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XVIIth PLENARY ASSEMBLY
DÜSSELDORF, 1990



INTERNATIONAL TELECOMMUNICATION UNION

REPORTS OF THE CCIR, 1990

(ALSO DECISIONS)

ANNEX TO VOLUME XII

TELEVISION AND SOUND TRANSMISSION
(CMTT)

CCIR INTERNATIONAL RADIO CONSULTATIVE COMMITTEE



Geneva, 1990



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92-61-04301-1

Geneva, 1990



ANNEX TO VOLUME XII

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(CMTT)

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⁽¹⁾ Recommendation 642, at present published in Volumes X-1 and XII, will not be published in the 1990 edition of Volume XII. Recommendation 642-1 will only be published in the 1990 edition of Volume X-1.

⁽²⁾ Reports 817-2 and AA/CMTT have been deleted and replaced by Report 1241 (see Document CMTT/1021).

SECTION CMTT A: TELEVISION TRANSMISSION STANDARDS AND PERFORMANCE OBJECTIVES

REPORT 965-1

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

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

(1982-1990)

1. Introduction

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

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

2. Maintenance values for the INTELSAT satellite system

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

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

525/60 Systems, full transponder: 53.3 dB

625/50 Systems, full transponder: 50.1 dB

525/60 Systems, half transponder: 48.7 dB

625/50 Systems, half transponder: 47.1 dB

3. Practical design values for a regional system

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

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

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

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



TABLE I

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

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

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

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

(4) Highest frequency: 5 MHz.

4. Measured values for a regional system

Measurements performed on the EBU leased network with the first generation of the EUTELSAT satellites (EUTELSAT I F2) and standard EUTELSAT earth stations (G/T = 39 dB) lead to the values presented in Table II for the whole transmission chain from CNCT to CNCT with single TV carrier per 72 MHz transponder.[CCIR, 1986-90a]

TABLE II

Parameters (defined in Recommendation 569)	Measurements from CNCT to CNCT*
- luminance bar amplitude error	$\pm 7\%$
- base line distortion	$\pm 3.5\%$
- 2T pulse/bar ratio error	$\pm 8.5\%$
- peak differential gain	$\pm 6.5\%$
- chrominance-luminance gain inequality	$\pm 11.5\%$
- signal-to-weighted random noise ratio	55 dB

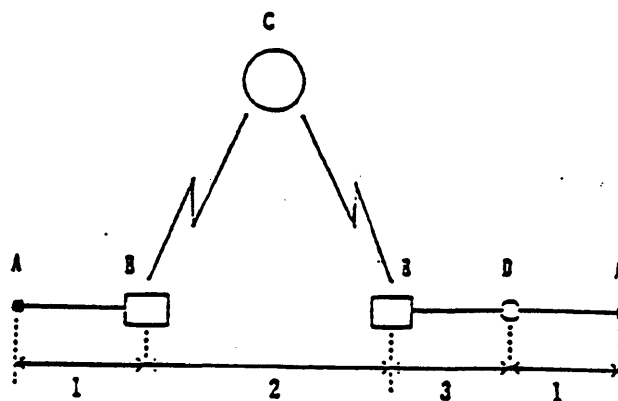
* 80% of the values (110) measured by the EBU are better than these given figures.

5. **Mixed terrestrial and satellite TV transmission circuits and their specified performance in Japan**

In Japan a new television transmission network is utilized in order to allow multi-destination transmission and to improve circuit flexibility. The 14/12 GHz band satellite communication system is being introduced systematically into the existing terrestrial television network [CCIR, 1986-90b].

A typical transmission circuit is shown in Figure 1 which consists of an earth station/satellite/earth station segment, a terrestrial circuit connecting the earth station to a terminal station and two local circuits.

The designed transmission performance for this circuit A to A in Figure 1 (see Table III) satisfies the performance objectives in Recommendation 567 [CCIR, 1986-90c]. In order to improve availability, two receiving earth stations with interconnecting terrestrial circuits and automatic selection are proposed.



1: Local television circuit	A: Program sending or receiving station
2: Satellite television circuit	B: Earth station
3: Terrestrial television circuit	C: Satellite space station
(third of hypothetical reference circuit)	D: Terminal station

FIGURE 1

Typical mixed terrestrial and satellite TV transmission circuits

TABLE III

Performance of video circuit utilizing fixed
satellite and terrestrial system

§ of Part D of Rec. 567	Parameters	(a) Design Objectives	(b) Measured Values
2.1	Nominal impedance	75/124Ω	75Ω
2.4	Nominal signal amplitude (V)	(UB/B) 1.0	(UB) 1.0
3.1	Insertion gain (dB)	0±0.5	-0.36
3.1.1	Insertion gain variation (1 s) (dB)	±0.3	0.05
3.1.1	Insertion gain variation (1 h) (dB)	±0.5	0.11
3.2.1	Signal-to-continuous random noise (dB)	53	59.6
3.4.1.1	Luminance non-linear distortion (%)	5	1.0
3.4.1.2	Chrominance non-linear distortion (amplitude) (%)	4	1.0
3.4.1.2	Chrominance non-linear distortion (phase) (deg.)	4	1.0
3.4.1.3	Differential gain (x or y) (%)	10	1.49
3.4.1.3	Differential gain (x or y) (deg.)	5	0.38
3.4.1.4	Chrominance-luminance intermodulation (%)	± 3	-0.4
3.4.2.1	Steady-state sync pulse non-linear distortion (%)	± 10	-4.2
3.5.1.2	Field-time waveform distortion (%)	± 6	-1.0
3.5.1.3	Line-time waveform distortion (%)	± 3	-2.1
3.5.1.4	Short-time waveform distortion (pulse/bar) (%)	100±12	104.2
3.5.1.4	Short-time waveform distortion (pulse lobes) (%)	3	1.5
3.5.3.1	Chrominance-luminance gain inequality (%)	± 10	0.9
3.5.3.2	Chrominance-luminance delay inequality (ns)	±100	97
3.5.4.1	Gain/frequency characteristics (0.15-6MHz) (dB)	± 1 (0.5-5MHz)	0.2 (0.5-5MHz)

Note - Columns (a) and (b) give the performance values for the entire circuit A to A (Figure 1).

REFERENCES

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

CCIR Documents

[1986-1990]: a. CMTT/9 (France); b. CMTT/42 (Japan); c. CMTT/233 (Japan).

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 1090-1*

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

(Study Programme 13G/CMTT)

(1986-1990)

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 the inclusion of terrestrial connections between the CNCT and earth stations reduces the S/N ratio 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.

4.1 Results obtained in France using a regional system (EUTELSAT I) with single TV carrier per 72 MHz transponder

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.

4.2 Performance expected on a regional system (EUTELSAT I) using two TV carriers per 72 MHz transponder

The EBU has carried out tests on the transmission of two TV programmes in a EUTELSAT I satellite repeater (72 MHz bandwidth) [CCIR, 1986-90a].

The signal quality obtained in these tests was considered by the EBU to be satisfactory and has prompted them to prepare for this type of operation.

In view of the constraints connected with multicarrier operation, in particular, adjacent channel interference, the transmitting earth stations (stations with G/T of 39 dB) have adopted an input backoff of 6 dB (i.e. a maximum e.i.r.p. of 76 dBW).

TABLE I - Video noise performance obtained over
TV transmission circuits via satellite

Satellite e.i.r.p. (dBW)	Trans- mitted e.i.r.p. (dBW)	(S/N) _w ratio (dB) received at large receiving antenna (G/T = 39 dB(K) ⁻¹)		(S/N) _w ratio (dB) received at small receiving antenna (G/T = 27 dB(K) ⁻¹)	
		Satellite ↙ ↘ Earth station Earth station	Satellite ↙ ↘ Earth station Earth → CNCT station	Satellite ↙ ↘ Earth station Earth station	Satellite ↙ ↘ Earth station Earth → CNCT station
45 (EUTELSAT I satellite) (limited coverage)	69	53	52	48	48
	69.7	53	52		
	75.5	57	55		
39 (EUTELSAT I satellite) (global coverage)	76	55 to 58 ⁽¹⁾	54 to 57 ⁽¹⁾	49 to 50 ⁽¹⁾	49 to 50 ⁽¹⁾
39.4 ⁽²⁾ (INTELSAT V satellite) (West spot beam edge)	69.3	—	—	53 ⁽³⁾	—

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

⁽²⁾ Allotted to the 36 MHz RF bandwidth of a 72 MHz West Spot beam transponder on INTELSAT V satellite.
The satellite e.i.r.p. toward the receiving earth station was estimated to be approximately 42 dBW.

⁽³⁾ Value obtained by using vehicle-equipped transportable receiving earth station having a system G/T of 27.0 dB(K)⁻¹.

Tables II, III, and IV give the conditions and the results of a study based on a theoretical model of the TWT of EUTELSAT I satellite (20W). The S/N values expected are given for the TV signal transmitted by a transportable earth station simultaneously with a main TV carrier (76 dBW e.i.r.p.) for reception by a station with a G/T of 39 dB or 35 dB. Both clear sky and rainy conditions are taken into account. The S/N values are calculated from the C/N value using the formula:

$$S/N = C/N + 37.5 \text{ dB.}$$

For typical e.i.r.p. values of transportable transmitting earth stations (68 to 75 dBW), it is expected that the S/N will lie between 47.4 dB and 54.4 dB (most favourable clear sky conditions). In the worst case considered in the tables (rainy weather and reception of the carrier from a transportable earth station with a G/T of 35 dB), a S/N of about 50 dB may be obtained with an e.i.r.p. of 75 dBW.

These values are similar to the 51 dB suggested in section 4 above and operational experience to date has confirmed that signals obtained in such conditions give acceptable quality.

