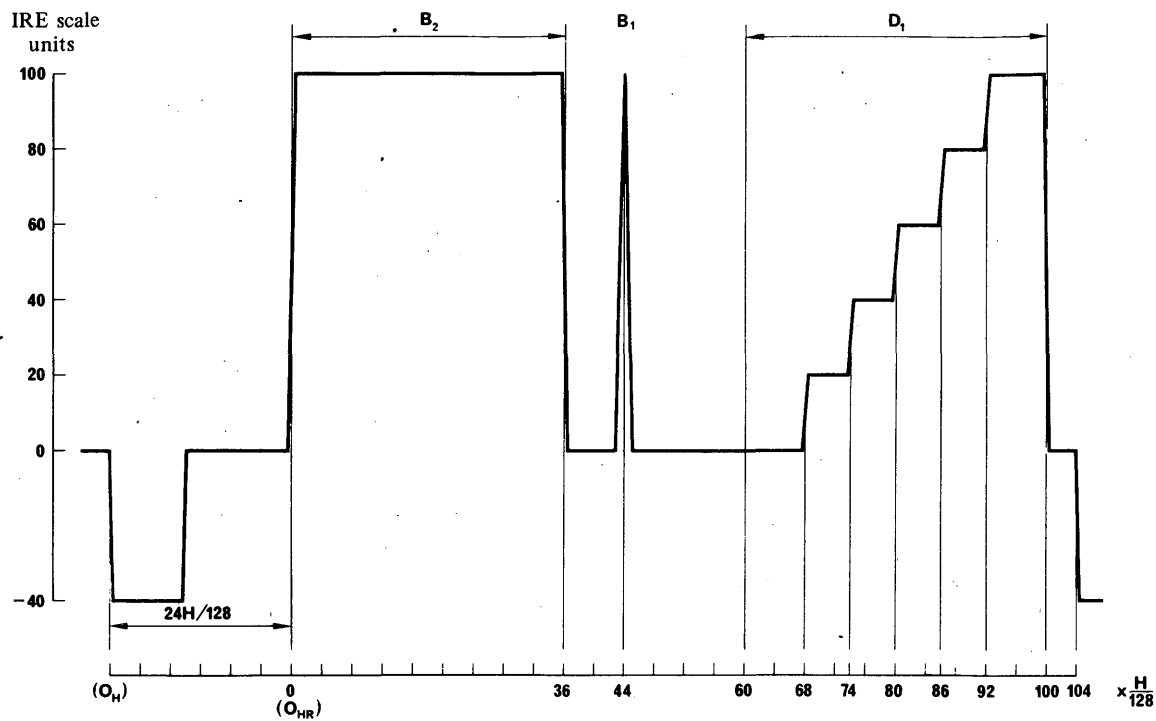


FIGURE 7 - Line 17 of field 1

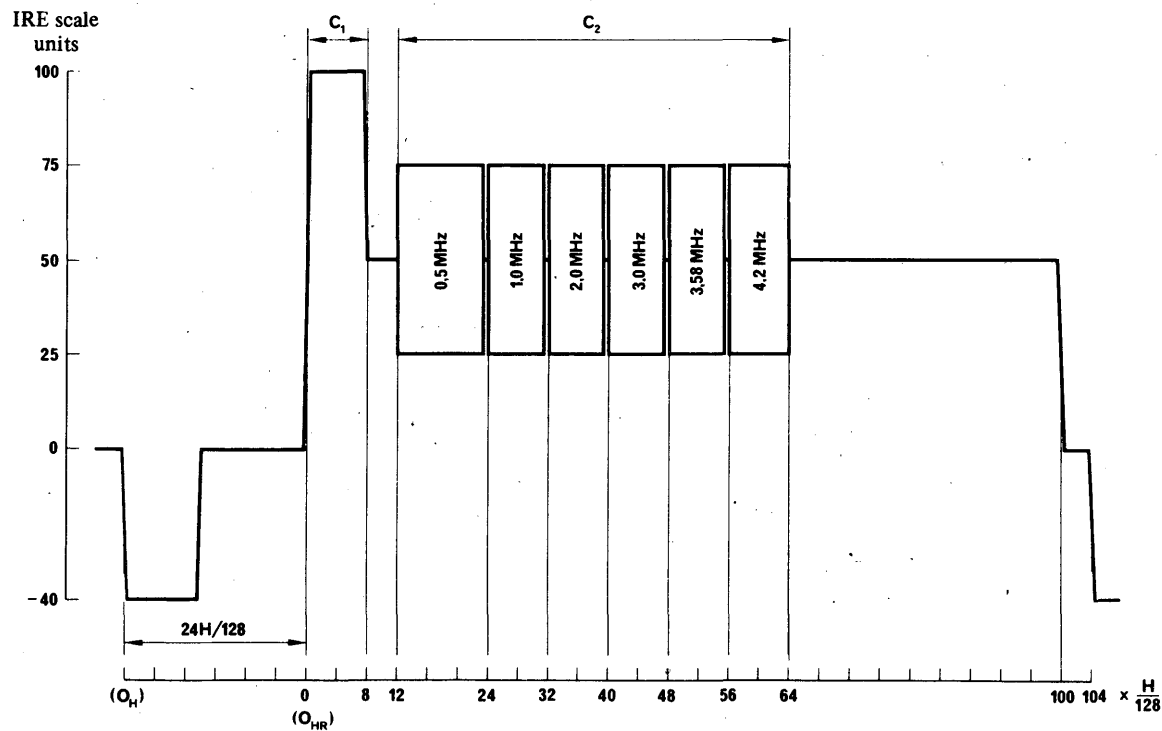


FIGURE 8 - Line 17 of field 2

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

BIBLIOGRAPHY

CCIR Documents
[1982-86]: CMTT/6 (United States of America).

REPORT 314-6

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

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

(1966-1970-1974-1978-1982-1986)

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,
- transmission of data for closed captioning (sub-titling) of the programme.

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, e.g. data carrying programme related closed captions, 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, e.g. data carrying programme related closed captions 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.

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., lines 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., line 22 (335) for the measurement of noise on international connections.

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

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

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

Closed captions (sub-titles) are programme related and can be in a form of teletext system data or, in North America, in a specific data format. The caption data in many countries is contained within the normal teletext service of which the line allocations are decided on a national basis. Some 625-line/50 PAL countries are now confining the caption data to line 21 and in some cases additionally to line 334 to simplify national and international exchange of sub-titled programmes (see Report 1080).

2.6 It may be noted that, if a simplified synchronizing waveform was 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 In the Federal Republic of Germany, a 2.5 Mbit/s bi-phase coded data is inserted in lines 16 and 329 (start code 10001010 10011001 equal for both lines) primarily for operational purposes. Data is inserted in line 329 for each separate section of a national circuit, whereas the data in line 16 refers to a national circuit as a whole. It is generated at the studio output and remains unblanked down to the home receiving equipment. This line also contains a 2-channel sound identification which indicates to the transmitter — and possibly to the receiver as well — whether the actual sound is mono, stereo or bilingual.

In addition, 4 bytes of data in line 16 are used for programme delivery, i.e. to identify each programme item. This enables domestic video recorders to record such an item synchronously with its actual transmission, even if its schedule has been changed.

Both Austria and Switzerland also intend to introduce the line-16 data programme labelling system [CCIR, 1982-86a].

2.10 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. The 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.

2.11 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 in 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 12-18 and 325-331 are used for the transmission of these data within the United Kingdom.

2.12 In Australia and New Zealand, lines 21 and 334 are reserved for data in a teletext system B format (see Recommendation 653) which is used to carry closed captions (sub-titles) for persons with impaired hearing. These data are regarded as part of the programme and special care is taken with video processing equipment so that the data as well as the programme can be delivered to broadcasting facilities and transmitted without corruption [CCIR, 1982-86b].

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 *

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	Field 2 only: 20	Both fields: 15, 16, 17, 18, 19, 21 Field 1 only: 20	Field 2 only: 20
Canada	Both fields: 17, 18	Both fields: 15, 16, 19, 20, 21	Both fields: 17

* The position of the line refers to the beginning of the field-blanking period.

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 20, field 2 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 In Japan, lines 16 and below of both fields may 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 transmit a network source identification (SID) signal on line 20, field 1, utilizing 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 hard of hearing.

3.9 In the United States of America, broadcasting organizations have been experimenting with the transmission of teletext signals on lines 15, 16, 17 and 18 at a data rate, in most cases of 5.72 Mbit/s.

3.10 In Canada, line 18 is designated as optional for test signals. However, users are encouraged to use only line 17 for transmission testing.

3.11 In Canada, line 19, fields 1 and 2, is used for a vertical interval reference signal (VIR) for the maintenance of the quality of the TV signal.

3.12 Canada has also allocated other lines for the following ancillary signals:

- line 15 and 16, both fields: for signals intended for use by the general public;
- line 18, both fields: for channel identification network monitoring, telemetry information for remote control and network communications;
- line 20: for source identification (SID). Signals intended for use by the general public may be used on a secondary basis;
- line 21: for closed captioning, i.e. to provide sub-title services for the hard of hearing. Signals intended for use by the general public may be used on a secondary basis.

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).

[1982-86]: a. CMTT/204 (Germany (Federal Republic of)); b. CMTT/209 (Australia).

BIBLIOGRAPHY

ANDERSON, W. N. [1977] Identification of programme sources on the Eurovision Network. *EBU Rev. Tech.*, 166.

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.

IRT [December, 1984] Video-Programm-System (VPS). Technische Richtlinie ARD/ZDF Nr. 8R2.

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).

[1982-86]: CMTT/7 (United States of America); CMTT/59 (Canada).

REPORT 818-2

AUTOMATIC SWITCHING SYSTEMS FOR TELEVISION CIRCUITS

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

(1978-1982-1986)

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 which is 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 and is given 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.

Under multipath fading conditions, the level of the higher frequencies of the baseband signal may be raised due to in-band distortion, and deterioration of video signal quality may occur before the noise level exceeds the switch threshold. In order to avoid this deterioration, the use of a supplementary switching method based on a continuity pilot inserted in the upper part of the baseband has been found effective. In this method, switching is initiated on an increase of the pilot level [CCIR, 1982-1986].

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.

Degradations 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).

[1982-86]: CMTT/19 (Japan).

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.

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.

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.

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).

REPORT 1094*

INFLUENCE OF AN ENERGY DISPERSAL SIGNAL ON THE PICTURE QUALITY OF SATELLITE TV TRANSMISSIONS AND ON THE AUTOMATIC MEASUREMENT OF CERTAIN PARAMETERS

(Study Programme 15D/CMTT)

(1986)

1. Introduction

Energy dispersal techniques are used to reduce the spectral energy density for transmission channels in the fixed-satellite service in order to reduce interference problems. This is achieved by superimposing a very low frequency dispersal signal on the baseband signal which modulates the transmitted carrier.

If the baseband signal is a TV signal, a triangular waveform is added to it for energy dispersal. The period of this waveform is that of the TV field and it is synchronized to the field frequency of the TV signal to be transmitted. Other energy dispersal techniques are also studied by Study Group 4 of the CCIR (see Report 384).

The influence of energy dispersal techniques on some impairments of a TV signal and on the reproducibility and correlation of related parameters (primarily differential phase and gain) measured by automatic measuring equipment currently in use is addressed in this Report.

* This Report should be brought to the attention of Study Group 4.

2. TV signal impairments

Due to the addition of an energy dispersal signal, a TV signal may be impaired in two ways:

- firstly, additional non-linear distortions may occur. These may vary in magnitude from line to line in the TV signal field according to the dispersal waveform voltage and are the result of a varying and asymmetric FM band limitation;
- secondly, the dispersal waveform must be removed from the TV signal at the receiving end by a clamping circuit. This may also cause additional distortions which vary with the dispersal waveform voltage.

3. Problems of automatic measurement

Regarding automatic measurement equipment, according to Recommendation 569, test signals are placed on specified lines in the vertical blanking interval and are repetitively evaluated.

Having synchronized the energy dispersal waveform to the TV signal field frequency, the results of automatic measurements are reproducible but may not be true in terms of the magnitude of the distortions affecting the active lines of the TV signal. This results from the strobing of only one line per field for measurement when the distortion varies line by line under the influence of the dispersal signal [CCIR, 1978-82a and b].

If different synchronization phases occur, the measured results can vary significantly depending on how the phase of the dispersal signal is locked to the TV signal. For optimum results, it is necessary to obtain correlation between the average amount of picture impairment and automatically measured distortion parameter magnitudes. This can be achieved by averaging two extreme results associated with the two extreme values of the energy dispersal signal. For this purpose, [CCIR, 1982-86] proposes a cyclic change of the phase lock of the commonly used dispersal signal. This requires a modified dispersal signal generator only. The receive terminal and automatic measuring equipment remain unchanged by this proposal.

4. Further work

The subject requires further study in order to establish an optimal way of overcoming the problems associated with energy dispersal techniques, i.e. to minimize the influence of such a process on picture quality and automatic measurement results.

REFERENCES

CCIR Documents

- [1978-82]: a. CMTT/2 (United Kingdom); b. CMTT/70 (United States of America).
[1982-86]: CMTT/212 (Germany (Federal Republic of)).

RECOMMENDATION 569-2

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

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

(1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that Recommendation 567 is the basic reference which defines the parameters which are to be measured and the test signal elements and measuring methods which are to be used, in order to determine the performance of a television transmission circuit;
- (b) that Reports 628 and 411 describe various techniques for automatic measurement and monitoring of the performance of television chains which make use of insertion test signals;
- (c) that operational measurements are commonly carried out using insertion test signals which are defined in Recommendation 473;

- (d) that Report 314 describes the allocation of lines in the field-blanking interval for special purposes;
- (e) that, although automatic measuring equipment exists that can make measurements in accordance with Recommendation 567, simplified automatic measuring equipment also exists that requires modifications of the measurement methods and definitions;
- (f) that such simplified automatic measurement of insertion test signals suits the requirements of operational staffs and makes the analysis of measurement results easier,

UNANIMOUSLY RECOMMENDS,

that, when simplified automatic measuring equipment is used to make measurements of an insertion test signal, and when a normalized form of presentation of the results is desired, the definitions used in quantifying the parameters of that signal should be those which are given in Annex I.

ANNEX I

1. Introduction

The need for each of the measurements described in this Recommendation (and possibly other measurements) will depend upon the type of plant in use and the policy of the administrations.

The test signals specified are those shown in Recommendation 473.

The definitions assume that the performance of the measuring equipment employed is such that any harmonic components of the incoming signal, 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:

- "peak differential gain", which is defined by either $+x$ or $-y$, depending upon which of these parameters has the larger magnitude.
- "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.

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.

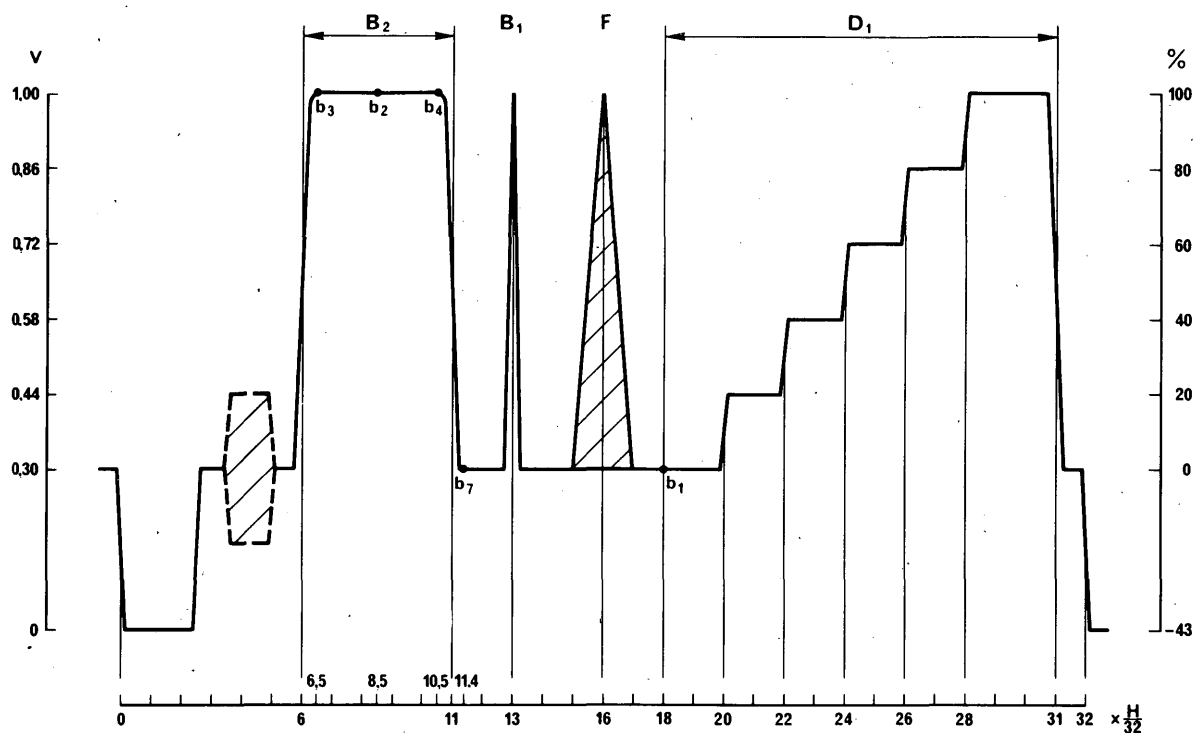


FIGURE 1 – Line 17 for 625-line systems

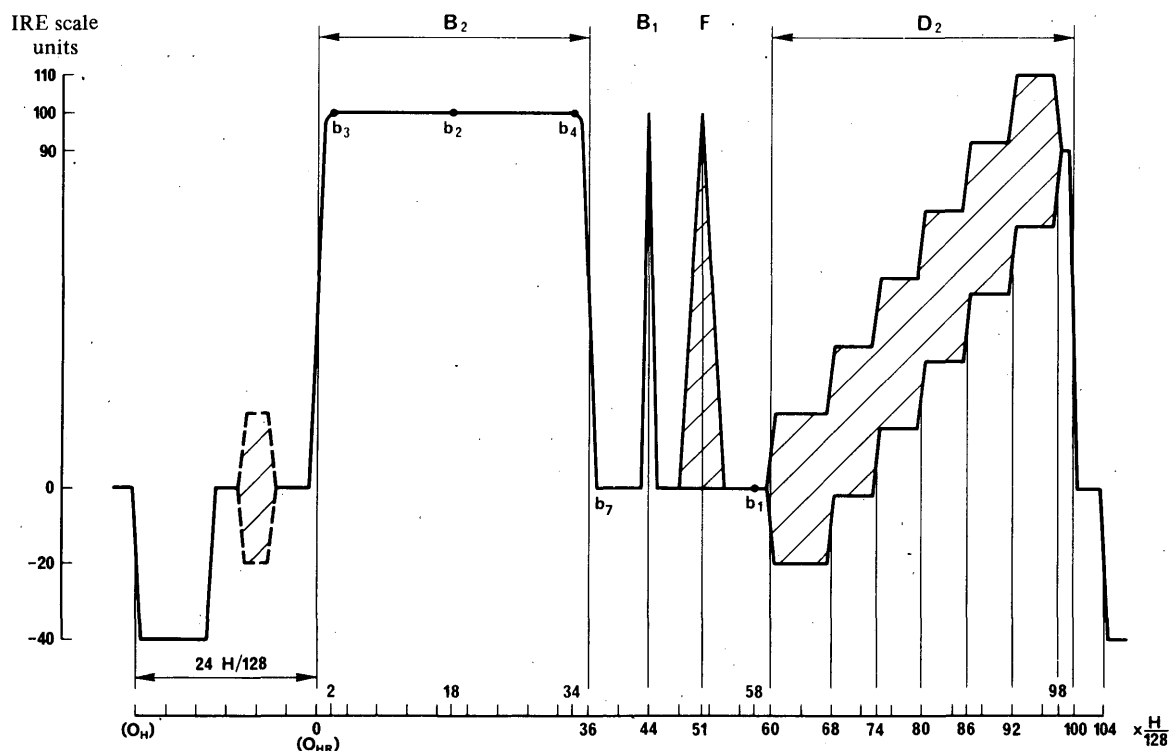


FIGURE 2 – Line 17 of field 1 for 525-line systems

Note. – The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

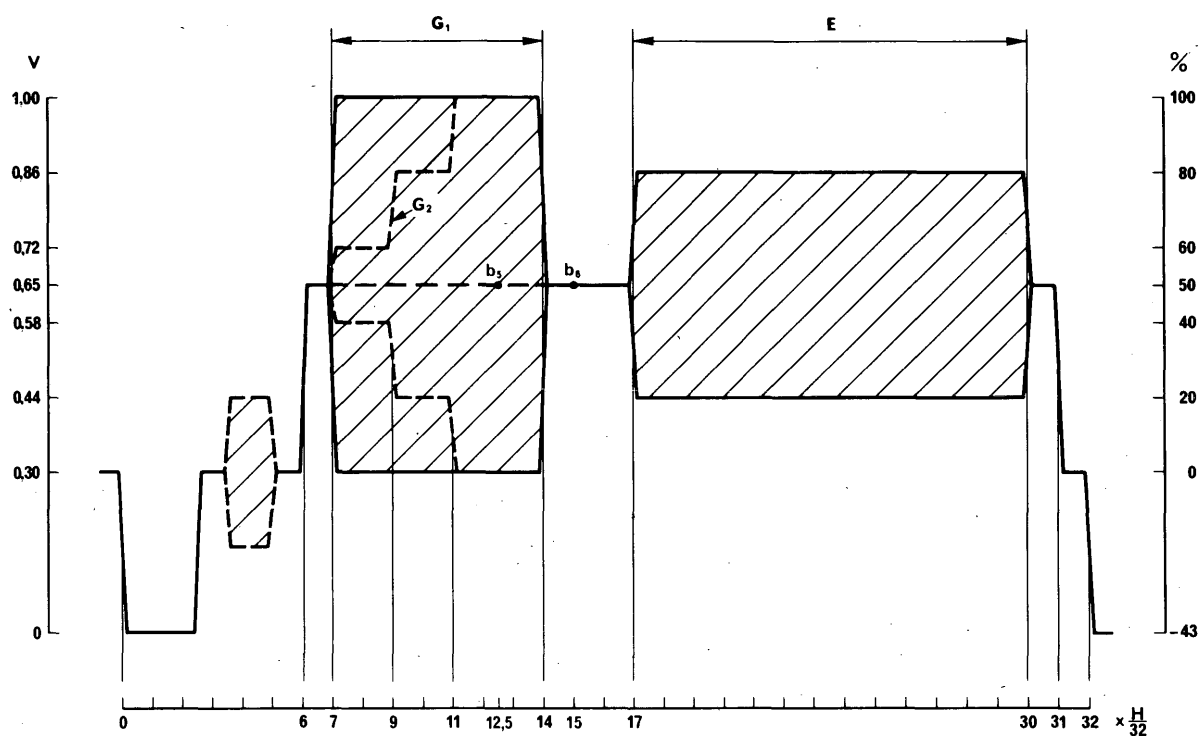


FIGURE 3 – Line 331 for 625-line systems

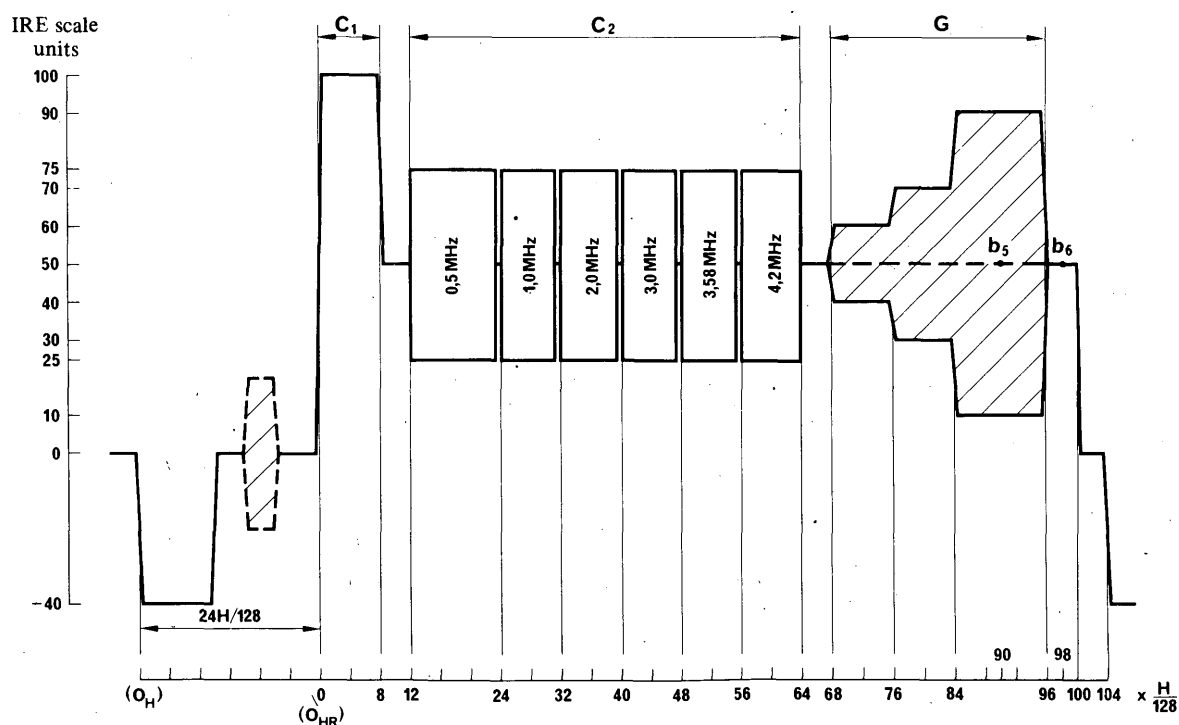


FIGURE 4 – Line 17 of field 2 for 525-line systems

Note. – The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

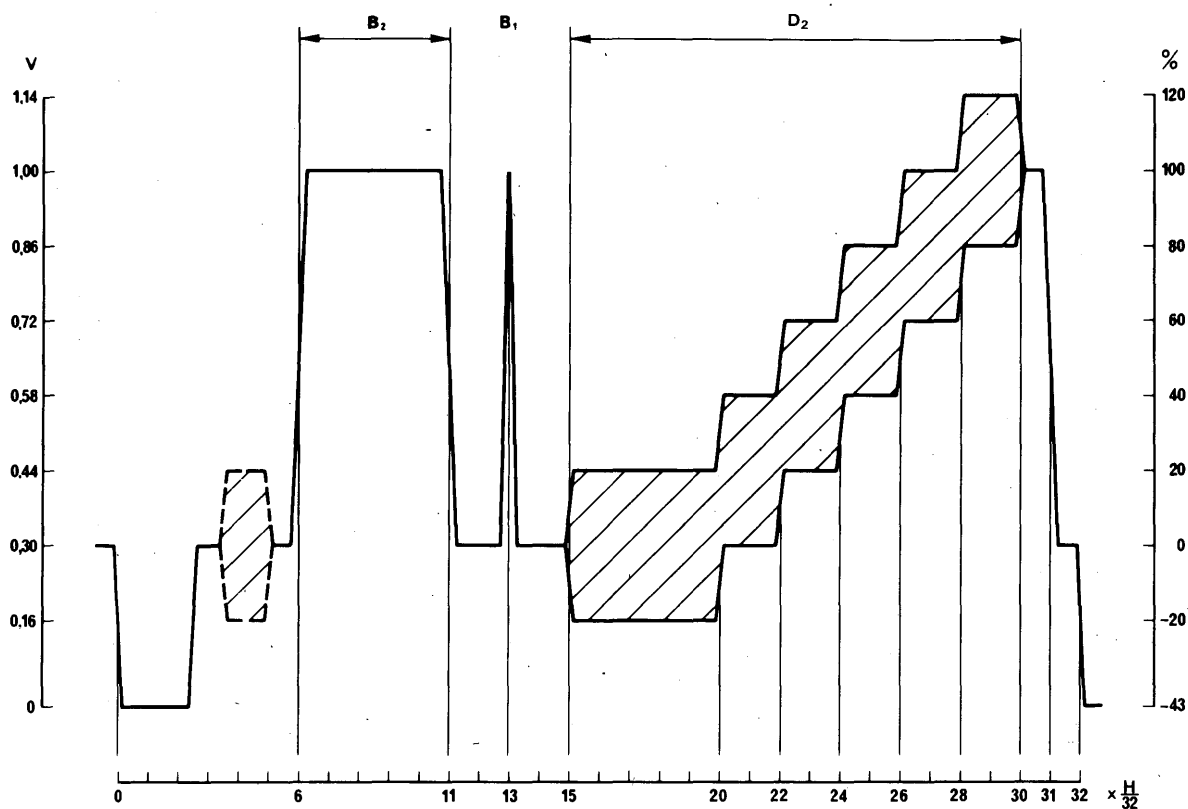


FIGURE 5 – Line 330 for 625-line systems

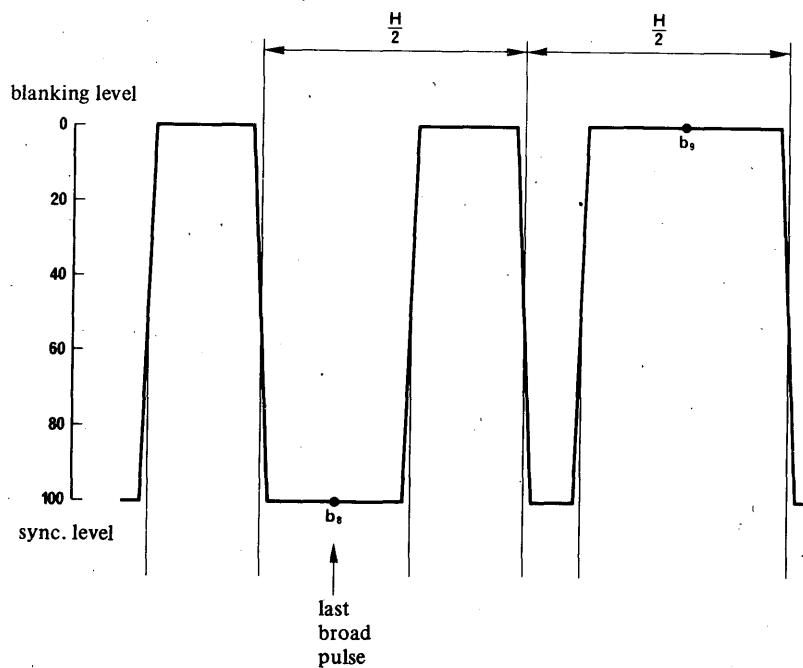


FIGURE 6 – Measurement points for sync. amplitude error during equalizing pulse period

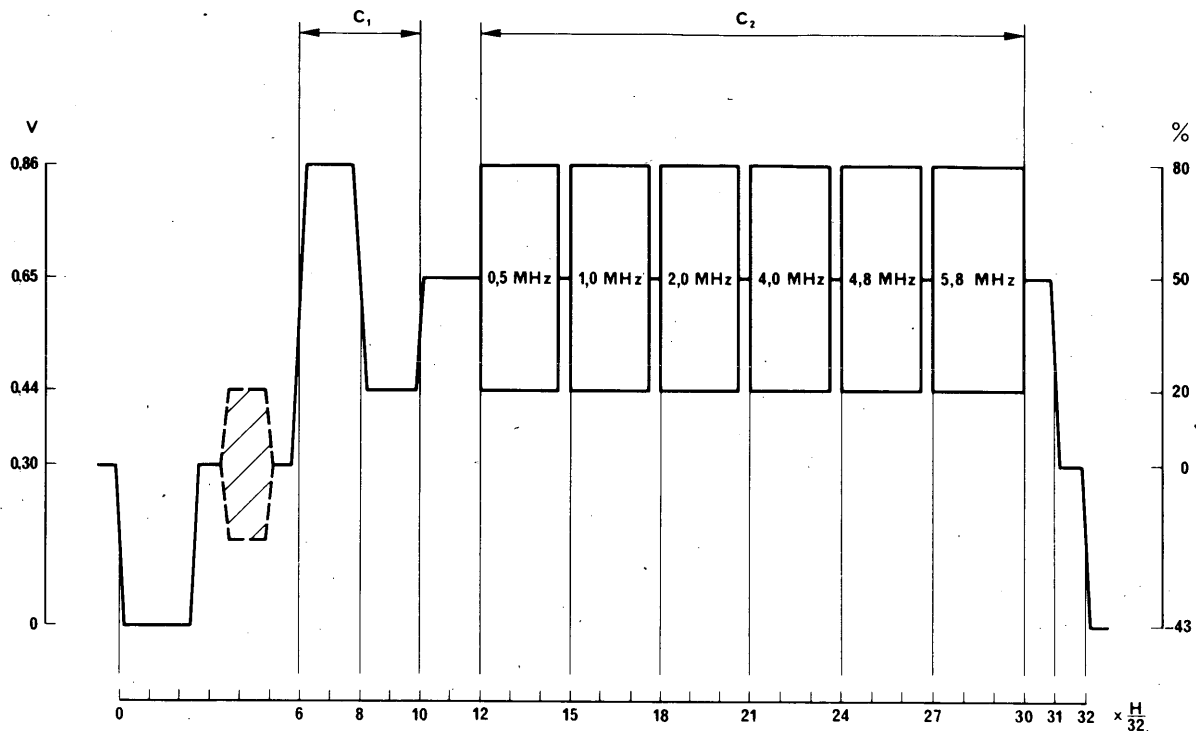


FIGURE 7 – Line 18 for 625-line systems

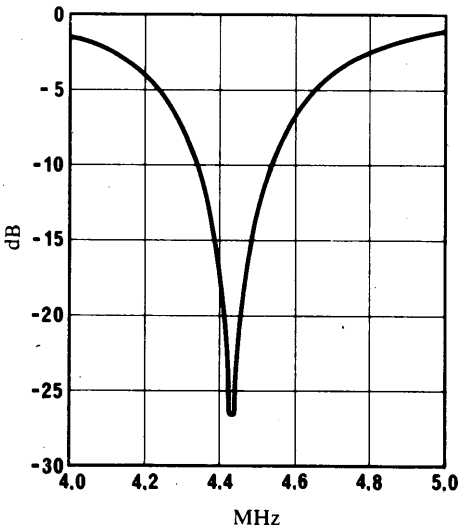


FIGURE 8 – Response of sub-carrier notch filter for 625-line systems

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

TABLE I – *Theoretical values of conversion factors in dB*
(rounded off to a tenth of dB)

		200 kHz high pass filter 20 dB/decade	Sub-carrier notch filter 3 dB bandwidth = 600 kHz
White noise	unweighted	0.3	0.7
	weighted	1.3	0.2
Triangular noise	unweighted	0.0	1.8
	weighted	0.0	1.3
De-emphasized triangular noise	unweighted	0.0	1.6
	weighted	0.0	0.9

REFERENCES

THOMSON, W. E. [1952] "Solution 3". *Proc. IEE*, Part III, **99**, 373.

CCIR Documents

[1974-78]: CMTT/78 (Italy).

[1978-82]: a. CMTT/270 (Germany (Federal Republic of)); b. CMTT/41 (France).

BIBLIOGRAPHY

CCIR Documents

[1978-82]: CMTT/304 (OIRT).

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 Whyte, 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.

REFERENCES

- ANDERSON, W. N. and MCKENZIE, G. A. [1970] Automatic control and supervision of television transmitting stations. EBU Doc. Tech. 3092.
- CASTELLI and GIUGLIARELLI [1968] Il controllo automatico delle reti di distribuzione televisive (The automatic monitoring of television distribution networks). *Alta Frequenza*, Vol. 37, 7, 633-646.
- D'AMATO, P. and TABONE, D. [1974] Misura automatica del rapporto segnale rumore nelle trasmissioni televisive (Automatic measurement of signal-to-noise ratio in television transmissions). *Elettronica e Telecomunicazioni*, 6.
- D'AMATO, P. [October, 1976] Distortion analyzer of insertion test signals made by RAI. Colloque International sur l'automatisation de la Radiodiffusion et les réseaux de transmission. Paris, France.
- EBU [1976] Doc. Tech. 3216-E.
- GROSSKOPF, H. [1971] Ein Verfahren zur Messung des Störabstandes in einer Zeile mit digitaler Anzeige. (A method of measuring the signal-to-noise ratio in one line with digital display). *Rundfunktechnische Mitteilungen*, 15, 56-60.
- IBA [1976] Independent Broadcasting Authority: The Lichfield experiment. E & D Report 106/76.
- ITA [1963] Independent Television Authority: A preliminary design study of a system suitable for monitoring television link performance over a telephone circuit. Interim Technical Memorandum 1/63.
- ITA [1964] Independent Television Authority: A system for remote monitoring of television signal quality by means of a telephone line. Interim Technical Memorandum 8/64.
- KRIVOSHEEV, M. I. [1962] New principles of design of television chain monitoring systems (in Russian). *Teknika kino i televideniya*, 10.
- KRIVOSHEEV, M. I. [1968] Automatic measuring and monitoring in a television chain (in Russian). *Electrosviaz*, 7.
- KRIVOSHEEV, M. I. [1976] *Fundamentals of Television Measurements* (in Russian). Second edition, Sviaz, Moscow.
- MARQUET, J. F. [19-21 October, 1976] Développement d'équipements de correction et de télémessure automatiques en vidéofréquence. Colloque international sur l'automatisation de la radiodiffusion et des réseaux de transmissions, Paris, France.
- RHODES, C. W. [June, 1977] An all digital measuring system for baseband video test signals. International Television Symposium, Montreux, Switzerland.
- SHELLEY, I. J. and SMART, D. L. [1971] Automatic measurement and control of unattended television transmitters. *JSMPT*, Vol. 80, 11.
- SHELLEY, I. J. and WHYTE, D. J. [1972] Automatic equalizers for analogue television signals. IEE Conf. Publ., 88.
- STEELE, F. H. [February, 1975] Some applications of digital techniques to broadcast data. IEE, London, *Electron. and Power*, Vol. 21, 3.
- WATSON, J. B. [1976] Digital automatic measuring equipment. IEE Conf. Publ., 145, 192-196.
- CCIR Documents*
- [1963-66]: a. XI/60 (USSR); b. XI/21 (CMTT/8) (Japan); c. CMTT/54 (United Kingdom).
- [1966-69]: a. CMTT/69; b. XI/162 (CMTT/143) (USSR).
- [1970-74]: a. CMTT/180 (Germany (Federal Republic of)); b. CMTT/30; c. CMTT/32; d. CMTT/31 (Germany (Federal Republic of)); e. CMTT/196 (Japan).
- [1974-78]: a. CMTT/268 (OIRT); b. CMTT/285 (France).
- [1978-82]: CMTT/280 (USSR).

RECOMMENDATION 570*

STANDARD TEST SIGNAL FOR CONVENTIONAL
LOADING OF A TELEVISION CHANNEL

(Study Programme 15E/CMTT)

(1978)

The CCIR,

CONSIDERING

- (a) that a common transmission path may be used by one or more television signals and one or more sound-programme channels or telephony channels;
- (b) that, due to distortion in the common path, the signals applied to one or more television channels may result in unwanted signals appearing in other channels;
- (c) that Report 375 draws attention to the need, where a sound-programme circuit is carried on a radio-relay system for television, for measurements or calculations to be made with the television channel loaded with a standard test signal;
- (d) that a similar problem exists for cable systems such as that described in CCITT Recommendation J.73 or that being studied in CCITT Question 20/XV (1976-80);
- (e) that a special test signal is required for use as a conventional load for a television channel when measuring, calculating or specifying the noise in other circuits which are carried in a common path with the television channel or channels;
- (f) that such a standard test signal for conventional loading of a television channel should preferably be one which is already in general use or which can readily be made available to users and manufacturers of the transmission systems concerned;
- (g) that a standard test signal for conventional loading of a television channel should preferably be typical of a wide range of video signals likely to be used in practice, and should include chrominance information and a field-frequency component,

UNANIMOUSLY RECOMMENDS

1. that a composite colour-bar test signal, appropriate to the colour television standard to be carried, should be used as the standard test signal for conventional loading of an analogue colour television channel;
2. that, using the nomenclature of Recommendation 471, the luminance and chrominance amplitudes of the colour bar test signal when used for conventional loading should be 100/0/75/0 for 625/50 standards;
3. that countries which operate on 525/60 standards should employ a colour bar signal of the split field type for conventional loading. The colour bars occupy the larger portion of the active lines and have the characteristic 75/7.5/75/7.5.

Accompanying the colour bars, on a split-field basis is a peak-white bar on all active lines not carrying the colour bars. For example, Electronic Industries Association (USA) RS-189-A colour bars;

4. that the colour bars should be arranged in order of descending luminance as commonly used.

Note 1. — The choice of standard test signals for conventional loading of analogue television channels which are used only for monochrome transmission, and digital channels for monochrome and colour television transmission, should be the subject of further study.

Note 2. — The colour bar test signal at an appropriate saturation is representative of the circuit load caused by the studio outputs. It is not typical, however, of some test signals, electronically-generated captions, or some data and other special signals, of which the disturbing effect on other channels on the same bearer path may be greater than for camera-generated signals. The disturbing effect due to the spectral components of these special signals should be tested by means of the signals themselves.

Note 3. — Administrations making measurements of the disturbing effect of a television signal on other signals which share the same bearer path, should take into account the relative stability of the carrier frequencies of the disturbing and disturbed circuits on the common bearer channel.

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

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 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.

* 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.

*** [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.

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.

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.

REFERENCES

CCIR Documents

[1970-74]: a. CMTT/35 (Germany (Federal Republic of)); b. CMTT/16 (United Kingdom);

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

[1978-82]: CMTT/62 (LM Ericsson).

BIBLIOGRAPHY

CCIR Documents

[1978-82]: CMTT/305 (OIRT).

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.

REFERENCES

- D'AMATO, P. [1972] Distorsioni della cromaticanza in un segnale PCM video lineare. *Elettronica e Telecomunicazioni*, 3, 111-118.
- FELIX, M. O. [February, 1976] Differential phase and gain measurements in digitized video signals. *SMPTEJ*, Vol. 85, 76-79.
- KRIVOSHEEV, M. I. and VILENCHIK, L. S. [1977] Izmereniya v sistemakh tsifrovogo televideniya (Measurements in digital TV systems). *Elektrosvyaz*, 1.
- THOMPSON, J. E. [September/October, 1975] Methods of digital coding for television transmission. *Roy. Telev. Soc. J.*, 384-391.
- CCIR Documents*
[1974-78]: a. CMTT/217 and Corr. 1 (USA); b. CMTT/257 (USSR).

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.

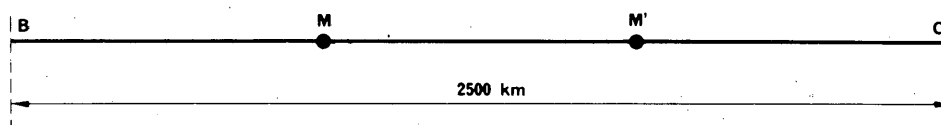


FIGURE 1 – *The hypothetical reference circuit for sound programme transmissions over a terrestrial system*

* 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.

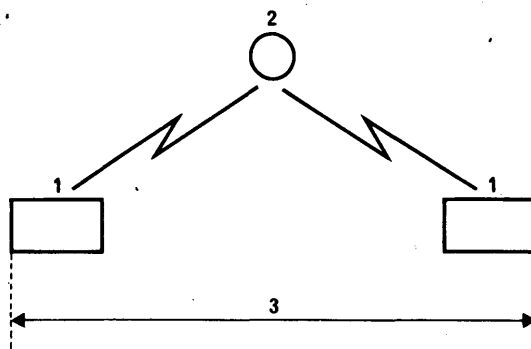


FIGURE 2 – *Hypothetical reference circuit for sound-programme transmissions over a system in the fixed-satellite service*

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

ANNEX I

ILLUSTRATION OF AN INTERNATIONAL SOUND-PROGRAMME CONNECTION

Figure 3 illustrates a typical international sound-programme connection in which:

- point A, to be considered as the sending end of the international sound-programme connection, may be the point at which the programme originates (studio or outside location),
- point D, to be considered as the receiving end of the international sound-programme connection, may be a programme-mixing or recording centre or a broadcasting station,
- the local sound-programme circuit AB connects point A to the sending terminal station, point B, of the international sound-programme circuit BC,
- the local sound-programme circuit CD connects point C, the receiving terminal station of the international sound-programme circuit BC to the point D.

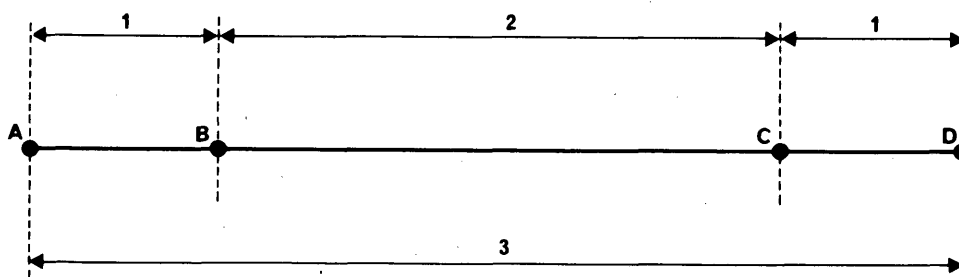


FIGURE 3 – *An international sound-programme connection*

- 1: Local sound-programme circuits
- 2: International sound-programme circuit
- 3: International sound-programme connection

The hypothetical reference circuit must not be considered identical to any of the sound-programme circuits illustrated above or to those defined for maintenance purposes in Volume IV, Fascicle 3 of the CCITT Red Book. However, some of these circuits may display the same structure as the hypothetical reference circuit. Such types of circuits are:

- an international sound-programme connection comprising three audio sections,
- a single sound-programme circuit made up of three audio sections.

In this case, the performance standards set for the hypothetical reference circuit may be applied to these circuits.

RECOMMENDATION 503-3 *

PERFORMANCE CHARACTERISTICS OF 7-kHz-TYPE (NARROW-BANDWIDTH) SOUND-PROGRAMME CIRCUITS **

Circuits for medium quality monophonic transmission

(Question 17/CMTT)

(1974-1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that it is necessary to set transmission standards for sound-programme circuits;
- (b) that quality requirements for the hypothetical reference circuit are established for analogue sound programmes;
- (c) that advantage should be taken of the technical evolution made possible by the introduction of digital techniques, particularly for mixed analogue and digital circuits,

UNANIMOUSLY RECOMMENDS

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out below.

1. Application

This Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation 502.

For estimation of the performance of circuits shorter or longer than the HRC, see Recommendation 605.

Note 1. – For all-digital circuits, a separate Recommendation might be envisaged after further study.

Note 2. – For further information, Report 496 may be consulted. This Report also draws attention to certain differences between CCIR and OIRT Recommendations.

2. Interface characteristics

2.1 Test conditions

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 Ω resistive.

* This Recommendation is also proposed by the CMTT as a revision of CCITT Recommendation J.23 and should be brought to the attention of CCITT Study Group XV.

** a) For the definition of absolute power, relative power and noise levels, see Recommendation 574.

b) For old 5 kHz, 6.4 kHz and 10-kHz-type systems still in use, see Recommendation 503-1 (Kyoto, 1978) and Recommendation 504-2 (Geneva, 1982) respectively.

2.2 Impedance

System input impedance	600 Ω , balanced *
System output impedance, provisionally	Low, balanced

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

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

2.3 Levels

Input maximum programme level	+ 9 dBm0s
Insertion gain (1 kHz at -12 dBm0s)	0 dB
Adjustment error, within	± 0.5 dB
Variation over 24 hours not to exceed	± 0.5 dB
Relative level (see CCITT Recommendation J.14)	+ 6 dBrs

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional trimming attenuators.

3. Overall performance

3.1 Common parameters

3.1.1 Gain/frequency response

Reference frequency:	1 kHz (nominal value)
The response shall be measured at:	-12 dBm0s

TABLE I

Frequency (kHz)	Response (dB)
$0.05 \leq f < 0.1$	+1 to -3
$0.1 \leq f \leq 6.4$	+1 to -1
$6.4 < f \leq 7$	+1 to -3

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

3.1.2 Group delay variation

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

kHz	$\Delta\tau$ (ms)
0.05	80
0.1	20
6.4	5
7	10

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

3.1.3 Noise

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

* The tolerance, permitted reactance and degree of unbalance need further study.

For radio-relay systems, the requirements shall be met for at least 80% of the total time of any 30-day period. For 1% of the time an additional impairment of 4 dB, and for 0.1% of the time an additional impairment of 12 dB is acceptable.

TABLE II

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
Idle channel noise, maximum (dBq0ps)	– 44	– 49
Programme-modulated noise, maximum (dBq0ps)	– 32	– 37

Programme-modulated noise usually occurs on sound programme circuits which are equipped with compandors (e.g. types of circuits corresponding to CCITT Recommendation J.31).

This noise value may be measured with the aid of an auxiliary sinusoidal test signal +9 dBm0s/60 Hz which has to be suppressed by a high-pass filter ($f_0 \leq 400$ Hz, $a \geq 60$ dB/60 Hz) before the measuring set.

Report 493 indicates that if a compandor is used, an improved signal-to-noise ratio is necessary to avoid objectionable effects with some programme material.*

Note. – For digital systems, appropriate values are under study. For further information, see Report 647.

3.1.4 Single tone interference

Level of any individual tone:

$$\leq -(73 + \psi) \quad \text{dBm0s}$$

where ψ is the weighting filter response factor (positive or negative) as given in Recommendation 468 at the particular frequency.

For sound-programme transmissions over carrier systems, occurrence of carrier leaks can be expected. For this reason, stop filters may be provided in the carrier frequency path which can be switched in, if required, to suppress the tones otherwise audible in the upper frequency range from 8 to 15 kHz. For a hypothetical reference circuit, it is recommended that the stop filters should have a 3 dB bandwidth of less than 3%, referred to the mid-frequency. The use of stop filters which affect frequencies below 8 kHz should be avoided.

3.1.5 Disturbing modulation by power supply

The level of the strongest unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains shall be less than –45 dBm0s with a test signal of 1 kHz at alignment level 0 dBm0s.

3.1.6 Non-linear distortion

3.1.6.1 Harmonic distortion

The total harmonic distortion (THD) shall be measured with the input signal at +9 dBm0s.

The duration for which a single tone is to be transmitted at this level should be restricted in accordance with CCITT Recommendations N.21 and N.23.

* Administrations are urged to supply additional information on an appropriate value.

The THD when measured with a true RMS meter shall not be greater than the following values:

TABLE III

Input frequency (kHz)	THD
$0.05 \leq f < 0.1$	2% (–25 dBm0s)
$0.1 \leq f \leq 2.0$	1.4% (–28 dBm0s)

Note. – If THD cannot be measured directly, compliance is considered to be fulfilled if the second or third harmonics are measured selectively and a calculated value, k , meets the requirement:

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

where:

k_2 : second harmonic coefficient and

k_3 : third harmonic coefficient.

3.1.6.2 *Intermodulation*

With input signals of 0.8 kHz and 1.42 kHz, at a level of +3 dBm0s, the third order difference tone at 0.18 kHz shall be less than 1.4% (–34 dBm0s).

3.1.6.3 *Distortion products measured by shaped noise* (under study. See Report 640-1 (Kyoto, 1978)).

3.1.7 *Error in reconstituted frequency* (applies only to FDM systems)

Not to be greater than 1 Hz.

Note. – A maximum error of 1 Hz is in principle acceptable where there is only a single transmission path between the signal source and the listener.

Where the broadcast network can involve two or more parallel paths, e.g. commentary and separate sound channels, or radio broadcast from different transmitters on the same frequency, unacceptable beats may occur unless zero error can be assured. The CCITT is studying methods of effecting this in all recommended systems.

3.1.8 *Intelligible cross-talk ratio*

3.1.8.1 The intelligible near-end and far-end cross-talk ratios between sound-programme circuits, or from a telephone circuit (disturbing) into a sound-programme circuit (disturbed) shall be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and shall not be less than the following values:

TABLE IV

Frequency (kHz)	Cross-talk attenuation
$f < 0.5$	Slope 6 dB/octave
$0.5 \leq f \leq 3.2$	74 dB
$f > 3.2$	Slope –6 dB/octave

3.1.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1. – It is understood that this value is defined between the relative levels applicable to telephone circuits. Administrations are invited to submit contributions or methods for measuring this parameter.

Note 2. — The attention of administrations is drawn to the fact that in some cases it may be difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or at low frequencies (e.g. below about 100 kHz) on certain coaxial cable carrier systems. If sub-standard performance is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

Note 3. — When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

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

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

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

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

3.1.9 Amplitude linearity

When a 1 kHz input signal is stepped from -6 dBm0s to $+6$ dBm0s or *vice versa*, the output level shall change accordingly by 12 ± 0.5 dB.

3.2 Additional parameters for stereophonic programme transmission

Not applicable, this section concerns 15 kHz type sound-programme circuits (see Recommendation 505).

3.3 Additional requirements for digital systems

3.3.1 If a test signal is harmonically related to the sampling frequency, measuring difficulties may arise. In this case, the nominal 1 kHz test signal must be offset. CCITT Recommendation O.33 recommends 1020 Hz.

3.3.2 Unbalance of the limitation level

The difference between the levels which lead to a limitation of the positive or negative half-wave of the test signal, shall not exceed 1 dB.

3.3.3 Intermodulation with the sampling signal

Intermodulation products (f_d) caused by non-linearities may occur in the sound channel when the sampling signal (f_0) is combined with the in-band audio signals (f_i) or out-of-band interfering signals (f_a).

3.3.3.1 In-band intermodulation

The following combination rule applies: $f_d = f_0 - nf_i$.

Only values with $n = 2$ or 3 are of importance.

The level difference between a 0 dBm0s signal (f_i) and the intermodulation products (f_d) shall not be less than 40 dB.

A restriction to the following f_i and f_d values is sufficient:

TABLE V

	$n = 2$		$n = 3$	
f_i (kHz)	5	7	3	5
f_d (kHz)	6	2	7	1

3.3.3.2 Out-of-band intermodulation

The following combination rule applies: $f_d = nf_0 \pm f_a$.

Only values with $n = 1$ or 2 are of importance.

The level difference between a 0 dBm0s signal (f_a) and the intermodulation products (f_d) shall not be less than 60 dB.

A restriction to the following f_a and f_d values is sufficient:

TABLE VI

	$n = 1$		$n = 2$	
f_a (kHz)	15	17	31	33
f_d (kHz)	1			

3.3.4 Further parameters

Characteristics for bit errors, clicks, jitter, etc. are under study (see Study Programme 18A/CMTT and Report 647).

BIBLIOGRAPHY

CCIR Documents

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

RECOMMENDATION 505-3*

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

Circuits for high quality monophonic and
stereophonic transmissions

(Question 17/CMTT)

(1974-1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that it is necessary to set transmission standards for sound-programme circuits;
- (b) that quality requirements for the hypothetical reference circuit are established for analogue sound programmes;
- (c) that advantage should be taken of the technical evolution made possible by the introduction of digital techniques, particularly for mixed analogue and digital circuits,

UNANIMOUSLY RECOMMENDS

that, with due regard to the application constraints, equipment for new circuits shall meet the requirements laid out below.

1. Application

This Recommendation applies to homogeneous analogue or mixed analogue-and-digital circuits.

The requirements below apply to the hypothetical reference circuit (HRC) defined in Recommendation 502.

* This Recommendation is also proposed by the CMTT as a revision of CCITT Recommendation J.21 and should be brought to the attention of CCITT Study Group XV.

** For the definition of absolute power, relative power and noise levels, see Recommendation 574.

For estimation of the performance of circuits shorter or longer than the HRC, see Recommendation 605.

Note 1. — For all-digital circuits, a separate Recommendation might be envisaged after further study.

Note 2. — For further information, Report 496 may be consulted. This Report also draws attention to certain differences between CCIR and OIRT Recommendations.

2. Interface characteristics

2.1 Test conditions

When circuit performance is to be measured, the system output shall be terminated by a balanced test load, nominally 600 Ω resistive.

2.2 Impedance

System input impedance 600 Ω balanced *

System output impedance, provisionally Low, balanced

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

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

This clause alone would however not rule out a large difference in the reactive parts of the output impedances of a stereophonic pair, and this in turn could lead to difficulties in meeting § 3.2.2. This aspect needs further study.

2.3 Levels

Input maximum programme level + 9 dBm0s

Insertion gain (1 kHz at -12 dBm0s) 0 dB

Adjustment error, within ± 0.5 dB

Variation over 24 hours not to exceed ± 0.5 dB

Relative level (see CCITT Recommendation J.14) + 6 dBrs

If the broadcast organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional trimming attenuators.

3. Overall performance

3.1 Common parameters

3.1.1 Gain/frequency response

Reference frequency: 1 kHz (nominal value)

The response shall be measured at: -12 dBm0s

TABLE I

Frequency (kHz)	Response (dB)
$0.04 \leq f < 0.125$	+0.5 to -2
$0.125 \leq f \leq 10$	+0.5 to -0.5
$10 < f \leq 14$	+0.5 to -2
$14 < f \leq 15$	+0.5 to -3

If broadcasting organizations wish to have closer tolerances, it is necessary for the receiving broadcasting organization to insert additional equalizers.

* The tolerance, permitted reactance and degree of unbalance need further study.

3.1.2 Group delay variation

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

kHz	$\Delta\tau$ (ms)
0.04	55
0.075	24
14	8
15	12

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

3.1.3 Noise

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

For radio-relay systems, the requirements shall be met for at least 80% of the total time of any 30-day period. For 1% of the time an additional impairment of 4 dB, and for 0.1% of the time an additional impairment of 12 dB is acceptable.

TABLE II

Noise	Transmission system	
	Analogue	Digital (3 codecs cascaded)
Idle channel noise, maximum (dBq0ps)	-42	-51
Programme-modulated noise, maximum (dBq0ps)	-30	-39

Programme-modulated noise usually only occurs on sound programme circuits which are equipped with companders (e.g. types of circuits corresponding to CCITT Recommendation J.31).

This noise value may be measured with the aid of an auxiliary sinusoidal test signal +9 dBm0s/60 Hz which has to be suppressed by a high-pass filter ($f_0 \leq 400$ Hz, $a \geq 60$ dB/60 Hz) before the measuring set.

Report 493 indicates that if a compander is used, an improved signal-to-noise ratio is necessary to avoid objectional effects with some programme material.*

Note. — For digital systems, appropriate values are under study. For further information, see Report 647.

3.1.4 Single tone interference

Level of any individual tone:

$$\leq -(73 + \psi) \quad \text{dBm0s}$$

where ψ is the weighting filter response factor (positive or negative) as given in Recommendation 468 at the particular frequency.

For sound-programme transmissions over carrier systems, occurrence of carrier leaks can be expected. For this reason, stop filters may be provided in the carrier frequency path which can be switched in, if required, to suppress the tones otherwise audible in the upper frequency range from 8 to 15 kHz. For a hypothetical reference circuit, it is recommended that the stop filters should have a 3 dB bandwidth of less than 3% referred to the mid-frequency. The use of stop filters which affect frequencies below 8 kHz should be avoided.

* Administrations are urged to supply additional information on an appropriate value.

3.1.5 *Disturbing modulation by power supply*

The level of the strongest unwanted side component due to modulation caused by low-order interference components from 50 Hz or 60 Hz mains shall be less than -45 dBm0s with a test signal of 1 kHz at alignment level 0 dBm0s.

3.1.6 *Non-linear distortion*

3.1.6.1 *Harmonic distortion*

The total harmonic distortion (THD) shall be measured with the input signal at $+9$ dBm0s for frequencies up to 2 kHz, and at $+6$ dBm0s for frequencies above 2 kHz up to 4 kHz.

The duration for which a single tone is to be transmitted at these levels should be restricted in accordance with CCITT Recommendations N.21 and N.23.

The THD when measured with a true RMS meter shall not be greater than the following values:

TABLE III

Input frequency (kHz)	THD	Second and third harmonic measured selectively
$0.04 \leq f < 0.125$	1% (-31 dBm0s)	0.7% (-34 dBm0s)
$0.125 \leq f \leq 2.0$	0.5% (-37 dBm0s)	0.35% (-40 dBm0s)
$2.0 < f \leq 4.0$	0.5% (-40 dBm0s)	0.35% (-43 dBm0s)

3.1.6.2 *Intermodulation*

With input signals of 0.8 kHz and 1.42 kHz, each at a level of $+3$ dBm0s, the third order difference tone at 0.18 kHz shall be less than 0.5% (-43 dBm0s).

Note. — Attention is drawn to the fact that in transmission systems using compandors, a 3rd order difference-frequency may occur which exceeds the specified limit of 0.5%. This may occur when the difference between the two fundamental frequencies is less than 200 Hz. Thus, the components due to 3rd order distortion will have frequencies which correspond to the difference between the two test frequencies. However, in these cases the subjective masking is such that a distortion up to 2% is acceptable.

For 15 kHz systems, intended for baseband transmissions on physical circuits only, and on modulation equipment in local loops, assuming no pre-emphasis, the following additional requirements apply:

TABLE IV

Input signals at $+3$ dBm0s each	Maximum difference-tone level at 1.6 kHz
5.6 kHz and 7.2 kHz	0.5% (-43 dBm0s) (second order)
4.2 kHz and 6.8 kHz	0.5% (-43 dBm0s) (third order)

3.1.6.3 *Distortion products measured by shaped noise* (under study. See Report 640-1 (Kyoto, 1978)).

3.1.7 *Error in reconstituted frequency* (applies only to FDM systems)

Not to be greater than 1 Hz.

Note. — A maximum error of 1 Hz is in principle acceptable where there is only a single transmission path between the signal source and the listener.

Where the broadcast network can involve two or more parallel paths, e.g., commentary and separate sound channels, or radio broadcast from different transmitters on the same frequency, unacceptable beats may occur unless zero error can be assured. The CCITT is studying methods of effecting this in all recommended systems.

3.1.8 *Intelligible cross-talk ratio*

3.1.8.1 The intelligible near-end and far-end cross-talk ratios between sound-programme circuits, or from a telephone circuit (disturbing) into a sound-programme circuit (disturbed) shall be measured selectively in the disturbed circuit at the same frequencies as those of the sinusoidal test signal applied to the disturbing circuit, and shall not be less than the following values:

TABLE V

Frequency (kHz)	Cross-talk attenuation
$f = 0.04$	50 dB
$0.04 < f < 0.05$	Oblique straight-line segment on linear-decibel and logarithmic-frequency scales
$0.05 \leq f \leq 5$	74 dB
$5 < f < 15$	Oblique straight-line segment on linear-decibel and logarithmic-frequency scales
$f = 15$	60 dB

3.1.8.2 The near-end and far-end cross-talk attenuations between a sound-programme circuit (disturbing circuit) and a telephone circuit (disturbed circuit) shall be at least 65 dB.

Note 1. — It is understood that this value is defined between the relative levels applicable to telephone circuits. Administrations are invited to submit contributions on methods for measuring this parameter.

Note 2. — The attention of administrations is drawn to the fact that in some cases it may be difficult or impossible to meet these limits. This may occur when unscreened pairs are used for a long audio-frequency circuit (e.g. about 1000 km or longer), or in certain carrier systems on symmetric pair cables, or at low frequencies (e.g. below about 100 kHz) on certain coaxial cable carrier systems. If sub-standard performance is to be avoided, such systems or parts of systems must not be used for setting up programme channels.

Note 3. — When 4000 pW0p or more noise is continuously present in the telephone channel (this may be the case in satellite systems, for example), a reduced cross-talk ratio of 58 dB between a sound-programme circuit and a telephone circuit is acceptable.

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

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

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

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

3.1.9 Amplitude linearity

When a 1 kHz input signal is stepped from -6 dBm0s to $+6$ dBm0s or *vice versa* the output level shall change accordingly by 12 ± 0.5 dB.

3.2 Additional parameters for stereophonic programme transmission

3.2.1 The difference in gain between A and B channels shall not exceed the following values:

TABLE VI

Frequency (kHz)	Gain difference (dB)
$0.04 \leq f < 0.125$	1.5
$0.125 \leq f \leq 10$	0.8
$10 < f \leq 14$	1.5
$14 < f \leq 15$	3.0

3.2.2 The phase difference between the A and B channels shall not exceed the following values:

TABLE VII

Frequency (kHz)	Phase difference
$f = 0.04$	30°
$0.04 < f < 0.2$	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
$0.2 \leq f \leq 14$	15°
$4 < f < 14$	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
$f = 14$	30°
$14 < f < 15$	Oblique straight-line segment on linear-degree and logarithmic-frequency scales
$f = 15$	40°

3.2.3 The cross-talk ratio between the A and B channels shall not be less than the following limits:

3.2.3.1 Intelligible cross-talk ratio, measured with sinusoidal test signal 0.04 to 15 kHz: 50 dB.

3.2.3.2 Total cross-talk ratio predominantly caused by intermodulation: 60 dB

This value is ascertained by loading one of the two channels with the sound-programme simulating signal, defined in Recommendation 571. In the other channel, the noise contribution due to intermodulation shall not be higher than -51 dBq0ps.

This leads to an increase of noise depending on the idle channel noise value. The tolerable increase is given in Table VIII.

TABLE VIII

Idle channel noise (dBq0ps)	-60	-57	-54	-51	-48	-45	-42
Tolerable increase of noise (dB)	9.5	7	4.8	3	1.8	1.0	0.5

3.3 Additional requirements for digital systems

3.3.1 If a test signal is harmonically related to the sampling frequency, measuring difficulties may arise. In this case, the nominal 1 kHz test signal must be offset. CCITT Recommendation O.33 recommends 1020 Hz.

3.3.2 Unbalance of the limitation level

The difference between the levels which lead to a limitation of the positive or negative half-wave of the test signal, shall not exceed 1 dB.

3.3.3 Intermodulation with the sampling signal

Intermodulation products (f_d) caused by non-linearities may occur in the sound channel when the sampling signal (f_0) is combined with the in-band audio signals (f_i) or out-of-band interfering signals (f_a).

3.3.3.1 In-band intermodulation

The following combination rule applies: $f_d = f_0 - nf_i$.

Only values with $n = 2$ or 3 are of importance.

The level difference between a 0 dBm0s signal (f_i) and the intermodulation products (f_d) shall not be less than 40 dB.

A restriction to the following f_i and f_d values is sufficient:

TABLE IX

	$n = 2$		$n = 3$	
f_i (kHz)	9	13	7	11
f_d (kHz)	14	6	11	1

3.3.3.2 Out-of-band intermodulation

The following combination rule applies: $f_d = nf_0 \pm f_a$.

Only values with $n = 1$ or 2 are of importance.

The level difference between a 0 dBm0s signal (f_a) and the intermodulation products (f_d) shall not be less than 60 dB.

A restriction to the following f_a and f_d values is sufficient:

TABLE X

	$n = 1$		$n = 2$	
f_a (kHz)	31	33	63	65
f_d (kHz)	1			

3.3.4 Further parameters

Characteristics for bit errors, clicks, jitter, etc. are under study (see Study Programme 18A/CMTT and Report 647).

Note. — The CCIR has issued Recommendation 572 which deals with the transmission of one sound-programme associated with an analogue television signal by means of time-division multiplex in the line synchronizing pulse. The system recommended is a digital one, using pulse code modulation. A sound-programme bandwidth of 14 kHz is provided.

BIBLIOGRAPHY

CCIR Documents

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

REPORT 496-4*

CIRCUITS FOR HIGH-QUALITY MONOPHONIC AND STEREOPHONIC TRANSMISSIONS**

(Questions 17/CMTT and 19/CMTT, Study Programme 19B/CMTT)

(1970-1974-1978-1982-1986)

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. 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.

* 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.

TABLE I – Limiting values of parameters of high-quality circuits for monophonic and stereophonic transmission

Parameter	OIRT 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	$\leq -(73 + \psi)$ ⁽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

(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 dBrs 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) Where ψ is the weighting filter response factor (positive or negative) as given in Recommendation 468 at the particular frequency.

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.

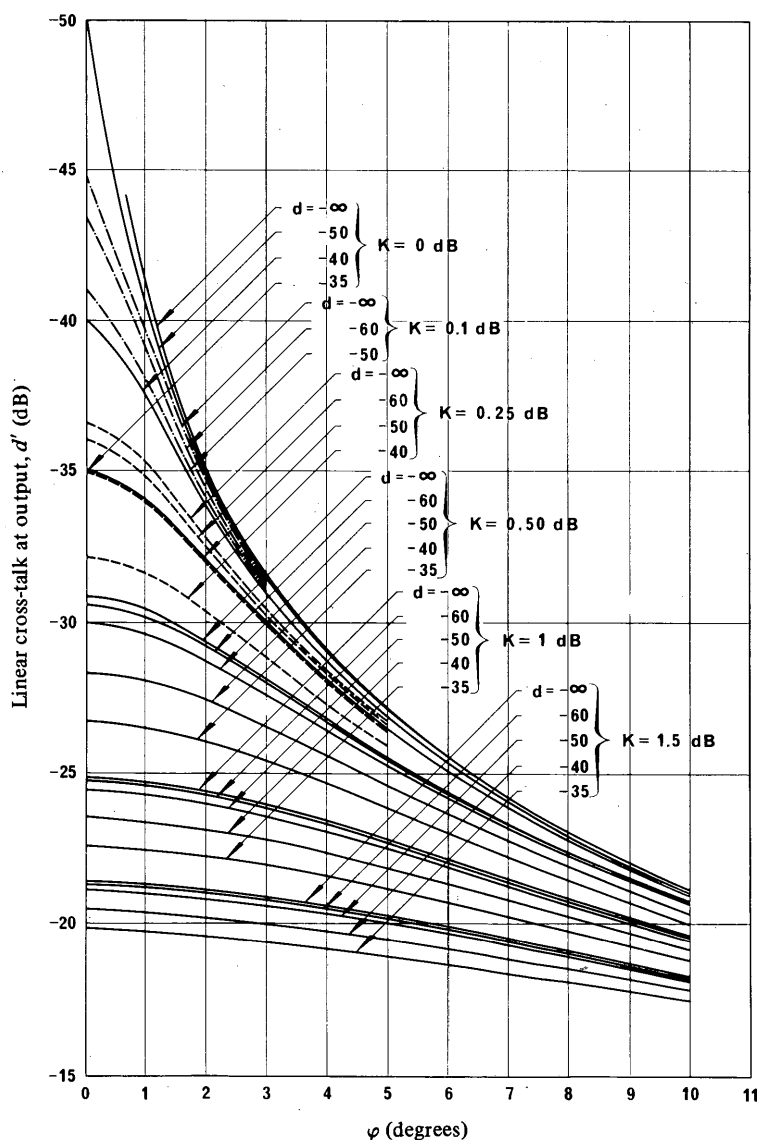


FIGURE 1 — Admissible cross-talk at the input of a transmitter-to-receiver chain

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.

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 can be 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 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

Spectral distribution of noise from system	Upper frequency limit of transmission of the circuit under consideration (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).

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.

The pre-emphasis curve calculated from this formula passes through the following points:

TABLE III

<i>f</i> (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 measurement of error in reconstituted frequency*

Recommendations 503 and 505 specify the error in reconstituted frequency to be not greater than 1 Hz and note that unacceptable beats may occur in broadcast networks involving two or more paths unless zero error can be assured.

The general subject of equipment for measuring the performance of sound-programme circuits is dealt with by CCITT Study Group IV. CCITT Recommendation O.111 specifies the essential characteristics of instruments for measuring the frequency shift caused by carrier transmission and the use of sinusoidal test frequencies having a 2 : 1 harmonic ratio for this purpose. However, when the harmonic ratio between the two test frequencies is small, it can take a long period of time to measure a small frequency shift.

Another measuring method has been proposed [CCIR, 1982-86a] which enables the frequency shift to be determined to a tolerance of ± 0.01 Hz within 3.6 s.

The method proposes that the ratio of the two measurement frequencies is made as large as possible, thus making the measuring time shorter while remaining within the bandwidth of all FDM (carrier) programme channels likely to be encountered [Thiele and Bonner, 1985].

Readings taken in Australia with this equipment indicate that sound-programme circuits can easily meet Recommendations 503 and 505.

2.4 *The measurement of non-linearity in the upper frequency range*

In a bandwidth-limited system such as a transmission channel, harmonic distortion measurements are not suitable for testing the amplitude linearity of the system.

For instance, in a 15 kHz channel, the measurement of third harmonic distortion above 5 kHz becomes meaningless.

Experience has shown that non-linearities may occur at high frequencies, even if the level of the signal is kept within the permitted limits (slew-rate limitations in operational amplifiers for example).

The non-linearity measurement at high frequencies may be performed by means of “difference-frequency distortion” methods which consist of simultaneously sending two sinusoidal signals. The frequencies, f_1 and f_2 ($f_2 > f_1$), are normally of the same order of magnitude and the amplitude levels are usually identical.

The particular difference components at frequencies $f_2 - f_1$ and $2f_1 - f_2$, which fall within the system bandwidth, are used to reflect the second and third order non-linearities, respectively. The use of such measuring methods has been proposed in [CCIR, 1982-86b and c].

The first text describes a method where the two measuring frequencies are very close together.

The second proposes frequencies further apart. The measuring equipment may use either selective filters to improve the protection against noise, or a low-pass filter to measure simultaneously distortions of various orders including other intermodulation components (e.g. intermodulation with PCM sampling frequencies, or in the FDM multiplex).

These two methods are summarized in Table IV. The method included in Recommendation 505 (using much lower frequencies) and another method [Thiele, 1983] are also shown. Significant differences exist regarding the maximum frequencies used, the relative spacing of these frequencies, and the reference level to which the distortion components are referred. Considering these differences, and the importance of applying such measurement methods, administrations are invited to contribute on this subject during the next study period.

2.5 *The use of limiters on sound-programme circuits*

Limiters for high quality sound-programme circuits:

Study Group 10 has prepared Recommendation 642 “Limiters for high-quality sound-programme signals”. The Recommendation is based on a United Kingdom proposal [CCIR, 1978-1982; 1982-86d].

TABLE IV — *Proposed methods of measurement*

Reference	Frequencies (Hz)		Proposed sending level of each tone ⁽¹⁾ (N_0)	Second order term		Third order term	
	f_1	f_2		Frequency (Hz)	Reference level	Frequency (Hz)	Reference level
Recommendation 505	5 600 4 200	7 200 6 800	+3 dBm0s +3 dBm0s	1600	N_0	1 600	N_0
[CCIR, 1982-86b]	13 960	14 040	−9 dBm0s	80	+9 dBm0s ($N_0 + 18$ dB)	13 880	+3 dBm0s ($N_0 + 12$ dB)
[CCIR, 1982-86c]	10 400	13 400		3000	$N_0 + 3$ dB	7 400	$N_0 + 3$ dB
[Thiele, 1983]	8 000	11 950		3950	N_0	4 050	N_0

⁽¹⁾ These levels depend significantly on the pre-emphasis used.

In Recommendation 505, the levels quoted are for no pre-emphasis.

In [CCIR, 1982-86b], the levels quoted are for 50 μ s pre-emphasis.