TABLE II

General conditions for the study

EUROBEAM RECEPTION - e.i.r.p. of the satellite 39 dBW - global beam	
TWO TV CARRIERS IN ONE TRANSPONDER:	
- one main carrier	
- one transportable earth station carrier	
SATELLITE RECEIVER CONTOUR (auxiliary carrier)	: 0.0 dB
SATELLITE SEND CONTOUR (auxiliary carrier)	: 4.2 dB
UP-LINK ATTENUATION	
OR: DOWN-LINK - clear skies	: 0.3 dB
- rainy weather	: 2.0 dB
RF BAND OF THE TRANSPORTABLE EARTH STATION CHANNEL	: 36.0 MHz
MAIN TV CARRIER INPUT BACK-OFF	: 6.0 dB
N.B. same weather conditions for both paths	
Reference TV CARRIER received by station	
with G/T = 39 dB and contour of 4.2 dB	

TABLE III

Receiving station G/T = 39 dB S/N (dB) from earth station to earth station		
Transportable earth station e.i.r.p. dBW	clear skies	rainy weather
67.0	46.4*	-
68.0	47.4*	45.7*
69.0	48.4	46.7*
70.0	49.4	47.7
71.0	50.4	48.7
72.0	51.4	49.7
73.0	52.4	50.7
74.0	53.4	51.7
75.0	54.4	52.7

* Reception ensured with a threshold extension FM demodulation.

TABLE IV

Receiving station G/T = 35 dB S/N (dB) from earth station to earth station		
Transportable earth station e.i.r.p. dBW	clear skies	rainy weather
69.0	45.3*	-
70.0	46.3*	-
71.0	47.3	45.6*
72.0	48.3	46.6*
73.0	49.3	47.6
74.0	50.3	48.6
75.0	51.3	49.6

* Reception ensured with a threshold extension FM demodulation.

Note - S/N of the main TV carrier, from earth station to earth station in all cases exceeds 53 dB.

4.3 Test results obtained by Japan using INTELSAT V satellite with single TV carrier per 72 MHz transponder

An international television transmission link was established, using 36 MHz of a 72 MHz bandwidth 14/11 GHz band spot-beam transponder on an INTELSAT V satellite. The transmitting and receiving earth stations were both transportable. The former is of a trailer-mounted type, the latter is of a vehicle-equipped type with a system G/T of 27 dB(K)^{-1} . Both were located at the broadcaster's premises in Seoul and Tokyo, respectively.

Test results obtained on this television transmission circuit showed that the transmission performance fully satisfied the performance objectives specified in Rec. 567 [CMTT 1986-90b]. A signal-to-weighted noise ratio $(S/N)_w$ of 53.0 dB was obtained for the video signal with an earth station e.i.r.p. of 69.3 dBW under clear sky conditions, employing INTELSAT 30MHz television parameters for 525-line television systems. These results are shown in Table I.

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).

[1986-90]: a. CMTT/10 (France); b. CMTT/167 (Japan).

REPORT 1237*

SATELLITE NEWS GATHERING

(Study Programme 13H/CMTT)

(1990)

1. Satellite news gathering requirements and specifications**1.1 Definition of SNG**

Satellite news gathering (SNG) operating in the fixed-satellite service (FSS) is used exclusively for broadcasting applications. The following definition for satellite news gathering (SNG) for international application is suggested for the system(s) considered in the work of the CCIR in order to differentiate such systems from the domestic SNG systems currently operating in the fixed-satellite service (FSS) in many countries. The proposed definition is:

"Highly-portable news-gathering satellite uplink system operating in the fixed-satellite service (FSS), capable of compliance with fixed-satellite service antenna side-lobe specifications, and of such small size and weight that it is capable of being transported in a scheduled airline, helicopter, automobile, or even being hand-carried to the site of a breaking event."

The proposed definition of the equipment is that it should be capable of uplinking the video programme with its associated sound**, or sound radio signals, and capable of providing two-way coordination (communication) circuits, and may provide for data transmission as well. Further, the equipment should be capable of being set up and operated by a crew of no more than two (2) people within a reasonably short time (for example, one hour).

Transportable earth stations (as described in Report 1090) are also applicable for meeting the SNG requirements when logistics dictate use of such systems and those systems meet the basic functional characteristics of the SNG systems.

* This Report should be brought to the attention of Study Groups 4, 8, 10 and 11.

** SNG (sound) may also be operated in the mobile-satellite service.

1.2 Requirements

1.2.1 Functional description

The main features of SNG systems are essentially defined by the uplink characteristics. Operations with the SNG uplink terminal assumes that the receiving side is appropriately dimensioned. To ensure system compatibility and efficient operation, it is necessary to standardize equipment characteristics and operating procedures.

Functions of the SNG system are to:

- transmit with a minimum of impairments, a video signal, regardless of the television standard;
- transmit with minimal distortions, the associated audio signals;
- provide limited receiving capability to point the antenna and to monitor the transmitted signals, where possible;
- provide two-way communication channels for operation and control, as described in §1.2.5.

1.2.2 Frequency bands appropriate for SNG

All frequency bands allocated to uplinks for the FSS may be used by SNG systems. However, since an SNG terminal requires a small antenna to provide high portability, the operating frequency band is extremely important. Experience has shown that the most appropriate frequency bands are either 14 GHz or 30 GHz, although SNG has been accomplished using frequency bands below 14 GHz.

1.2.3 Link performance

1.2.3.1 Video transmission

In general, video transmission characteristics for SNG shall comply with Recommendation 567 except for the signal-to-noise ratio. The signal-to-noise performance for SNG may be relaxed if acceptable to the user.

1.2.3.2 Sound transmission

The number of associated sound channels for an SNG system is suggested to be two. All audio signals associated with the video signal shall be transmitted using the vision carrier. Possible solutions include subcarriers and sound-in-sync (SIS) techniques. Transmission performance of audio signals shall comply with Recommendation 505, except for the noise objectives.

1.2.3.3 RF performance

Since the SNG terminal operates mainly in the FSS, RF parameters shall comply with the Radio Regulations, the applicable local radio regulations and be consistent with any coordination agreements which apply to a specific satellite being used. Major items to be considered are as follows:

(a) *Off-axis e.i.r.p. density*

According to Recommendation 524-2 (MOD I), the off-axis e.i.r.p. density in any direction within 3 degrees of the geostationary satellite orbit should not exceed the following values:

Angle off-axis (deg)	Maximum e.i.r.p. (dB(W/40 kHz))
$2.5 < \phi < 7$	$(39 - 25 \log \phi)$
$7 < \phi < 9.2$	18
$9.2 < \phi < 48$	$(42 - 25 \log \phi)$
$48 < \phi < 180$	0

However, satellite system operators may have more stringent requirements.

(b) *Polarization discrimination*

Some satellites use overlapping channels with orthogonal polarization discrimination (frequency reuse). The cross-polarization discrimination should be +35 dB within 1 dB off the main axis of the beam and +30 dB elsewhere. However, SNG terminals generally operate with low e.i.r.p. so a corresponding reduction in cross-polar performance may be accepted by the satellite system operator.

(c) *E.i.r.p.*

The necessary e.i.r.p. of the SNG terminal depends on the required uplink carrier-to-noise ratio (C/N) and satellite G/T. However e.i.r.p. is often limited by the off-axis e.i.r.p. density limits as indicated in (a).

For example, with an uplink e.i.r.p. of 70 dBW, the off-axis e.i.r.p. density limit can be met for an SNG antenna of about 1.8 m (G = 46 dBi) with the two possible following sets of parameters (i.e. antenna radiation patterns and energy dispersal):

29 - 25 log ϕ and 1 MHz or
32 - 25 log ϕ and 2 MHz.

It has been demonstrated that there is no difficulty for an SNG terminal to comply with CCIR Recommendation 524.

1.2.3.4 *Energy dispersal*

The use of analogue television requires the addition of energy dispersal to reduce interference. Recommendation 446 and Report 384 are relevant.

1.2.3.5 *Necessary RF bandwidth*

The necessary RF bandwidth for SNG can be determined by taking into account the signal format, the required signal-to-noise ratio, the e.i.r.p., the number of associated carriers for auxiliary circuits and the available bandwidth of the transponder.

1.2.3.6 *Communication requirements for supervision and coordination*

SNG uplink signals are often originated from remote areas. In these cases, communication using the public switched telephone network (PSTN) is difficult or impossible. The SNG terminal should therefore be equipped to provide all of its own communications through the satellite to both the satellite operator's communication control centre and the broadcaster's premises, with the following facilities:

- Between SNG terminals and the satellite operator's communication control centre: a minimum of one two-way narrow band, voice/data communication circuit. This should be available at all times and not be restricted to the transponder booked times. It is desirable that these circuits are provided in the same transponder as the programme vision and sound.
- Between the SNG terminal and the broadcaster's premises: four (4) two-way, narrow band, voice/data communication circuits. These circuits should be available for a short time before and after as well as during the transponder booked times. It is essential that these communications are provided in the same transponder as the programme vision and sound.

1.2.4 *Equipment**

1.2.4.1 *General*

SNG terminals consist of the following main units:

- Antenna and feed system with polarization adjustment;
- Antenna mount with azimuth/elevation adjustment;
- High power amplifier (HPA) for vision/sound and auxiliary (voice/data) communication channels;
- Baseband/modulation equipment and IF to RF upconverter;
- Two-way voice/data communication equipment;
- System local/remote control panel;
- Video/audio/RF monitoring equipment;
- Optional power generator.

1.2.4.2 *Antenna and feed system*

The antenna should be small in diameter (less than 2 m), light weight, easy to assemble and easy to transport.

The antenna radiation patterns in the plane of the orbital arc at the earth station location shall comply with §1.2.3.3. The cross-polarization discrimination shall comply with §1.2.3.3.

Repeated assembly and disassembly of the antenna shall not affect the radiation and cross-polarization discrimination performance.

*The figures cited are relevant to the 14 GHz band unless otherwise specified.

The feed system shall have transmit and receive ports and should be optimised for the uplink frequency band of 14.0 to 14.5 GHz and the corresponding downlink frequency band(s).

For operation with frequency reuse (dual linear polarization) satellites, polarization adjustment controls of adequate resolution must be provided to optimize the cross-polarization alignment.