REFERENCES

- THIELE, A. N. [June, 1983] Measurement of non-linear distortion in a band-limited system. *J. Audio Eng. Soc.*, Vol. 31, 6, 443-445.
- THIELE, A. N. and BONNER, D. J. [30 September-4 October, 1985] Measuring error in reconstituted frequency multiplexed sound program circuits. Convention Digest, 684-686. IREE International Electronics Convention and Exhibition (IREECON '85), Melbourne, Australia.
- CCIR Documents*
- [1966-69]: CMTT/165 (Italy).
- [1970-74]: CMTT/184 (OIRT).
- [1978-82]: CMTT/239 (United Kingdom).
- [1982-86]: a. CMTT/208 (Australia); b. CMTT/175(10/213) (Sweden); c. CMTT/182(10/224) (France); d. CMTT/35(10/26) (United Kingdom).

BIBLIOGRAPHY

CCIR Documents

- [1966-69]: CMTT/10 (EBU); CMTT/61 (France); CMTT/63 (Italy); CMTT/128 (United Kingdom); CMTT/133 (Netherlands); CMTT/135 (Japan); CMTT/145 (Italy); CMTT/162 (Federal Republic of Germany); CMTT/163 (France); CMTT/164 (France)

REPORT 491-3

CHARACTERISTICS OF SIGNALS SENT OVER SOUND-PROGRAMME CIRCUITS

(Study Programme 17A/CMTT, Kyoto, 1978)

(1970-1974-1978-1986)

1. Characteristics of test signals

The first point of the former Study Programme 17A/CMTT (Kyoto, 1978) was concerned with the characteristics of the test signals used by broadcasting authorities on sound-programme circuits and considered the possibility of new types of test signals.

In [CCIR, 1966-69a], it was also suggested that new test signals might be wanted, in particular for non-linearity tests and for stereophonic circuits. It also pointed 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 in Recommendation 571, Report 497 and in Report 640-1 (Kyoto, 1978).

2. Mean power and distribution of instantaneous levels of programme signals

The second point of the Study Programme 17A/CMTT (Kyoto, 1978) was 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, 1966-69b] 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 should be noted that Recommendation 571 and Report 497 have proposed 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, 1970-74a]) 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):
 $-15 + 10 \log 12 \approx -4 \text{ dBm}_0$
- conventional load:
 $-1 + 4 \log 12 = +3.3 \text{ dBm}_0$
- peak level exceeded with a probability of 10^{-5} times:
 $+19 \text{ dBm}_0$.

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, 1970-74a]) 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 compandors 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, 1970-74b]. 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.

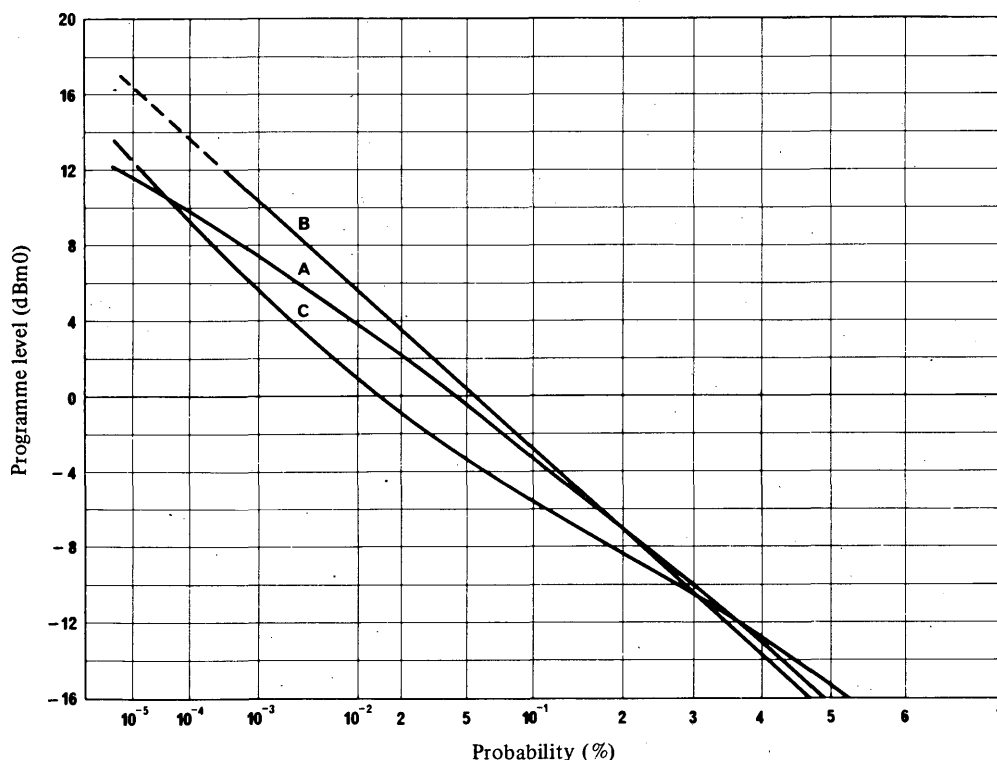


FIGURE 1 – Cumulative distribution
Cases A, B and C: Distribution of the programme level in 100% of the time

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, 1970-74b] 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, 1970-74b]. 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].

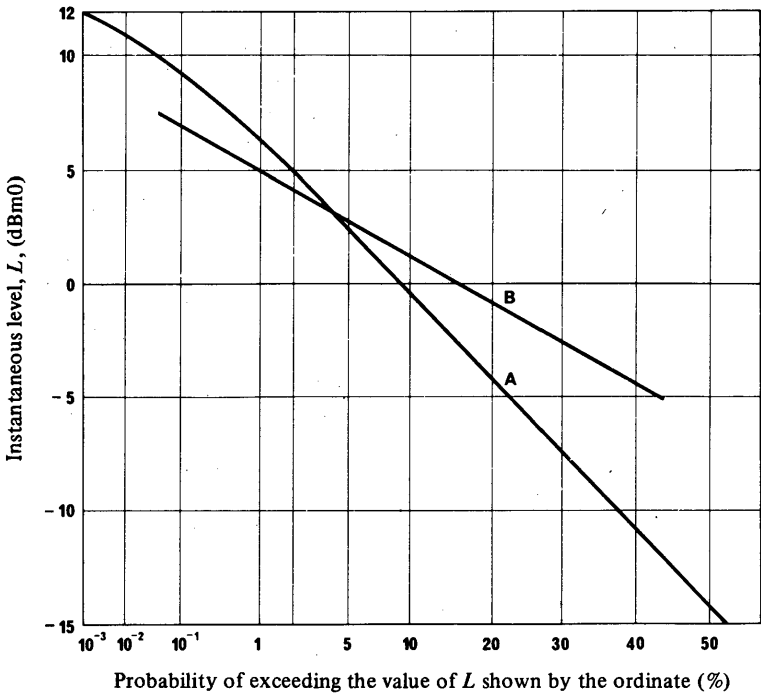


FIGURE 2 – Distribution of instantaneous level, L , expressed in equivalent sine-wave powers of the signal in a programme channel at audio frequencies (Curve A)

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]:

TABLE V

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 (this is the nominal peak indicated value used in the BBC), as indicated upon a peak programme meter (see Note) and no additional programme signal limiters were employed.

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.

Note. – The characteristics of this meter are shown as item 4 in the table of CCITT Recommendation J.15.

3. Spectral power distribution

Study Programme 17A/CMTT (Kyoto, 1978), § 6 was concerned with the spectral power distribution of sound-programme signals.

The following information has been made available.

3.1 [CCIR, 1966-69c] gives some spectral power distributions 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.

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.

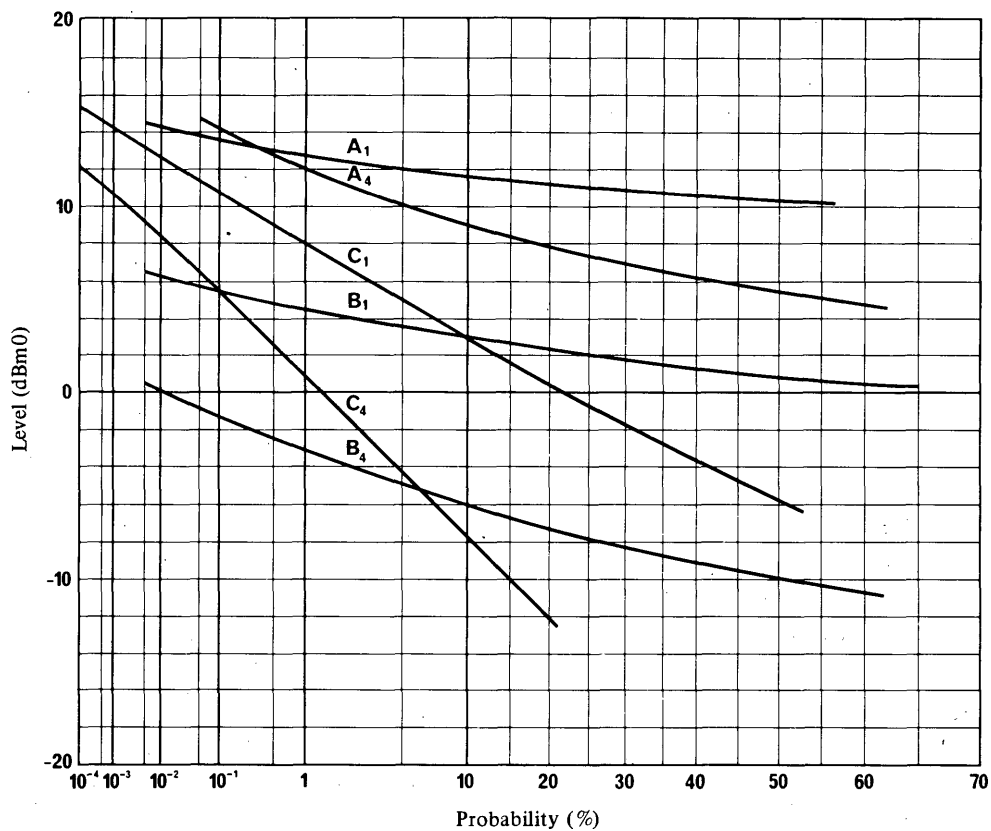
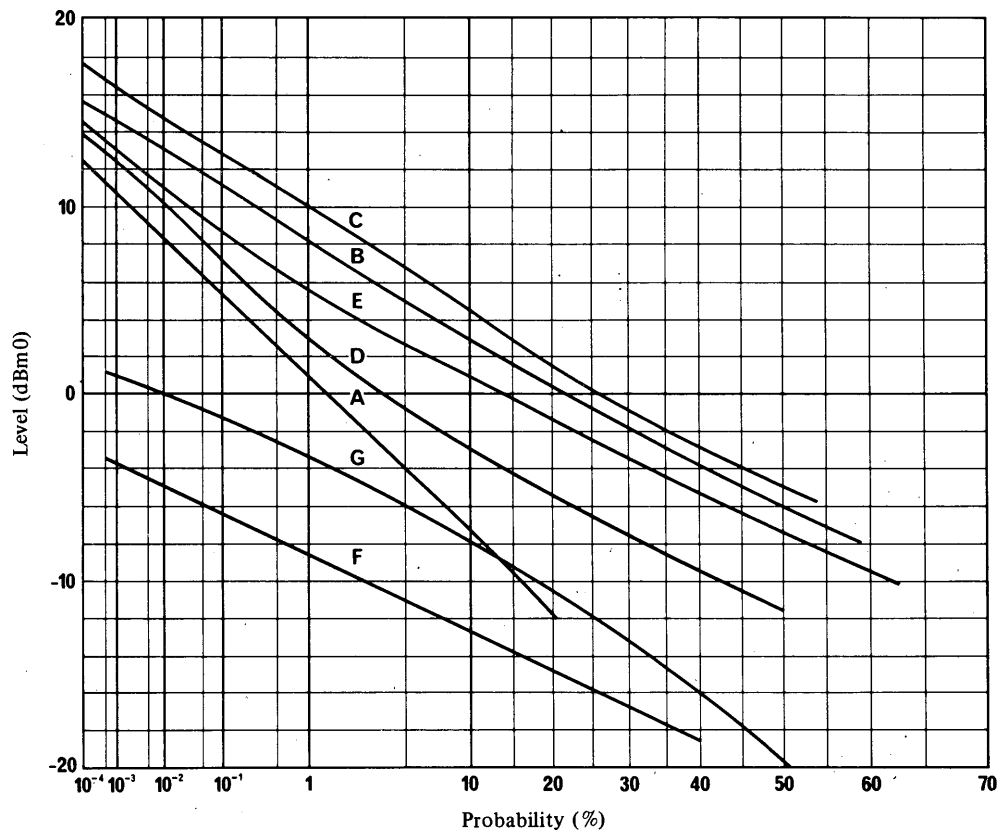


FIGURE 3 – *Characteristics of unprocessed signals*

- 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)

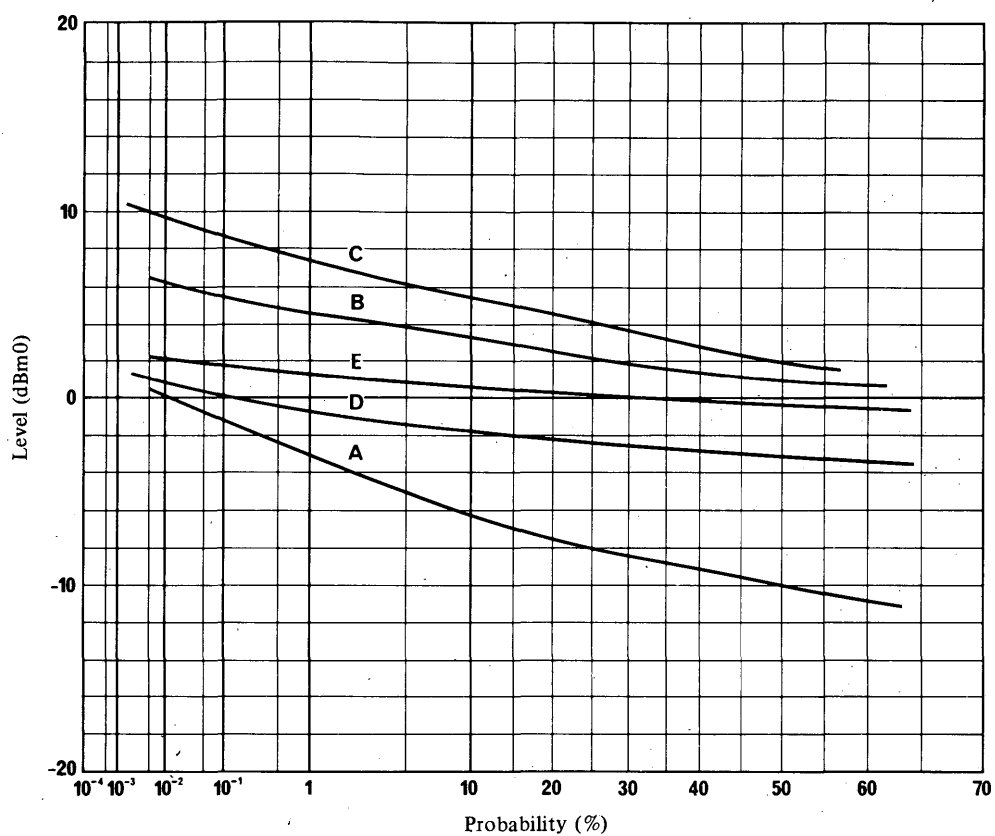


FIGURE 5 – 1-Minute mean powers

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

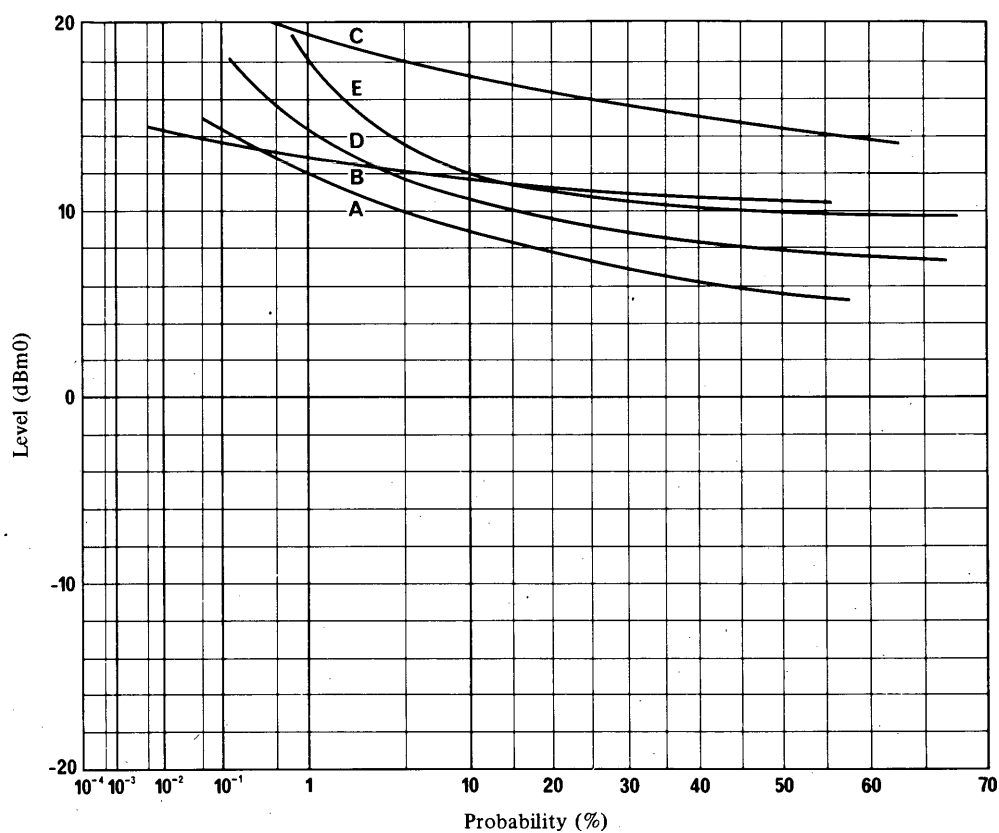


FIGURE 6 – Peak powers occurring within each 1-minute period

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

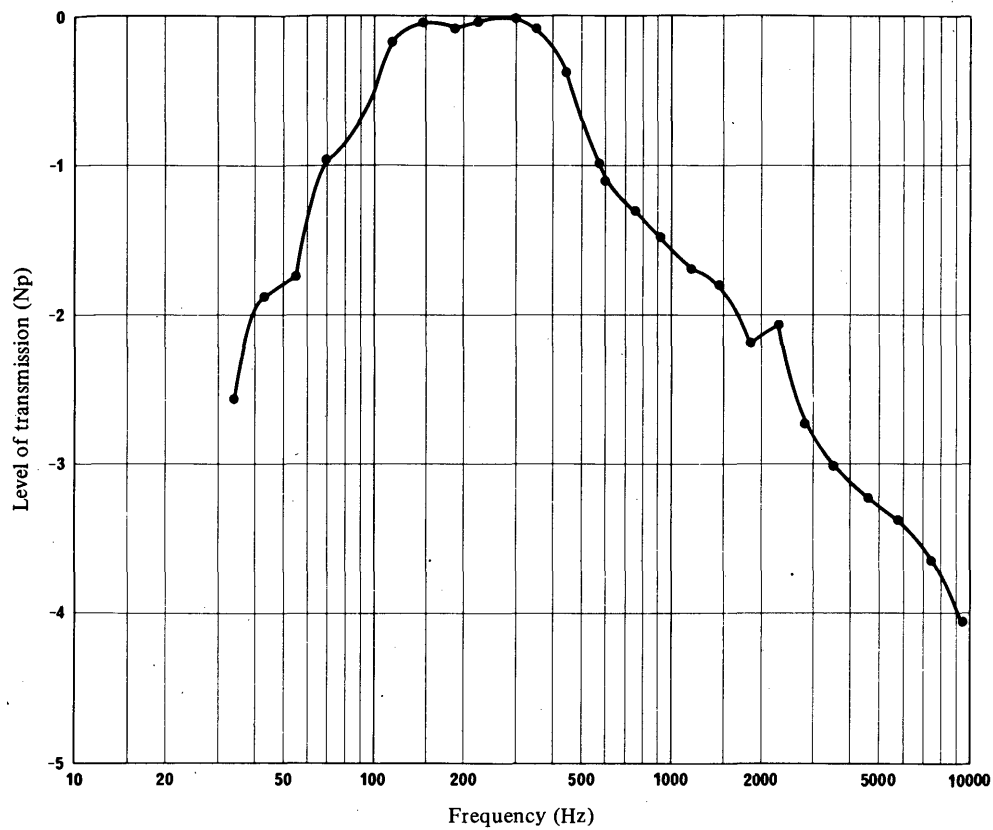


FIGURE 7 – Averaged spectral characteristics of level of transmission of male speech

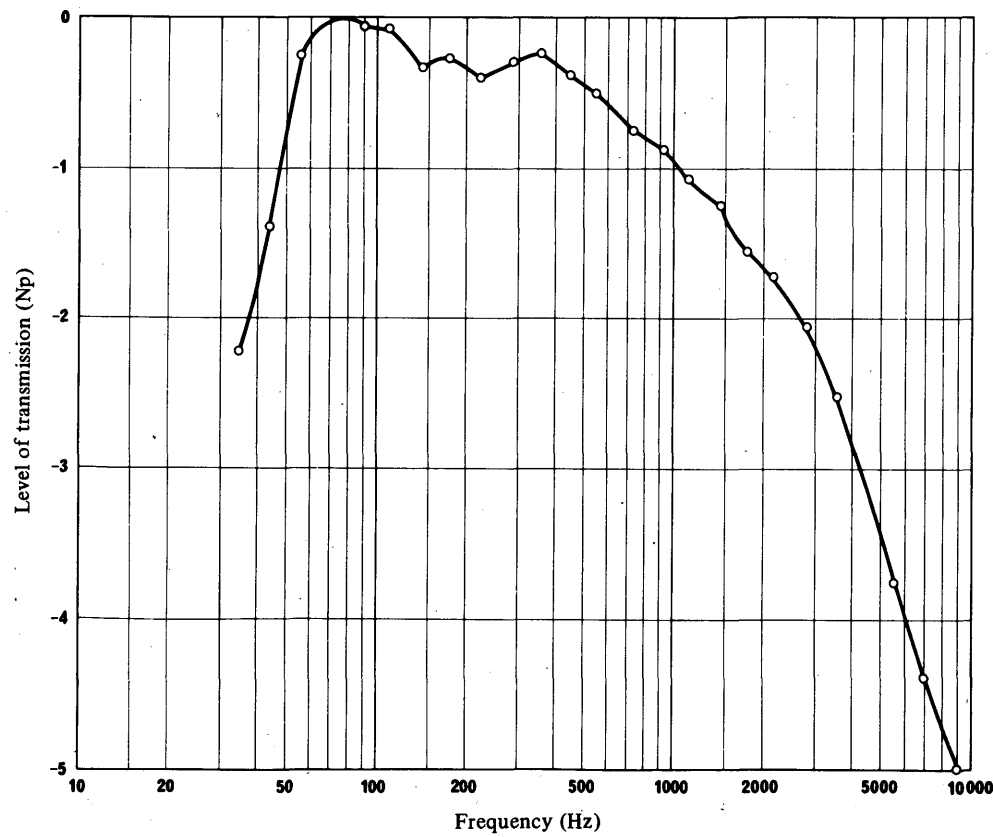


FIGURE 8 – Averaged spectral characteristics of level of transmission of a symphony orchestra

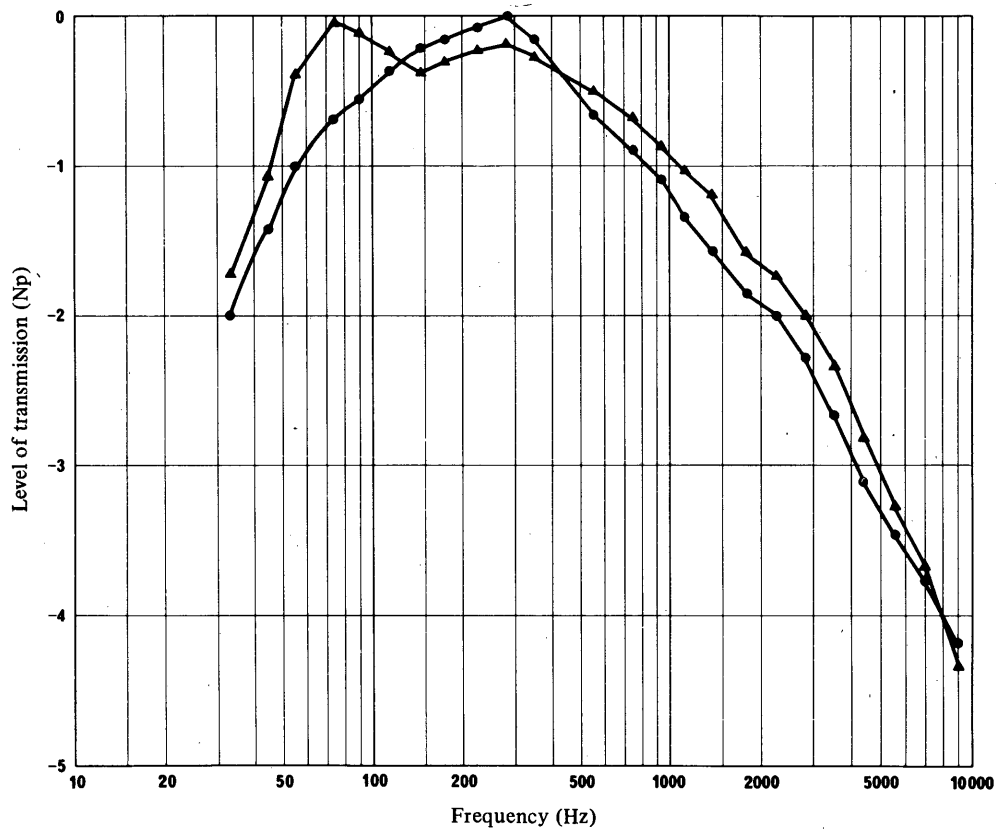


FIGURE 9 – Averaged spectral characteristics of level of transmission of daily broadcasting programme, taking account of the incidence of transmissions with different characteristics

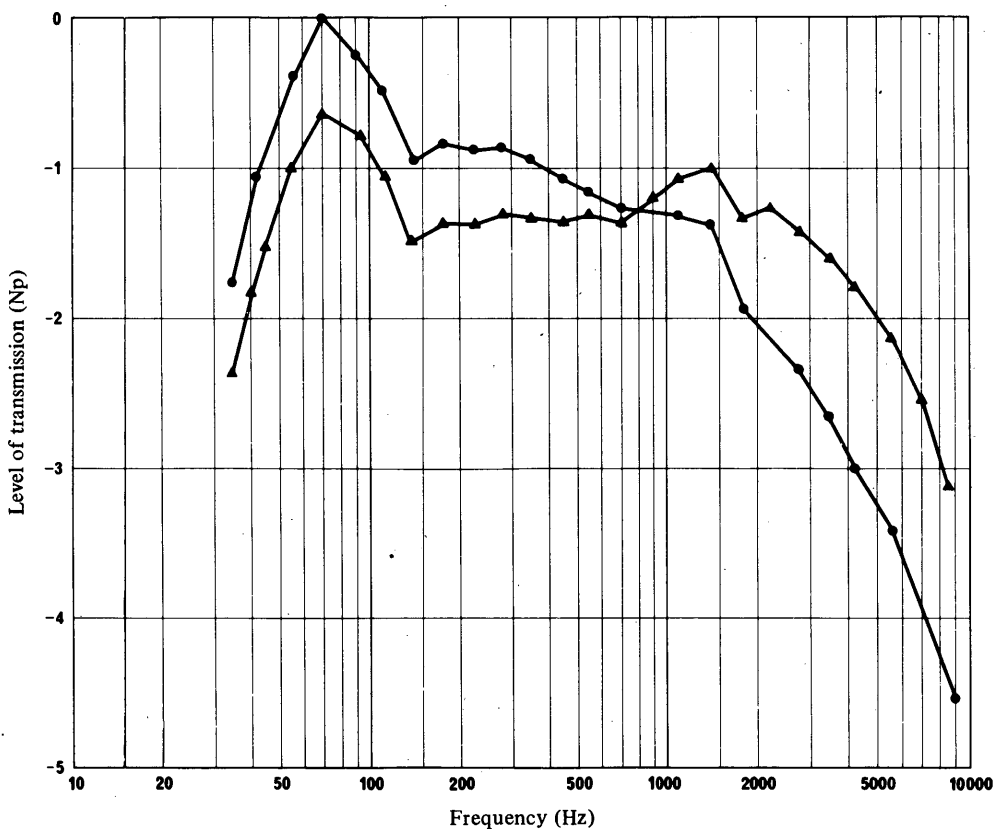


FIGURE 10 – Averaged spectral characteristics of level transmission of a light orchestra with the pre-emphasis circuit switched off and with it switched on

3.2 Figures 11 and 12 show the spectral average power distribution for some different programme items recorded on magnetic tape [CCIR, 1966-69d].

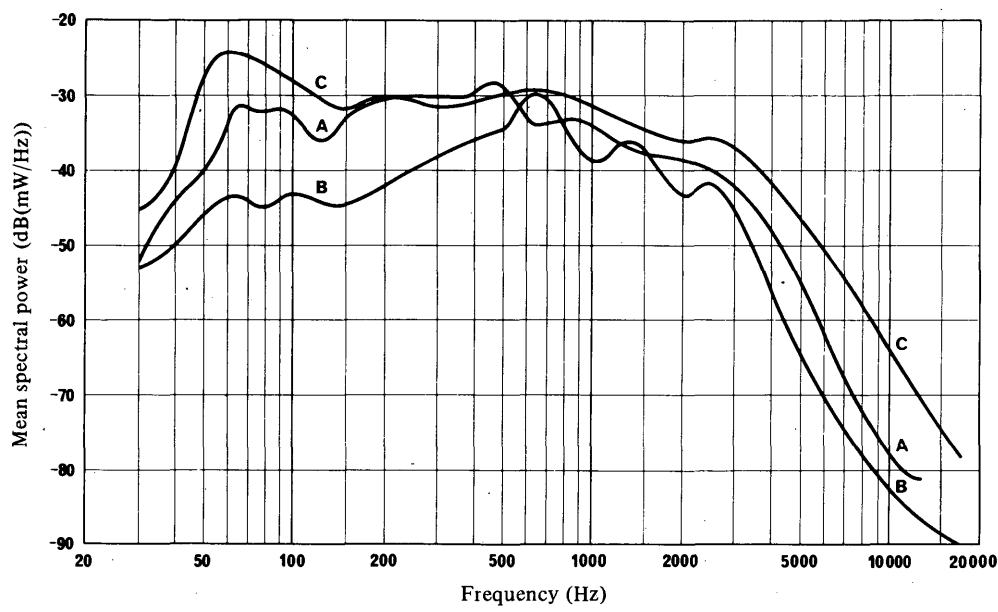


FIGURE 11 – Mean spectral power of sound signals

Curve A : classical music Curve B : opera Curve C : modern music

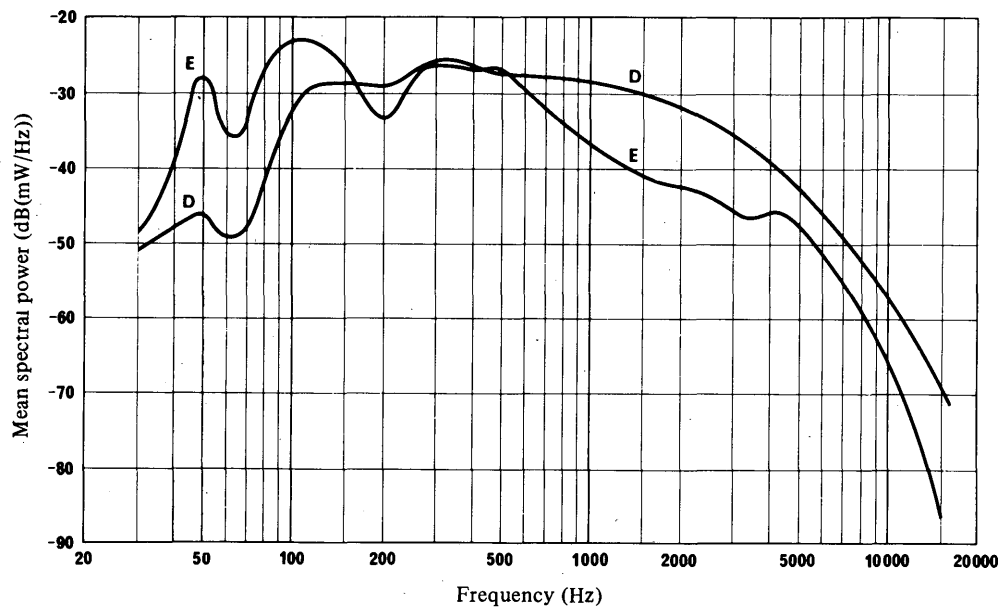


FIGURE 12 – Mean spectral power of sound signals

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, 1970-74c] 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, 1970-74d].

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) was 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.

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].

It has been reported that close agreement, in terms of the statistical distributions, of the model and actual sound-programme signals has been confirmed with data shown in this Report [Ehara, 1982].

REFERENCES

- EHARA, S. [January, 1977a] Some applications of programme-model signals. NHK Lab. Note No. 207.
- EHARA, S. [July, 1977b] A stochastic process model of orchestra music and speech. 9th ICA, Madrid 4/9-VII-1977.
- EHARA, S. [January, 1982] Spectral amplitude and power distributions of sound-program signals. NHK Lab. Note No. 269.
- JAKUBOWSKI, H. [1971] Analyse des Programm-Materials des Hörrundfunks (Analysis of sound broadcasting programme material). *Rundfunktechn. Mitt.*, Vol. 13, 6.
- WEISS, W. [July, 1964] Statistische Untersuchungen an Sprachschwingungen. *NTZ*, Heft 7.
- CCIR Documents*
- [1966-69]: a. CMTT/31 (LM Ericsson); b. CMTT/22 (United Kingdom); c. CMTT/153 (USSR); d. CMTT/158 (Italy).
- [1970-74]: a. CMTT/52 (Germany (Federal Republic of)); b. CMTT/27 (Switzerland); c. CMTT/36 (Germany (Federal Republic of)); d. CMTT/47 (France).
- [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

2.1 Introduction of the model

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 a continuous spectrum.

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.

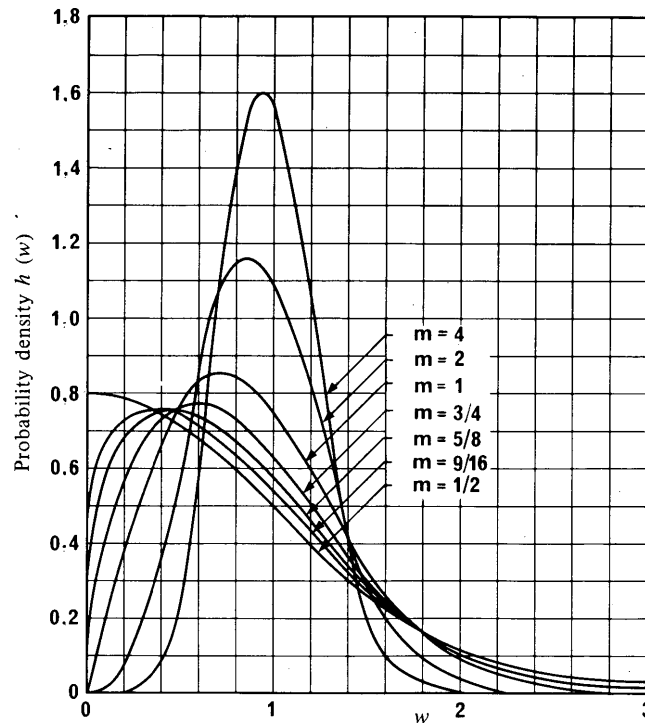


FIGURE 13 – The probability density function of the M-distribution

2.2 Probability density function of the model

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 model's probability density function $f(u)$.

2.3 Cumulative distribution function of the model

The cumulative distribution function of the $U(t)$ signal, $F(u)$, is obtained by integrating its probability density function, $f(u)$.

$$F(u) = \int_{-\infty}^u f(u) du \quad (5)$$

The agreements of the theoretical and experimental results can be shown more conveniently in the form of cumulative distribution, because the measurement of the cumulative distribution is much easier than that of probability density.