1.2.4.3 *Antenna mount and azimuth/elevation adjustments*

A rugged antenna mount with azimuth and elevation controls and associated graduated scales should be provided.

The mount should be designed to accurately deploy the antenna and keep it pointed at the satellite even during high wind conditions.

1.2.4.4 *High power amplifier (HPA)*

The HPA for transmission of the vision/sound carrier should be rated at approximately 300 W. The resulting e.i.r.p. will then be of the order of 70 dBW when used with a 1.8 m antenna.

To keep the waveguide losses to a minimum, the HPA should be located as close to the antenna as possible. The HPA should therefore be designed and packaged to operate under wide environmental conditions.

To improve the availability of the uplink, or when higher power is needed, a second HPA can be provided. This HPA can be operated in a one-for-one redundant configuration or in a phase-combined configuration.

A separate solid state 14-GHz band power amplifier may be required for the auxiliary communication system unless the auxiliary communication hardware is combined at IF with the vision/sound.

1.2.4.5 *Baseband/modulating equipment and upconverter*

This unit shall include: a video low pass filter which can be bypassed, switchable pre-emphasis networks, selectable IF bandpass filters, one or more frequency agile sound subcarrier modulators, dispersal generator, IF power control, and a linear video FM modulator. The DC component of the video signal should be preserved to take advantage of the available bandwidth.

The upconverter should be frequency agile with a suitable frequency step size for the satellite system used. The phase noise and frequency stability should be adequate to allow upconversion of the auxiliary communication signals. To preclude accidental interference to other traffic on the same satellite, it is recommended that the up-converter be equipped to inhibit transmission in the event of off-frequency operation.

1.2.4.6 *System local/remote control panel*

All controls such as main carrier deviation, operating carrier frequency, subcarrier frequencies, subcarrier deviation, pre-emphasis, dispersal, power adjustment, etc., should be readily accessible.

It may be desirable to provide remote access to the power on/off and raise/lower controls from the satellite operator's communication control centre or from the broadcaster's premises.

1.2.4.7 *Video/audio/RF monitoring equipment*

To allow the SNG terminal operator to properly set up the terminal, monitoring equipment is essential.* To simulate the satellite transponder it is advantageous to provide a loop-test translator and video/audio receiver to verify the performance of the uplink equipment.

Due to the low G/T of the SNG antenna system and the particular operational configuration of the satellite system, it is normally not possible to monitor the transmitted video and audio signals at full quality. In such cases, narrower IF bandwidth monitoring of the satellite receive signal may provide limited monitoring capability.

1.2.4.8 *Size and weight objectives*

A complete, highly portable SNG terminal, designed specifically to be transported anywhere in the world, must be packaged in shock-resistant shipping cases in order to withstand rough handling. The total dimensions (length + width + height) should meet the International Airline Transportation Authority (IATA) regulations for carriage on scheduled airline flights as accompanied baggage.

The weight of each shipping case of the SNG terminal must enable the cases to be handled easily by a two-man crew (maximum weight of 45 kg per case including equipment is suggested as a good target).

1.2.5 *Auxiliary communications*

1.2.5.1 *Communication channels for supervision and coordination*

SNG terminals require two-way communication channels, in addition to the vision and associated sound, to provide for communications capability with the satellite operator's communications control centre and the broadcaster's facilities.

An example is shown in Figure 1 for the case of multiple SNG terminals accessing occasional use 36-MHz transponders.

The SNG terminals are assumed to be located in the downlink footprint.

If the SNG terminal is not in the downlink footprint other solutions, as described in §1.2.5.4, are required.

It should be noted that several domestic systems are presently in operation using various communication techniques including multiple analogue SCPCs within the same transponder as the video signal and/or narrow-band carriers in a separate transponder from the video in the same satellite and with the same polarization.

If an SNG terminal is accessing a leased transponder, wide-band digital carriers may be employed (e.g., 128 kbit/s, 256 kbit/s or 1.544 Mb/s).

*Where monitoring of the satellite transmit signal is possible, a spectrum analyzer could be used to optimize the antenna pointing and polarization.

The configuration (see §1.2.5.2, §1.2.5.3 and Fig. 1) is proposed to efficiently provide an SNG operator with the minimum communication facilities in an occasional-use SNG environment (i.e., short notice, short duration, high traffic demand) on shared-use transponders.

1.2.5.2 *Channels between SNG terminal and satellite operator*

Liaison with the satellite operator's communications control centre should be available at all times and should not be restricted to the duration of the transponder booking. For this purpose it is desirable that coordination channels, in each direction, be provided preferably in the same transponder as the programme vision and sound. Single-channel-per-carrier (SCPC) techniques may be used to provide these channels. (These are shown in Figure 1 as Carriers 1 and 2.)

Carrier 1 (see Fig. 1) provides communication from the satellite operator's communications control centre to the SNG terminals. This carrier should always be present and may be used for the following purposes:

- Voice and data,*
- Satellite channel identification,*
- Alignment of SNG terminal,*
- Safety in the event of misalignment,*
- Verification of channel availability,
- Activation of SNG transmitter for control purposes,*
- Tracking down-converter reference signal for receiving SCPC carriers.

It may be desirable to supplement this carrier with a separate pilot carrier* to accommodate some of these functions, although it is considered preferable to use Carrier 1 for this purpose.

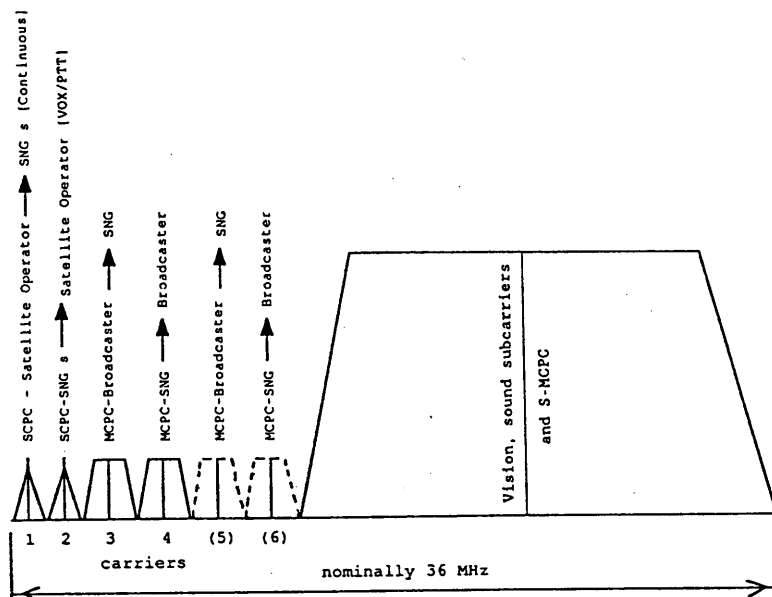
Carrier 2 (see Fig. 1) provides communication from the SNG terminals to the satellite operator's communications control centre. Since, at the same time, more than one SNG terminal may wish to communicate via this carrier, voice operated switching/"press to transmit" equipment should be used. This channel could also carry an SNG terminal identification code.

If it is necessary to provide these carriers elsewhere on the satellite, and linear polarization is being used, they should be provided on both polarizations to avoid the need for a dual polarization feed on SNG terminals.

1.2.5.3 *Channels between SNG terminal and broadcaster's premises*

For the purpose of communicating with the broadcaster's premises, typically up to four (4) two-way (duplex) voice/data channels are required per broadcaster. These channels generally operate for brief periods before and after transponder bookings and during actual programme transmissions.

*See also §1.2.6.3 of this Report



Note 1: Not drawn to scale, nor in order.

Note 2: Carriers 1, 3 and 5 may be allocated in other satellite repeaters if appropriate coverage with the SNG terminal is located outside the downlink footprint.

Figure 1 - Example of satellite carrier loading

Two-way channels can be multiplexed onto a pair of multiple-channels-per-carrier (MCPC) carriers. (Carriers 3 and 4 (see Fig. 1) represent these MCPC carriers, one in each direction, between the SNG terminal and the broadcaster's premises.)

Carriers 5 and 6 (see Fig. 1) represent the MCPC carriers, one in each direction, between the "next" SNG terminal in the booking sequence and the broadcaster's premises. Carriers should only be used in these numbered pairs. While this represents the international operational practice, it does not preclude additional SCPC channels from being used in domestic or national circumstances.

These "two-way" channels between the SNG terminal and the broadcaster's premises could be used for:

- production coordination;
- engineering coordination;
- programme-related data transmission;
- more than one broadcaster;
- more than one language.

1.2.5.4 Auxiliary communications when outside the downlink footprint

When the uplink is in the 14 GHz band but outside the downlink footprint and/or PSTN connections are not available, a possible solution would be to use INMARSAT services or, alternatively, the 6/4 GHz band, either one way or two way.

1.2.5.5 *Technical considerations*

The technical parameters and other considerations that cannot be defined at this time and require further investigation are:

- (a) SCPC/MCPC carrier levels;
- (b) SCPC/MCPC/visual carrier frequency spacing;
- (c) Modulation techniques (using analogue or digital techniques);
- (d) Carrier combining configuration (e.g. pre-HPA or post-HPA);
- (e) Technical consideration for transmission of communication channels in the case of a double-hop situation (e.g. IF repeat);
- (f) Phase noise and frequency stability requirements to be consistent with any SCPC system to be utilized;
- (g) Limitations to uplink operation while not in the downlink footprint;
- (h) A possible solution to the problem in (g) could be SCPC/MCPC transmitted over a wide, or even a global, coverage area (14/11 GHz, 6/4 GHz, or other frequency bands);
- (i) Possibility of using subcarriers (associated with the main TV carrier) to realize the channels from the SNG terminal to the broadcaster's premises;
- (j) Costs, size and weight of equipment;
- (k) Current equipment availability;
- (l) Importance of consensus on the technical specification between all concerned.

1.2.5.6 *Communication channels for SNG (sound)*

For SNG sound broadcasting, many of the concepts contained in §1.2.5 apply.