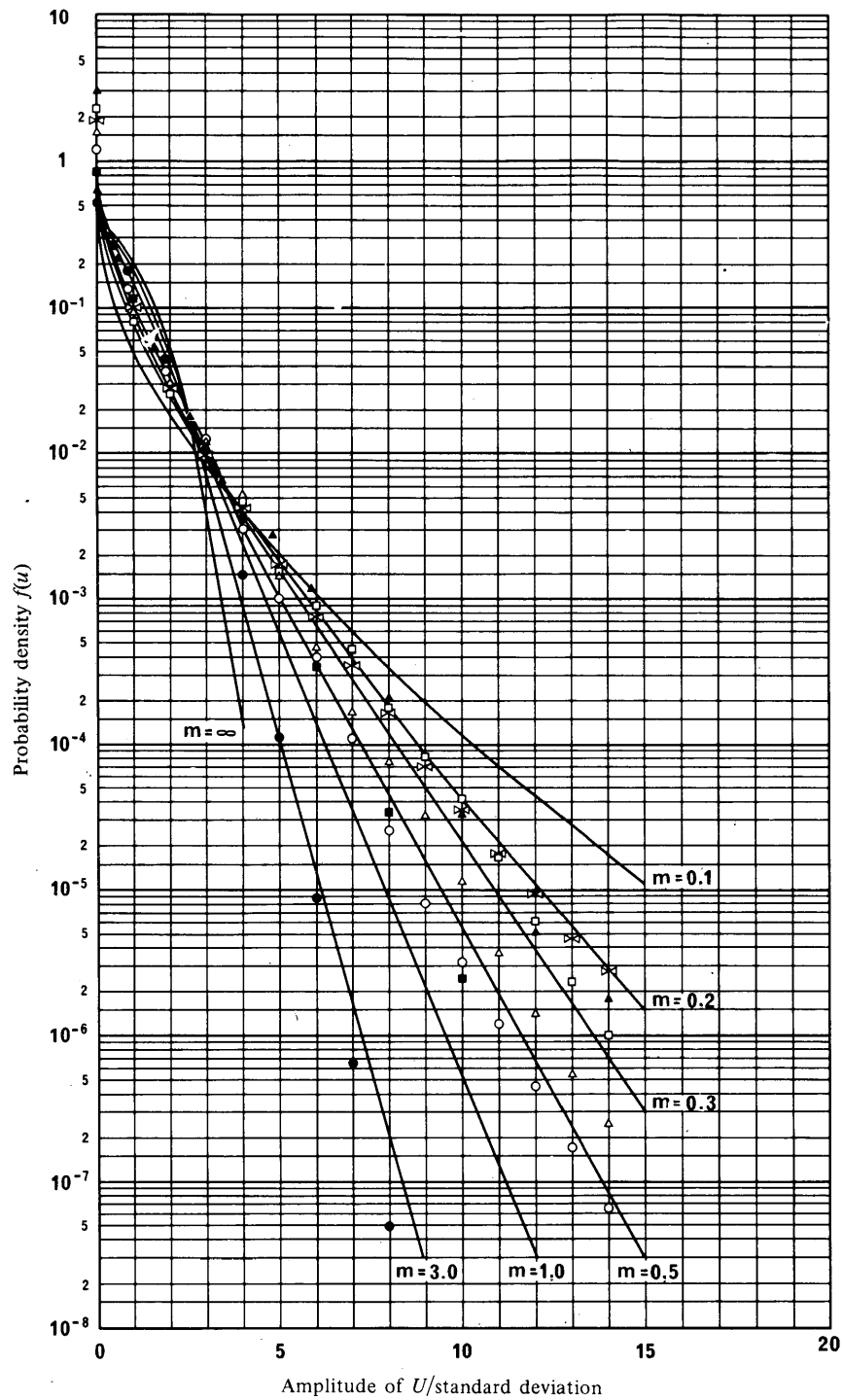


FIGURE 14 – The probability density functions of the $U(t)$ signal and sound-program signals

- 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) }

Figure 15 shows the calculated $F(u)$ curves for instantaneous amplitude, and the measured data described in this Report. The measured data in this figure were taken from curve C of Fig. 1, Curve A of Fig. 2 with Table V, and curves C_1 and C_4 of Fig. 3, and transferred onto this figure after normalization with respect to the signal level.

Figures 16 and 17 show other curves and plots for the 1 min mean power and the 1 min peak power respectively. These figures were made in a similar way to that of Fig. 15 using data contained in this Report.

Details of the process applied for the verification of the agreement can be found in [Ehara, 1982].

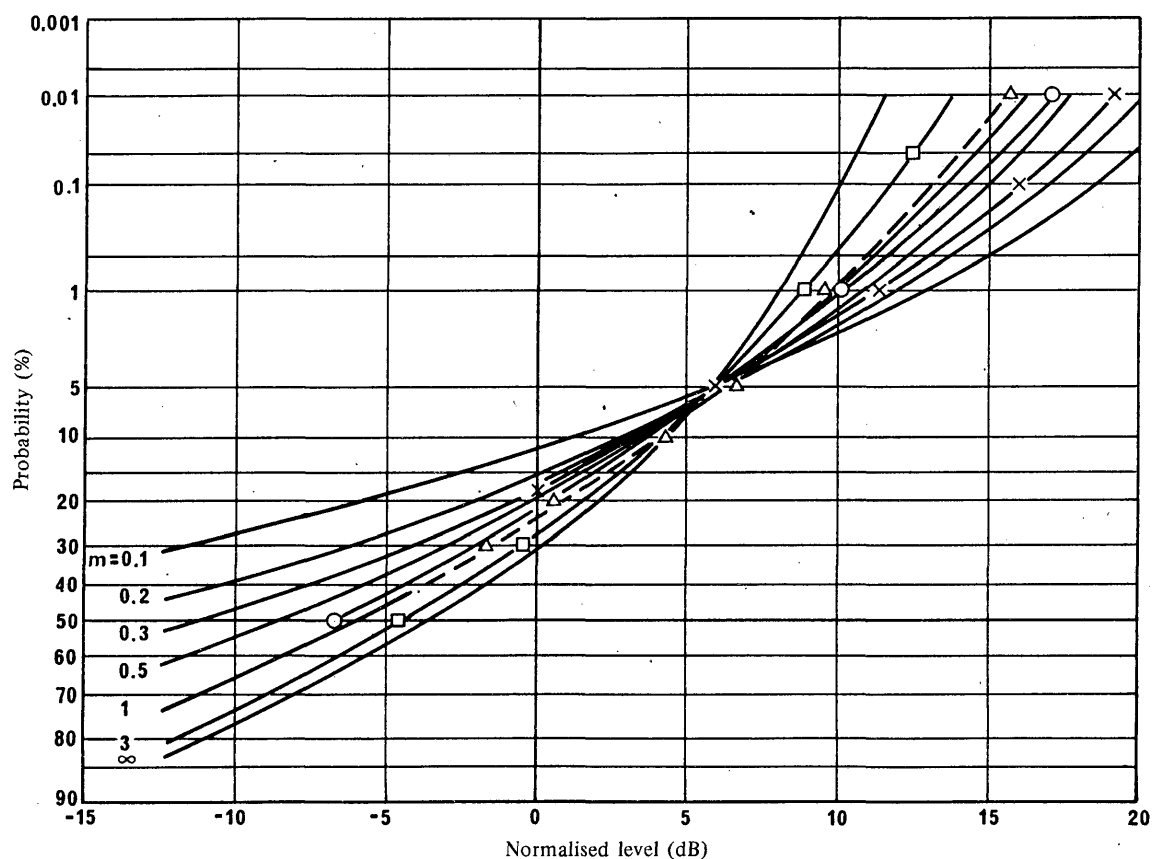


FIGURE 15 – Cumulative distribution function and data plots for instantaneous amplitude (full wave) or power

Data plotted were taken from:

- : curve C of Fig. 1
- Δ : curve A of Fig. 2 and Table V
- : curve C_1 of Fig. 3
- X : curve C_4 of Fig. 3 and Table III-B

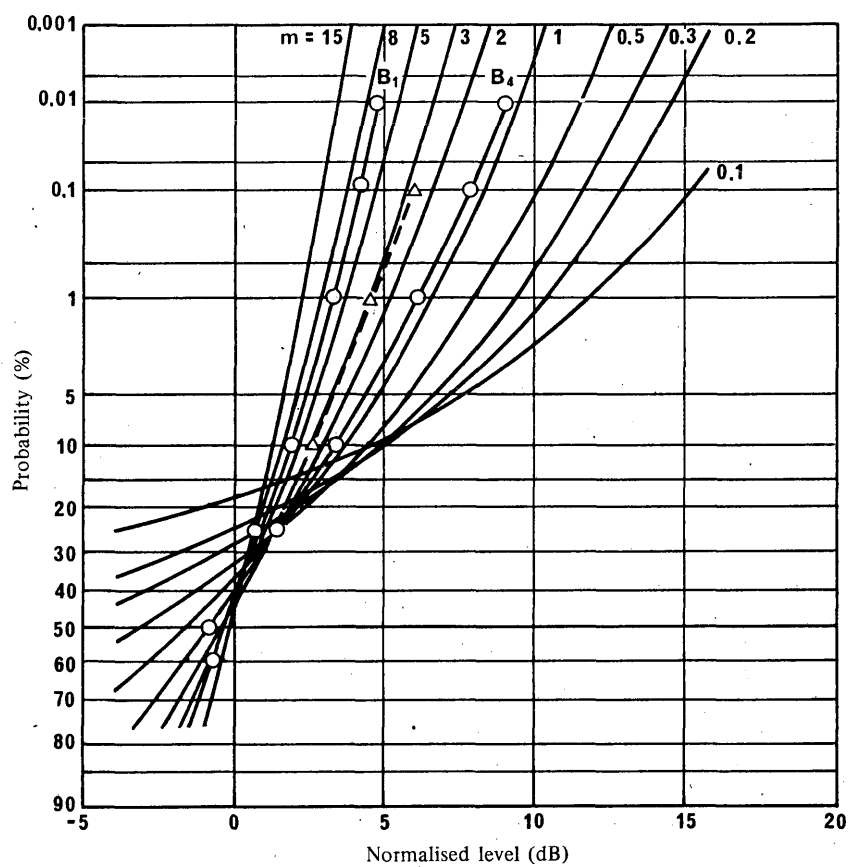


FIGURE 16 – Cumulative distribution function and data plots
for 1 min mean power

Data plotted were taken from:

- : curves B_1 and B_4 of Fig. 3
- Δ : Tables III-A and IV

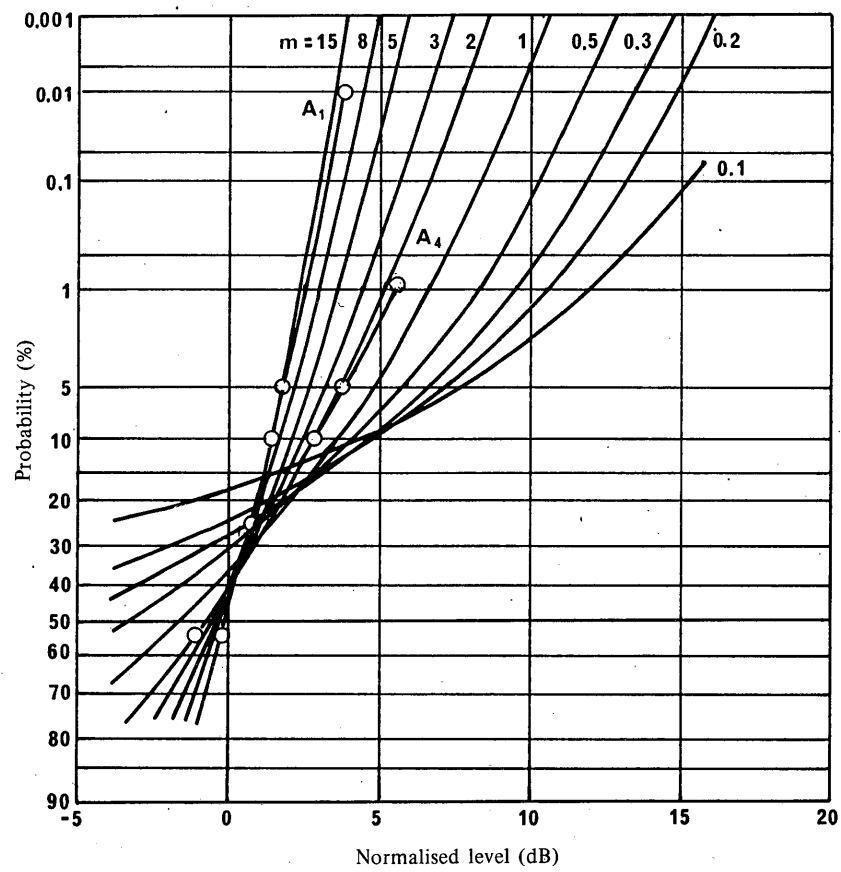


FIGURE 17 – Cumulative distribution function and data plots
for 1 min peak power

Data plotted were taken from:

○ : curves A_1 and A_4 of Fig. 3

2.4 Parametric representation of the statistical distributions

With the results of the close agreements described in § 2.2 and 2.3, it has been made clear that the statistical distributions of sound-programme signals can be represented by the shape parameter, m . The only other parameter necessary for specifying the model signal is the long-term mean power, P , which relates to the signal level or sound pressure level (SPL).

Values of m and P obtained for measured data described in this Report are shown in Table VI. These values may be used for the generation of a model signal.

TABLE VI — Values of parameters representing the statistical distributions of the measured signals

Data taken from	Signals dealt with	Values of shape parameter, " m "	Values of long-term mean power, " P " (dB)
Fig. 1 Curve A Curve B Curve C	Instantaneous amplitude (full wave)	0.5	−4.4
		0.2	−2.8
		0.9	−5.8
Fig. 2 Curve A Curve B		1.0	−2.2
		∞	0
Fig. 3 Curve A ₁ Curve A ₄	1 min peak power	11 3	10.4 6.0
Fig. 3 Curve B ₁ Curve B ₄	1 min mean power	6 1.5	1.0 −9.2
Fig. 3 Curve C ₁ Curve C ₄	Instantaneous power	3.0 0.3	−0.8 −10.8
Table III-A	1 min mean power	3	−3.4
Table III-B	Peak power	∞	−0.7
Table IV	Mean power	3	−8.9
Fig. 14 Mozart, No. 29 Beethoven, No. 2 Beethoven, No. 9 Tchaikovsky, No. 6 Popular music Speech (positive) Speech (negative)	Instantaneous sound pressure	0.6 0.4 0.25 0.2 3 0.25 0.6	77 (SPL) 79 (SPL) 84 (SPL)

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 continuous spectrum, the signal

$$V(t) = \sum_{i=1}^n C_i \cos(\omega_i t + \phi_i) \quad (6)$$

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. 18.

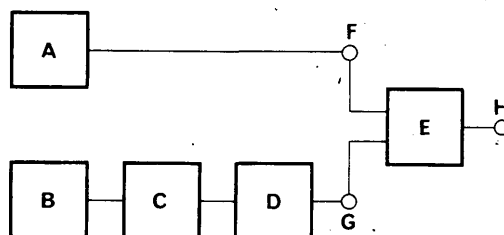


FIGURE 18 – A schematic block diagram for generating signals $U(t)$, $V(t)$, and $W(t)$

- 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. 18.

REFERENCES

EHARA, S. [January, 1982] Spectral amplitude and power distribution of sound-program signals. NHK Lab. Note No. 269.

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

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 duplexed 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 (see Note 1) at the audio output. The frequencies of the injected sinusoids were such as to produce tones of 2 kHz (see Note 2), 6 kHz (see Note 2), 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 (measured with a sound level meter conforming to IEC Publication 123, with time-constant “slow”).

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.

Note 1. — 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.

Note 2. — 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.

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 9).

<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 1(a) shows the result, for one modem, of two separate tests, A and B, carried out by six experienced observers; while Fig. 1(b) shows the corresponding results obtained in a single test by six inexperienced observers. Fig. 1(c) gives the results from six experienced observers, obtained for four modems in cascade.

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. 1(a) and 1(b). The corresponding limiting condition for four modems in cascade, Fig. 1(c), 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. 1(a) and 1(c), 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. 2(a), 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. 2(b) and Fig. 2(c) 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 2(d) 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.

8. Conclusion

It is concluded that the study can be best continued under the Question 17/CMTT.

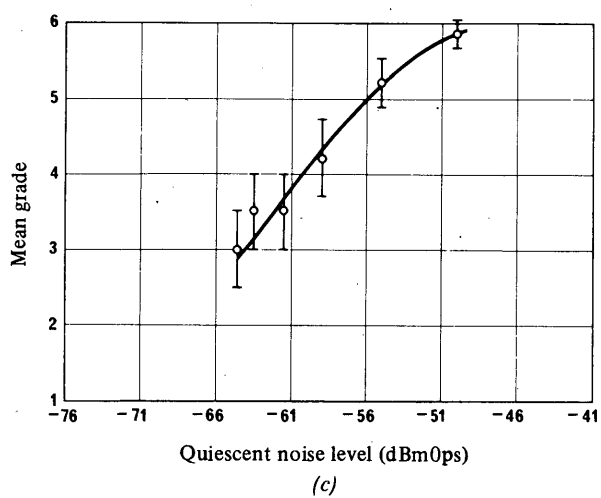
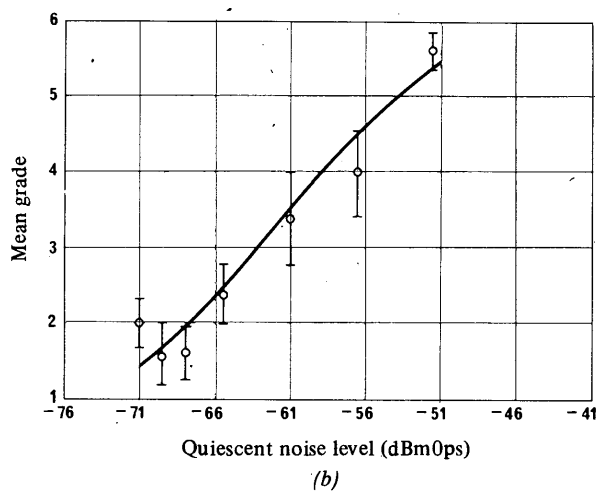
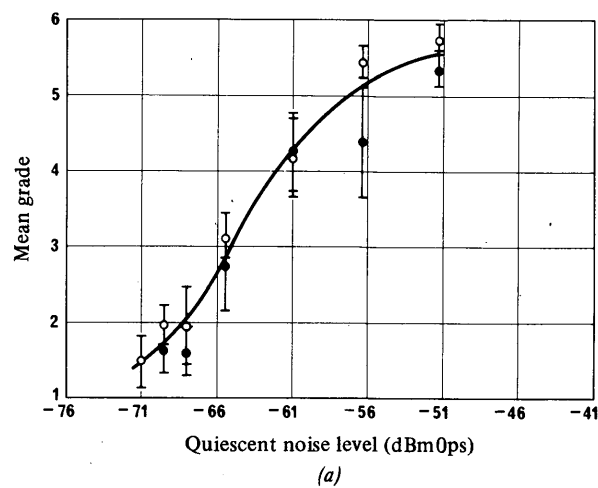


FIGURE 1 – Impairment versus quiescent noise level:
simulated white noise on bearer circuit

○ standard deviation of mean ($\pm$)

(a) one modem: experienced observers { ○ Test A
● Test B

(b) one modem: inexperienced observers

(c) four modems in cascade: experienced observers

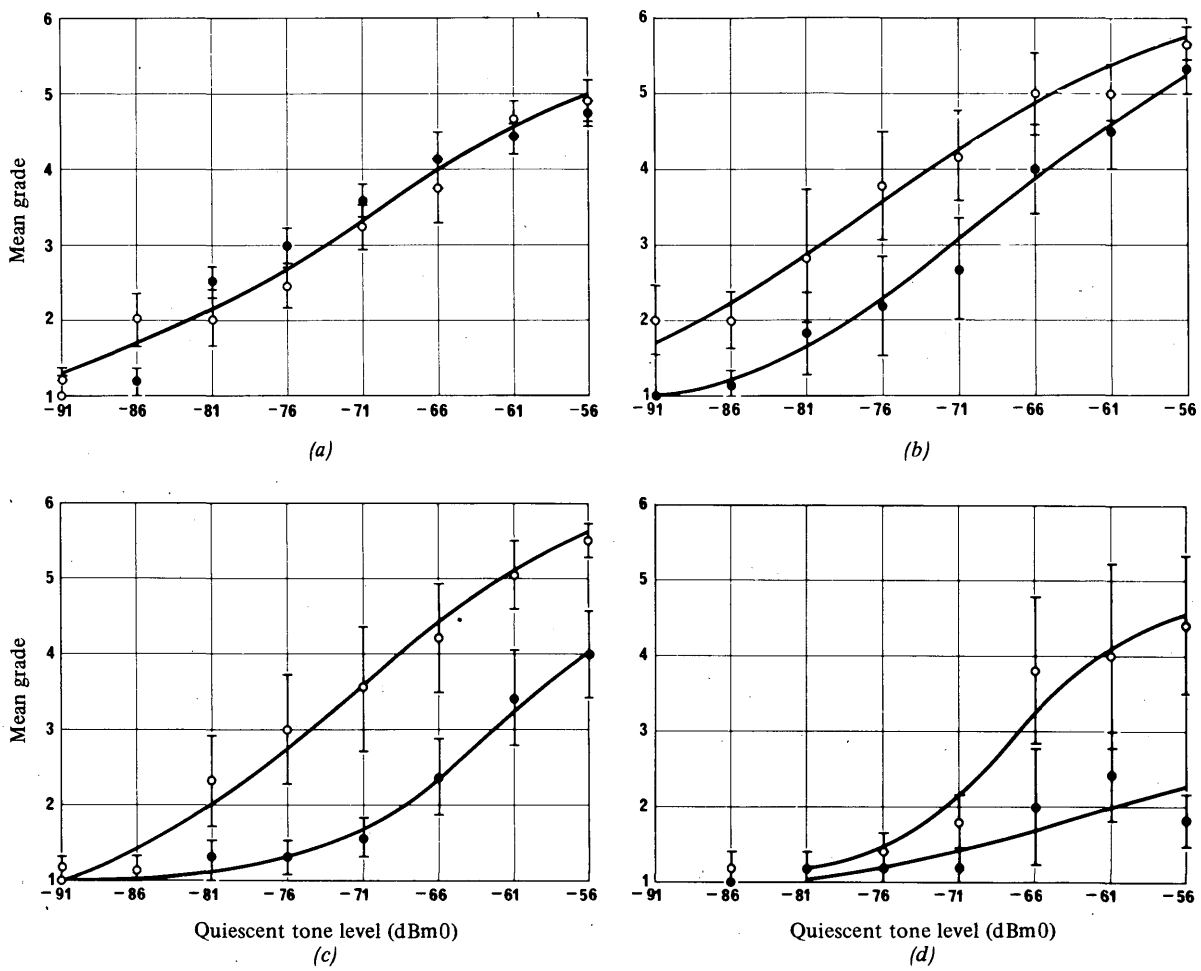


FIGURE 2 – Impairment versus quiescent tone level: simulated carrier leak

○ 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 605-1

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS
SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT

(Study Programme 17D/CMTT)

(1982-1986)

The CCIR

UNANIMOUSLY RECOMMENDS

that for the estimation of transmission performance of sound-programme circuits shorter or longer than the hypothetical reference circuit the following rules be applied:

Laws of addition

1. Comments on the use of the laws of addition

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

n : number of audio-to-audio sections,

L : length of the circuit,

l : 2500 km.

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

2. Law pertaining to circuit configuration

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

Let D_3 : design objective or the parameter derived therefrom and indicated in Table II, permitted value for three homogeneous sections of the hypothetical reference circuit,

and D_n : performance, or the parameter mentioned above, to be estimated for n sections.

The following expressions apply:

- for absolute values and logarithmic deviations:

$$D_n = D_3 (n/3)^{1/h} \quad (1)$$

- 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, especially to cover the case of mixed analogue-and-digital circuits.

TABLE I

n	Formula (1)			Formulas (2) and (3)		
	$\left(\frac{n}{3}\right)^{1/h}$			$\frac{20}{h} \log \left(\frac{n}{3}\right)$		
	$h = 1$	$h = 3/2$	$h = 2$	$h = 1$	$h = 3/2$	$h = 2$
1	0.33	0.48	0.58	−9.5 dB	−6.4 dB	−4.8 dB
2	0.67	0.76	0.82	−3.5	−2.3	−1.8
3	1.00	1.00	1.00	0.0	0.0	0.0
4	1.33	1.21	1.15	2.5	1.7	1.2
5	1.67	1.41	1.29	4.4	3.0	2.2
6	2.00	1.59	1.41	6.0	4.0	3.0
7	2.33	1.76	1.53	7.4	4.9	3.7
8	2.67	1.92	1.63	8.5	5.7	4.3
9	3.00	2.08	1.73	9.5	6.4	4.8
10	3.33	2.23	1.83	10.5	7.0	5.2
11	3.67	2.38	1.91	11.3	7.5	5.6
12	4.00	2.52	2.00	12.0	8.0	6.0
13	4.33	2.66	2.08	12.7	8.5	6.4
14	4.67	2.79	2.16	13.4	8.9	6.7
15	5.00	2.92	2.24	14.0	9.3	7.0

TABLE II

	Parameter	D_3 expressed in	Applicable formula	h (tentative value)
Parameters relevant to mono and stereo circuits	Insertion gain (1.0 kHz)			
	Adjustment error	dB	1	2
	Variation during 24 h	dB	1	2
	Gain/frequency response	dB	1	3/2
	Group delay/frequency response referred to minimum	ms	1	1
	Maximum weighted noise level	dBq0ps	4	2
	Single tone interference level	dBm0	2	3/2
	Ratio of disturbing modulation by power supply to reference signals	dB	3	3/2
	Non-linear distortion	%	1	3/2
	Error in reconstituted frequency	Hz	1	3/2
	Intelligible crosstalk ratio	dB	3	3/2
	Error in amplitude/amplitude response	dB	1	3/2
Additional parameters relevant to stereo circuits	Difference in gain between A and B channels	dB	1	(1)
	Phase difference between A and B channels	degree	1	(1)
	Intelligible crosstalk ratio between A and B channels	dB	3	3/2
	Non-linear crosstalk ratio between A and B channels	dB	3	3/2

(1) Some administrations have found $h = 3/2$ appropriate. Tests conducted in Japan [CCIR, 1978-82], however, showed better agreement with quadratic law of addition ($h = 2$) than $3/2$ law of addition.

REFERENCES

CCIR Documents

[1978-82]: CMTT/233 (Japan).

REPORT 821-2 *

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS
SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT **

(Study Programme 17D/CMTT)

(1978-1982-1986)

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.

RECOMMENDS 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.

Administrations are also invited to make contributions on the laws of addition applicable to mixed analogue-and-digital circuits.

* 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 I

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: $60 \text{ dB} \hat{=} -54 \text{ dBm} \hat{=} 1.55 \text{ mV}$

section 2: $58 \text{ dB} \hat{=} -52 \text{ dBm} \hat{=} 1.95 \text{ mV}$

$$a_s \approx \sqrt{1.55^2 + 1.95^2} \text{ mV} \approx 2.49 \text{ mV}$$

Total value: noise level $2.49 \text{ mV} \hat{=} -49.9 \text{ dBm}$

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, Geneva, 1982)

(1970-1982)

The CCIR

UNANIMOUSLY RECOMMENDS

that the ratio of a sine-wave test signal applied to the sound programme circuit to the highest level unwanted side component due to modulation caused by interfering signals from power supply sources should be greater than 45 dB.

Note 1. — This limit is identical to that which is considered tolerable for other types of transmission (FM and AM-VF telegraphy, facsimile transmission, speech, telephone signalling and data transmission).

Note 2. — This limit applies only where the interfering signals are the usual low order mains frequency harmonics. For modulation by much higher frequencies a more stringent limit is likely to apply.

Note 3. — For design purposes this limit should be taken as applying to the hypothetical reference circuit of length 2500 km.

RECOMMENDATION 659*

DIGITAL TRANSMISSION OF SOUND PROGRAMMES — GENERAL PRINCIPLES

(Question 18/CMTT and Study Programmes 18A/CMTT, 18C/CMTT, 18D/CMTT and 18E/CMTT)

(1986)

The CCIR,

CONSIDERING

- (a) that it is desirable to use common standards for the transmission of high-quality sound;
- (b) that it is desirable to use as few code conversions as possible in the international digital sound exchange procedure;
- (c) that to facilitate the exchange of signals, it is desirable to render the transmission interfaces as transparent as possible to the transmitted message content;
- (d) that the interface bit rates should take into account the hierarchical levels recommended by the CCITT;
- (e) that it may be efficient to apply bit-rate reduction and error-protection methods to lower transmission costs;
- (f) that quality and availability should not be restricted by the signal processing equipment or methods employed in the transmission circuit,

* This Recommendation should be brought to the attention of CCIR Study Group 10 and CCITT Study Groups XV and XVIII.

UNANIMOUSLY RECOMMENDS

1. that a sound programme originating in digital form should preferably be maintained in digital form for transmission;
2. that it should be possible to transmit digitally coded studio signals at the quality level required for each particular application;
3. that all the components of an individual sound programme (e.g. the left and right channel of a stereophonic programme signal) should be transmitted as one composite digital signal.

RECOMMENDATION 660*

**TRANSMISSION OF ANALOGUE
HIGH-QUALITY SOUND-PROGRAMME SIGNALS ON MIXED
ANALOGUE-AND-DIGITAL CIRCUITS USING 384 kbit/s CHANNELS**

(Question 18/CMTT, Study Programmes 18A/CMTT, 18C/CMTT, 18D/CMTT and 18E/CMTT)

(1986)

The CCIR,

CONSIDERING

- (a) that high-quality sound-programme signals on mixed analogue-and-digital circuits must satisfy the performance requirements of Recommendation 505;
- (b) that the number of analogue-to-digital and digital-to-analogue conversions as well as the number of digital coding techniques should be kept to a minimum, in order for the international sound programme connection to provide the same or higher degree of performance than possible with a homogeneous analogue connection;
- (c) that the CCITT has recommended a 384 kbit/s H0 bearer channel in Recommendation I.412 which has the capacity to transport sound-programme signals on an integrated services digital network (ISDN);
- (d) that digital-to-digital conversions using the same encoding law on tandem sound-programme circuits with different primary rates should not introduce any signal degradation,

UNANIMOUSLY RECOMMENDS

that the transmission of sound-programme signals on 384 kbit/s channels, with an analogue interface at the input and output of the international connection, should satisfy the analogue-and-digital requirements defined in § 1 and 2 respectively of this Recommendation.

Note 1. — Equipment characteristics for the coding of analogue high-quality sound-programme signals using 384 kbit/s channels have been defined in CCITT Recommendation J.41. Characteristics for equipment offering digital access at 384 kbit/s into 2048 kbit/s are given in CCITT Recommendations G.735 and G.737.

Note 2. — Other coding techniques are proposed and are the subject of Question 18/CMTT, for example, coding of analogue sound-programme signals at 316 kbit/s, insertion into a 320 kbit/s channel and multiplexing of six high-quality sound-programme channels within 2048 kbit/s. Techniques which may be used by bilateral agreement of the administrations concerned are listed in Table I of Report 647.

1. Analogue requirements

Transmission of high-quality sound-programme signals on mixed analogue-and-digital circuits shall satisfy the performance requirements of Recommendation 505.

1.1 Single codec performance

The analogue-to-analogue transmission performance of a single codec shall be equivalent to one-third the performance characteristics of the hypothetical reference circuit (see Recommendation 505). Where applicable, the laws of addition given in Recommendation 605 can be used to calculate the single codec performance with $n = 3$ analogue-to-analogue sections.

* This Recommendation should be brought to the attention of CCIR Study Group 10 and CCITT Study Groups XV and XVIII.

1.2 Tandem codec performance

The hypothetical reference circuit (HRC) given in Recommendation 502 is divided into three sections of equal length which could be either analogue or digital. Each digital section should only contain one encoder and one decoder. Therefore, in a mixed analogue-and-digital HRC, Table I indicates the number of codecs which may be connected in tandem.

TABLE I — Number of tandem codecs on a mixed analogue-and-digital sound programme circuit

Analogue sections	Tandem codecs
0	3
1	2
2	1
3	0

2. Digital requirements

Digital-to-digital conversions using the same encoding law as given in Table II or Table III of this Recommendation on tandem sound-programme circuits with different primary rates should not introduce any signal degradation. Digital-to-digital conversions using different encoding laws, as given in Tables II and III, should cause minimum signal degradation.

TABLE II — Coding table for instantaneous companding of sound- programmes signals *

Normalized analogue input (1)	Normalized analogue output (1)	Compressed digital code	Segment No.	Effective resolution (bits)
8160 to 8192 4096 to 4128	8176 4112	895 768	1	9
4080 to 4096 2048 to 2064	4088 2056	767 640	2	10
2040 to 2048 1024 to 1032	2044 1028	639 512	3	11
1020 to 1024 512 to 516	1022 514	511 384	4	12
510 to 512 256 to 258	511 257	383 256	5	13
255 to 256 128 to 129	255.5 128.5	255 128	6	14
127 to 128 0 to 1	127.5 0.5	127 0		

* The positive half of the coding table only is shown.

(1) The top code word of ± 8192 corresponds to an overload level of +15 dBm0s at the 0 dB insertion loss frequency (2.1 kHz) of the CCITT Recommendation J.17 pre-emphasis circuit with 6.5 dB loss at 800 Hz.

TABLE III – Coding table for near-instantaneous companding of sound-programme signals

Range	Normalized analogue input (1)	Normalized analogue output (1)	Compressed digital code	Effective resolution (bits)
4	+8176 to +8192 0 to +16 –16 to 0 –8192 to –8176	+8184 +8 –8 –8184	+511 0 –1 –512	10
3	+4088 to +4096 0 to +8 –8 to 0 –4096 to –4088	+4092 +4 –4 –4092	+511 0 –1 –512	11
2	+2044 to +2048 0 to +4 –4 to 0 –2048 to –2044	+2046 +2 –2 –2046	+511 0 –1 –512	12
1	+1022 to +1024 0 to +2 –2 to 0 –1024 to –1022	+1023 +1 –1 –1023	+511 0 –1 –512	13
0	+511 to +512 0 to +1 –1 to 0 –512 to –511	+511.5 +0.5 –0.5 –511.5	+511 0 –1 –512	14

(1) The top code word of ± 8192 corresponds to an overload level of +12 dBm0s at the 0 dB insertion loss frequency (2.1 kHz) of the CCITT Recommendation J.17 pre-emphasis circuit with 6.5 dB loss at 800 Hz.

2.1 Sampling frequency

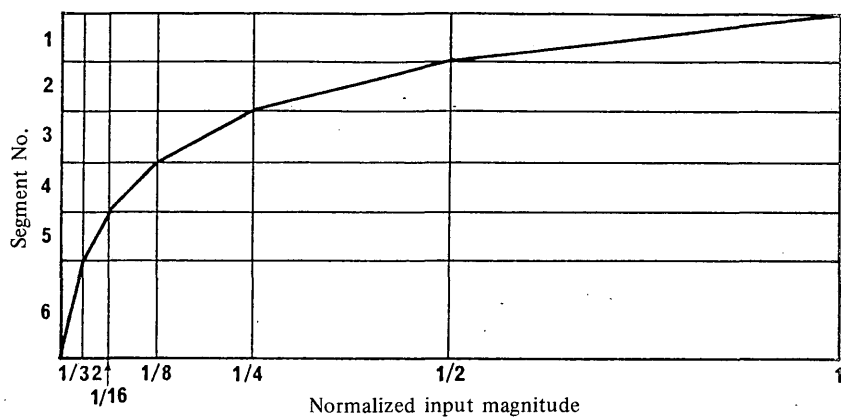
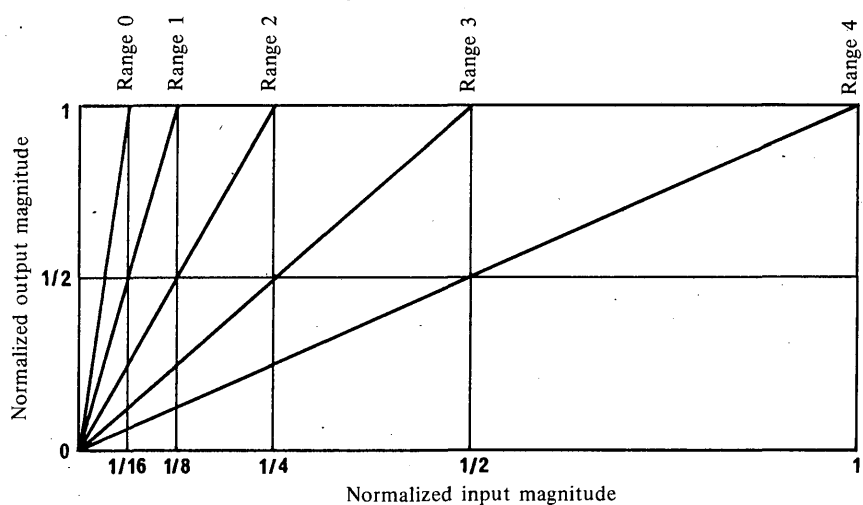
The sampling frequency shall be 32 kHz. The associated tolerance shall be $\pm 5 \times 10^{-5}$ as specified in CCITT Recommendations G.732 and G.733 for primary PCM multiplex equipment. This sampling frequency conforms with that given in Recommendation 606.

2.2 Encoding method

The recommended encoding laws are based on a uniformly quantized 14-bit per sample PCM technique with companding and shall employ either method (a) or method (b) with the appropriate rules of precedence as defined below:

- (a) eleven-segment 14-to-11 bit instantaneous A-law companding. The companding characteristic is illustrated in Fig. 1;
- (b) five-range 14-to-10 bit near-instantaneous companding. The companding characteristic is illustrated in Fig. 2.

Digital paths between administrations which have adopted different systems should carry signals encoded according to method (a). Where both administrations have adopted the same method, that method should be used on digital paths between them. Any necessary transcoding will be done by administrations using method (b).

FIGURE 1 – *Instantaneous companding characteristics*FIGURE 2 – *Near-instantaneous companding characteristics*

2.3 Coding table

The coding table for the instantaneous A-law compandor is specified in Table II and the coding table for the near-instantaneous compandor is specified in Table III.

2.4 Bit error protection

2.4.1 Error detection

Error detection shall be used to detect erroneous sound programme samples which can be caused by random bit errors, burst errors, frame loss or frame slips.

2.4.2 Error concealment and correction

Error concealment or correction techniques shall be used to ensure that the subjective quality of the sound-programme signal is not degraded below grade 4.5 as defined in Recommendation 562. This quality shall be maintained for a random bit error ratio of 10^{-6} as defined in CCITT Recommendation G.821. Further study is necessary to define a requirement for burst errors (see Report 648).

2.4.3 *Frame loss/recovery*

Frame loss/recovery on digital channels causes interruptions to sound-programme signals. The tolerable limits for these interruptions require further study (see Reports 642 and 647).

2.5 *Jitter*

Jitter characteristics from the digital interface can produce impairments to the decoded analogue signal after passing through a digital system. This requires further study as outlined in Report 648.

2.6 *Transmission bit rate*

2.6.1 *Monophonic transmission*

The transmission bit rate for one sound-programme channel shall be 384 kbit/s, including all ancillary bits as necessary.

2.6.2 *Stereophonic transmission*

Two separate 384 kbit/s channels shall be used to form a stereophonic (stereo-pair) sound-programme circuit. Both sound-programme channels shall be routed together over the same transmission path to avoid transmission delay differences.

2.7 *Network access*

Sound-programme signals coded into 384 kbit/s digital channels should interface to the ISDN at 1544 kbit/s or 2048 kbit/s as specified in CCITT Recommendation I.412.

REPORT 647-3 *

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS **

(Question 18/CMTT)

(1974-1978-1982-1986)

1. *Introduction*

This Report includes a description of the composition of digital networks and gives details on several coding methods. Annex II contains information concerning subjective quality assessments for signals encoded with instantaneous or near-instantaneous companding systems and subjective comparisons with other 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 647-2 of study period 1978-1982 and new contributions received during study period 1982-1986, 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 and e; CCIR, 1982-86a, b and c].

CCITT Recommendations G.821, G.823 and G.824 provide details of the error and jitter limits of digital networks.

2.1 *Integrated services digital network (ISDN)*

CCITT Study Group XVIII has developed a number of Recommendations relating to the ISDN. These are published in the Series I Recommendations, CCITT Red Book, Fascicle III.5. Some of the Series I Recommendations impact on CMTT activities for the transmission of sound-programme signals as described below.

* 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.

2.1.1 *Services (general)*

Three Recommendations deal with service capabilities. Sound-programme and television signal transmission are examples for services to be carried by an ISDN.

Recommendation I.210 describes the principles of telecommunication services supported by an ISDN. Two general classes of services are defined, bearer services and tele-services.

Recommendation I.211 specifies bearer services, that is services which provide the capability for the transmission of signals between user/network interfaces and involving only low layer attributes.

Recommendation I.212 specifies tele-services, that is services which provide the complete capability for communication between users of specified user terminals and involving low and high layer attributes.

Services are therefore defined from the viewpoint of network functions/capabilities.

In the case of bearer services, the network mainly provides signal transport functions, e.g. interfaces, transmission, routing.

In the case of tele-services, the network, in addition to signal transport functions, could also provide functions which relate to the contents of the signal carried, e.g. encoding standards conversion, bit-rate reduction.

Both bearer services and tele-services are characterized by attributes and their corresponding values.

Sound-programme and television signal transmission could thus be provided as both a bearer service or a tele-service, depending on the attributes requested by a customer.

Typical low layer attributes are the information transfer rate, information transfer capability and access channel structure and rate, whereas typical high layer attributes are layers 4 to 7 protocol functions. A leased line, for example, is considered a bearer service with a corresponding "communication establishment" attribute, such as "reserved" or "permanent".

2.1.2 *Services (< 2 Mbit/s)*

For the work in the CMTT, the most important services at present are certain bearer services of Recommendation I.211 as follows:

- 384 kbit/s: unrestricted, circuit-mode, 8 kHz structured,
- 1536 kbit/s: unrestricted, circuit-mode, 8 kHz structured,
- 1920 kbit/s: unrestricted, circuit-mode, 8 kHz structured.

The definitions for service attributes and values are contained in Recommendation I.130.

These bearer services could be employed for the transmission of one or several monophonic or stereophonic digital sound-programme channels, as specified in Recommendation 660.

Recommendations I.130, I.210, I.211 and I.212 will be developed further to include, amongst others, broadband services.

2.1.3 *User-network interface*

Three Recommendations specify ISDN user-network interfaces:

- Recommendation I.410 on general aspects,
- Recommendation I.411 on reference configurations,
- Recommendation I.412 on channel structures and access capabilities.

These Recommendations are the key to understanding of access to the ISDN.

Recommendation I.412 – ISDN user network interfaces, interface structures and access capabilities – defines a limited set of both channel types and channel structures for ISDN user/network physical interfaces. The definitions for channels, channel types and interface structures are essential to understand the ISDN user/network interface specification and are therefore reproduced below:

- a *channel* represents a specified portion of the information-carrying capacity of an interface;
- channels are classified by *channel types*, which have common characteristics;
- the channels are combined into *interface structures*. An interface structure defines the maximum digital information-carrying capacity across a physical interface.

The following channels and interface structures are of interest to the CMTT:

- H0 – 384 kbit/s,
- H11 – 1536 kbit/s,
- H12 – 1920 kbit/s.

All H channels (H0, H11, H12) appear at the ISDN user-network interface by means of the “primary rate user/network interface”, that is 1544 kbit/s and 2048 kbit/s respectively. Recommendation I.431 specifies the layer 1 characteristics for the primary rate user/network interfaces.

The Recommendations for user/network interfaces provide for the bidirectional transmission of H0 (384 kbit/s), H11 (1536 kbit/s) and H12 (1920 kbit/s) channel signals or combinations of B (64 kbit/s) and H0 channel signals. It is important to note that H0 is the only H channel (bit rate) specified which is suitable for the transmission of digital monophonic sound-programme signals. H11 and H12 are suitable for multiple H0 channels. For transmission of stereophonic sound-programme signals, means will have to be provided for maintaining the synchronism of the left and right components.

No user/network interface at 384 kbit/s is envisaged.

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, 1978-82f, g and h] or part of separate dedicated networks [CCIR, 1978-82h, i and j].

The concept of “drop-insert” facilities given in § 2.3 is appropriate to the above application.

2.3 *Network access*

Examples of network access envisaged for sound-programme signals [CCIR, 1982-86b] 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. In the case of a network using a 1544 kbit/s based hierarchy, a number of bits in the frame should be reserved for service functions. In the case of a network using a 2048 kbit/s based hierarchy, there is in principle no restriction on the use of the full capacity for the transmission of sound-programme signals, although compatibility with a recommended frame structure could offer some advantages. Access to the ISDN is specified in CCITT Recommendations I.412 and I.431 and for primary rate access (H11, H12 channels) it is restricted to 1536 and 1920 kbit/s respectively;
- 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);
- information from CCITT Study Group XV provides details of the characteristics of equipment to be used for the digital transmission of sound-programme signals (see CCITT Recommendations J.41 and J.42);
- external access equipment operating at 2048 kbit/s and offering digital sound-programme access at 384 kbit/s is specified in CCITT Recommendations G.735 and G.737.

As mentioned in § 2.1.3, no user/network interface is envisaged at 384 kbit/s: interface with 384 kbit/s H0 channels is at the primary hierarchical level (1544 kbit/s or 2048 kbit/s). The United Kingdom has suggested in [CCIR, 1982-86d] that there are some advantages in using the primary hierarchical level for international exchange of high-quality digital stereophonic sound signals. If this was agreed, it would be necessary to define methods for multiplexing high-quality stereophonic sound and ancillary signals into the primary levels of the two digital hierarchies to make full economic use of the available bit rates.

The United Kingdom has further proposed [CCIR, 1982-86e] that for internal connection between digital studio equipment and interface equipment concerned with access to the ISDN at the primary level (or a component thereof), use of the studio standard CCIR digital interface should be considered (see Recommendation 647).

The Federal Republic of Germany uses a method of multiplexing two sound channels at 480 kbit/s and two data channels at 16 kbit/s to a 1024 kbit/s interface giving access to a dedicated 2048 kbit/s primary level [CCIR, 1982-86f]. For this interface, a coding method is also specified [CCIR, 1982-86g].

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 (see CCITT Recommendation G.703). These restrictions have been taken into account when defining the encoding law (character signals) for sound-programme signals (see CCITT Recommendations J.41, J.42).

Furthermore, when circuits are leased at the full hierarchical bit rates, no spare capacity is available for administrations to provide, for example, 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 appear 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, 1978-82k; CCIR, 1982-86h] 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, if necessary, during a quiet audio passage, resulting in no audible disturbance.

3. Coding methods

Recommendation 660 specifies the coding method to be used for digital transmission of analogue sound-programme signals at 384 kbit/s.

Several other 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, 1978-82f, g, h, j, k, l, m, n, o, p, q, r, s, t, u, v, w, x, y and z; CCIR, 1982-86i, k, l and m]. Table I contains an updated survey of the most important parameters of some of these proposals.

The coding methods 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 companders 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 ($10 \log 12/8$).

TABLE 1 – Proposed coding methods

	Coding methods using different companding principles								
	Near-instantaneous					Instantaneous			
	A1	A2	A3	A4	A5	B1	B2	B5	B6
Nominal bandwidth (kHz) Pre/de-emphasis	0.04-15 ⁽¹⁾ ⁽²⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁵⁾	0.04-15 ⁽¹⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁵⁾	0.04-15 CCITT Rec. J.17 6.5 dB/ 0.8 kHz	0.04-15 CCITT Rec. J.17 6.5 dB/ 0.8 kHz	0.05-7 CCITT Rec. J.17 6.5 dB/ 0.8 kHz	0.04-15 ⁽¹⁾ ⁽⁴⁾	0.04-15 ⁽¹⁾ None	0.04-15 ⁽¹⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁵⁾ +15 ⁽⁷⁾	0.05-7 ⁽³⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁵⁾ +15
Overload point ⁽⁶⁾ (dBm0s)	+12	+12	+12	+12	+12	+12	+12	+15 ⁽⁷⁾	+15
Sampling frequency (kHz)	32	32	32	32	16	32	32	32	16
Companding law	5 ranges	5 ranges	5/3 ranges ⁽⁸⁾	5/3 ranges ⁽⁸⁾	5/3 ranges ⁽⁸⁾	13 segments	7 segments	11 segments	11 segments
Bit rate reduction (bits)	14/10.1	14/10.1	14/9.5	14/9.5	14/9.5	14/10	13/11	14/11	14/11
Finest resolution (bits/sample)	14	14	14	14	14	14	13	14	14
Corresponding noise (dBq0ps)	−65	−65	−65	−65	−65	−66	−55	−62	−60
Coarsest resolution at +9 dBm0s/ <i>f</i> ₀ (bits/sample)	10	10	10	10	10	8	10	9	9
Corresponding noise (dBq0ps)	−41	−41	−41	−41	−39	−30	−37	−32	−30
Resolution at +9 dBm0s/60 Hz (bits/sample)	12	12	12	12	12	10	10	12	12
Corresponding noise (dBq0ps)	−55	−55	−55	−55	−53	−42	−37	−50	−48
Source coding (kbit/s)	323	323 ⁽⁹⁾	308	308	} for 7 two 1 circuits 316 320	320	352	352 ⁽¹⁰⁾	176 ⁽¹⁰⁾
Error protection (kbit/s)	12 13	11 ⁽⁹⁾	7	7		16	32	32	16
Framing and signalling (kbit/s)	3 2	4 ⁽⁹⁾	1	1		0.66	0	0	0
Service bit rate (kbit/s)	338	338	316	316		336.66	384	384	192
Transmission bit rate (kbit/s)	~340 ⁽¹¹⁾ 384	340 ⁽¹¹⁾ 384	320	320		336.66 ⁽¹¹⁾ 384	384	384	192
Countries and CCIR references	France [CCIR, 1978-82s, t, u, aa, ab]	UK [CCIR, 1978-82h]	USSR [CCIR, 1982-86j, n)	USSR [CCIR, 1982-86n]	Italy [CCIR, 1970-74d; 1978-82l, p]	Japan [CCIR, 1978-82x, aa, ab]	Switzerland, Federal Republic of Germany, the Netherlands, Denmark, Norway, Canada, USA [CCIR, 1978-82f, m, n, o, r, w, y, z, aa, ab, ac; 1982-86m, o]	Federal Republic of Germany, the Netherlands, [CCIR, 1978-82f, m, n, o, z]	

f_0 : zero loss frequency of pre-emphasis.

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) 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).

- (5) For convenience, the characteristic given in CCITT Recommendation J.17 has been reproduced in § 2.2.2 of Report 496. At this time, pre- and de-emphasis is not used in the United States of America. If used for international purposes, the Recommendation J.17 characteristic and insertion loss indicated in column B5 will be employed.
- (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) One administration considers that with the use of a limiter at the interface between the radio broadcasting studio and the transmission circuit, it would be desirable to adopt an overload point of +12 dBm0s, which would allow a more efficient use of the dynamic range available on sound-programme circuits.
In Canada and the United States of America, sound-programme levels originate from a +8 dBrs relative level point and have an overload level of +13 dBm0s. The net result is that the signal to quantizing noise values shown in column B5 are 2 dB higher for codecs used in Canada and the United States of America. The remaining transmission characteristics are identical to the codec defined in column B5 which has an overload level of +15 dBm0s at a +6 dBrs relative level point.
- (8) First number of ranges for odd samples, second number for even samples.
- (9) These values are approximate. For more details see [CCIR, 1978-82h].
- (10) 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.
- (11) Dedicated frame.

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 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)	$n_o = +12 \text{ dBm0s}$			$n_o = +15 \text{ dBm0s}$
		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

In Table III the calculated noise values are associated to the proposed coding methods listed in Table I to facilitate an objective comparison.

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							
	−24	−18	−12	−6	−2	0	+6	+12 +15
			−15	−9	−3	+3	+9	dBm0s
A1 A2 A3	−65		−59	−53	−47	−41		
A4 (1)	−65		−59	−53	−47	−41		
A5 (1)	−63		−57	−51	−45	−39		
B1	−66	−60	−54	−48	−42	−36	−30	
B2 (2)	−55		−49		−43		−37	
B5	−62		−56	−50	−44	−38	−32	
B6	−60		−54	−48	−42	−36	−30	

(1) [CCIR, 1982-86n].
(2) μ law coding and no pre-emphasis are used in this system.

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, 1978-82ad]. 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, 1978-82ae]. Thus the choice of 32 kHz as the sampling frequency appears to be satisfactory [CCIR, 1970-74e, 1978-82aa]. 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 has been unanimously recommended as the sampling frequency (see Recommendations 606 and 660).

The EBU believes that the use of emphasis, for reducing the effect of quantizing noise, is essential when using a 14/10 near-instantaneous compandor and proposes to use the emphasis specified in CCITT Recommendation J.17. The EBU further considers that there is no advantage in introducing emphasis in a 14-bit (or more) linear system for the purpose of reducing the audibility of quantizing noise [CCIR, 1982-86p].

From Table I it can be seen that with only one exception all proposals are based on a maximum 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. The 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 have even greater sensitivity to overloading than those for analogue transmission.

Recommendation 642 on limiters for high quality sound-programme signals has been adopted in 1986 based on a United Kingdom proposal [CCIR, 1978-82af; 1982-86q]. However, 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.

On transcoding between different coding methods a method has been described [CCIR, 1982-86l] for transcoding between A-law and near-instantaneous systems. Measurements of the signal-to-noise ratio produced suggest a result similar to that obtained with an A-law system alone [CCIR, 1982-86r].

The information communicated by CCITT Study Group XVIII on wideband speech coding [CCIR, 1982-86s] is of particular interest for the transmission of commentary speech or medium-quality sound-programme signals.

To that effect, the CMTT has adopted a new Study Programme 18G/CMTT on standards for the digital transmission of sound-programmes on one or two 64-kbit/s channels.

A document has been presented [CCIR, 1982-86t] describing and proposing two systems of 7 and 15 kHz, based on sub-band adaptive differential pulse code modulation (SB-ADPCM) which would enable medium quality sound-programme signals to be transmitted on two 64 kbit/s channels.

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.

Proposals for adding quality parameters concerning digital transmission [CCIR, 1982-86u and v] have been incorporated in Recommendations 503 and 505. One administration has proposed values for maximum weighted noise. These figures are given in the Recommendations but are still regarded as provisional and require further study.

	Recommendation 503	Recommendation 505
Idle channel noise (dBq0ps)	-49	-51
Programme modulated noise (dBq0ps)	-37	-39

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, § 3.1.1. 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, 1978-82q], 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, 1978-82q], 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 maximum resolution. A formula to calculate idle channel noise is given in [CCIR, 1978-82p]. The value to be applied to codecs which do not use 14 bits for maximum 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, 1978-82n] proposed the use of the test signal specified in Recommendation 505, § 3.1.3, 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 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	X
0.125 to 7.5	Y

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 to 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, 1978-82r], 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 to 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 of 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, 1978-82q], 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, 1978-82q], 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 consider that subjective assessments are important especially with several codecs in tandem, and have presented the results of tests as a confirmation that their proposed coding method produces a negligible impairment even when 3 or 4 codecs are used in tandem. The results of these 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

At the Final Meeting of the CMTT (Geneva, 1985), the texts of two new Recommendations on the subject of digital transmission of sound-programme signals were approved. Recommendation 660 addresses the transmission of analogue high-quality sound-programme signals on mixed analogue-and-digital circuits using 384 kbit/s channels. The other Recommendation 659 deals with digital transmission of sound programmes and gives some general principles for the future, when the network interfaces are all digital.

ANNEX II

SUBJECTIVE QUALITY ASSESSMENTS FOR SOUND-PROGRAMME SIGNALS ENCODED BY MEANS OF INSTANTANEOUS COMPANDING SYSTEMS AND COMPARISONS WITH NEAR-INSTANTANEOUS AND OTHER COMPANDING SYSTEMS

1. Introduction

Based on tests performed on their proposed coding method, some administrations (Italy, Japan, the Federal Republic of Germany and Canada) have presented the results of subjective assessments of 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 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 four administrations should be considered separately.

The Canadian Administration has also presented the results of comparison by subjective assessment of a number of companding systems (see § 2.5 of this Annex).

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 for programme-modulated noise and for granular distortion. For the 7 excerpts considered the average value of impairment due to programme-modulated noise was 4.5 for 1 codec and 4.3 for 4 codecs in tandem. If granular distortion is additionally weighted the equivalent values were 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 shown that unacceptable impairments would be caused for sound-programmes which contained rather high-frequency components by the insertion of pre-/de-emphasis. The Administration of Japan decided therefore not to use any pre-/de-emphasis. The following 5 programme excerpts were included in the tests:

- pianoforte solo,
- male speech,
- symphony orchestra,
- drums,
- cymbals.

When pre-/de-emphasis was used, the most critical test occurred with the excerpt of cymbals. With 3 codecs in tandem (all without pre-/de-emphasis) a very small degradation was found.

2.3 *Federal Republic of Germany* [CCIR, 1978-82ag]

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 (Kyoto, 1978). 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 (Table I, column B5 (Geneva, 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 (Table I, column 4 (Kyoto, 1978)) are 3.6 and 2.6 respectively and were regarded as unacceptable.

2.4 Canada [CCIR, 1982-86w]

The tests were performed by the Canadian Broadcasting Corporation in 1984. The results were taken as a basis for recommending the use of the coding method detailed in column B5 in Table I of this Report, the use of pre-/de-emphasis (see CCITT Recommendation J.17 set for a pre-emphasis loss of 6.5 dB at 800 Hz) and for fixing a lower limit for the bit error ratio performance of digital bit streams. Evaluation of results comparing transmissions with and without pre-/de-emphasis over 2, 4 and 6 codecs in tandem, indicated a subjective advantage in favour of pre-/de-emphasis. Imperceptible impairments were produced with 2 codecs in tandem and perceptible but not annoying impairments were reported with up to 6 codecs in tandem when pre-/de-emphasis was used. A test of the subjective effect of random errors over one codec also indicated some advantage in favour of pre-emphasis at bit error ratios of 10^{-5} and 10^{-4} . At 10^{-3} the impact of pre-emphasis was found to be minimal. The tests included programme excerpts containing flutes, piano, orchestra and cymbals.

2.5 Canada [CCIR, 1982-86x]

The tests were performed by the National Research Council of Canada in 1984 on a number of digital coding methods for transmission of high-quality digital audio signals in a direct broadcasting satellite environment. The coding methods included 14 bit linear, A-law, μ -law instantaneous companding, near-instantaneous as well as adaptive delta modulation (ADM) sampled at 204 kHz and 330 kHz. All coding methods were assessed with and without CCITT Recommendation J.17 emphasis. The test results showed that the use of Recommendation J.17 pre-emphasis, before the digital codecs, produced a clear improvement to the reproduced sound quality. The instantaneous companding systems were found to give slightly poorer performance than the near-instantaneous and ADM methods. However, the ADM was found to be noisier at low frequencies than the instantaneous companding system. The ADM was found to give similar performance to the near-instantaneous system for high-frequency material when sampled at 330 kHz, but worse when sampled at 204 kHz.

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/82 (Germany (Federal Republic of)); g. CMTT/92 (France); h. CMTT/240 (United Kingdom); i. CMTT/90 (France); j. CMTT/91 (France); k. CMTT/231 (Japan); l. CMTT/271 (Italy); m. CMTT/234 (the Netherlands); n. CMTT/251 (Germany (Federal Republic of)); o. CMTT/258 (Germany (Federal Republic of)); p. CMTT/275 (Italy); q. CMTT/276 (Canada); r. CMTT/297 (Denmark); 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/35 (France); ab. CMTT/110 (CCITT Study Group XVIII); ac. CMTT/143 (USA); ad. CMTT/247 (EBU); ae. CMTT/10 (EBU); af. CMTT/239 (United Kingdom); ag. CMTT/237 (Germany (Federal Republic of)).
- [1982-86]: a. CMTT/61 (CCITT Study Group XVIII); b. CMTT/62 (CCITT Study Group XVIII); c. CMTT/67 (CCITT Study Group XVIII); d. CMTT/172 (United Kingdom); e. CMTT/167 (United Kingdom); f. CMTT/214 (Germany (Federal Republic of)); g. CMTT/202 (Germany (Federal Republic of)); h. CMTT/34 (EBU); i. CMTT/48 (France); j. CMTT/10 (CCITT Study Group XV); k. CMTT/12 (USA); l. CMTT/41 (United Kingdom); m. CMTT/46 (Canada); n. CMTT/186 (USSR); o. CMTT/220 (Canada); p. CMTT/55 (EBU); q. CMTT/35(10/26) (United Kingdom); r. CMTT/42 (United Kingdom); s. CMTT/194 (CCITT); t. CMTT/226 (Switzerland); u. CMTT/3 (Germany (Federal Republic of)); v. CMTT/4 (Germany (Federal Republic of)); w. CMTT/223 (Canada); x. CMTT/241 (Canada).

BIBLIOGRAPHY

- BBC [1973] BBC Report 1973/41.
- BENNETT, D. [1984] Subjective evaluation of 14-11 bit A-law companded digital audio codec performance. Development Report 6366. Canadian Broadcasting Corporation (CBC), Montreal, Canada.
- BILLIA, G. P. and DECINA, M. [1974] Digital coding and transmission of high quality sound programmes. *Alta Frequenza*, Vol. 43, 1, 29E-29 to 39E-39.

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).

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 1095

TRANSMISSION OF SOUND-PROGRAMME SIGNALS OF DIGITAL STUDIO QUALITY

(Question 18/CMTT, Study Programme 18A/CMTT)

(1986)

1. Introduction

Recommendation 646 specifies the coding of sound-programme signals in studios. The sampling rate is 48 kHz and the resolution is at least 16 bits per sample, which gives a consequential bit rate of at least 768 kbit/s for a monophonic signal and 1536 kbit/s for a stereophonic signal.

In a second Recommendation, 647, an interface for the interconnection of different items of digital audio equipment within the studio, is specified. This interface has a gross bit rate of 1536 kbit/s for a monophonic signal or 3072 kbit/s for a stereophonic signal.

For sound-programme transmission, a sampling frequency of 32 kHz is specified in Recommendation 606.

2. Proposals for transmission formats and interfaces

2.1 Investigations carried out in the Federal Republic of Germany by the Institut für Rundfunktechnik have shown that 16 bits per sample can be reduced to 14 bits per sample without any noticeable degradation by a near-instantaneous floating point process, described in [CCIR, 1982-86a]. With an additional bit rate per sample for error protection and 16 kbit/s for the data signal, the channel bit rate then consists of 496 kbit/s. Two of these channels are combined with a frame alignment signal to form a multiplexed signal of 1024 kbit/s, which constitutes the bit rate of the interface. The structure of the frame is almost the same as that specified in CCITT Recommendation G.732 for the primary hierarchical level of 2048 kbit/s. It should be noted that the frame repetition rate is 4 kHz instead of 8 kHz. The frame format is shown in Fig. 1.

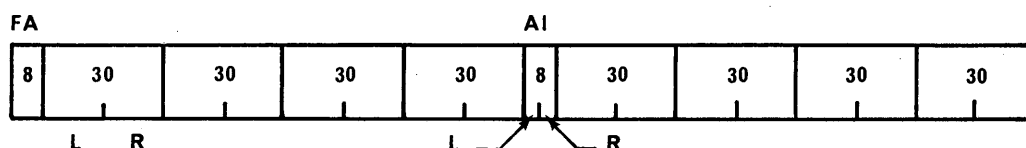


FIGURE 1 – Frame for the 1024 kbit/s signal with a length of 256 bits and a 4 kHz repetition rate

FA : frame alignment signal
AI : additional data signal
L : left channel
R : right channel

A method for the synchronous insertion of two such signals into the 2048 kbit/s frame structure is described in [CCIR, 1982-86b], in which only the net bit rate (without frame alignment signals) is transmitted.

This transmission technique requires the use of dedicated 2048 kbit/s links and is intended for operation in the Federal Republic of Germany's direct broadcasting satellite system, TV-SAT, in which one of the transponders will be allocated to a multiplex of 16 digital stereophonic sound-programme channels.

2.2 The United Kingdom has proposed [CCIR, 1982-86c] that for internal connection between digital studio equipment and for interface equipment concerned with access to the ISDN at the primary level (or a component thereof), use of the EBU/Audio Engineering Society standard digital interface should be considered.

2.3 In [CCIR, 1982-86d], Italy raises the question that for high-quality sound-programme exchange the ISDN offers no access level at a suitable bit rate as the lowest possible network access point for the user is at 1920 kbit/s.

One access method outside the ISDN which can be applied to the primary level at 2048 kbit/s is proposed in § 2.1.

2.4 [CCIR, 1982-86e] from Sweden describes a prototype multiplexing system for sound and data that can be interfaced to any transmission channel with a bit rate of not more than 4000 kbit/s. The existing system is equipped with a 2048 kbit/s interface that is compatible with CCITT Recommendation G.703, but interfaces to other access levels can be implemented as desired.

Because of the general multiplexing method adopted, any type of sound coding can be used. Codecs including linear 16 or 14 bits per sample PCM formats operating at 48, 32 or 16 kHz sampling frequencies have been developed.

A new digital programme network that is currently being studied in Sweden proposes to utilize the flexibility offered by the new multiplexing system, for example, sound codecs of the type described in [CCIR, 1982-86a] will be studied.

REFERENCES

CCIR Documents

[1982-86]: a. CMTT/202 (Germany (Federal Republic of)); b. CMTT/214 (Germany (Federal Republic of)); c. CMTT/167 (United Kingdom); d. CMTT/211 (Italy); e. CMTT/190 (Sweden).

REPORT 648-3

DIGITAL SOUND-PROGRAMME TRANSMISSION IMPAIRMENTS AND METHODS OF PROTECTION AGAINST THEM*

(Study Programmes 18A/CMTT and 18D/CMTT)

(1974-1978-1982-1986)

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.

Although for high error and slip rates these tests are significant, for very low rates, the results will be affected by the fact that the listener's attention decreases with increasing test duration. It should also be noted that even in the absence of transmission impairments the analogue studio apparatus might introduce some additional clicks. It therefore seems probable 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 is suggested that for very low error and slip rates, the quality objective should be expressed in terms of rate of occurrence of clicks.

It seems 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 a 32×10^3 sample/s repetition rate of PCM sound samples is assumed, then a random error ratio 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 ratio of 10^{-9} in any digital path.

In CCITT Recommendation G.821, the CCITT specifies error performance objectives for a 64 kbit/s circuit-switched connection in the ISDN, used for voice traffic or as a "bearer channel" for data type services. The 64 kbit/s circuit-switched connection referred to, is an all digital hypothetical reference connection (HRX) and is given in Figure 1/G.821 of CCITT Recommendation G.821. It encompasses a total length of 27 500 km and is a derivative of the standard hypothetical reference connection given in Fig. 1/G.801 of CCITT Recommendation G.801.

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 in terms of the rate of eight-bit byte slips on an international connection of 27 500 km in length or a corresponding bearer channel.

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, and 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.

* For the definition of absolute power, relative power and noise levels, see Recommendation 574.

In [CCIR, 1970-74a] it is proposed that an error mitigation technique be used together with an A-law instantaneous compandor. 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 ratio 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].

[CCIR, 1970-74b] discusses the effect of transmission errors in near-instantaneous companding and gives results of protection studies 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 I – 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. Figures 1 and 2 give these results and also propose tolerance limits based on impairment that are 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. In Recommendation G.823, the CCITT provides guidelines and limits to be applied in the planning of networks and in the design of equipment. At present, jitter is specified at the hierarchical interfaces of 64, 2048, 8448, 34 368 and 139 264 kbit/s. It is also desirable to have means for objectively measuring jitter on the decoded analogue signals; IEC Publication 386 (concerning the measurement of wow and flutter in recording systems) may help further studies on this aspect.

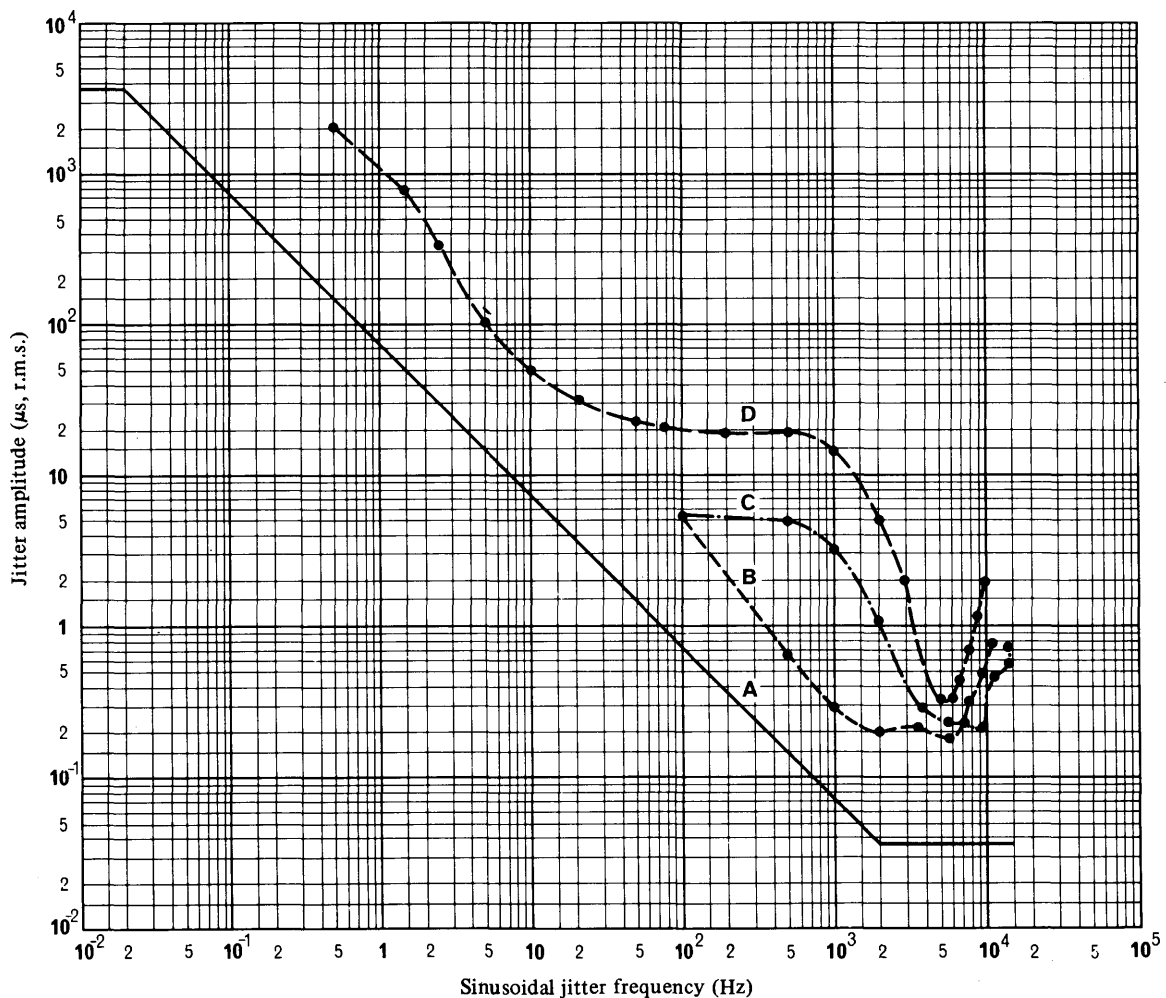


FIGURE 1 – *Proposed maximum allowable r.m.s. amplitude of sinusoidal timing jitter in high-quality sound signals*

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])

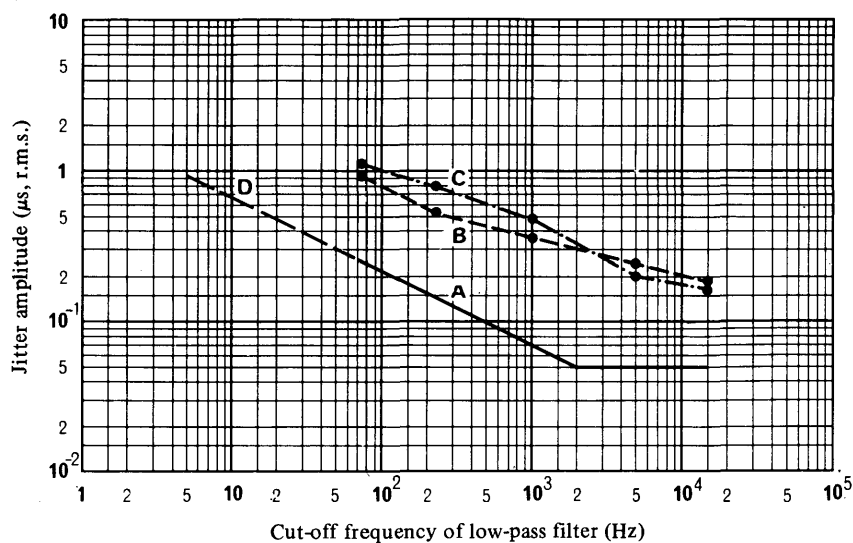


FIGURE 2 – Proposed maximum allowable r.m.s. amplitude of random timing jitter in high-quality sound signals

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

REFERENCES

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).

SECTION CMTT D: METHODS OF OPERATION AND ASSESSMENT OF PERFORMANCE OF SOUND CHANNEL TRANSMISSIONS

Recommendations and Reports

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.

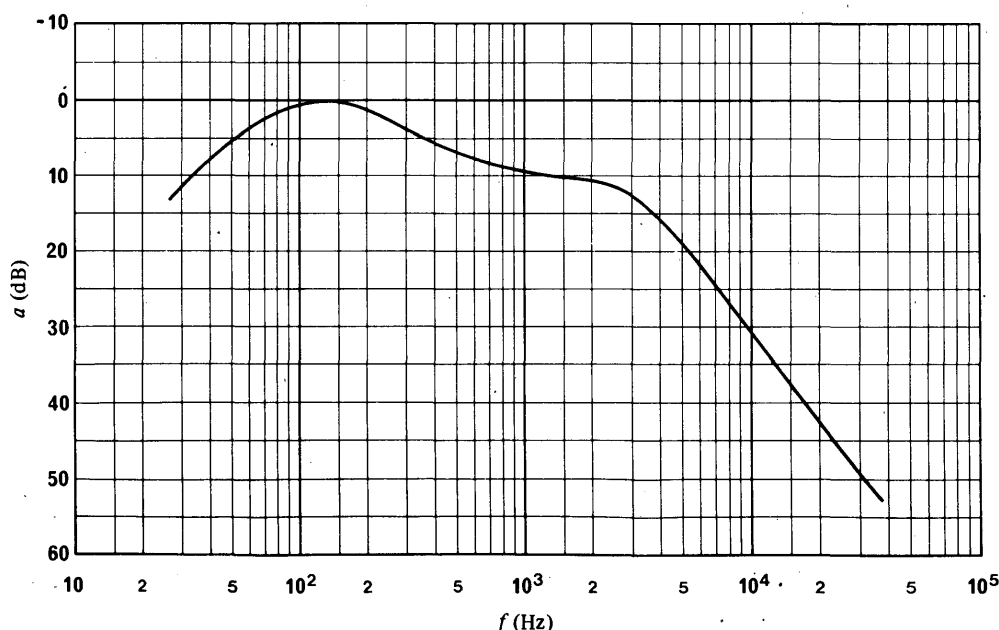


FIGURE 1 — Insertion-loss/frequency

* 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.

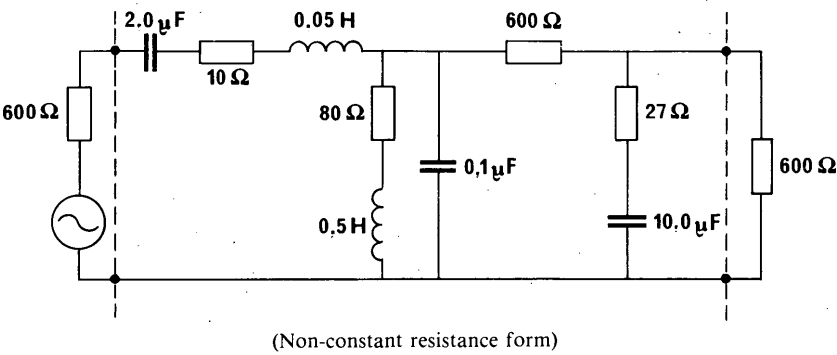


FIGURE 2

TABLE I

Frequency (Hz)	Relative insertion-loss (dB)	Tolerance (± 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).

* 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.

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.

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.