The minimum requirements for an SNG (sound) terminal are as follows:

- One two-way voice communication channel between the SNG terminal and the satellite operator's communications control centre.
- Two two-way voice/data communication channels between the SNG terminal and the broadcaster's facilities.

1.2.6 *Installation and operational features*

1.2.6.1 *Antenna installation and alignment*

An approximate mechanical alignment of the antenna of the SNG terminal with the calculated azimuth, elevation and polarization will facilitate the acquisition of the satellite. Particularly when operating at low elevation angles, the satellite signal could be susceptible to rain fading which would degrade the transmission and could result in difficulties with the antenna alignment.

Frequency reuse satellites employ either linearly or circularly polarized antennas. Therefore, the antenna feed of the SNG terminal must be aligned to match the polarization of the satellite otherwise severe interference to other traffic in cross-polarized channels and severe signal degradation of the wanted signal may occur. Transmission shall only be initiated after antenna alignment to the correct satellite has been verified.

In most instances SNG terminals do not require any antenna repointing during programme transmissions. However, if the satellite has excessive drift in station keeping, some realignment of the antenna may be required. The extent of realignment will also depend on the size (beamwidth) of the SNG antenna.

All antenna adjustments, such as elevation, azimuth and polarization, should preferably be carried out from one convenient position, behind the antenna.

1.2.6.2 *Other design considerations*

The SNG terminal can be designed with all the equipment next to the antenna or with the HPA and LNA at the antenna and the other equipment located some distance (e.g. less than 50 m) away from the antenna. The latter option, which is operationally preferable, allows the SNG terminal to be operated and monitored from indoors after antenna alignment. The HPA must be located as close as possible to the antenna to minimize the waveguide losses.

The system should be designed to accept redundant high power amplifiers.

System interconnections must be reduced to a minimum to allow quick and easy setup under all weather conditions; all external connectors must be waterproof.

It is desirable that the SNG terminal can accept a primary power input of 100-250 V/50-60 Hz, and power consumption not greater than 2.5 kVA.

The design of the shipping cases should allow the terminal to be operated in direct rain with the lids removed. The cases must provide a controlled environment for the electronics to allow proper operation during extremes of both high and low temperatures.

The SNG terminal must be designed to offer the maximum possible safety to the operator.

1.2.6.3 *SNG system enhancements*

To reduce the cost, enhance the reliability and safety, and simplify the operation of the SNG terminal, certain operational features may be implemented:

- Provision to uniquely identify the satellite and transponder by utilizing a pilot, modulated with a code, on the edge of the transponder.
- The pilot signal could be used for initial alignment of the antenna of the SNG terminal.

- Provision for remote enabling of the SNG uplink transmitter from the satellite operator's control centre through use of a "transmit enable code" on the pilot. This allows the satellite operator to share control over transmission.
- In the event of antenna mispointing, the enable code would not be received and the high power amplifier would be turned off.
- By using a tracking downconverter on the pilot signal the cost of the SCPC equipment may be reduced.
- By calibrating the power of the SNG equipment SCPC channels and calibrating the main carrier power during the SNG equipment approval process, the required uplink e.i.r.p. can be determined prior to enabling the main power amplifier.
- Provision to uniquely identify the SNG terminal by modulating an identification code on the calibrated SCPC channel. This code could, for example, define country of origin, organization, unit number, and certification number.

1.2.7 *Reliability/availability*

There are two aspects to be considered: mechanical reliability and operational availability.

1.2.7.1 *Mechanical reliability*

A high degree of mechanical reliability is achieved by designing the packaging of the various sub-systems of the SNG equipment to withstand rough handling in shipping and deployment in adverse weather conditions. The reliability of equipment is greatly improved by mounting all equipment on sub-frames which are supported by shock-absorbing mounts within the transport cases to protect from shocks and vibrations encountered during transportation. The shipping cases should be designed to provide weatherproofing.

If a segmented antenna is used, the individual segments should be capable of rigid assembly with captive lock pins. The segmented antenna system must be able to withstand stresses and wear resulting from the frequent assembly/disassembly process. It is important to maintain the physical shape of the reflector under frequent and continued usage conditions. The antenna system should be calibrated periodically to ensure that the side-lobe characteristics have not changed after continued field use.

Strong wind conditions affect the pointing accuracy of the antenna and may be controlled by placing weights on the antenna support or guying the antenna to some substantial structure or the ground. It is important to ensure that the antenna shape is not distorted by the use of any system of guys. At very high wind velocities, it may be advisable to stow the antenna.

It is desirable to design and package all equipment to operate under the following conditions: temperature -30°C to $+50^{\circ}\text{C}$, relative humidity 0 to 100%, solar radiation 1100 W/m^2 .

1.2.7.2 *Operational availability*

System performance availability should be measured under "clear sky" conditions. Because of the highly portable nature of the SNG terminal, it is sometimes not possible to provide for severe precipitation fade protection.

Precipitation fades in the 6 GHz band are relatively insignificant, whereas they can be of the order of 3 to 4 dB or more in the 14 GHz band, under adverse weather conditions. Fades in excess of 10 dB for several minutes can be expected in areas where heavy precipitation is not unusual. In the event of rain fades, transmissions may have to be repeated. In higher frequency bands such fades can be even greater.

1.2.8 *Radiation hazard*

The small physical size of the uplink system, and the likelihood of it being sited where it is readily accessible to the general public, require measures to be taken to ensure that exposure to excessive radiation is avoided.

Various countries specify differing maximum permissible radiation levels, therefore the values for the country in question should be used for determining the unsafe area. Safety standards generally distinguish limits for "public" and "occupational" exposure.

The unsafe area should be clearly defined by markers placed in accordance with the values for "public exposure" limits.

The equipment should be designed so that all adjustments can be done before the uplink power is applied, or are so placed that they can be done in safety with power applied.

All high power connections (waveguide, co-axial connectors, etc.) should be checked for integrity before switching on the high power amplifier. SNG terminals must be equipped with a calibrated radiation meter to check compliance with the safety standards around the antenna and components carrying high level radio frequency energy.

1.3 *Examples of link performance*

1.3.1 *Examples of the link performance for various satellite systems*

Table I shows typical link performance for 6 GHz (INSAT) and for 14 GHz (AUSSAT, EUTELSAT, JCSat/SCC) satellites.

1.3.2 *Examples of link performance for Intelsat satellites*

Tables II, III and IV show typical performance examples for 6/4 GHz, 14/11 GHz and cross-strapped transponders using Intelsat satellites.

1.3.2.1 *Performance results and modulation parameters*

To satisfy SNG transmission requirements, a minimum peak-to-peak luminance-to-weighted thermal noise ratio of 45 dB is assumed. Tables II, III and IV show examples for television transmission through Intelsat using System M (NTSC, 525/60) with 7.5 MHz peak video deviation, deemphasis in accordance with Rec. 405 and noise weighting in accordance with Rec. 421. The transmission bandwidth is assumed to be 17.5 MHz.

TABLE I - Examples of the link performance for various satellite systems

PARAMETER		SYSTEM			
		JCSAT/SCC	INSAT	AUSSAT	EUTELSAT
<i>Up-link</i>					
Frequency	(GHz)	14.25	6.0	14.1	14.25
SNG e.i.r.p.	(dBW)	70.0	58.0	70.0	70.0
Antenna diameter	(m)	1.8	2.5	1.8	1.8
Transmitter power	(W)	300	100	300	300
Spreading loss	(dB)	162.7	162.0	162.7	162.7
Miscellaneous loss ⁽¹⁾	(dB)	6.0	0.0	0.2	0.3
P.f.d. at satellite	(dBW/m ²)	-98.7	-104.0	-92.9	-93.0
Satellite G/T ⁽²⁾	(dB(K ⁻¹))	10.0	-4.5	0.5	3.5
Uplink C/N	(dB)	20.8	8.6	17.0	19.9
<i>Down-link</i>					
Frequency	(GHz)	12.5	4.0	12.35	11.3
Satellite e.i.r.p. per carrier	(dBW)	50.0	26.1	29.5	41.0
Spreading loss	(dB)	162.7	162.0	162.7	162.7
Miscellaneous loss ⁽¹⁾	(dB)	1.0	0.5	0.4	0.2
P.f.d. at receiving earth station	(dBW/m ²)	-113.7	-135.9	-133.6	-121.9
Receiving earth station G/T	(dB(K ⁻¹))	30.0	31.7	38.0	35.0
Downlink C/N	(dB)	27.0	16.0	15.2	24.5
<i>Overall</i>					
Overall C/N	(dB)	19.9	8.0	12.5	18.6
Video bandwidth	(MHz)	4.2	5.0	5.5	5.0
IF bandwidth	(MHz)	28.5	28.0	30.0	30.0
Frequency deviation	(MHz p-p)	21.0	24.0	22.3	19.0
Improvement factor (includes preemphasis and weighting)	(dB)	37.8	39.1	39.4	34.3
Overall weighted S/N	(dB)	57.6	47.0	51.9	52.9

(¹) Miscellaneous loss includes atmospheric loss and/or the effect of rain attenuation, depending on the case.

(²) All values refer to near beam centre, except for INSAT, which refers to the -3 dB contour.

Note: Effects of interference are not considered.

TABLE II - 14 GHZ-band cases

Transponder Bandwidth	Earth Station Standard ⁽¹⁾	14 GHz Band ⁽²⁾			
		Elliptical east spot up/ elliptical east spot down		Circular west spot up/ circular west spot down	
		C/N (dB)	S/N (dB)	C/N (dB)	S/N (dB)
36 MHz	C	17.9	49.3	21.4	52.8
36 MHz	E3	16.6	48.0	20.2	51.6
72 MHz	C	16.9	48.3	21.4	52.8
72 MHz	E3	15.2	46.6	20.2	51.6

(¹) The receive earth stations used are:

C : Standard "C", $G/T = 37 \text{ dB(K)}^{-1}$

E3 : Standard "E3", $G/T = 34 \text{ dB(K)}^{-1}$.

(²) All calculations based on beam centre.