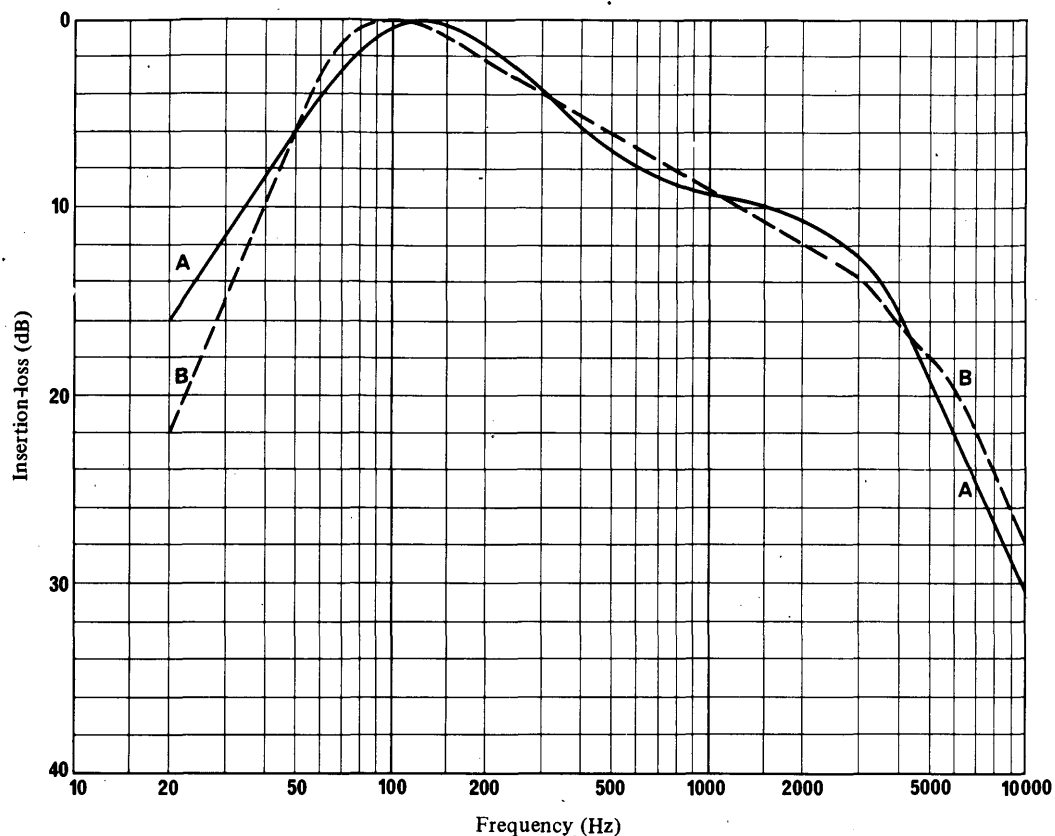
Note. — In Recommendation 559, for the objective measurement of radio-frequency protection in LF, MF and HF broadcasting, a similar test signal based on filter-shape white noise is recommended. In this case not only the insertion loss of the shaping filter is specified but also the frequency spectrum 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

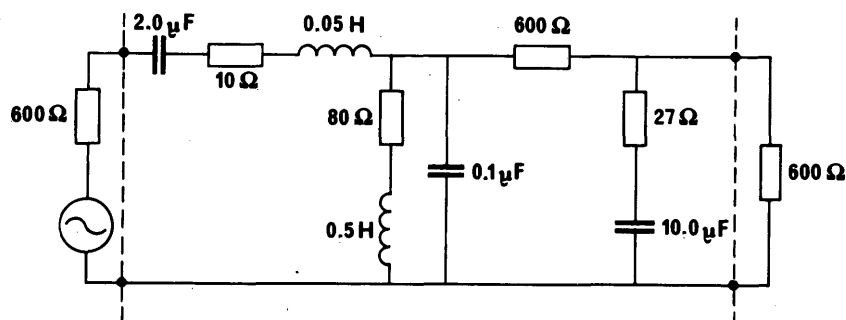
Note 1. — The integrated loss of the network complying with Table I has been calculated as 14.4 dB for white noise in the band 0 to 15 kHz.

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.

REFERENCES

HPJ (*Hewlett-Packard Journal*) [September, 1967] Various articles on pseudo-random noise.

PFYFFER, H. [1971] Discussion de quelques problèmes de mesure de bruit sur les systèmes de transmission. *PTT Techn. Mitt.*, 3, 94-106.

CCIR Documents

[1974-78]: a. CMTT/34 (United Kingdom); b. CMTT/198 (United Kingdom); c. CMTT/254 (Germany (Federal Republic of)).

BIBLIOGRAPHY

FISCHER, G. and RASCH, J. [1965] Die bei Fernsprech- und Rundfunkübertragung auftretenden elektrischen Leistungen unter Berücksichtigung von Preemphasis und Kompander (Electrical powers occurring in telephone and programme transmission, allowance being made for pre-emphasis and companders). *NTZ*, Vol. 18, 4, 205-209.

HOLBROOK, B. D. and DIXON, J. T. [1939] Load rating theory for multi-channel amplifiers. *BSTJ*, 624-644.

JAKUBOWSKI, H. [1971] Analyse des Programm-Materials des Hörrundfunks (Analysis of sound-broadcasting programme material). *Rundfunktechn. Mitt.*, Vol. 15, 6.

VON GUTTENBERG, W. and HOCHRATH, H. [1959] Untersuchungen über die Geräuschverminderung mittels Pre- und Deemphasis bei Rundfunkübertragung (Inquiries into noise reduction by pre-emphasis and de-emphasis in programme transmission). *NTZ*, Vol. 12, 9, 467-474.

RECOMMENDATION 645*

TEST SIGNALS TO BE USED ON INTERNATIONAL SOUND-PROGRAMME CONNECTIONS

(Questions 50/10 and 19/CMTT, Study Programmes 50B/10 and 50E/10)

(1986)

The CCIR,

CONSIDERING

- (a) that many impairments in international programme exchange on sound-programme connections are attributed to different national test signal definitions;
- (b) that some existing definitions are found in different Recommendations of the CCITT and the CCIR;
- (c) that for clarification, a list of those definitions should be available,

UNANIMOUSLY RECOMMENDS

that for an international sound-programme connection only the test signals defined below should be used:

1. Alignment signal (AS)

Sine-wave signal at a frequency of 1 kHz, which is used to align the international sound-programme connection. The signal level corresponds to 0 dBu0s (see Note) (i.e. 0.775 V r.m.s. at a zero relative level point). In accordance with CCITT Recommendation N.13, the period of sending the alignment signal should be kept as short as possible — preferably to less than 30 s.

Note. — The notation “dBu0s” is defined in Recommendation 574. Other related texts of the CMTT use the notation “dBm0s” also defined in Recommendation 574. The CMTT will be reviewing its texts during the next study period to ensure the correct and consistent use of these notations.

2. Measurement signal (MS)

Sine-wave signal at a level 12 dB below the alignment signal level which should be used for long-term measurements and measurements at all frequencies (see CCITT Recommendations N.12, N.13, N.21 and N.23).

3. Permitted maximum signal (PMS)

Sine-wave signal at 1 kHz, 9 dB above the alignment signal level, equivalent to the permitted maximum programme-signal level. The sound-programme signal should be controlled by the sending broadcaster so that the amplitudes of the peaks only rarely exceed the peak amplitude of the PMS.

Note. — Under these conditions a peak programme meter will indicate levels not exceeding the level of the permitted maximum signal.

A numerical example may serve to clarify this definition. The alignment signal has an r.m.s. voltage of 0.775 V and a peak amplitude of 1.1 V at a zero relative level point. The instantaneous peak amplitude of the sound-programme signal at this point should only rarely exceed 3.1 V.

Although it is intended that the peaks of the sound-programme signal should not exceed the permitted maximum signal level, an overload margin must be provided so that rare excursions of the sound-programme signal above the permitted maximum signal level may be tolerated.

Note. — Annex I describes the response of peak programme and vu meters to these test signals.

ANNEX I

ALIGNMENT USING THE RECOMMENDED TEST SIGNALS
WITH PEAK PROGRAMME METERS AND VU METERS

1. Broadcasters have evolved, over a period of forty years, procedures for using both types of meter to control programme levels. These procedures are satisfactory to the organizations using them, so that they produce neither over-modulation, leading to distortion, nor under-modulation, leading to impairment from noise.

Although different kinds of programme material deflect the two meters differently, the organizations using them have evolved techniques that produce satisfactory level control and artistic balance within the programme.

* The CMTT and Study Group 10 will coordinate the future development of this Recommendation. This Recommendation should be brought to the attention of Study Group IV of the CCITT.

2. The sensitivity of peak programme meters (PPM) is such that a sine wave signal at the alignment level, 0 dBu0s, indicates "Test" on an EBU PPM (this corresponds to "4" on the BBC PPM and "-9" on the PPMs of the Federal Republic of Germany and the OIRT (see Fig. 1)).

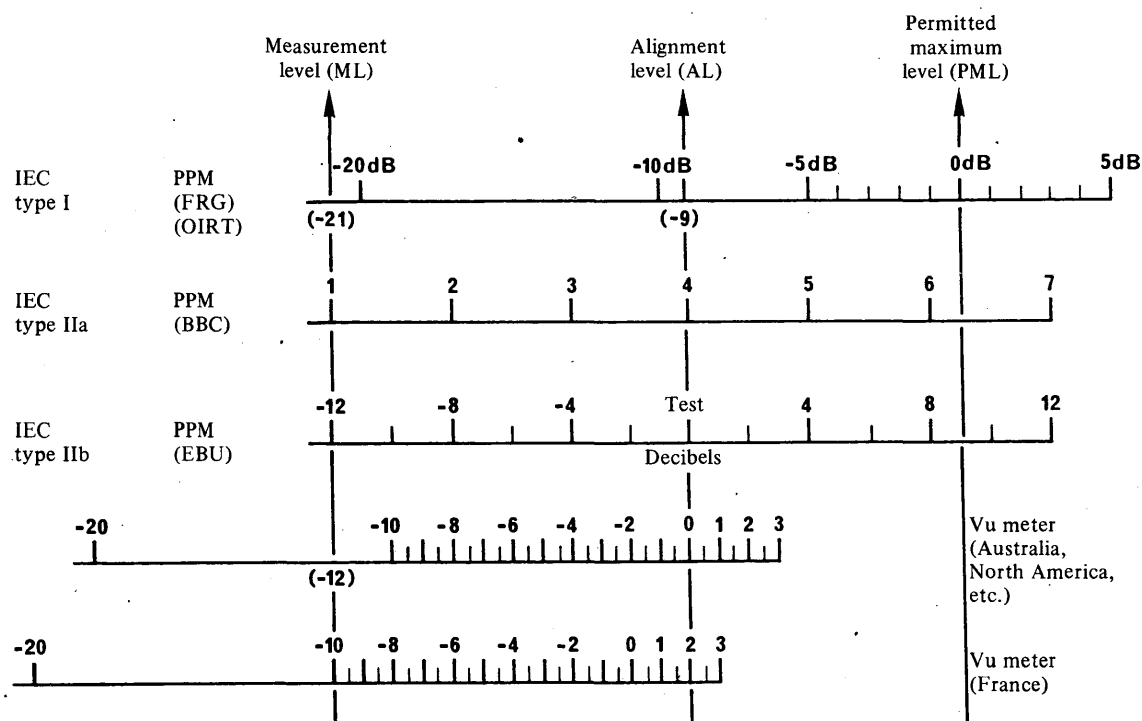


FIGURE 1 – Indications produced by various types of programme meter with the recommended test signals.

Note. – Meter indications are schematic – not to scale.

3. The sensitivity of the vu meter is such that a sine wave signal at the alignment level, 0 dBu0s, produces nearly full indication, 0 vu in Australia and North America and +2 vu in France (see Fig. 1).

4. The PPM reads "quasi-peak", that is, its peak indication on programme signals reads a little lower than true peaks. Operators are instructed to make the programme peaks give the same indication as a sinusoidal tone at +9 dBu0s (+8 dBu0s in some organizations). The true peaks of the programme are higher than indicated by up to 3 dB. When, additionally, operator errors are taken into account, the true peaks of the programme signal may reach the amplitude of a sinusoidal tone at +15 dBu0s.

5. The vu meter indicates the mean level of the programme, which is generally much lower than the true peak. Operators are instructed to make programmes peak generally to the 0 vu reading. Experience has shown that the true programme peaks are higher than indicated by between +6 dB and +13 dB, depending on the programme material. When, additionally, operator errors are taken into account, the true peaks of the signal may be up to 16 dB higher than indicated, corresponding to the peak amplitude of a sinusoidal tone at +16 dBu0s, or alternatively +14 dBu0s when application of the alignment level signal results in +2 vu indication.

6. Thus, although the dynamic characteristics of the two meters are different, the highest peak levels encountered after control using either meter are very similar.

7. Thus, an international connection between broadcasters will be correctly aligned regardless of the type of meter employed when a sinusoidal signal at alignment level, 0 dBu0s, produces the indication appropriate to that level at both the sending and receiving ends of the circuit.

To avoid any confusion between alignment level and other levels that might be used, it is recommended that the three level tone test signal described in Recommendation 661 be used for the alignment of an international sound-programme connection.

Figure 1 illustrates the indications given by a number of programme level meters when the recommended test signals are applied to them.

RECOMMENDATION 661*

SIGNALS FOR THE ALIGNMENT OF INTERNATIONAL SOUND-PROGRAMME CONNECTIONS

(Question 19/CMTT)

(1986)

The CCIR,

CONSIDERING

- (a) that Recommendation 645 defines three test signals to be used on international sound programme connections;
- (b) that no level information can be derived from a single-level test signal concerning its relationship to the levels defined in Recommendation 645;
- (c) that many impairments in international programme exchanges can be traced to misinterpretations of the single-level test signals;
- (d) that some administrations while employing analogue FDM circuits cannot accommodate high-level sinusoidal test signals,

UNANIMOUSLY RECOMMENDS

that international sound-programme connections should be identified and aligned using the definitions in § 1, the test signal format in § 2 and the measurement methods in § 3 of this Recommendation.

1. Definitions

1.1 Source identification

An announcement should be used to identify the originating point of the test signals and should be preferably as short as possible. It is suggested that such an announcement contain at least the following information:

- name of originating organization,
- location,
- country.

The sound-programme signal should be controlled by the sending broadcaster so that the amplitudes of the peaks only rarely exceed the peak amplitude of the permitted maximum (sine-wave test) signal.

1.2 Test signal and level definitions

1.2.1 Alignment signal (AS)

Sine-wave signal at 1 kHz** at a level of 0 dBm0s, which is used to align the international sound-programme connection.

1.2.2 Measurement signal (MS)

Sine-wave signal at 1 kHz** at a level 12 dB below the alignment signal level, which should be used for long-term measurements and measurements at all frequencies.

* This Recommendation should be brought to the attention of CCIR Study Group 10 and CCITT Study Group IV.

** This frequency is nominal, and 1020 Hz recommended by CCITT Recommendation O.33 may be used.

1.2.3 Permitted maximum signal (PMS)

Sine-wave signal at 1 kHz*, 9 dB above the alignment signal level, equivalent to the permitted maximum programme-signal level.

2. Test signal format

2.1 A three-level sinusoidal test signal at a reference frequency of 1 kHz* should be used to check the alignment of international sound programme connections. These three levels should be combined with the source identification and be repeated cyclically as specified in the format shown in Fig. 1 for monophonic and stereophonic connections.

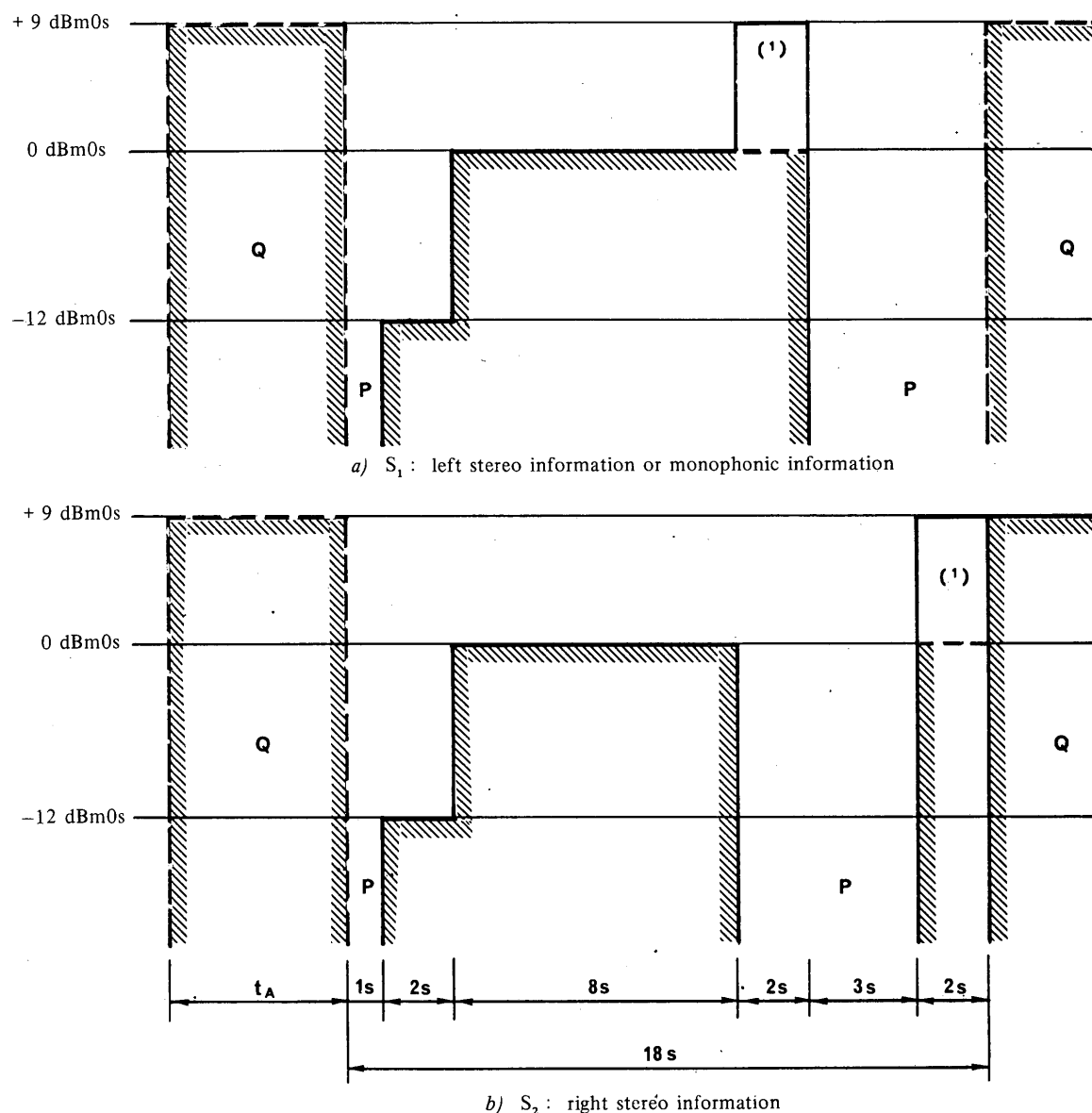


FIGURE 1 – Format for the three-level test signal for sound programme connections

Cycle duration = $t_A + 18$ s

Q: station announcement

t_A : duration of the station announcement.
It varies depending on the length of the message

P: signal pauses

(1) See § 2.2

* This frequency is nominal, and 1020 Hz recommended by CCITT Recommendation O.33 may be used.

2.2 During an initial period, until the CCIR decides otherwise, the portion of the test signal designated at +9 dBm0s in Fig. 1 will be replaced by a signal at 0 dBm0s. The resulting two-level test signal is required until all transmission systems are capable of carrying sinusoidal signals at +9 dBm0s without producing excessive channel loading or cross-talk into other channels.

2.3 Some organizations may not have automatically generated test levels as defined in § 2.1 and § 2.2 above. In these cases, the alignment level of 0 dBm0s at 1 kHz (see Note) should be used for the alignment of international sound-programme connections.

Note. — This frequency is nominal, and 1020 Hz recommended by CCITT Recommendation O.33 may be used.

3. Measurement methods

The fundamental concept of the test signals defined in this Recommendation is to provide organizations with accurate and well-defined levels [Thiele, 1984]. These levels are intended to provide rapid identification of level errors as well as to allow operational personnel sufficient time to make the necessary level adjustments at the appropriate points in the international sound-programme connection. Alignment of the connection is made by adjusting the alignment signal to the appropriate point on the programme level meter as defined in Annex I to Recommendation 645. Identification of left and right channels is provided as shown in Fig. 1. Organizations should give consideration to Reports 292 and 820 when establishing measurement procedures.

The three-level test signal defined in § 2.1 and the two-level test signal defined in § 2.2 also make provision for a brief subjective and/or objective noise measurement in the signal pauses (P) as shown in Fig. 1. These measurement pauses are not intended to replace the maintenance practices defined in the CCITT Series N Recommendations but rather to confirm that there are no gross noise or cross-talk impairments on the circuit.

REFERENCES

THIELE, A. N. [September, 1984] Three-level-tone test signal for setting audio levels. AES Australian Convention, Melbourne, Australia.

RECOMMENDATION 642 *

LIMITERS FOR HIGH-QUALITY SOUND-PROGRAMME SIGNALS

(Questions 46/10, 19/CMTT and 20/CMTT, Study Programme 46B/10)

(1986)

The CCIR,

CONSIDERING

- (a) that on high-quality analogue sound-programme circuits it is desirable to keep the signal level as high as is reasonably practicable in order to maximize signal-to-noise ratio but that overloading can lead to signal distortion and consequent programme quality impairment;
- (b) that on digital sound-programme circuits, similar considerations apply;
- (c) that techniques exist to design low-distortion limiters without overshoot (for example, by the use of delay lines);
- (d) that in certain circumstances interference may be caused by excessively high signal levels being applied to transmission equipment;
- (e) that the level of some sound signal components (most commonly those at the higher audio frequencies) may be raised at the system input by the introduction of pre-emphasis — for example, CCITT pre-emphasis for sound-programme links and 50 µs pre-emphasis for VHF FM broadcast transmission;
- (f) it is generally undesirable to subject sound signals to more than one limiting process, as unacceptable impairment may result. The connection of limiters at a number of different points in a sound-programme circuit is therefore inadvisable,

* Study Group 10 and the CMTT will coordinate the future development of this Recommendation.

UNANIMOUSLY RECOMMENDS

1. that low-distortion limiters should be employed at the interfaces between studios and sound-programme circuits. It is recognized that the use of limiters enables more efficient use to be made of the available dynamic range on sound-programme circuits. Moreover, the responsibility for their use should be clearly defined so that no more than one limiter is inserted in a sound-programme connection between broadcasters;
2. that if pre-emphasis is applied, the limiter should take account of this. An example of a limiter which does this is described in Annexes I and II.

ANNEX I

VARIABLE-EMPHASIS LIMITERS

Limiters are commonly employed at the programme inputs of frequency modulated sound-programme transmitters in order that the carrier deviation, and hence the signal-to-noise ratio at the receiver, may be kept as high as is practicable, whilst avoiding over-deviation and the consequent risk of audible distortion or of causing interference. Similarly, limiters may be used at the sending ends of point-to-point sound-programme transmission circuits, analogue or digital, so that the signal-to-noise ratios at the receiving ends are optimized, by permitting the signal levels on the circuits to be kept high without risk of distortion caused by over-loading.

In the above cases, the sound signals are commonly subjected to high-frequency pre-emphasis. However, programme levels are normally controlled using a meter (see Report 292) at a point in the circuit which is not subject to pre-emphasis (see Note). In consequence high-level, high-frequency, components are likely to cause overmodulation — even on properly controlled programmes — unless a limiter is employed to prevent this.

Note. — It has been suggested [CCIR, 1978-82b] that a pre-emphasized level metering and control point should be employed but in tests carried out by the BBC with 50 μ s pre-emphasis, the overall effect was found to be an undesirable potential under-modulation of the broadcast transmitter to the extent of about 4 dB.

In a conventional limiter, gain variations affect all sound signals equally, and limiter action brought about by high-amplitude high-frequency components may cause obvious and objectionable level fluctuations of low- and medium-frequency components in the reproduced programme. This effect, commonly known as “gain ducking” or “limiter-cut-back”, may be reduced by allowing a wide margin between the nominal maximum programme peaks and the limiting level. Such a practice is considered to be undesirable, as the listening signal-to-noise ratio would then be less than it otherwise could be. Tests have shown [Manson, 1973, 1975] that the need for this wide margin may be avoided by the employment of a frequency-selective form of limiter. Such frequency-selective or “variable-emphasis” limiters have attracted interest in the CMTT [CCIR, 1978-82a] and their use for controlling signal level at the inputs of digital sound-programme circuits is under consideration following the presentation of a draft Recommendation at the Final Meeting in 1981 (see Report 647).

REFERENCES

- MANSON, W. I. [1973] Frequency-dependent limiters for f.m. sound transmitters. BBC Research Department Report 1973 25.
- MANSON, W. I. [1975] Frequency dependent limiters for f.m. sound broadcasting: optimisation of attack period and of delay-length in the variable de-emphasis stage. BBC Research Department Report 1975/22.

CCIR Documents

[1978-82]: a. CMTT/239 (United Kingdom); b. CMTT/66 (LM Ericsson).

ANNEX II

DESCRIPTION OF A VARIABLE-EMPHASIS LIMITER FOR USE ON HIGH-QUALITY SOUND-PROGRAMME TRANSMISSION CIRCUITS

A block diagram illustrating one form of a variable-emphasis limiter which has been found to be suitable for use on 50 μ s pre-emphasized signals (a different value of pre-emphasis time constant may be used if appropriate) is shown in Fig. 1.

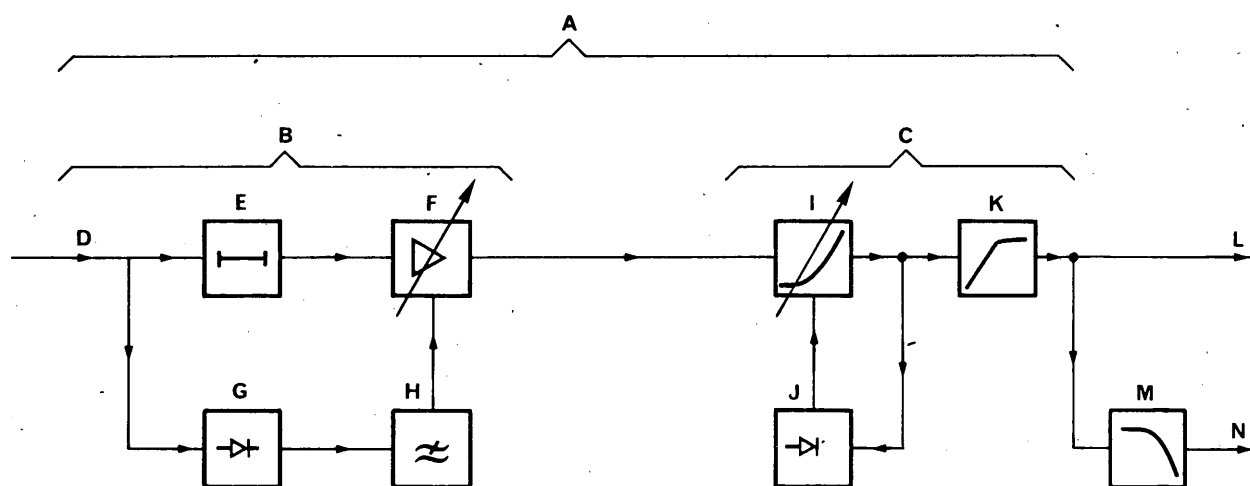


FIGURE 1 – Block diagram illustrating one possible variable-emphasis limiter arrangement

- A: variable-emphasis limiter (complete)
- B: 1st limiter stage (flat-spectrum, delay-line design)
- C: 2nd limiter stage (variable-emphasis design)
- D: audio input
- E: delay line
- F: variable-gain amplifier, with law appropriate for control provided by G
- G: control signal rectifier, smoothing and time-constant circuits
- H: shaping network (low-pass filter)
- I: variable-pre-emphasis circuit, 50 μ s quiescent
- J: control-signal rectifier, smoothing and time constant circuits
- K: peak clipper
- L: audio output, 50 μ s pre-emphasis protected
- M: 50 μ s de-emphasis network (optional)
- N: audio output, 50 μ s pre-emphasis protected signal with 50 μ s de-emphasis applied, if required (i.e. for monitoring)

In the first stage, the incoming audio signal is limited by a “flat-spectrum” limiter using delay-line and gain change-rate control techniques [Shorter *et al.*, 1967] to prevent transient overshoot.

In the second stage it is limited by a variable pre-emphasis circuit. This imposes 50 μ s pre-emphasis so long as the resulting output signal does not exceed the prescribed maximum value, but momentarily reduces the amount of pre-emphasis as necessary to ensure that the prescribed maximum output level is not exceeded when high-level high-frequency programme signal components are present.

Subjective tests [Manson, 1973] show a definite preference for a variable-emphasis limiter rather than a flat limiter. Further tests [Manson, 1975] have indicated that if a peak clipper, set about 0.75 dB above the limiting level, is employed at the output of the limiter as shown in Fig. 1, it is not necessary for the variable emphasis stage to include a delay line.

REFERENCES

- MANSON, W. I. [1973] Frequency-dependent limiters for f.m. sound transmitters. BBC Research Department Report 1973/25.
- MANSON, W. I. [1975] Frequency-dependent limiters for f.m. sound broadcasting; optimisation of attack period and of delay-length in the variable de-emphasis stage. BBC Research Department Report 1975/22.
- SHORTER, D. E. L., MANSON, W. I. and STEBBINGS, D. W. [1967] The dynamic characteristics of limiters for sound-programme circuits. BBC Engineering Division Monograph No. 70.

REPORT 820-1 *

**RELATIVE VALUES OF SOUND-PROGRAMME SIGNAL LEVELS
ESTABLISHED WITH THE VU METER AND WITH A
PEAK-PROGRAMME METER**

(Study Programme 17A/CMTT, Kyoto, 1978)

(1978-1986)

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 dBrs), 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 dBm0s.)

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 dBm0s 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 dBrs (+4 dBrs in France) 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. A 0 dBm0s sine-wave test signal at this point, generally called the "absolute level", has a level of +8 dBm.

* This Report should be brought to the attention of Study Group 10 and Study Group IV of the CCITT.

For a 0 dBm0s 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*

In the absence of limiters and compression devices, a test signal of 0 dBm0s 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 in North America circuits may also be aligned with a test tone with an electrical value of 0 dBm inserted at the +8 dBrs point. In terms of standard sound-programme terminology outlined in Report 292, this is a -8 dBm0s test tone. In terminology associated with telephony, the +8 dBrs point is in fact a 0 dBr point and the referred level (for 0 dBm) is 0 dBm0. This is a source of misunderstanding which can be overcome by the rigid use of proper terminology for referred and relative levels which differentiates between sound-programme and telephony practices.

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.

3.3 *A three-level test signal*

A test signal has been specified in Recommendation 645 as a means to assist in the identification of various test levels experienced in practice. The test signal comprises a 1 kHz tone presented cyclically at +9 dBm0s, 0 dBm0s and -12 dBm0s; and a stored speech message peaking to the permitted maximum programme level.

3.4 *Preferred monitoring instrument*

The CMTT considers that programme meters conforming to IEC Publications 268-10 and 268-10-A are suitable for measurements of levels in international sound-programme connections.

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 connected.

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. This, in part, is due to differences in reaction to programme signals of peak programme meters and vu meters. It is possible that over a properly aligned circuit, programme signals with high peak-to-average level characteristics will appear high on peak-programme meters when set up on vu meters. Conversely, programme signals with low peak-to-average characteristics will appear high on vu meters when set up on peak-programme meters. This residual error due to meters should not generally exceed 2 or 3 dB when the programme levels are controlled by expert operators.

A more important factor for interconnection is the establishment of the proper relative level at the interconnection point between circuits using the two methods of level measurement. Incorrect levels between circuits connected with North America will be overcome by considering that the standard reference level point in North America is generally agreed as a +8 dBrs point, i.e., when the standard sine-wave reference level (0 dBm0s) is set to +8 dBm. Appropriate level normalization to match the relative levels of the circuit at the point of interconnection will ensure that level discrepancies will be limited to those caused by the residual differences between the two types of meters used.

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.

BIBLIOGRAPHY

- CCIR Documents*
 [1982-86]: CMTT/222 (Canada).

REPORT 642

INTERRUPTION TO SOUND-PROGRAMME SERVICES

(Study Programmes 17C/CMTT and 19B/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 (Geneva, 1982)), 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 822-2 *

TRANSMISSION OF SOUND-PROGRAMME SIGNALS OVER LONG DISTANCES **

Routine automatic testers for sound-programme circuits

(Question 19/CMTT)

(1978-1982-1986)

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, O.32 and O.33 provide comprehensive coverage of automatic measurement methods.

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 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.

* 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.

On the other hand, in addition to the amplitude/frequency response and weighted and unweighted noise, the RAI equipment permits the measurement 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.

Automatic monitoring systems for sound-broadcasting circuits without any interruption of the normal programme signal, have been developed by the Federal Republic of Germany [CCIR, 1978-82] and by the OIRT. The system of the Federal Republic of Germany has been in use in the Federal Republic of Germany for several years and is described in part C.

In the USSR, a complete system for automatic monitoring of the performance characteristics of a broadcast channel, including the whole transmission chain and transmitters, is under consideration [CCIR, 1974-78].

In the OIRT, two different monitoring systems are used (i.e. in the USSR and in the German Democratic Republic) [CCIR, 1982-86a and b]. These systems are described in part D.

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.

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 and there is insufficient time available before the next programme transmission to complete the test. In this situation, 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 $2\frac{1}{2}$ 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 suitably chosen frequencies in order to give the best results for various types of equipment, e.g. cables, radio links, broadcast transmitters.

The chosen frequencies are: 0.063, 1, 4.5, 10 and 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, in all cases, be the most suitable and it is therefore possible to select, by means of a control placed on the front panel, a "measurement programme", i.e. a deflected level for each test tone or the suppression of some of them.

The measurement programmes may be pre-arranged by setting 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 display 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, including the internal switching devices.

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 starts to make the measurements on the subsequent frequency groups.

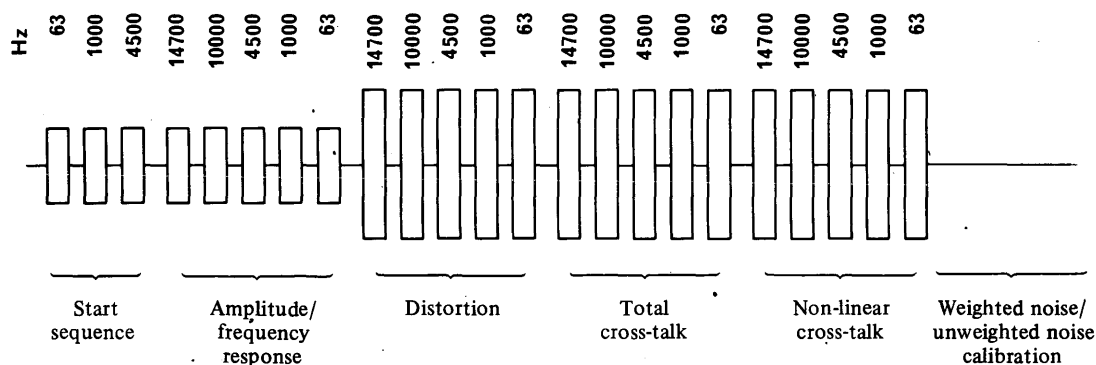


FIGURE 1

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 connected to an output amplifier which feeds the circuit under test.

The nominal output level is +9 dBm and it is adjustable by 1 dB steps according to the characteristics of the circuit under test.

Additionally, the transmitter logic circuit can insert an attenuator and an amplitude-correcting network according to the measurements to be carried out and the "programme" chosen.

A pause occurs at the end of all the frequency groups, while the circuit is still connected to the measuring system. During this time interval, the receiving unit carries out a noise measurement, after which the counter resets a 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 unit but, during cross-talk measurements, it is automatically switched to the other channel. For the complete test of a stereophonic circuit, two measurement cycles are needed. During the first cycle the transmitter unit is connected to channel L, while during the second, it is connected to channel R (or vice versa).

In order to carry out the measurement of harmonic distortion, a bandstop filter for the frequency under examination is inserted and the cross-talk measurements are undertaken on the channel not fed by the transmitter unit. The total cross-talk is measured with the bandstop filters out of circuit while the non-linear cross-talk measurements include these filters. A psophometric filter is inserted when measuring the weighted noise.

Between the non-linear cross-talk and noise measurements, 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.

The control 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 to 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 stopped in the middle of a measuring cycle is avoided.

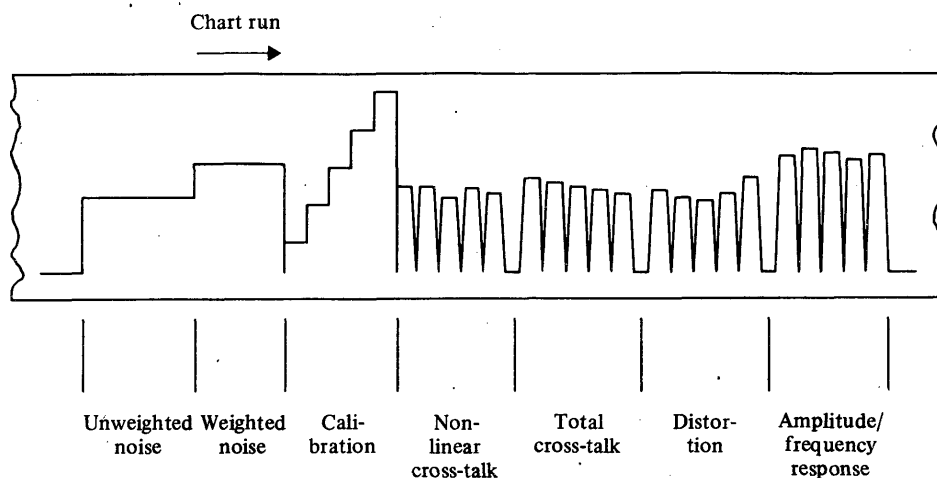


FIGURE 2

C. AUTOMATIC MONITORING EQUIPMENT USED IN THE FEDERAL REPUBLIC OF GERMANY

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.