Note: Antenna diameter = 1.8 m, transmit e.i.r.p. = 70 dBW (300 W HPA)

TABLE III - Cross-strap cases

Transponder Bandwidth	Earth Station Standard ⁽¹⁾	14 GHz-band elliptical spot beam to			
		4 GHz-band zone cross strap ⁽²⁾		4 GHz-band hemi cross strap ⁽²⁾	
		C/N (dB)	S/N (dB)	C/N (dB)	S/N (dB)
36 MHz	A	16.8	48.2	16.8	48.2
36 MHz	A(R)	13.1	44.5	13.1	44.5
72 MHz	A	17.0	48.4	17.0	48.4
72 MHz	A(R)	13.5	44.9	13.5	44.9

(¹) The receive earth stations used are:

A : Standard "A", $G/T = 40.7 \text{ dB(K)}^{-1}$

A(R) : Revised Standard "A", $G/T = 35 \text{ dB(K)}^{-1}$

(²) All calculations based on beam centre.

Note: Antenna diameter = 1.8 m, transmit e.i.r.p. = 70 dBW (300 W HPA)

TABLE IV - 4 GHz-band cases

Transponder bandwidth	Earth Station Standard	Global 4 GHz-band ⁽²⁾		Zone 4 GHz-band ⁽²⁾		Hemi 4 GHz-band ⁽²⁾	
		C/N (dB)	S/N (dB)	C/N (dB)	S/N (dB)	C/N (dB)	S/N (dB)
36 MHz	A	8.9	40.3	17.0	48.4	14.8	46.2
36 MHz	A(R)	7.3	38.7	13.5	44.9	11.9	43.3
72 MHz	A	8.3	39.7	14.7	46.1	13.5	44.9
72 MHz	A(R)	5.9	37.3	10.2	41.6	9.7	41.1

(¹) The receive earth stations used are:

A : Standard "A", $G/T = 40.7 \text{ dB(K)}^{-1}$

A(R) : Revised Standard "A", $G/T = 35 \text{ dB(K)}^{-1}$

(²) All calculations based on beam centre.

Note: Antenna diameter = 3.0 m, transmit e.i.r.p. = 68.5 dBW (350 W HPA)

1.3.2.2 14 GHz-band cases

Performance examples for 14-GHz band transmission, assuming a transmitting antenna diameter of 1.8 m and an e.i.r.p. of 70 dBW are given in Table II.

1.3.2.3 "Cross-strap" cases

Performance examples for "cross-strap" transmission, that is, 14-GHz band uplink, 4 GHz-band downlink, are shown in Table III, assuming a transmitting antenna diameter of 1.8 m and an e.i.r.p. of 70 dBW.

1.3.2.4 4 GHz-band cases

Performance examples for transmission from a relatively small 6 GHz band SNG terminal are shown in Table IV.

1.4 Experimental and operational experience

Operational experience is increasing rapidly with the use of SNG for coverage of world events such as Papal visits, summit conferences, national elections, natural disasters, and particular programme production needs. The experience is being gained on a world-wide basis as these events dictate.

While the equipment currently available is being used for the coverage of the actual events, additional experimental work is being carried out in the areas of:

- overall performance under difficult operating conditions;
- portability and operating convenience of the equipment;
- the communications requirements;
- the technical and operating standards and practices.

Plans are already in place to obtain more experimental experience in the coming months.

2. Uniform operating procedures required for temporary authorization for satellite news gathering

2.1 Introduction

SNG differs from most other forms of satellite transmissions in a number of ways. For example, the requirement for SNG typically is identified only days, possibly hours, before transmission. It lasts typically for no more than a few days, or at the most, weeks. Nevertheless, the SNG operator has to comply with the regulations of the host country and with a number of procedures which are designed to ensure the proper management and protection of the space segment and frequency spectrum.

The regulatory framework in which an SNG operation takes place has a dual effect on its operational effectiveness. In order to carry out its intended function, the SNG operator must have access to temporary agreements and/or authorizations in a timely and cost-effective manner. The operator's needs range from frequency authorization, to coordination with the space segment entity, to tariffs and administrative costs, to the necessary supporting lines of communication. This section describes the nature of the operational information that is required to assist the SNG capability.

Given that SNG requirements are occasional and/or temporary and that coverage for an unplanned fast-breaking news event is a valuable worldwide service, expeditious approval for activation of portable earth stations is essential.

The successful application of SNG technology requires uniform agreement on standard technical approaches and recognized operating procedures. (See § 5.4.) The frequency and number of programme sound channels as well as the number of auxiliary, data and coordination simplex and duplex channels should be uniformly adopted.

2.2 Earth station approval

Earth station approval is necessary to allow the responsible body to ensure compatibility of the SNG terminal with the space segment. To meet this requirement, administrations are requested to consider procedures to permit the SNG terminal to be brought into service as quickly as possible. Administrations are urged to investigate the possibility that an SNG terminal whose technical performance is approved by the space segment providers be accepted on a uniform basis and they are encouraged to complete administrative procedures in close cooperation with SNG operators as expeditiously as possible. A technical report demonstrating the measured performance characteristics should be prepared and be available to the administration. The following technical characteristics should be documented as a minimum:

- Transmit gain as a function of frequency;
- Transmit off-axis gain;
- Transmit main beam e.i.r.p.;
- Transmit beamwidth and polarization;
- Transmit main beam spectral density for the worst 4 kHz;
- Transmit off beam spectral density for the worst 4 kHz;
- Maximum energy dispersal;
- Receive G/T as a function of frequency;
- Cross polarization isolation;
- Receive and transmit frequency agility within the operating bands;
- Spurious emissions (in-band and out-of-band).
- Manufacturers' model numbers, modulation characteristics and frequency stability.

2.3 *Frequency assignment and coordination*

Frequency coordination procedures are derived from international and national regulations. In order to assess the acceptability of an SNG terminal in this respect, the responsible body may require the same information noted above, plus details of the geographical location of the SNG terminal and the anticipated transmission times.

The SNG operator requires in a timely manner radio regulatory authorization by the host administration for operation of the SNG terminal. Co-ordination between the proposed SNG terminal and existing terrestrial radio services is a necessary prerequisite to manage radio interference and allow authorization of the SNG terminal. Frequency coordination for SNG terminals is more difficult in the 6 GHz band, since this band is shared between fixed satellite and fixed terrestrial services and many fixed terrestrial links exist. For reasons of portability and ease of co-ordination, use of higher frequency bands (e.g. 14 GHz and 30 GHz) is beneficial. A portion of the 14 GHz band is not shared with fixed service using radio-relay systems making SNG coordination much easier. In some countries, however, the 14 GHz band is used for fixed and other terrestrial purposes and coordination is, therefore, necessary. The frequency band 29.5-30.0 GHz is assigned exclusively to the FSS (earth to space) on a primary basis. With respect to future technological improvements this band may become suitable for SNG applications. Where coordination is required between the SNG station and the terrestrial links of the host country, it may be different because of the temporary nature of SNG.

2.4 *Space segment booking*

The SNG operator needs to have a quick and clear understanding of what space segment will be available for his purpose in a timely manner (e.g. within less than 24 hours). This information needs to include:

- transponder characteristics, (satellite identifier);
- amount of bandwidth and power;
- earliest available time of access.

The SNG operator may require direct contact with the space segment provider on a continuous basis.

2.5 *Support communications*

In order to facilitate the effective operation of the SNG terminal, support communication channels may be required. Separate, independent transmission support channels are highly desirable. These additional facilities may include required point-to-point microwave, telephone communication systems, two-way simplex/duplex radio, wireless microphones and mobile satellite terminals for voice and data.

In the case of the radio-related services, temporary authorization may be needed to use frequencies at the desired location. This authorization needs to be obtained in a timely fashion, and to the extent possible, in advance of actual equipment deployment. Also, it is necessary to have information with respect to the technical acceptability of particular equipment which might be used.

In addition, it may be necessary typically to have a few or many telephone lines. These are critical to the proper functioning of the SNG. The SNG operator will require cost-effective (including a clear definition of tariffs and services) and reliable access to such telephone lines, and will need to have knowledge of the actual circuit availability involved.

2.6 *Radiation hazards*

It is essential to protect the public and personnel from hazardous radiation. Many administrations have established standards for safe exposure to radio (non-ionizing) radiation which are a function of frequency, power level and duration of exposure.

SNG operators should comply with permitted radiation standards (health and safety) established by the host country. Where the host has not established its own standards the World Health Organization (WHO) standards should be used.*

Operators must carefully consider the siting and configuration of specific installations, and the access of both occupational personnel and the general public, when applying applicable maximum exposure criteria.

A danger area around the SNG terminal shall be identified, checked and clearly marked.

2.7 *Importation and customs*

The SNG Operator should have a sufficient understanding of the importation and customs system of the host country. This is particularly important when there is frequent news gathering and where facilities in that country cannot be used.

* WHO develops health criteria in conjunction with the International Non-Ionizing Radiation Committee of the International Radiation Protection Association.

2.8 *Contact point for information, guidance and approval*

Each administration or relevant organization should, if possible, establish a designated point of contact (DPC) for SNG, which should be available for 24 hours per day, seven days per week.

This contact point should be available for assistance to facilitate temporary authorization of SNG earth stations owned by foreign operators through intermediating exchange of information necessary for authorization procedures and frequency coordination and providing guidance for the administrative procedures of the host country.

It is considered necessary to exchange the information shown in Table V.

TABLE V

<i>Information to be given to the Administration</i>	<i>Information to be received from the Administration</i>
1. Service characteristics (e.g. TV standard, number and type of audio channels, number of telephone support circuits)	1. Enquiry reference
2. Time and duration of service	2. Ordering entity
3. Uplink and downlink location	3. Other authorizing sections to be routed to
4. Intended space segment	4. Custom contact points
5. SNG earth terminal characteristics (e.g. existing application, identification code, frequency band, maximum e.i.r.p., antenna transmit pattern, modulation methods)	5. Tariff information
6. Transmission support (e.g. microwave, etc.)	6. Safety standards

3. *Satellite parameters relevant to satellite news gathering (television)*

3.1 *Introduction*

This section gives information on several characteristics of current and planned satellites relevant to the design and operation of SNG terminals for international connection and to the performance possible.