D. AUTOMATIC MONITORING SYSTEMS USED BY THE OIRT

1. Automation of the control processes used for the monitoring of sound broadcasting channels during transmission by means of data signals

1.1 Scope of application

The equipment provides for the automation of the control processes used for monitoring a 15 kHz sound broadcasting channel by means of a data channel located within the sound transmission channel.

1.2 Introduction

Efficient automatic monitoring of a sound broadcasting channel requires the provision of data signals within the nominal frequency band of the sound broadcasting channel. It is proposed that the following signals should be transmitted:

- synchronizing signal,
- programme identification signals,
- remote-control signals,
- data on the quality of the sound broadcasting channel.

The data signals should be combined in a data channel located in the upper frequency portion of the sound transmission channel that is to be checked and sent from the programme source to the final user.

1.3 *Requirements for the data and sound channels*

The data channel is superimposed on the supervised channel. For a 15 kHz type sound transmission channel, the following values are proposed:

- amplitude-frequency response from 40 Hz to 14.5 kHz ± 0.2 dB
- attenuation of the data channel in the sound frequency band ≥ 50 dB
- total harmonic distortion $\leq 0.2\%$
- permitted maximum level $+ 9$ dBm0s

1.4 *Requirements for the data signal*

- bit rate 50 bit/s
- data structure: 1 data frame consisting of:
 - 6 bits for synchronization
 - 6 bits for programme identification
 - 6 bits for remote control
- data transmission speed 2 frames/s
- data coding bi-phase code/split-phase-level
- carrier frequency 15 000 Hz
- carrier frequency modulation phase modulation
- carrier frequency level -66 dBm0

2. **Automatic monitoring of the quality of sound transmission channels of all types during transmission**

Taking into consideration that subjective monitoring of the sound transmission quality in sound broadcasting is difficult to realize, that the measurement of quality parameters of the channels during the sound broadcasting signal transmission causes difficulties, and that in case of international transmissions any type of channel (with a 7 kHz, 10 kHz or 15 kHz bandwidth) may be used, the OIRT Technical Commission proposes, that during the transmission of sound signals over international circuits the following should be ensured:

2.1 Automatic checking of:

- level stability,
- exceeding of the admissible interference level,
- transmission interruption.

2.2 The check should be effected at points A, B, C, D (Recommendation 502, Annex I) of the international sound transmission circuit as well as on the national sections according to need.

2.3 Automatic checking of the quality should be made with the aid of a pilot signal inserted into the sound transmission channel during programme and technical pauses in the sound-programme signal transmission of at least 1.5 s duration.

2.4 The pilot signal frequency should lie in the middle of the effectively transmitted frequency band (e.g. 800 Hz).

2.5 For purposes of identification, the pilot signal frequency should be chosen from among certain given frequencies.

2.6 The level of the pilot signal inserted into the sound transmission channel during pauses should provide the following ratios of permitted maximum sound signal level to pilot signal level:

- for the section of the monophonic channel in the broadcasting centre and over the international sound transmission channel used for a monophonic sound programme transmission: 78 dB;
- for the section of the stereophonic channel in the broadcasting centre and the international stereophonic programme transmission channel: 82 dB.

2.7 Equipment for automatic checking of the quality of sound channels should ensure checking of the following quality parameters:

- interruption of the channel or reduction of insertion gain by 12 dB during a 3 to 5 s period;
- variation of insertion gain by ± 3 dB, during periods of at least 300 ms, for more than three times per hour;

Supervision of the sound signal parameters:

- sound signal falling below -43 dB during the transmission time;
- reduction of the signal level by 6 dB during 4 to 5 min.

REFERENCES

CCIR Documents

[1974-78]: CMTT/109 (USSR).

[1978-82]: CMTT/263 (Germany (Federal Republic of)).

[1982-86]: a. CMTT/187 (OIRT); b. CMTT/188 (OIRT).

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 (see Note) 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.

Note. — The first zero crossing of the spectral power density function is at the sampling-frequency.

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.

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).

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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 (see Note), 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.

Note. — 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 488-4*

**TRANSMISSION OF SOUND AND VISION SIGNALS BY TIME-DIVISION MULTIPLEX
OR FREQUENCY-DIVISION MULTIPLEX**

(Study Programmes 21A/CMTT and 21B/CMTT)

(1970-1974-1978-1982-1986)

1. Introduction

It is a common practice to convert sound and vision signals into a form specially suited for the effective use of a specific transmission circuit.

In particular, 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. Furthermore, such systems enhance the utilization efficiency of a circuit's frequency bandwidth. If sufficient bandwidth is available, multiplexing of more than one vision signal might be possible. At the output of the dedicated transmission circuit, the original input signals have to be reconstituted. The CMTT has been studying these techniques and the results which have been obtained so far are summarized below.

Note. — 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.

In the United Kingdom, a time-division multiplex equipment [CCIR, 1970-74a] for providing a single high-quality music channel from binary coded pulses transmitted during the line synchronizing pulse period was produced many years ago and is now widely used in the United Kingdom and elsewhere [CCIR, 1974-78a and b]. It operates satisfactorily over long international television connections (see Recommendation 603) when these are suitably equalized. Tests have shown however [CCIR, 1974-78c] that for half bandwidth satellite transponder operation, the sound pulse amplitude should be reduced by 6 dB. Other information is contained in [Shorter, 1969] and in Table I.

The United Kingdom has recently developed a new time-division multiplex equipment [CCIR, 1982-86a] which provides two high-quality sound channels, each of 15 kHz bandwidth with a cross-talk ratio more than adequate for the use of separate programme signals such as dual language transmission. Each audio signal is sampled at 32 kHz, not synchronously with the video signal line frequency. The 14-bit audio data samples are reduced to 10 bits by near-instantaneous companding [CCIR, 1978-82a] and multiplexed into a common bit-stream. These data are inserted into the line synchronizing pulse in the form of a marker pulse plus 22 quaternary (4-state) symbols, each symbol representing 2 bits. Tests have confirmed a satisfactory immunity from the effect of transmission distortions over national and international circuits including satellite links [Holder *et al.*, 1984]. The physical size of the new system has been made the same as that of the original single channel equipment, using LSI techniques to accommodate the NICAM processor [Robinson, 1985]. Other information is contained in the References and Table I.

* This Report should be brought to the attention of Study Group 9.

TABLE 1 – Summary of TDM systems

Country and CCIR reference	United Kingdom [CCIR, 1970-74a]	United Kingdom [CCIR, 1982-86a]	USSR [CCIR, 1966-69]		Japan [CCIR, 1970-74c]	Federal Republic of Germany [CCIR, 1970-74b; 1974-78d; 1978-82c]	German Democratic Republic [CCIR, 1974-78g]
Video system	625-line (PAL) 525-line (NTSC)	625-line (PAL) Adaptable to all systems	625-line (SECAM)		525-line (NTSC)	625-line (PAL)	Adaptable to all systems
Audio-modulation system	Pulse-code modulation	Pulse-code modulation with 14/10 near-instantaneous companding	Pulse-width modulation		Pulse-code modulation (Ternary of colour burst)	Pulse-code modulation (Binary NRZ code) with 14/11 instantaneous companding	Pulse-code modulation (Quaternary NRZ code)
Video waveform modification	Increased duration of line-frequency equalizing pulses	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	Increased back porch of line blanking interval by reduced duration of line-sync. pulse	a) Line-sync. pulse decreased in duration b) Line-blanking inserted through field blanking c) Colour burst modified
Audio bandwidth (kHz)	14	15	6	10 to 14	14	15	15; 7; 3.4
Number of audio channels	1	2	1	1	2	2 (15 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	65 dBq0ps	60 (video S/N ratio: 41 dB)	53 (with compandor) (video S/N ratio: 55 dB)	52	– 58 dBq0ps (idle channel noise)	63 dB } idle and } channel 65 dB } noise (weighted according to Recommendation 468)
Audio S/N dependent upon video S/N	No	No	Yes	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. For the definition of noise levels, see Recommendations 574.

In the USSR, 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 USSR, 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].

[CCIR, 1970-74b] 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, 1978-82b] a modification of the system is described which improves the mode of operation and increases transmission reliability. The modified equipment uses the source coding given in Report 647, Table I, column B5 [CCIR, 1978-82c]. For insertion of two sound signals, the back porch has to be extended by reducing the line sync. pulses. 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. Additional information on the transmission performance with low signal-to-noise ratios in an FM satellite link can be found in [CCIR, 1982-86b].

[CCIR, 1970-74c] 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].

[CCIR, 1974-78f] 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.

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.

3.1 *Frequency-division multiplex of one composite signal with sound signals*

The method of diplexing sound channels on one or more sub-carriers can be implemented without any change of the video waveform. Any resultant degradation can be limited to acceptable values for terrestrial circuits up to 6400 km and for satellite circuits. This method can be valuable for circuits using frequency modulation. Sound sub-carriers are generally frequency modulated but other types of modulation may be used.

Where frequency modulation is used, it is important that the received levels of the sub-carriers are sufficient to provide an adequate margin above the FM threshold [CCIR, 1982-86c]. Additional information can be found in Report 632.

The use of digital modulation will result in multi-channel sound capability with only one sub-carrier.

Particular solutions using sub-carriers have been described in [CCIR, 1974-78f] in which two methods currently being employed in the United States of America 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. 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.

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. Other sub-carrier frequencies can also be used, but an analysis must be performed on the system intermodulation characteristics and their impact on video S/N ratio.

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 504-2 (Geneva, 1982). Test results on transmission characteristics and subjective assessment were satisfactory [CCIR, 1978-82d].

Other solutions using this technique have been described in [CCIR, 1978-82d] in which a method currently being employed in France is mentioned (similar systems will be employed in Norway [CCIR, 1978-82f], the Federal Republic of Germany and the Netherlands). This uses a 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.

3.2 *Frequency-division multiplex of one multiplexed analogue component system with additional sound or data signals*

The D2-MAC/packet system which is described in Report 1073 has the useful characteristic that its nominal baseband bandwidth of 8.4 MHz can be reduced, as desired, down to 5 MHz with a proportional reduction in the bandwidth of the decoded luminance and colour difference signals. Therefore, it is fully compatible with those existing networks which use FDM techniques to multiplex additional sound and data signals. At the same time, it ensures a good match between picture quality and efficient use of the available bandwidth of the transmission circuit. The insertion or extraction of these sound or data signals does not require any processing of the D2-MAC/packet, thus simplifying operations on the network. These multiplexing techniques using the method described in § 3.1 [CCIR, 1978-82e] have been tested successfully on the French TDF microwave network [CCIR, 1982-86d].

4. Simultaneous transmission of more than one vision signal over a single transmission circuit

In Italy, an investigation has been carried out by the RAI to assess the possibility of exploiting the network of contribution links, whose channel spacing conforms to Recommendation 383, with baseband signals having a bandwidth up to about 10 MHz. The results have been satisfactory [CCIR, 1982-86e]. A foreseen application consists of the simultaneous transmission of two colour TV programmes on a single circuit. A previous design of the system [Barbieri, 1982] was based on the separate transmission of a multiplex of time-compressed luminance and colour-difference components; a new development is now in progress in order to exploit the full available bandwidth of the transmission circuit.

In Japan, a system has been developed and used for ENG which simultaneously transmits two video signals over a single transmission circuit [CCIR, 1982-86f]. On the transmitter side, the two video signals are compressed horizontally by one-half and combined to form the left and right sides of a composite signal for transmission. On the receiver side, each half of the composite signal is expanded two-fold to reconstruct the original video signals.

5. 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, 1978-82g].

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.

6. Compatibility with sound programme signals on digital paths

[CCIR, 1970-74d] 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.

TABLE II — Summary of FDM systems

Country and CCIR references	United States [CCIR, 1974-78f]		Japan [CCIR, 1978-82d]	France [CCIR, 1978-82e]	Norway [CCIR, 1978-82f]
Video system	525-line (NTSC)		525-line (NTSC)	625-line (SECAM, PAL)	625-line (PAL)
Audio modulation system	Digital 4-PSK	FM	FM	Digital 4-PSK	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.

REFERENCES

- BARBIERI, G. F. [September, 1982] Simultaneous transmission of two TV programmes on a single link. IEE Conf. Publ. No. 220. Ninth International Broadcasting Convention (IBC 82), Brighton, UK.
- ESIN, V. T., TOVBIN, M. N. and SKLIZKOV, V. M. [1968] A method for the transmission of the picture and sound signals on the same channel in colour television (in Russian). *Tekhnika kino i Televideniya*, 6.
- FUJIO, T. and KOMOTO, T. [1971] Transmission of the television sound signal with the colour synchronizing signal. *EBU Rev.*, 128-A, 145-148.
- HOLDER, J. E., SPENCELEY, N. M. and CLEMENTSON, C. S. [21-25 September, 1984] A two-channel sound in syncs transmission system. IEE Conf. Publ. No. 240. Tenth International Broadcasting Convention (IBC 84), Brighton, UK.

- KATAEV, S. I. and ZUBAREV, Y. B. [1965] Transmission of picture and sound components in combined frequency band (in Russian). *Tekhnika kino i Televideniya*, 7.
- MATSUMOTO, SAMEJIMA and KUROKAWA [1980] Television programme relay system for simultaneous transmission of picture signal and stereophonic or bilingual sound signal. *Japan Telecomm. Rev.*, Vol. 22, 1.
- ROBINSON, J. [July/August, 1985] An l.s.i.-based p.c.m. processor for high-quality sound transmission. *J. IERE*, Vol. 55, 7/8, 247-252.
- SEVALNEV, L. A. and TSIRLIN, V. M. [1969] Transmission of the sound component over satellite communication circuits (in Russian). *Vestnik Sviazi*, 1.
- SHORTER, D. E. L. [1969] The distribution of television sound by PCM signals incorporated in the video waveform. *EBU Rev.*, Part A, 113, 13-18.
- TALYZIN, N. V., KANTOR, L. Y. and TSEITLIN, M. Z. [1967] "Orbita" earth station for receiving television programmes from artificial earth satellites (in Russian). *Elektrosviaz*, 11.
- YASCHENKO, K. A. and ZVEREV, Y. B. [1964] Transmission of television sound component by time division multiplexing of the video signal by phase modulated pulses (in Russian). *Elektrosviaz*, 8.

CCIR Documents

- [1966-69]: CMTT/144 (USSR).
- [1970-74]: a. CMTT/6 (United Kingdom); b. CMTT/197 (Germany (Federal Republic of)); c. CMTT/12 (11/34) (Japan); d. 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/44 (United States of America); g. CMTT/187 (German Democratic Republic).
- [1978-82]: a. CMTT/240 (United Kingdom); b. CMTT/82 (Germany (Federal Republic of)); c. CMTT/79 (Germany (Federal Republic of)); d. CMTT/53 (Japan); e. CMTT/90 (France); f. CMTT/6 (Norway); g. CMTT/52 (Japan).
- [1982-86]: a. CMTT/239 (United Kingdom); b. CMTT/215 (Germany (Federal Republic of)); c. CMTT/5 (United States of America); d. CMTT/181 (France); e. CMTT/232 (Italy); f. CMTT/156 (Japan).

BIBLIOGRAPHY

- FUJIO, T. and KOMOTO, T. [1973] A colour TV "sound-in-vision" system. *NHK Tech. J.*, Vol. 25, 1.

CCIR Documents

- [1974-78]: CMTT/258 (Italy).

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].

REFERENCES

SHERRY, L. A. [1976] An assessment of teletext data transmission over the UHF television network. Independent Broadcasting Authority (United Kingdom) Teletext Transmission Working Group Note 10.

CCIR Documents

[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.

* 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.

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.

REFERENCES

CCIR Documents

[1978-82]: CMTT/63 (LM Ericsson).

REPORT 823-1

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)

(1978-1986)

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,
- transmission of data for closed captioning (sub-titling) of the programme.

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.

National and international exchange of programmes with accompanying sub-title data is expected to increase. Where required, it would be desirable for transmission systems to pass this data without corruption and, if regeneration is necessary, without line shift or more than one frame delay. The possibility therefore exists for the development of a method for the standards conversion of sub-title data to accompany the standards conversion or transcoding of the video (see Report 1080).

2. Allocation of lines

Report 314 proposes the allocation of lines tabulated below:

Purpose	Standard			
	525 lines		625 lines	
	Line No.	Signal (see Recommendation 473)	Line No.	Signal (see Recommendation 473)
International ITS	(¹)	Figs. 5 and 6	17-18-330-331 (²)	Figs. 1, 2, 3, 4
International noise	—	(³)	22-335	—
Data signals	—	(⁴)	16-329	(⁴)
National(⁵)	15 both fields (⁶) 16-17-18-19-20-21 both fields	(⁷)	19-20-21-332-333-334	—

(¹) The majority of the administrations reserve line 17 for insertion of international test signals. Report 314 provides information on the current allocation of lines reserved for special signals.

(²) This should not prejudice their use for broadcasting signals such as teletext on a national basis.

(³) It has been suggested that 525-line noise measurement might be possible using the bar element B₂ of the ITS on field 1.

(⁴) In the EBU and the USA, experiments are being carried out with digital signals.

(⁵) Must be removed if so requested by an authority at the receiving end of the international circuit.

(⁶) Canada and the United States of America.

(⁷) Line 19 is reserved for Vertical Interval Reference signals in Canada and the United States of America.

In some countries, line 21 and often the corresponding line of the alternate field (line 334 in a 625/50 system) are now dedicated for closed caption data which provides sub-titles for the hard of hearing [CCIR, 1982-86a].

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 at present the minimum number of signals which it is desirable to process by an international converter, is as follows:

3.1 *For a 525-line input:* the ITS on line 17 of fields 1 and 2 for Canada and Japan and line 20 of field 2 for the United States of America [CCIR, 1982-86b].

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 (differential phase only in the PAL case) 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 § 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.

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.

CCIR Documents

[1982-86]: a. CMTT/209 (Australia); b. CMTT/9 (USA).

REPORT 1096 *

TRANSMISSION OF TELEVISION SIGNALS USING MULTIPLEXED ANALOGUE COMPONENTS

(Question 24/CMTT)

(1986)

1. Introduction

For various applications, the use of television signals comprising time-compressed multiplexed analogue components (MAC) instead of conventional composite signals such as NTSC, PAL, or SECAM has been proposed. Various systems of this kind have been developed for satellite broadcasting, e.g. B-MAC, C-MAC, D2-MAC, and are described in Report 1073. When a television signal of this kind is conveyed over a transmission circuit, it has the advantage of not being subject to impairments associated with a colour sub-carrier. However, some forms of distortion, not considered relevant to composite signals, may become important and a greater transmission bandwidth may be required. Some MAC systems provide a baseband multiplex of vision, sound and data signals in which case the same modems used for terrestrial transmission of composite signals may be used, provided that they have sufficient bandwidth.

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

2. Results of transmission tests with MAC signals

Tests described in [CCIR, 1982-86a] have been carried out in France by TDF concerning the transmission of the D2-MAC/PACKET system as described in Report 1073 on microwave circuits of up to 70 hops. Any clamps, designed for composite video signals were removed for the transmission of the MAC signals. Performance was found to be satisfactory provided the circuit bandwidth at baseband exceeded 8.4 MHz. In the tests of the sound/data channel, bit error ratio (BER) degradation was found to be comparable to that caused by a decrease in S/N of not more than 1 dB. The pre-emphasis characteristic employed is described in Recommendation 405, curve C (819 lines).

[CCIR, 1982-86b] gives information about an investigation carried out in Italy by the RAI. Its purpose was to assess the possibility of transmitting television signals of greater bandwidth than conventional systems in the network of contribution links with channel spacing according to Recommendation 382. The preliminary results show the possibility of exploiting the network with signals having a bandwidth up to 10 MHz. A potential application would be to transmit the time-compressed and multiplexed analogue components for programme contributions from one studio centre to another; the actual bandwidth of the luminance and of the two simultaneous colour-difference components would be close to that of Recommendation 601.

Tests conducted by AUSSAT in Australia [CCIR, 1982-86c] have shown that B-MAC signals can be successfully transmitted and received using satellite FM terminal equipment and terrestrial microwave link equipment. Modifications to make the video modulator for the satellite circuit, designed for PAL operation, compatible with B-MAC have been successful. The terrestrial microwave equipment used pre-emphasis described in Recommendation 405 (curve B) and was found to operate satisfactorily without modification, although improved performance was realized by the addition of a baseband group delay equalizer. Performance monitoring using standard insertion test signals (ITS) was still possible after modifications to both the PAL-RGB decoders and the equipment monitoring system software.

Circuits subject to long-time waveform distortion, or spurious mains frequency noise (hum) will require new clamps designed specifically for MAC signals. These new clamps should also be capable of operating with composite signals, preferably without manual intervention.

3. Methods appropriate to measurements on transmission circuits carrying MAC signals

A clear description of what is meant by the transmission circuit is essential for defining the measurement problem. Figure 1 shows a studio encoder driving the transmission circuit, and a studio decoder driven by the transmission circuit. MAC is a multiplex of luminance, chrominance, and sound/data signals. The video inputs to the encoder are the luminance component and the two colour difference components. These are also present at the output of the decoder. The transmission problem of concern to the CMTT deals with the transmission circuit between the MAC encoder and the MAC decoder. It will also be required to develop new test methods and test signals appropriate for testing both encoder and decoder.

3.1 Vision

A number of organizations are engaged in preliminary studies on the measurement and assessment of time-multiplexed component systems and this Report is intended to draw attention to the reported information on these studies. It is generally agreed that the test signals and methods of Recommendation 567 should be used wherever they are appropriate. However, the measurement of sub-carrier related distortions is considered as having little or no relevance to circuits carrying MAC signals. Weighted measurements of continuous random noise using the unified weighting network are inappropriate. These tests must be replaced by new tests with appropriate weighting networks, and test signals. Consideration must also be given to the particular measurement requirements of circuits carrying scrambled MAC signals (see Report 1091).

A preliminary review of these problems together with some suggested new test waveforms have been given by James and Marshall [1984]. Preliminary discussion on the choice of test signals, supported by the results of some measurement, has been reported by Wilcock [1984]. More recent discussions of the choice of test signals have been given by Holder *et al.* [1985] and James [1985], while a test-signal generator for producing suitable waveforms has been described by Harm [1985]. Ferbrache [1985] and Baker [1985] have described an ingenious method of vector display within a studio, which permits the comparison of pairs of all three of the component signals.

The measurement of linear waveform distortion in component systems has been reviewed by Macdiarmid and Patel [1985], who conclude that the methods of Recommendation 567 supplemented by the response to a T sine-squared pulse are suitable, provided that the T and the $2T$ pulse widths are chosen appropriate to the channel bandwidths. An alternative test signal to the T pulse has been proposed [CCIR, 1982-86d] in order to avoid the use of a measurement filter as described in former Recommendation 451. The proposed test signal is a close approximation to the T -pulse after filtration by an ideal low-pass filter of cut-off frequency $1/2T$. It can be generated in a reproducible manner by digital synthesis. In the case of transmission circuit testing, the appropriate time interval, T , is influenced by the time-compression factors and studio component bandwidths.

A feature of the above references is that the proposals for test signals and measurements are all very similar; differences are mainly in points of detail. However, the choice of the test signals and measurements to be recommended must await experience of measurements on component systems and information on how the measured parameters are related to the subjective quality of pictures.

3.2 Sound/data

No contributions on measurement methods for sound/data signals in MAC have been received. Further study is required.

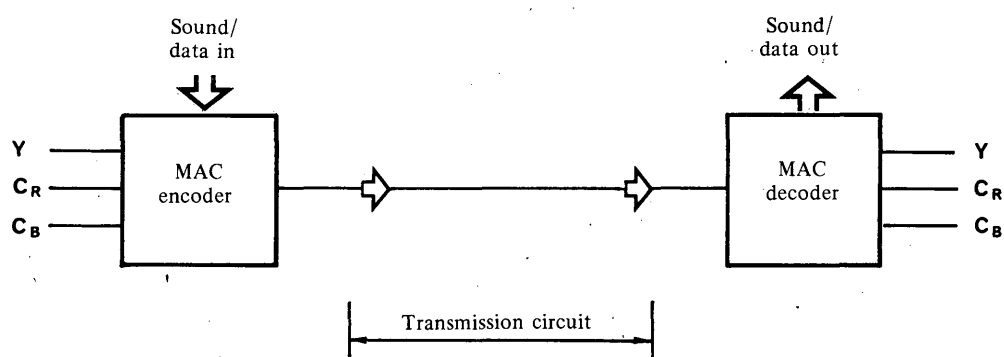


FIGURE 1 – Transmission of MAC signals

REFERENCES

- BAKER, D. [15 February, 1985] A new and unique method for measuring video analog component signal parameters. 19th Annual SMPTE Television Conference, San Francisco, CA, USA.
- FERBRACHE, R. [1985] New and unique method for measuring video analogue component signal parameters. Colloquium Digest, 1985/23, 5/1-5/2. IEE Colloquium on Component TV Measurements and Their Relevance.
- HARM, H. [1985] The development of a video component signal generator. Colloquium Digest, 1985/23, 4/1-4/4. IEE Colloquium on Component TV Measurements and Their Relevance.
- HOLDER, E., BOWD, D. M. and FERNE, A. C. [1985] Measuring techniques and test equipment for component analogue systems. Colloquium Digest, 1985/23, 1/1-1/5. IEE Colloquium on Component TV Measurements and Their Relevance.
- JAMES, A. [1985] Measurements in a component environment. Colloquium Digest, 1985/23, 2/1-2/3. IEE Colloquium on Component TV Measurements and Their Relevance.
- JAMES, A. and MARSHALL, P. J. [21-25 September, 1984] Measurements in a television component environment. IEE Conf. Publ. No. 240, 383-391. Tenth International Broadcasting Convention (IBC 84), Brighton, UK.
- MACDIARMID, I. F. and PATEL, B. R. [1985] Measurement of linear waveform distortion in component systems. Colloquium Digest, 1985/23, 3/1-3/5. IEE Colloquium on Component TV Measurements and Their Relevance.
- WILCOCK, P. E. [21-25 September, 1984] Analogue component video systems: measurement, instrumentation and alignment. IEE Conf. Publ. No. 240, 392-394. Tenth International Broadcasting Convention (IBC 84), Brighton, UK.

CCIR Documents

[1982-86]: a. CMTT/180 (France); b. CMTT/232 (Italy); c. CMTT/207 (Australia); d. CMTT/163 (United Kingdom).

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 Recommendations 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, 1978-1982 and 1982-1986.

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.

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, 1978-1982 and 1982-1986.

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;
- (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, 1974-1978 and 1982-1986.

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 (Study Period 1981-1984) and Recommendation G.229, which consider 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 monochrome 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;

* 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.

- (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,

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.

STUDY PROGRAMME 13G/CMTT*

USE OF MOBILE AND TRANSPORTABLE TRANSMITTING EARTH STATIONS FOR OUTSIDE TELEVISION BROADCASTS OVER SATELLITES

(1986)

The CCIR,

CONSIDERING

- (a) that in point-to-point television transmission via satellite, quality objectives are fixed on the basis of large-sized transmitting and receiving stations;
- (b) that the introduction of mobile and transportable transmitting stations is essential for television operation and provides a satisfactory technical solution for problems of outside television broadcasting;
- (c) that the special characteristics needed for mobile and transportable stations may necessitate acceptance of transmission performance objectives differing from those specified in Recommendation 567;
- (d) that it may be necessary to provide, on the same satellite bearer, the auxiliary signals required for the operation of mobile and transportable transmitting stations and that these auxiliary signals may have an effect on the transmission quality of the television signals,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. determination of overall transmission objectives;
2. determination of the additional requirements for operation and control of mobile and transportable transmitting stations.

Note. — See Report 1090.

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;

* This Study Programme should be brought to the attention of Study Groups 4, 8 and 11.

** 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.

(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 Recommendations 604, 658 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;
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 Recommendations 604 and 658 and Reports 646, 819, 823, and 967.

* 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.

** Studies under this Study Programme should be carried out in close collaboration with Study Group 11.

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 Recommendations 604 and 658.

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;
- (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 (this includes the case where the digital coding format of the signal may be changed between sections);
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 Recommendations 604 and 658.

* This Study Programme should be studied in close coordination with Study Programme 16A/CMTT.

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 639, 646 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;
3. the effect of such limits on the planning of the international long-distance digital network which may convey broadcast television signals.

Note. — See Recommendation 658 and Report 1093.

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,

* This Study Programme should be studied in close coordination with Study Programmes 14A/CMTT, 14B/CMTT, 14C/CMTT and 16A/CMTT.

** The results of this Study Programme should be brought to the attention of CCITT Study Group XVIII.

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.

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 Recommendation 473 and Reports 314 and 819.

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;
- (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,

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-2/CMTT

AVAILABILITY OF AN INTERNATIONAL
TELEVISION CIRCUIT OR CHAIN

(1974-1978-1982-1986)

The CCIR,

CONSIDERING

- (a) 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);
- (b) 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 (picture and associated sound). Identification of any differences between analogue, digital or mixed transmissions in this respect;
2. analysis of availability data gathered from actual television transmission circuits and chains;
3. availability and reliability objectives for transmission in an international television circuit or chain. Compatibility between these objectives in the event that it proves impossible to establish a single objective independent of the type of modulation (digital, analogue or mixed);
4. possible division of the availability and reliability objectives for transmissions in progress between the components of an international television circuit or chain;
5. in particular cases in which overall availability and reliability objectives for the television broadcasting services prove difficult to attain with transmission media having characteristics which meet the quality standards for programme exchange defined by the CCIR or the CCITT, the means which might be applied separately or jointly to reduce unavailability attributable to transmission.

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-1/CMTT

AUTOMATIC MEASUREMENT AND MONITORING
ON TELEVISION CHAINS

(1978-1986)

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;
- (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;
- (d) that auxiliary signals added to a TV-signal (e.g. energy dispersal signals) may influence automatic measurement,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the use that can be made of test signal elements (see Recommendation 473) 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;
8. the minimization of the influence of auxiliary signals added to the TV-signal on the results of automatic measurement.

Note. — Information relevant to this Study Programme is contained in Recommendation 569 and Reports 411 and 1094.

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;
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, 1978-1982 and 1982-1986.

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,

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 (Study Period 1981-1984) and Recommendation G.229 which considers 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?

* This Study Programme should be brought to the attention of Study Groups 4 and 9 of CCIR and CCITT Study Group XV.

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,

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.

* This Study Programme should be studied in close co-ordination with Study Programme 14B/CMTT.

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, 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 17-1/CMTT

**STANDARDS FOR ANALOGUE OR MIXED ANALOGUE-AND-DIGITAL TRANSMISSION
OF SOUND-PROGRAMME SIGNALS OVER LONG DISTANCES**

(1970-1978-1986)

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 and mixed analogue-and-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 hypothetical reference circuits composed of analogue or of mixed-analogue-and-digital sections:

1. what are the characteristics of the signals 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 1. — At present the performance characteristics of certain types of sound-programme circuits are laid down in Recommendations 503 and 505. For more information see Reports 496 and 647.

Note 2. — See also Recommendations 502 and 605 and Reports 491, 820 and 821.

Note 3. — Studies relating to digital encoding of analogue sound-programme signals are addressed under Question 18/CMTT.

STUDY PROGRAMME 17D-1/CMTT

ESTIMATION OF TRANSMISSION PERFORMANCE OF SOUND-PROGRAMME CIRCUITS SHORTER OR LONGER THAN THE HYPOTHETICAL REFERENCE CIRCUIT

(1974-1986)

The CCIR,

CONSIDERING

- (a) that Recommendation 502 defines the hypothetical reference circuit for sound programme transmission;
- (b) that Recommendation 605 provides guidelines to estimate the transmission performance of sound-programme circuits shorter or longer than the hypothetical reference circuit;
- (c) that some values are provisional and others are under study;
- (d) that the laws of addition in Recommendation 605 and Report 821 are only applicable to entirely analogue circuits and not to a mixed analogue-and-digital circuit;

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the possibility to finalize provisional values for entirely analogue circuits and the specification of the values for those parameters which have been left for further study;
2. the possibility to specify the laws of addition for transmission parameters on mixed analogue-and-digital paths and their values.

Note. — See Table II of Recommendation 605 and Report 821.

STUDY PROGRAMME 17G/CMTT

PERFORMANCE CHARACTERISTICS OF ANALOGUE OR MIXED ANALOGUE-AND-DIGITAL SOUND-PROGRAMME CIRCUITS

(1986)

The CCIR,

CONSIDERING

- (a) that Recommendations 503 and 505 specify performance characteristics for medium-quality and for high-quality sound-programme circuits respectively;
- (b) that certain parameters and values require further studies,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the additional parameters and values which need to be specified for analogue or mixed analogue-and-digital circuits to complete Recommendations 503 and 505.

STUDY PROGRAMME 17H/CMTT

TRANSMISSION OF ANALOGUE SOUND-PROGRAMME SIGNALS
OVER BROADCASTING-SATELLITE FEEDER LINKS

(1986)

The CCIR,

CONSIDERING

- (a) that analogue sound-broadcasting by direct broadcasting satellites is likely to occur in the near future;
- (b) that broadcasting of the sound-programme signals takes place in the same format as the signal transmission on the feeder link,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the characteristics which need to be specified for broadcasting-satellite feeder links to ensure satisfactory performance for the transmission of analogue sound-programme signals.

QUESTION 18-2/CMTT

STANDARDS FOR DIGITAL TRANSMISSION OF
SOUND-PROGRAMME SIGNALS OVER LONG DISTANCES

(1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that the CCITT is studying the standards for 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 connections and circuits composed entirely of digital transmission systems:

1. what are the characteristics of the signals and the coding methods to be used for 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 1. — The coding methods include those to be used for mixed analogue-and-digital transmission systems addressed under Question 17/CMTT.

Note 2. — See Recommendations 660 and 661 and Reports 647 and 1095.

Note 3. — Study Programme 18J/CMTT deals with the definition of a hypothetical reference connection for digital sound-programme transmission.

STUDY PROGRAMME 18A-2/CMTT

STANDARDS FOR THE DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

(1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that Recommendation 660 recommends the encoding methods for high quality sound-programme signals on mixed analogue-and-digital circuits using 384 kbit/s channels;
- (b) that Report 647 describes other methods of encoding than those recommended in Recommendation 660,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the need, if necessary, for the specification of coding methods other than those recommended in Recommendation 660 for the conversion of analogue sound-programme signals into digital coded form and, if required, the characteristics that should be recommended;
2. the methods which should 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;
 - drop and insert facilities on the international sound-programme connection;
 - composition of international sound-programme connections determining the necessary number of conversion processes in tandem;
 - particular constraints in relation to digital transmission impairments (e.g. errors, jitter, slips);
 - the use of pre-/de-emphasis;
 - maintenance aspects;
 - economical considerations.

Note 1. – The studies should take into account:

- the work being carried out in CCIR Study Groups 10 and 11 and CCITT Study Groups IV, XV and XVIII;
- the corresponding requirements defined by CCIR Study Group 10 for the processing and the quality of such signals.

Note 2. – See Recommendations 606, 646, 647, 659 and 660 and Reports 648 and 1095.

STUDY PROGRAMME 18B-1/CMTT

STANDARDS FOR THE USE OF AN AUXILIARY DATA
CHANNEL TIME-MULTIPLEXED WITH DIGITAL
SOUND-PROGRAMME SIGNALS

(1978-1986)

The CCIR,

CONSIDERING

- (a) that in principle, it is possible to insert an auxiliary data channel, time-multiplexed with digital sound-programme signals, in digital transmission paths;
- (b) that such an auxiliary data channel might be used for various purposes, for example for conveying:
 - signals associated with the programme (source identification, time codes, etc.);
 - supervision and control signals (type of coding, location of transcoder, etc.);
 - technical information on the signal (sound level, stereo phase, noise level, signal spectrum at the source, etc.),

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the possibility that these signals can be time-multiplexed with, and demultiplexed from, the digital sound bit stream, without increasing digital transmission impairments (e.g. error-ratio, slips);
2. the format of signals which should be recommended for international use and the minimum repetition rate required for certain types of information;
3. the transmission capacity, type of insertion (synchronous or asynchronous) and multiplexing frame characteristics of the auxiliary data channel which should be recommended.

Note. — See Recommendation 647 and Report 1095.

STUDY PROGRAMME 18C-1/CMTT

STANDARDS FOR MULTIPLEXING AND ROUTING OF DIGITAL SOUND-PROGRAMME SIGNALS IN DEDICATED DIGITAL NETWORKS

(1982-1986)

The CCIR,

CONSIDERING

- (a) that broadcasting organizations may prefer to make use of dedicated digital networks;
- (b) that in those cases the broadcasting organizations could combine two or more digital sound-programme signals (including commentary, radio data signals, etc.) by suitable multiplexing;
- (c) that this multiplexing format used by broadcasters need not necessarily be the same as those recommended by the CCITT to be used for the digital networks;
- (d) that in determining a suitable multiplexing structure, automatic routing of digital sound-programme signals needs to be considered,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the characteristics for multiplexing digital sound-programme signals which should be recommended;
2. the most suitable frame structure to permit a flexible allocation of time slots.

Note. — See Recommendations 659 and 660.