3.2 Uplinks

3.2.1 Orbit spacing

Satellite systems may use orbit spacings as low as 2° . For actual SNG operations the particular established satellite configurations in the orbital arc appropriate for the required circuit need to be taken into account.

Details of current satellites are available from the IFRB.

Some administrations require that a transmitting satellite earth station have an antenna radiation pattern designed to ensure that sidelobe peaks do not exceed $G = (29 - 25 \log \phi)$ dBi, at least in the direction of the geostationary orbit. Further information is given in §1.2.3.3.

3.2.2 Frequency bands

Frequency bands identified as operationally preferable for SNG uplinks are those above 10 GHz. Currently available satellites use the bands near 14 GHz and near 30 GHz. There is preference for use of the 14 GHz band for which the technology is sufficiently mature.

3.2.3 Uplink service areas

Uplink service areas fall into two general categories:

- national
- international.

Nations spanning large areas sometimes use large satellite antenna beamwidths which would be more appropriately categorised as "continental".

3.2.4 Satellite G/T

The predominant factor governing G/T performance is the size of the uplink service area. National beams generally have higher gain antennas, more suitable to the low e.i.r.p.s from SNG terminals.

The more common configuration is for large uplink service areas - national and international - resulting in G/T values of zero or in some cases negative at the edge of the beam.

To be as flexible as possible, the SNG terminals should be capable of working to satellite G/T values as low as $-6 \text{ dB(K}^{-1})$, even if with lower performance. It is desirable for normal operation to take account of satellite values of $0 \text{ dB(K}^{-1})$.

3.2.5 Satellite gain

Satellites are often equipped with variable gain settings to achieve a higher e.i.r.p. Lower gain values have been shown to provide improved overall performance for SNG applications.

As it is not always possible to change satellite gain settings for short-term use, SNG terminals should be capable of operating with nominal satellite gain settings and the downlink receiver should be appropriately dimensioned.

3.2.6 *Bandwidth*

It may be possible to carry SNG on a shared basis with other television signals, and/or SCPC communication channels, depending on the transponder bandwidth available and its intermodulation characteristics.

3.2.7 *Channel plans for satellites*

Satellites often make use of overlapping channels with orthogonal polarization separation. Account needs to be taken of interference in relation to each specific satellite, in particular the polarization discrimination requirements.

3.2.8 *Future design possibilities*

Steerable antennas

Advantages of steerable satellite beams have been demonstrated by Intelsat. This allows a narrow beam antenna to be directed to the SNG earth station location to provide improved performance.

3.3 *Connectivity*

Satellites generally have some levels of connectivity (on-board switching) flexibility ranging from switching to spare components to variable interconnection of input amplifiers, output amplifiers and antenna beams. Use of variable connectivity depends on the satellite operator and may be related to the level of reliability and the inter-relationship of the satellite hardware affected by any change.

Where available, reconfiguration of connectivity may be useful in establishing the desired circuit. In some satellites it may be possible to connect 14 GHz band uplink capacity to, for example, 4 GHz band downlink circuits, however, two-way communication channels can sometimes be affected.

Improved flexibility in future satellites may be of benefit to SNG operations.

3.4 *Downlink*

3.4.1 *Downlink service area*

It is necessary that the downlink service area encompasses the intended receiving site. For ease of establishment, for communications and for monitoring, it is also highly desirable that the SNG terminal site be capable of reception of signals from the satellite.

4. *Satellite news gathering for sound broadcasting*

4.1 *Need for SNG (sound)*

Broadcasting organizations may need an application of SNG (sound) for the coverage of news and other events. Situations warranting such applications are:

- non-availability of normal terrestrial links having required baseband capacity (30 Hz-10 kHz), either due to remoteness of the location or due to disruption, as in the case of a natural disaster;

- news dispatches requiring high reliability, with minimum human intervention;
- requirement of instantaneous broadcast coverage of a news event.

4.2 Available techniques

The operational requirement for SNG (sound) can be realized by utilizing portable uplink terminals operating with dedicated or leased space segment capacity from any satellite system compatible with the technical characteristics of the SNG terminal.

In order to allow origination of carriers from different geographical locations, SNG (sound) may utilize an SCPC system operated in:

- dedicated transponder mode, where multiple carriers exist within the same transponder;
- edge of transponder operation, where narrow-band, low-level carriers may be added at the edge of a transponder band in addition to a TV signal.

A hand-carried personal communications device (PCD) would provide a baseband of 100 Hz-3 kHz (telephone quality). In this case, independent satellites such as the Inmarsat Standard A system and future systems utilizing amplitude companded single sideband (ACSSB), narrow-band frequency modulation (NBFM) and voice coder/decoder (vocoder) technology could be applicable.

4.3 Quality objective

For SNG applications, the performance objective for the sound programme signals at the receive earth station of the broadcaster shall be:

- S/N : 50 dB (weighted), or higher if digital technique is used;
- Gain/frequency response: } as per Recommendation
- Non-linear distortion: } 504-2 (Geneva, 1982)

When received at the SNG terminal, the quality objective given above could be relaxed by the user.

4.4 Appropriate frequency band

Practically all the bands allocated to the FSS can be utilized for SNG (sound). In addition, this service can also be operated in the mobile-satellite service (MSS) bands, by employing a PCD (see §4.2).

SNG terminals as defined in this Report are feasible in the bands indicated below:

6/4 GHz or 6/2.6 GHz

These bands are being used extensively in existing satellite systems and provide intercontinental, continental or national coverage, however, the SNG terminals would be considered transportable.

14/11 GHz, 14/12 GHz or 30/20 GHz

The 14/11 GHz and higher frequency bands are suitable, even though these bands have the disadvantage of greater attenuation due to rain. In temperate climates, however, the use of these bands would allow a further reduction of the size of the antenna and can open the way to the ideal situation where just one reporter carrying a small suitcase could access a satellite from anywhere within the service area.

In the case of relaxed quality objectives (§4.3), the 1.6 GHz band could be used with a PCD.

4.5 Technical characteristics

In order to allow flexibility of operation, SNG terminals should be compatible to operate with a wide range of spacecraft parameters. However, for the nominal case, the typical satellite and operating parameters are given in Tables VI and VII, respectively.

TABLE VI - Example of satellite characteristics for SNG (sound)

Frequency up/down (GHz)	G/T (Edge of coverage area) (dB(K ⁻¹))	Saturation power flux-density (single carrier) (dB(W/m ²))	E.i.r.p. (saturation) (Edge of coverage area) (dBW)
6/2.6	-5	-87	42
6/4	-5	-85	32
14/11	-3	-85	36
30/20	+0.5	-82	48

The antenna must be as small as possible. Since the terminal is required to receive downlink audio, the antenna size cannot be reduced arbitrarily by increasing the rating of the high-power amplifier (HPA). Furthermore, Rec. 524-2 (MOD I) specifies limits on the off-axis power spectral density (PSD) in the 6 and 14 GHz bands and this imposes constraints for the reduction of antenna size. Report 1001 also deals with the off-axis PSD for the frequency range 10-15 GHz.

In the case of a 6/4 GHz band SNG terminal, the above limitations can be satisfied by utilizing a 2.4 m antenna. This will yield a receive system G/T of the order of 19 dB(K⁻¹) at 4 GHz. Adaptive energy dispersal may be required with an FM-SCPC carrier. Use of a solid state power amplifier (SSPA) instead of a travelling wave tube amplifier (TWTA), wherever practicable, would make the SNG terminal more compact.

In the case of the 14/11, 14/12 and 30/20 GHz bands, even smaller antennas, up to 1.2 m, could be used. The receive system G/T of the SNG terminal in these bands is required to be of the order of 17 dB(K⁻¹).

The following features of the SNG terminal will add to its flexibility of operation:

- adjustable e.i.r.p. settings,
- ability to change the modulation deviation characteristic at the IF modulator stage,
- synthesizer-tuned upconverter,
- adjustable setting for polarization.

TABLE VII - Example of SNG (sound) system characteristics

Frequency up/down (GHz)	Antenna diameter (m)/transmit gain (dBi)	Transmitter power (dBW)	Bandwidth (kHz)	C/N up (dB)	Receive system G/T (dB(K ⁻¹))	C/N down (dB)	Post-detection S/N (dB) (weighted)
(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)
6/2.5	2.4/41.5	12.5	200	24.76	(a) 9 (b) 17	7.5 15.5	44.5 52
6/4	2.4/41.5	14	200	26.26	(a) 18.6 (b) 29	4.4* 13.8	40.4 50.6
14/11	1.8/46.4	9.4	200	21.6	(a) 18.5 (b) 29	5.5* 15	41.4 51.1
14/11	2.5/50	20	(2 Mbit/s) ~2000	18	24	12	>70
30/20	1.2/49.5	9.8	200	21.6	(a) 16.8 (b) 29	4.6* 15.8	40.6 51.7

(1) G/T at (a) indicates the value for SNG terminal whereas (b) indicates the minimum value at the broadcaster's base station.

(2) Use of a threshold extension device has been assumed for demodulation at the SNG terminal.

* predetection bandwidth: 52 dBHz.

4.6 Modulation methods

This report is based on FM-SCPC, since this type of modulation is quite effective in a power-limited environment. Nevertheless, other modulation methods such as PCM/PSK, delta-modulated PSK, etc., could be used if permitted by the system.

In the case of PCD, real-time transmission will be limited to a baseband of telephone quality. Additional processing such as that obtainable with a 2.4 kbit/s vocoder yields higher reliability at the cost of voice quality while still maintaining intelligence. More processing, such as store forward, to yield bandwidth compression would allow 4.8 kbit/s vocoder quality at the cost of longer transmissions. Inmarsat standard "A" provides telephone quality and the system uses a 1.2 m dish at 1.6 GHz.

4.7 Communication requirements for supervision and coordination

SNG uplink signals are often originated from remote areas. In these cases, communications using public switched telephone network (PSTN) is difficult or impossible. The SNG system should, therefore, include equipment to support independent, two-way voice/data communications between:

- the SNG terminal and the control centre of the satellite operator;
- the SNG terminal and the broadcaster's headquarters.

5. Minimum requirements applicable to SNG (television) terminals

5.1 General functional requirements

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

5.2 Transmission performance requirements

5.2.1 Video signal

Shall be compliant with Recommendation 567-2, except possibly for signal-to-noise ratio.

5.2.2 Audio signal

Shall be compliant with Recommendation 505-3, except possibly for signal-to-noise ratio.

5.3 Equipment requirements

5.3.1 Antenna

5.3.1.1 Off-axis e.i.r.p density

Shall be compliant with Recommendation 524-2 (MOD I) or the satellite operator's requirements, whichever is more stringent.

5.3.1.2 On-axis e.i.r.p. (14 GHz)

When using a 1.8 m antenna and a 300-W HPA, the on-axis e.i.r.p. is nominally 70 dBW.

5.3.1.3 Polarization discrimination requirements

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

5.3.2 *Video/audio exciter*

This unit should include:

- a video low pass filter and switchable pre-emphasis networks, dependent on the television standard;
- selectable IF bandpass filters, dependent on the transponder bandwidth;
- a minimum of two frequency agile sound subcarrier modulators, adjustable between 5.5 and 8.0 MHz;
- a dispersal generator, the deviation of which can be easily adjusted, if required, between 0.5 and 3.0 MHz p-p;
- power control, with a minimum range of 30 dB;
- a linear video FM modulator capable of deviating the IF carrier between 7.0 and 15.0 MHz peak and capable of preserving the DC component of the video signal to take advantage of the available bandwidth;
- a frequency agile upconverter (14.0 to 14.5 GHz) with a 250-kHz step size. The phase noise and frequency stability should be adequate to allow upconversion of the auxiliary communication signals.

5.3.3 *Communication equipment*

The minimum requirement for SNG communication equipment should be consistent with the technical considerations given in §1.2.5.5.

5.4 *Requirements for operational procedures*

When the SNG operator determines that there is a need to operate in another country he must comply with the operational procedures of the host country and the satellite system which he intends to use. The easiest way to accomplish this is to use an SNG operator's guide which would be available from the host Administration and from each satellite system.

The contents of these guides should be as harmonized as possible.

5.4.1 *Space segment guide*

The minimum information to be contained in a satellite space segment provider's guide for SNG users is:

- General description;
- Definition of terms used in the user's guide;
- Description of the approval procedures to access the appropriate satellite;
- Booking procedures to obtain the required space segment;
- Radio regulatory and frequency clearance matters relevant to the planned transmission;
- List of contact points in the individual countries;
- Pre-transmission preparatory tests;
- Satellite access procedures;

- Operations control coordination and communications procedures and requirements;
- Operating conditions.

5.4.2 *Administration guide*

The minimum information to be contained in an Administration's guide to SNG operators in that country is:

- General description of the specific situation and rules for SNG operations (e.g., radiation hazard);
 - How to contact the designated point of contact (DPC);
 - The procedure on how to obtain authorizations for SNG and related services;
 - References to satellite operators' guides;
 - Possible services that are available, e.g., for support communications;
 - Tariff information.
-

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,

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

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

The complete theoretical expressions for the conversion factors, F (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.

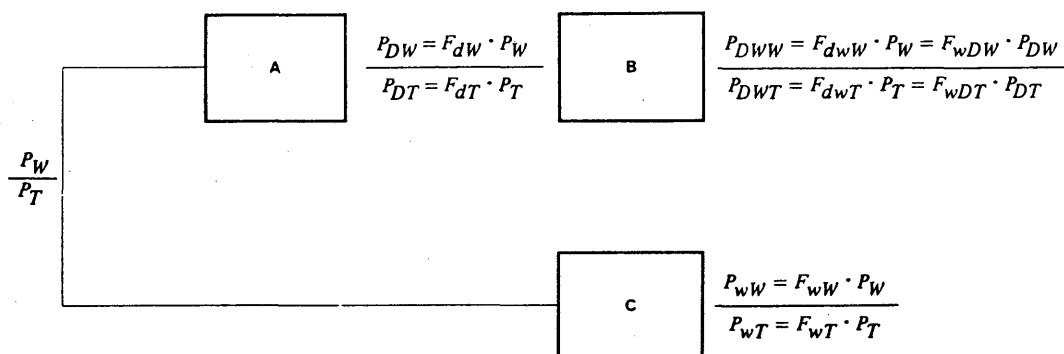


FIGURE 1 - Block diagram with conversion factors F

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		6.2 6.8 7.4	10.3 11.5 12.2	2.7 2.7 3.3	9.9 10.9 11.7	2.3 2.3 3.3	12.8 13.8 14.8
		-0.4	2.9						
		-0.4	2.9						
5.0 5.0 5.0/6.0 ⁽³⁾	B, C, G, H I (luminance) I (chrominance)	625-lines		8.5 6.5 4.8	16.3 12.3 0.8	3.1 2.3 9.4	14.3 10.9 1.3	1.0 0.2 7.3	16.3 12.9 3.3
		-2.1	2.0						
		-2.1	2.0						
5.0 6.0 6.0/5.0 ⁽³⁾	Unified D, K, L Unified	-2.1	2.0	7.4 9.3 8.2	12.2 17.7 14.6	3.0 3.5 3.2	11.2 15.8 13.3	0.9 1.9 1.6	13.2 18.1 15.6
		-1.6	2.3						
		-1.6	2.3						
6.0	Unified	-1.6	2.3	8.0	12.8	3.3	12.0	1.7	14.3

⁽¹⁾ 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

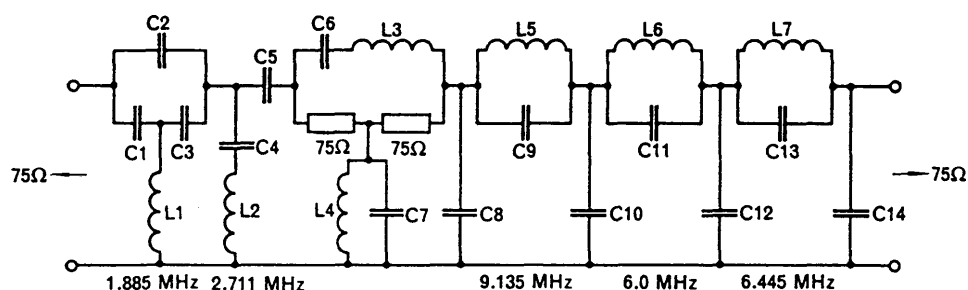
FIGURE 2 - Weighting network design diagram
(non-constant resistance)

TABLE OF VALUES

Component	Value	Tolerance	Component	Value	Tolerance
C1	496.0	$\pm 1\%$	C12	311.4	$\pm 1\%$
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.

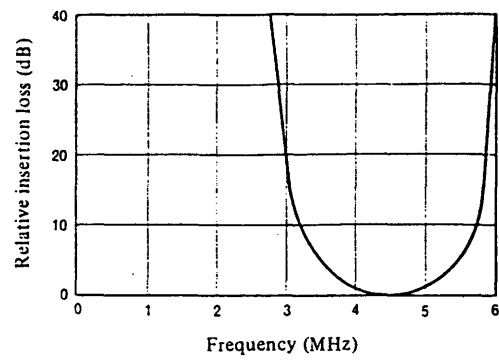


FIGURE 3 – Characteristics of the weighting network

REPORT 816-3

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

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

(1978-1982-1986-1990)

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 Measurements of circuits in Canada indicate a severe form of field time distortion (called superstreaking), which is not adequately measured by standard test signals (signals A and B2) in the format usually used. Although subjectively similar to field and line time distortions, the definitions of these parameters in Recommendation 567 do not properly describe the distortion causing superstreaking. The nature, cause, method of measurement and proposed measurement objectives are given in [CCIR 1986-90 a].

C.6 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.

C.7 [CCIR, 1986-90b] discusses a new approach to the selection of test signals for the assessment of TV channel characteristics. New test signals have been synthesized which enable much greater accuracy to be achieved in the assessment of TV channel characteristics while preserving the advantages of the signals in current use.

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.

D.5 A contribution to the CMTT and CCIR Study Group 11 proposes the use of digital noise suppression on the still parts of the television picture [CCIR, 1986-90c]. This document sets out the findings of theoretical and experimental studies and the methodology of calculating the effectiveness of noise suppression.

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).

[1986-90]: a. CMTT/49 (Canada); b. CMTT/189 (USSR); c. CMTT/96 (USSR).

BIBLIOGRAPHY

DVORKOVICH, V.P. [1988] - Optimizatsiya izmeritelnykh signalov dlya otsenki kharakteristik televizionnogo kanala (Optimization of test signals for the assessment of television channel characteristics) *Radiotekhnika*, No. 2, pp. 21-29.

DVORKOVICH, V.P. [1988] - Optimalye metody izmerenii parametrov televizionnogo signala (Optimum methods of measuring television-signal parameters) *Radio i televidenie*, OIRT, No. 6, pp. 17-21.

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.