STUDY PROGRAMME 18D-1/CMTT*

PERFORMANCE OBJECTIVES FOR INTERNATIONAL DIGITAL CONNECTIONS AND CIRCUITS** CARRYING SOUND-PROGRAMME SIGNALS

(1982-1986)

The CCIR,

CONSIDERING

- (a) that the quality of sound-programme signals carried by international digital connections and circuits will be affected by:
 - the digital sound coding method;
 - the error control strategy;
 - the performance of the international digital network (bit error ratio, error distribution, slip rate and jitter amplitude);

* This Study Programme should be brought to the attention of CCITT Study Groups XV and XVIII.

** The terms "connection" and "circuit" should be read in conjunction with Study Programme 18J/CMTT.

(b) that the influence of the parameters mentioned in CONSIDERING (a) on the quality of sound-programme signals has yet to be quantified,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the impairments to sound-programme signal quality caused by errors, slip and jitter that may be introduced by an international long-distance digital connection or 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 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 should take into account the relevant CCITT Study Group XVIII Recommendations on the characteristics of digital networks.

Note 2. — See Recommendations 659, 660 and Report 648.

STUDY PROGRAMME 18E-1/CMTT*

SPECIFIC REQUIREMENTS FOR THE INTEGRATED SERVICES DIGITAL NETWORK (ISDN) CARRYING SOUND-PROGRAMME SERVICES

(1982-1986)

The CCIR,

CONSIDERING

- (a) that the CCITT has prepared numerous Recommendations (Series I) for the ISDN in particular Recommendations specifying user network interfaces, bearer services, teleservices;
- (b) that the CCITT is studying the network characteristics to be recommended for the transmission of wideband services over the ISDN;
- (c) that the CCITT Series I Recommendations may need to be amended to take into account the transmission of digital sound-programme signals;
- (d) that the CMTT is studying the standards to be recommended for digital transmission of sound-programme signals over the ISDN,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the information relating to the transmission of digital sound-programme signals which should be passed to the CCITT to be taken into account by that body when preparing Recommendations concerning network characteristics (e.g. synchronization, digit sequence integrity, signal formats);
2. the operational requirements, attributes and values for bearer services and teleservices carrying digital sound-programme signals over the ISDN.

Note. — See Recommendations 659 and 660.

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

STUDY PROGRAMME 18F/CMTT

TRANSMISSION OF DIGITAL SOUND-PROGRAMME SIGNALS
WITH DIGITAL STUDIO QUALITY

(1986)

The CCIT,

CONSIDERING

- (a) that CCIR Study Group 10 has expressed the view that there will be a future need for the exchange of digital sound-programme signals with studio quality consistent with Recommendation 646;
- (b) that to permit full, unrestricted studio processing of sound-programme signals, the channel status and users data, defined in Recommendation 647 have to be combined with the sound-programme signals;
- (c) that the use of ISDN to provide switched or dedicated circuits may offer some advantages although the use of other dedicated circuits cannot be excluded;
- (d) that the exchange of digital sound-programme signals between networks based on different digital hierarchies will become more frequent,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the methods which should be recommended for the transmission of digital sound-programme signals with studio quality consistent with Recommendation 646;
2. the methods which should be recommended to convey channel status data and users data associated with digital sound-programme signals as defined in Recommendation 647;
3. the possibilities for using the ISDN as well as dedicated networks for the transmission of digital sound-programme signals;
4. the methods which should be recommended to facilitate the exchange of digital sound-programme signals between networks based on different hierarchies (see also CCITT Recommendation G.802).

STUDY PROGRAMME 18G/CMTT*

STANDARDS FOR THE DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS
ON ONE OR TWO 64 kbit/s CHANNELS

(1986)

The CCIR,

CONSIDERING

- (a) that for international sound-programme exchanges, circuits of medium-quality with 7 kHz bandwidth are already recommended (CCIR Recommendation 503, CCITT Recommendation J.23) and an equipment with analogue interface to the broadcasters and digital access to the primary level is recommended (CCITT Recommendations J.42 and G.737);
- (b) that for international sound-programme exchanges, it might be acceptable, for the sake of economy, to use one or two 64 kbit/s digital channels which permit the transmission of commentary or medium-quality sound-programme digital signals;

Note. — Medium-quality denotes a performance which is lower than that specified for high-quality sound-programme circuits (15 kHz), but possibly superior to that specified for 7 kHz sound-programme circuits, and corresponding to a value of 3.5 or 4 of the 5-grade quality and impairment scale recommended in Recommendation 562.

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

- (c) that, to facilitate the insertion, transmission and switching of medium-quality sound-programme signals in circuits forming part of the integrated services digital network (ISDN), it would be desirable to use one or two 64 kbit/s channels;
- (d) that CCITT Study Group XVIII has prepared a draft Recommendation for 7 kHz audio coding within 64 kbit/s (CCITT Recommendation G.72X);
- (e) that the CCITT has requested the CMTT to investigate applications of the audio coding algorithm defined by Study Group XVIII,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the performance requirements which should be recommended for the connections (see Study Programme 18J/CMTT) carrying commentary or medium-quality sound-programme signals;
2. the possibilities for the international exchange of sound programmes, using the audio coding algorithm defined by CCITT Study Group XVIII, and any necessary modifications to the algorithm;
3. the methods of transcoding which should be recommended for signals derived from and/or converted to the digital studio standard;
4. the methods which should be recommended for ensuring the digit sequence integrity of the bit stream recovered from the ISDN, for use when two 64 kbit/s channels are used to provide a single medium-quality audio connection.

STUDY PROGRAMME 18H/CMTT

TRANSMISSION OF DIGITAL SOUND-PROGRAMME SIGNALS OVER BROADCASTING SATELLITE FEEDER LINKS

(1986)

The CCIR,

CONSIDERING

- (a) that digital sound-broadcasting by direct broadcasting satellites is planned for the near future;
- (b) that the feeder link should use the same format that is specified for direct broadcasting satellites;

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the characteristics which need to be specified for direct broadcasting satellite feeder links to ensure satisfactory performance for the transmission of digital sound-programme signals.

Note. — The signal format suitable for digital sound-broadcasting by direct broadcasting satellites is being studied by Joint Working Group 10-11S (see Report 953).

STUDY PROGRAMME 18J/CMTT*

**DEFINITION OF A DIGITAL HYPOTHETICAL REFERENCE CONNECTION
FOR DIGITAL SOUND-PROGRAMME TRANSMISSION**

(1986)

The CCIR,

CONSIDERING

- (a) that the CMTT has not yet defined a hypothetical reference connection (HRX) for digital sound-programme transmission;
- (b) that the CCITT in Recommendation G.801 has defined three HRX models ranging from 10 000 km to 27 500 km;
- (c) that the CCITT "standard HRX" in Recommendation G.801 does not encompass digital sound-programme connections and therefore may need to be appropriately modified in composition;
- (d) that several administrations have already introduced first generation switching and transmission equipment for an all-digital sound-programme connection,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the HRX model(s) for digital sound-programme transmission which need to be defined in order to facilitate the definition and apportionment of network performance objectives to the constituent elements prior to the derivation of equipment design objectives;
2. the lengths and complexity (e.g. transcoders, multiplexers, switches, demultiplexers) of the HRX that are required for the various grades of sound-programme quality (e.g. commentary, medium, high, and studio qualities);
3. the need to specify different HRXs for different bit rates.

Note. — See Recommendations 659 and 660, Report 1095 and CCITT Recommendation G.801.

STUDY PROGRAMME 18K/CMTT**

**STANDARDS FOR DIGITAL INTERFACES BETWEEN
STUDIO AND DEDICATED OR INTEGRATED
SERVICES DIGITAL NETWORKS**

(1986)

The CCIR,

CONSIDERING

- (a) that the digital sound-programme signals coding standard recommended for studio applications (see Recommendation 646) is different from that specified in Recommendations 606 and 660;
- (b) that it may be necessary to provide conversion between the digital studio standard and those more suitable for long-distance transmission involving, for example, bit rate reduction;

* This Study Programme should be brought to the attention of CCITT Study Groups XV and XVIII.

** This Study Programme should be brought to the attention of CCIR Study Group 10 and CCITT Study Group XVIII.

- (c) that CCIR Study Group 10 has already recommended a serial interface for studio applications (see Recommendation 647);
- (d) that the bit-rate reduction process, if any, is under the prime responsibility of the CMTT (see Decision 18-5);
- (e) that the transmission formatting, including error protection, should be studied by the CMTT in close cooperation with CCITT Study Group XVIII (see Decision 18-5),

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the interfaces which are necessary between studios and dedicated networks or integrated services digital networks;
2. the characteristics which need to be specified for these interfaces.

QUESTION 19-2/CMTT*

METHODS OF MEASUREMENT, TEST SIGNALS AND OPERATIONAL REQUIREMENTS FOR ANALOGUE OR MIXED ANALOGUE-AND-DIGITAL SOUND-PROGRAMME TRANSMISSION

(1978-1982-1986)

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 or mixed analogue-and-digital transmission circuits 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 circuit and connection composed of analogue or mixed analogue-and-digital transmission circuits:

what methods of measurement, test signals, and operational requirements can be recommended for checking the transmission performance?

Note. — See Recommendations 642, 645, 661 and Report 822.

STUDY PROGRAMME 19B-1/CMTT**

AVAILABILITY OF INTERNATIONAL ANALOGUE AND MIXED ANALOGUE-AND-DIGITAL SOUND-PROGRAMME TRANSMISSION CIRCUITS

(1974-1986)

The CCIR,

CONSIDERING

- (a) that the availability objectives of transmission systems are under study in several Study Groups (e.g. Study Groups 4 and 9 of the CCIR, and Study Groups XV and XVIII of the CCITT);
- (b) that sound-programme transmission may require availability and reliability objectives different from those applying to the transmission of other signals such as telephone or video signals;

* This Question should be brought to the attention of CCITT Study Groups IV and XV.

** This Study Programme should be brought to the attention of CCIR Study Group 9 and CCITT Study Groups II, XV and XVIII.

- (c) that the studies on the availability and reliability of sound-programme transmission systems may be different according to whether analogue or mixed analogue-and-digital circuits are used;
- (d) that compatibility between the availability and reliability objectives of analogue or mixed analogue-and-digital circuits would be desirable;
- (e) that Report 642 specifies what constitutes an interruption in sound-programme services,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the type and magnitude of faults which are to be considered as interruptions of the sound-programme signal transmission and the differences, if any, between analogue and mixed analogue-and-digital transmission in this respect (see also Study Programme 20C/CMTT);
2. the availability and reliability objectives which should be recommended for a hypothetical sound-programme reference circuit and in the event that it proves impossible to establish a single objective independent of the type of bearer channel, the compatibility of the various objectives;
3. the apportionment of availability and reliability objectives which should be given to the constituent elements of the hypothetical sound-programme reference circuit (see Recommendation 502);
4. the methods which should be recommended to meet the availability and reliability objectives;
5. the methods which should be recommended to determine availability and reliability objectives for circuits longer or shorter than the hypothetical reference circuit (see Recommendation 502).

Note. — See Report 642.

STUDY PROGRAMME 19D-1/CMTT*

ANALOGUE AND MIXED ANALOGUE-AND-DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

Methods of monitoring and measurement

(1978-1986)

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;
- (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 it may be useful to introduce methods for the measurement of performance and the monitoring of quality of analogue and mixed analogue-and-digital sound-programme transmission circuits;
- (d) that traffic demands may make it necessary to perform measurement of performance and monitoring of quality during programme transmission periods without affecting the sound-programme signal;
- (e) that for some parameters, the measuring methods are still under study,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the significant differences between methods of setting up and monitoring sound-programme signal levels on international circuits, and the possible effects upon the sound-programme signal;
2. the methods which should be used to minimize the influence on the level of the sound-programme signal due to different monitoring methods;

* This Study Programme should be brought to the attention of CCIR Study Groups 9 and 10 and CCITT Study Groups IV and XV.

3. the methods which should be adopted for the automatic monitoring of the sound-programme circuit without affecting the sound-programme signal;

4. the methods which should be adopted for measuring, either automatically or manually, the quality parameters of the analogue or mixed analogue-and-digital sound-programme circuits, during programme transmission periods, without appreciable impairment of the sound-programme itself.

Note. — See Recommendation 571 and Report 497.

QUESTION 20-2/CMTT*

METHODS OF MEASUREMENT, TEST SIGNALS AND OPERATIONAL REQUIREMENTS FOR DIGITAL SOUND-PROGRAMME TRANSMISSIONS

(1978-1982-1986)

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 circuit and connection entirely composed of digital transmission systems:

what methods of measurement, test signals and operational requirements can be recommended for checking the transmission performance?

Note. — See Recommendation 642. The attention of the CMTT is also drawn to CCITT Recommendation G.803 "Maintenance of digital networks".

STUDY PROGRAMME 20A-1/CMTT**

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

Methods of monitoring and measurement

(1978-1986)

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 which are to be recommended for checking the performance over long-distance transmission links when digital modulation is used;
3. the extension or variation of such methods needed for checking the performance of digitally coded sound-programme signals which are multiplexed with an analogue video signal.

Note. — See Report 968.

* This Question should be brought to the attention of CCITT Study Groups IV, XV and XVIII.

** This Study Programme should be brought to the attention of CCIR Study Group 10 and CCITT Study Groups IV and XV.

STUDY PROGRAMME 20B-2/CMTT*

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS

Operational characteristics

(1978-1982-1986)

The CCIR,

CONSIDERING

that operational practices and characteristics for digital sound-programme connections (i.e. switching, spare connections) may be different from those on analogue connections,

UNANIMOUSLY DECIDES that the following studies should be carried out:

the operational characteristics of international digital sound-programme connections which have to be defined to maintain satisfactory performance.

STUDY PROGRAMME 20C/CMTT**

AVAILABILITY OF INTERNATIONAL DIGITAL SOUND-PROGRAMME CONNECTIONS

(1986)

The CCIR,

CONSIDERING

- (a) that the availability objectives of transmission systems are under study in several Study Groups (e.g. Study Groups 4 and 9 of the CCIR and Study Groups XV and XVIII of the CCITT);
- (b) that digital sound-programme transmission may require availability and reliability objectives different from those applying to the transmission of other signals such as telephone or video signals,

UNANIMOUSLY DECIDES that the following studies should be carried out:

1. the type and magnitude of faults which are considered as interruptions of the sound-programme signal (see also Study Programme 19B/CMTT);
2. the availability and reliability objectives for digital sound-programme transmission on a hypothetical reference connection;
3. the apportionment of availability and reliability objectives which should be given to the constituent elements of a hypothetical reference sound-programme connection;
4. the methods which should be recommended to meet the availability and reliability objectives.

* This Study Programme should be brought to the attention of CCITT Study Group XV.

** This Study Programme should be brought to the attention of CCIR Study Groups 4 and 9 and CCITT Study Groups II, XV and XVIII.

QUESTION 21-2/CMTT*

**STANDARDS FOR TRANSMISSION OF SIGNALS WITH
MULTIPLEXING OF VIDEO, SOUND AND
OTHER TYPES OF SIGNAL**

(1978-1982-1986)

The CCIR,

CONSIDERING

- (a) that various operational constraints in transmission circuits, particularly those stemming from television service requirements, may 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 (an example of such systems is provided by the simultaneous transmission of sound and picture signals over radio-relay systems);
- (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 video signal is in analogue form and the associated signals are in either digital or analogue 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 video signal and the associated signals are in digital form?

Note. – See Report 969.

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;

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

(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-1/CMTT

TRANSMISSION OF SOUND AND PICTURE SIGNALS BY TIME-DIVISION MULTIPLEX OR FREQUENCY-DIVISION MULTIPLEX

(1978-1986)

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;
- (e) that the use of a multiplexing system can assist in efficient use of the frequency spectrum,

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. techniques which can be used to transmit more than one picture signal over a single transmission circuit;
3. standards of performance that can be obtained by the use of these techniques;
4. 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.

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 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;

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

- (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);
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 22-1/CMTT

STANDARDS, OPERATIONAL CHARACTERISTICS, TEST SIGNALS AND METHODS OF MEASUREMENT RELEVANT TO SYSTEMS FOR THE TRANSMISSION OF HIGH-DEFINITION TELEVISION SIGNALS OVER LONG DISTANCES

(1978-1986)

The CCIR,

CONSIDERING

- (a) that transmission of wideband television signals is possible over cable systems, terrestrial radio links and by communication satellites;
- (b) that because high-definition television standards may be adopted for use in various media, an exchange of information between relevant parties in the CCITT and in the CCIR is desirable;

- (c) that Study Group 11 is studying the standards which should be recommended for high-definition television systems under Question 27/11;
- (d) that problems of long-distance transmission should be involved in the choice of standards for distribution of high-definition television signals;
- (e) that the existing CCIR Recommendations do not seem to provide a complete solution to these problems,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of high-definition television signals over an international television connection:

1. what forms should the relevant hypothetical reference circuits take;
2. what are the characteristics of the signals and of the circuits which must be considered;
3. what methods of bit-rate reduction are suitable when digital transmission is involved;
4. what methods of measurement, test signals and operational characteristics can be recommended for checking and monitoring the transmission performance;
5. what tolerances must be imposed to ensure satisfactory transmission?

Note. — See Report 1092.

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 1. — 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.

Note 2. — See Report 1091.

QUESTION 24/CMTT*

TRANSMISSION OF TELEVISION SIGNALS USING
MULTIPLEXED ANALOGUE COMPONENTS (MAC)

(1986)

The CCIR,

CONSIDERING

- (a) that television systems using multiplexed analogue components as described in Report 1073 have been proposed for various applications;
- (b) that such systems may require greater baseband bandwidth than systems using composite signals as described in Report 624;
- (c) that means will have to be provided to convey such signals from one point to another;
- (d) that existing transmission circuits may have to be modified or new circuits provided,

UNANIMOUSLY DECIDES that the following question should be studied:

for the transmission of television signals using multiplexed analogue components over hypothetical reference circuits or chains entirely composed of analogue transmission systems:

1. what are the characteristics of circuits capable of carrying these signals;
2. what are the possibilities of modifying existing circuits to make them capable of carrying these signals;
3. what are the characteristics of any necessary interfaces between circuits carrying these signals;
4. what are the characteristics of any converters that may be necessary to convert between these signals and composite signals as described in Report 624;
5. what methods of measurement, test signals and operational characteristics can be recommended for checking the transmission performance?

Note. — See Report 1096.

DECISION 18-5**

DIGITAL SYSTEMS FOR THE TRANSMISSION OF
SOUND-PROGRAMME AND TELEVISION SIGNALS

(1974-1976-1978-1981-1983-1986)

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;

* This Question should be brought to the attention of Study Groups 4, 9, 10 and 11.

** This Decision should be brought to the attention of CCIR Study Groups 10 and 11 and CCITT Study Groups IV, XV and XVIII.

- (d) that CCIR Study Groups 10 and 11 have the primary responsibility for specifying the overall performance, apportioning and methods of assessment of systems in the broadcasting service;
- (e) that the CMTT is studying the performance requirements applicable to long-distance digital transmission circuits for sound-programmes and television, either for contribution or distribution circuits;
- (f) that the CMTT has to consider the transition period during which technological changes will take place;
- (g) 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 IWP CMTT/1 shall be a representative of the Administration of the United Kingdom;
3. that IWP CMTT/1 shall be composed of the Chairmen and Vice-Chairmen of the CCIR and representatives of the 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, the Netherlands, United Kingdom, Sweden, Switzerland, the USSR and Yugoslavia;
4. that, as far as possible, the work of IWP CMTT/1 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.

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 the CCIR Study Groups concerned.

The IWP CMTT/1 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 IWP CMTT/1 proposes that particular parts of studies concerning digital techniques should receive emphasis by Study Groups in the manner given in Table I.

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 Groups XV and XVIII).
- 2.* ** Coding methods, including redundancy reduction for foreseen application of sound broadcasting and television, and standards to be adopted for (see Note 1):
 - 2.1 studio installations including recording { sound (SG 10)
television (SG 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 (SG 10)
television (SG 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	(SG 10)	{ (CMTT).
3.2 television	(SG 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 CMTT/1.)
5. Study of transmission methods over various transmission media, including supervision, for:
 - 5.1 studio installation, including recording { sound (SG 10)
television (SG 11);
 - 5.2 terrestrial radio-relay links (SG 9);
 - 5.3 cable transmission links (SG XV);
 - 5.4 satellite transmission links (SG 4);
 - 5.5 broadcasting from terrestrial transmitters and from satellites { sound (SG 10)
television (SG 11).

* Close cooperation between the Interim Working Party CMTT/1 and Study Group XVIII is essential for these studies.

** Study Group XV is studying coding methods for visual telephony and Study Group XVIII is studying wideband speech coding.

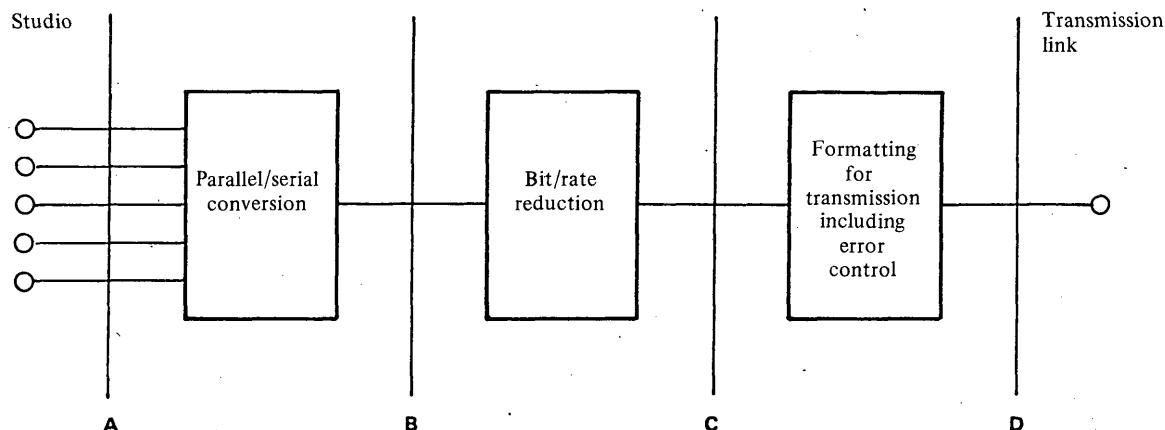
*** Aspects of compatibility relating to the exchange of recordings are matters for Study Groups 10 and 11.

6. PCM and digital multiplex equipment for sound and television programmes, including insertion of programme signals into telephony digital links (Study Group XV).
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 (see Note 1):
 - 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 (SG 10)
 - television (SG 11);
 - 11.1.2 terrestrial radio-relay links (SG 9);
 - 11.1.3 cable transmission links (Study Groups IV and XV);
 - 11.1.4 satellite transmission links (SG 4);
 - 11.1.5 broadcasting from terrestrial transmitters and from satellites
 - sound (SG 10)
 - television (SG 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 (see Note 1):
 - 12.1 sound programmes (SG 10);
 - 12.2 television (SG 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 (SG 10)
 - television (SG 11).
 - 13.2 switching in the transmission network (programme switching or otherwise) (CMTT).
14. Interfaces
 - 14.1 within studios and within broadcasting transmitters (Study Groups 10 and 11).
 - 14.2 within transmission links (CMTT).
 - 14.3 between studios and transmission links and between transmission links and broadcasting transmitters (CMTT), see Note 2.

Note 1. — The basic representations of sound and vision signals by digital codes and the associated quality requirements are studied and recommended by Study Groups 10 and 11. Irreversible transformations of these basic representations into other representations required for transmission should be subject to subjective and objective evaluation criterias defined by Study Groups 10 and 11 before their recommendation by the CMTT.

Note 2. — Digital interfaces for sound-programme and television signals.

Digital interfaces between studios and transmission circuits may involve either serial or parallel transmission of information and, in general, may involve processing for bit-rate reduction and formatting to suit the transmission circuit. These stages are illustrated in the following diagrams.



The interface at the output of the studio may be either A, B or C. Depending on the requirements of the transmission links, the block between interfaces C and D may, or may not, be required.

The process of bit-rate reduction shown between interfaces B and C may be carried out in the studio prior to parallel/serial conversion. This block may also include some processes for error protection.

Taking account of the above, Study Groups 10 and 11 (for sound and television respectively) should have the prime responsibility for interfaces A and B but the CMTT should have the opportunity to comment on them before final agreements are reached. Similarly, for interface C, the CMTT should have the prime responsibility but Study Groups 10 and 11 should have the opportunity to comment before final agreements are reached. In the case of interface D this should be studied with close coordination between the CMTT and CCITT Study Group XVIII but Study Groups 10 and 11 should have the opportunity to comment before final agreements are reached. This interface will also be of interest for access to the ISDN.

DECISION 67

DIGITAL TRANSMISSION OF COMPONENT-CODED TELEVISION SIGNALS

(1986)

The CMTT,

CONSIDERING

- (a) that Decision 18-5 of the CMTT states that Study Group 11 has the primary responsibility for specifying the overall performance, apportioning and methods of assessment of systems in the broadcasting service, and that the CMTT is responsible for the performance requirements for the digital transmission of television signals;
- (b) that the general principles for the digital transmission of component television signals, originated in conformance with Recommendation 601, are contained in Recommendation 604;
- (c) that Interim Working Party 11/7 is studying bit-rate reduction algorithms and their performance in respect of picture quality, processing capability and resistance to transmission errors;
- (d) that bit-rate reduction equipments will be important for the transmission of television signals over distribution and contribution connections, and that the CMTT is urgently required to provide information on the performance requirements for the digital transmission of television signals;

- (e) that in Recommendation G.702, CCITT Study Group XVIII has recommended the hierarchical bit rates for digital networks;
- (f) that in Recommendation G.802, CCITT Study Group XVIII has recommended a specific digital hierarchy for interworking between networks using different hierarchies;
- (g) that for the ISDN, CCITT Study Group XVIII plans to specify broadband services and related broadband channels, channel structures and access capabilities;
- (h) that within the CMTT, some interest has been expressed in a bit rate, intermediate between existing hierarchical levels, for digital television transmission,

DECIDES

1. that an Interim Working Party IWP CMTT/2 should be set up. The specific terms of reference of the Interim Working Party will be to draft Reports and Recommendations for presentation to the CMTT on the methods of encoding 525-line and 625-line television signals in component form for transmission, with particular emphasis on the following factors:
 - the methods for use on contribution circuits and distribution circuits should be considered, and if it is decided that different methods are required for the different applications, the question of compatibility between these methods should be addressed;
 - in considering the coding methods, account should be taken of the ease with which these can be adapted for use at all the corresponding intra-regional and interregional digital transmission rates recommended by the CCITT;
 - the required performance objectives in terms of jitter, transmission errors and transmission bit slip;
 - the definition of a digital interface between each type of encoding equipment and the digital transmission circuit;
 - the method of multiplexing any accompanying digital sound and/or ancillary signals with the digital television signal;
 - the possible use of encryption;
2. that appropriate parts of the work should be carried out in close collaboration with Interim Working Party 11/7 and CCITT Study Group XVIII to ensure compatibility between the results of the studies;
3. that IWP CMTT/2 should, as far as possible, work by correspondence. However, it may meet upon the proposal of its Chairman, following consultation with the Chairman of the CMTT and the Director of the CCIR. In such a case, it would be advantageous for IWP CMTT/2 and IWP 11/7 to meet at the same place and time;
4. that the progress of work of IWP CMTT/2 should be reported to the CMTT at its Interim Meeting, and its results presented at the Final Meeting of the 1986-1990 study period;
5. that the Chairman, the two Vice-Chairmen, and the composition of Joint Interim Working Party CMTT/2 shall be as shown in Annex I.

ANNEX I

The following Administrations, have indicated their participation in the work of Interim Working Party CMTT/2:

Administrations:

Germany (Federal Republic of)
 Australia
 Canada
 Denmark
 United States of America
 France
 Italy
 Japan
 Netherlands
 German Democratic Republic
 United Kingdom
 USSR

Chairman of Interim Working Party CMTT/2:

A representative of the Administration of the
Federal Republic of Germany

Vice-Chairmen of Interim Working Party CMTT/2:

A representative of the Administration of Japan
A representative of the Administration of Canada

DECISION 68

**TELEVISION AND SOUND-PROGRAMME SIGNALS
IN THE BROADBAND ISDN**

(1986)

The CMTT,

CONSIDERING

- (a) that the CMTT is competent to set the standards for transmission of broadcast quality television and sound-programme signals;
- (b) that CCITT Study Group XVIII is studying the services to be carried by the ISDN;
- (c) that CCITT Study Group XVIII has set up a Task Group on broadband aspects of ISDN which planned to complete its work by June, 1986;
- (d) that the Task Group on broadband aspects of ISDN requires information on the required capacity for broadband channels transmitting digital television and sound-programme signals;
- (e) that Study Group XVIII will require guidance on the use that broadcasting authorities may wish to make of the services provided by the ISDN,

DECIDES

1. that an Interim Working Party IWP CMTT/3 be established with the following terms of reference:
 - 1.1 to study the capacity and the characteristics required for broadband channels transmitting broadcast quality television and sound-programme signals in the ISDN (see Study Programme 18E/CMTT);
 - 1.2 to study the potential use that may be made of the ISDN by broadcasting authorities, taking account of the services and facilities (including access arrangements) proposed by CCITT Study Group XVIII;
 - 1.3 to maintain liaison with CCITT Study Group XVIII and any other bodies it may set up relative to its work on the definition of the ISDN on any matters which lie within the competence of the CMTT;
2. that, as far as possible, the work of IWP CMTT/3 shall be carried out by correspondence;
3. that the work shall be carried out in close coordination with CCITT Study Group XVIII and IWP CMTT/2;
4. that IWP CMTT/3 shall be empowered to transmit its results directly to Study Group XVIII, subject to these results being subsequently approved by the CMTT at its next meeting;
5. that IWP CMTT/3 shall in the first instance provide appropriate and timely input directly to the above-mentioned Task Group of CCITT Study Group XVIII so that this input may be incorporated in the Task Group report to its parent Study Group;
6. that the Chairman of IWP CMTT/3 shall be a representative of the EBU;
7. that the Administrations and International Organizations listed in Annex I will participate in the work of IWP CMTT/3 and that the Special Rapporteur of the CMTT to CCITT Study Group XVIII should be a member.

ANNEX I

LIST OF ADMINISTRATIONS AND INTERNATIONAL ORGANIZATIONS
PARTICIPATING IN THE WORK OF INTERIM WORKING PARTY CMTT/3*Administrations:*

Germany (Federal Republic of)
Australia
Canada
Denmark
Spain
United States of America
France
Italy
Japan
Netherlands
United Kingdom
USSR

International Organizations:

EBU

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ALPHABETICAL INDEX OF KEY WORDS AND TERMS OF VOLUME XII

A

Absolute level (Rep.820)

ACC (see Automatic chroma control)

Acceptance test method (Rec.567)

Adaptive prediction (Rep.967)

A-law instantaneous compandor (Rep.648)

Alignment (Rec.661)

Alignment signal (AS) (Rec.661)

Amplitude/frequency response (Rec.473, Rep.496)

Amplitude linearity (Rec.503, Rec.505)

Analogue transmission, sound-programme (see Sound-programme transmission, analogue)

Analogue transmission, television (see Television transmission, analogue)

Analogue-to-digital conversion (Rep.646)

APL (see Average picture level)

Automatic chroma control (ACC) (Rep.635)

Automatic measurement, television transmission (Rec.569, Rep.411, Rep.1094) (see also Insertion test signal)

- average picture level (APL) (Rec.569)
- signal-to-quasi peak-to-peak noise ratio (Rec.569)
- signal-to-random noise ratio (Rec.569)
- signal-to-unweighted random noise ratio (Rec.569)

Automatic monitoring (Rep.411, Rep.822)

Automatic switching systems (Rep.818)

- impairments due to switching equipment (Rep.818)
- switching thresholds (Rep.818)

Automatic testers (Rep.822)

Availability (Rep.639, Rep.642, Rep.967)

- sound-programme transmission (Rep.642)

Average picture level (APL) (Rec.567, Rec.569, Rep.636)

B

Bar tilt (Rec.569)

BER (see Bit error ratio)

Bi-directional connections (Rep.817)

Bit error objectives (Rep.967)

Bit error protection (Rec.660)

Bit error ratio (or Bit error rate) (BER) (Rep.817, Rep.967, Rep.969)

Bit errors (Rep.646, Rep.648, Rep.967, Rep.968)

Bit rate (Rec.660, Rep.646, Rep.647)

Bit rate hierarchies (Rep.817)

Bit rate reduction techniques (Rec.604, Rep.646, Rep.1093)

- DPCM (differential PCM) (Rep.646)
- Hadamard transform block (Rep.646)
- interframe coding (Rep.646)
- intrafield coding (Rep.646)
- predictive coding (Rep.646)

Bit slips (Rep.646, Rep.648, Rep.967)

Blanking level (Rec.567)

C

Cascade (Rep.635)

CCITT (International Telegraph and Telephone Consultative Committee) (Rec.473, Rec.502, Rec.503, Rec.505, Rec.567, Rec.571, Rec.572, Rec.590, Rec.604, Rec.606, Rec.645, Rep.412, Rep.491, Rep.493, Rep.496, Rep.497, Rep.637, Rep.643, Rep.646, Rep.647, Rep.648, Rep.816, Rep.817, Rep.820, Rep.821, Rep.822, Rep.965, Rep.967, Rep.970)

CCITT Recommendations (Rec. G.223 (Rep.491, Rep.821), Rec. G.226 (Rep.821), Rec. G.700 (Rep.817), Rec. G.702 (Rep.647, Rep.817), Rec. G.703 (Rep.647), Rec. G.704 (Rep.817), Rec. G.721 (Rep.817), Rec. G.732 (Rep.606, Rep.647), Rec. G.733 (Rep.606, Rep.647), Rec. G.735 (Rep.817), Rec. G.737 (Rep.817), Rec. G.739 (Rep.647), Rec. G.751 (Rep.817), Rec. G.752 (Rep.817), Rec. G.800 (Rep.817), Rec. G.801 (Rep.648), Rec. G.811 (Rep.647), Rec. G.821 (Rep.648, Rep.817, Rep.967), Rec. G.822 (Rep.647, Rep.648), Rec. G.823 (Rep.648, Rep.817), Rec. G.900 (Rep.817), Rec. I.110 (Rep.817), Rec. I.130 (Rep.647, Rep.817), Rec. I.210 (Rep.647, Rep.817), Rec. I.211 (Rep.647, Rep.817), Rec. I.212 (Rep.647, Rep.817), Rec. I.410 (Rep.647, Rep.817), Rec. I.411 (Rep.647), Rec. I.412 (Rep.645, Rep.647), Rec. I.431 (Rep.647), Rec. J.11 = Rec.502 (Rec.505), Rec. J.14 (Rep.820), Rec. J.15 (Rep.491, Rep.822), Rec. J.17 (Rec.572, Rep.491, Rep.496, Rep.497, Rep.647, Rep.820), Rec. J.18 (Rec.503, Rec.505), Rec. J.19 = Rec.571 (Rec.503, Rec.505, Rep.497), Rec. J.21 = Rec.505 (Rec.571), Rec. J.22 (Rec.571), Rec. J.23 = Rec.503 (Rec.571), Rec. J.31 (Rec.503, Rec.505, Rep.491, Rep.493), Rec. J.34 (Rec.503), Rec. J.41 (Rep.647, Rep.817), Rec. J.42 (Rep.647, Rep.817), Rec. J.61 = Rec.567, Rec. J.62 = Rec.568, Rec. J.63 = Rec.473, Rec. J.65 = Rec.570, Rec. J.66 = Rec.572, Rec. J.73 (Rec.570, Rep.643), Rec. J.77 (Rep.637, Rep.643), Rec. N.12 (Rec.571), Rec. N.13 (Rec.571), Rec. N.21 (Rec.505, Rep.820, Rep.821), Rec. N.23 (Rec.505, Rep.647, Rep.820), Rec. N.62 (Rep.965), Rec. N.67 (Rec.473), Rec. O.31 (Rec.505, Rep.822), Rec. O.32 (Rep.822), Rec. P.53 (Rep.496))

CCITT Study Groups

- Study Group IV (Rep.496, Rep.497, Rep.643, Rep.820, Rep.821, Rep.822)
- Study Group XV (Rec.503, Rec.505, Rec.570, Rec.571, Rec.659, Rec.660, Rep.493, Rep.643, Rep.647, Rep.817, Rep.820, Rep.822)
- Study Group XVIII (Rec.502, Rec.604, Rec.659, Rec.660, Rep.646, Rep.647, Rep.817, Rep.967)

Characteristics, international television circuits (Rep.816)

Characteristics of video signals (Rep.970)

Chrominance bar signal (Rec.473)

Chrominance signal (Rec.567)

Chrominance-luminance delay (Rep.635)

Chrominance-luminance gain (Rep.635)

Circuits shorter or longer than HRC (Rec.605, Rep.821)

- laws of addition (Rec.605, Rep.821)

CISPR (International Special Committee on Radio Interference) (Rep.496)

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Codecs (Rec.660, Rep.647, Rep.819)

Coding (Rec.660, Rep.604, Rep.646, Rep.647, Rep.648)

- component coding (Rep.604, Rep.646)
- composite coding (Rep.646)

Coding (*cont'd*)

instantaneous coding (Rep.647)
 interframe coding (Rep.646)
 intrafield coding (Rep.646)
 near-instantaneous coding (Rep.647)
 predictive coding (Rep.646)
 table (Rec.660)

Colour bar (Rec.570, Rep.634)

Colour burst (Rec.567)

Colour sub-carrier (Rec.473)

Colour television channel (Rec.570)

Comb-filtering (Rep.646)

Companding (Rep.647)

Companding, instantaneous (Rep.647)

Companding, near-instantaneous (Rep.647, Rep.648, Rep.968)

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