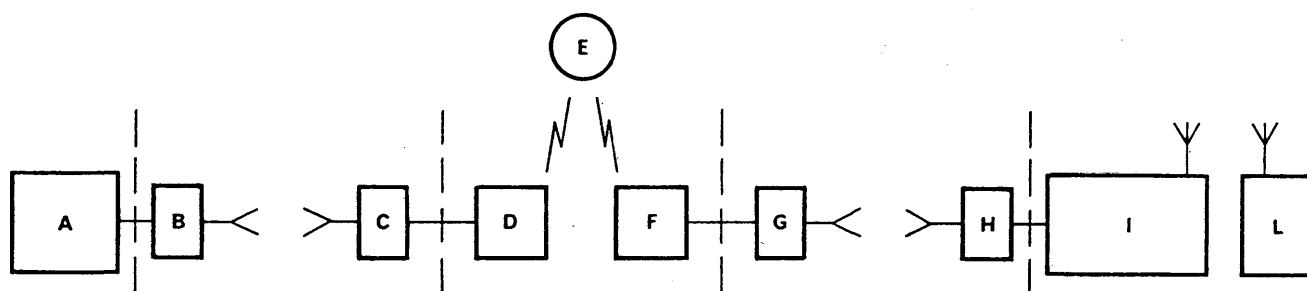


FIGURE 1 - Television system considered

- 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 ⁽³⁾	± 10 ⁽³⁾	— ⁽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 ⁽³⁾

⁽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 1 – 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 $\pm 3\%$	Mesured by using a vectorscope
	Phase error	Within $\pm 3^\circ$	
Linear distortion	Field-time	Within $\pm 1\%$	
	Line-time	Within $\pm 3\%$	
	Short-time ($2T$ pulse-bar)	Within $100^{+6\%}_{-10\%}$	
	Short-time ($2T$ 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	

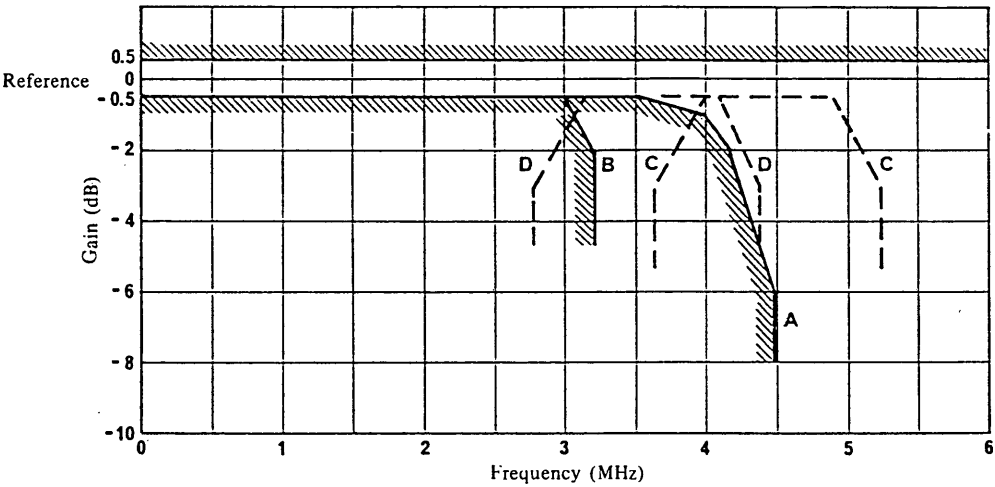


FIGURE 1 – Steady-state gain/frequency characteristics

— 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	> 50 dB	} Total output noise figure
	Weighted	> 56 dB	
Non-linear distortion	Colour saturation	Within $\pm 2\%$	Measured by using a vectorscope
	Phase error	Within $\pm 2^\circ$	
Linear distortion	Field-time	Within $\pm 1\%$	
	Line-time	Within $\pm 3\%$	
	Short-time (2 T pulse-bar)	Within $100 \pm 5\%$	
	Short-time (2 T pulse response)	Within 2%	
	Steady-state gain/frequency characteristics	See Fig. 2	

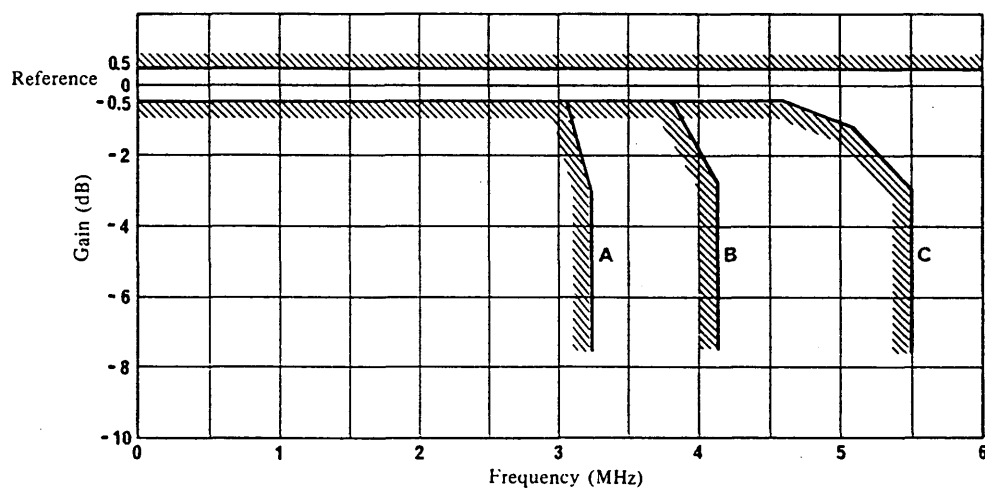


FIGURE 2 — Steady-state gain/frequency characteristics (luminance only)

Conversion mode:

A: SECAM → PAL, SECAM → NTSC, SECAM → SECAM

B: NTSC → NTSC, NTSC → PAL, NTSC → SECAM, PAL → NTSC

C: PAL → PAL, PAL → SECAM

REPORT 1238

TRANSMISSION PERFORMANCE OF ANALOGUE
COMPONENT TELEVISION SIGNALS

(Study Programme 24A/CMTT)

(1990)

Introduction

The emergence of a large number of television systems using analogue component signals raises the problem of the definition of new test signals and new measurement methods to assess their quality [CCIR, 1986-90]. The test signals and measurement methods in Recommendation 567 were defined to describe the quality of composite television signals. Many of these test signals are not suitable for component signals. In order to avoid any inconsistency in CCIR Reports and Recommendations, it is proposed, despite the wide range of applications involved, to consolidate all proposals concerning analogue component television signal quality measurement in a single report. It would then be possible to define the most universal test signals and measurement methods for analogue component television signals:

- separate RGB or Y, C_R, C_B component signals for 525 and 625 line television systems or high definition television systems;
- time multiplexed signals for 525 and 625 line television systems or high definition television systems.

For the definitions of a connection and of reference circuits, refer to Part A of Recommendation 567.

This report contains four parts listed below:

Part A: Definition of parameters

Part B: Measurement methods and test signals

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

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

Part A - Definition of parameters

This part is to include definitions for each system which are necessary for the understanding of the report itself. Information should be taken from existing CCIR texts to briefly describe the systems (e.g. specification of transmission systems for satellite broadcasting service, reports on high definition television).

This part should also contain transmission performance characteristics. Information could be taken from Recommendation 567 and Report 1218.

Part B - Measurement methods and test signals

This part is to include information from the methods and signals proposed in Reports 1096 and 1218—and from other suitable sources.

Part C - Design objectives and tolerances applicable to the hypothetical reference circuit

Administrations are invited to contribute on this topic.

Part D - Performance of circuits shorter or longer than the hypothetical reference circuit

Administrations are invited to contribute on this topic notably in order to verify the possible application of the laws of addition specified in Recommendation 567 to component signals.

REFERENCES

CCIR Documents

[1986-90]: CMTT/212 (France).

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 1092-1

TRANSMISSION OF HIGH-DEFINITION TELEVISION SIGNALS

(Question 22/CMTT)

(1986-1990)

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), multiple sub-sampling encoding (MUSE) [CCIR, 1982-86a] and MAC/packet (HD-MAC). The bandwidths of the signals used in these systems are shown in Table I and Table II.

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].

In the high efficiency TCM (HE-TCM) system which is now under study, a bandwidth compressed analogue signal is obtained by an adaptive sub-Nyquist sampling with intrafield and interframe interpolations. This reduces the bandwidth of 1125-line HDTV signal to 8 MHz [Kishimoto *et al.*, 1987; Kishimoto and Sawada, 1986; CCIR, 1986-90a].

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 compressed analogue signal is obtained by phase-alternative sub-Nyquist sampling with intrafield and motion compensated interframe interpolations. This reduces the bandwidth of 1125-line HDTV signal to 8 MHz [Ninomiya, *et al.*, 1987]. Details of the MUSE system are described in [CCIR, 1986-90b].

In the HD-MAC format compatible with the MAC/packet standard which is now under study, a reduced baseband signal is obtained by means of sub-sampling and pre- and post-filtering techniques. Digital data incorporated in the MAC/packet format will provide the receiver with information on the reconstruction algorithms to be used [Melwig, 1986; Arragon *et al.*, 1986; Boyer and Melwig, 1987]. The HD-MAC signal thus occupies a 12 MHz passband [CCIR, 1986-90c,d].

In the case of digital transmission, component signals of HDTV, such as Y , P_R and P_B , can be employed at codec input/output. These component signals are time-division multiplexed in digital form for transmission [CCIR, 1986-90e].

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.

3. Digital transmission of HDTV signals

3.1 General

The digitization of transmission networks is progressing and the development of bit rate reduction coding systems for HDTV signals is an urgent matter.

At the present time it is important that the reduced signal bit rate should coincide with the hierarchical levels of digital transmission networks. In the near future it is also important that the bit rates should coincide with the channel rates of the broadband ISDN, which are now being studied in CCITT.

The bit rates of some possible formats are shown in the annex for the proposed 1125/60 system. For the proposed 1250/50 system, a bit rate reduction technique is now under study and it may be applied directly in order to meet the 140 Mbit/s hierarchical level.

Performance objectives in terms of such parameters as quantization, sub-sampling, bit error, jitter and slips should be studied, based on the subjective assessment of HDTV picture quality.

It is important to note that a digitally sampled representation of HDTV studio signals is being considered in Study Group 11 in relation to Recommendation 601 for existing conventional television systems.

Broadband-ISDNs are being studied as a next generation network by the CCITT. In Document [CCIR, 1986-90f] the following consideration is given. An asynchronous transfer mode (ATM) network has been adopted by CCITT Study Group XVIII as the target solution for the B-ISDN. An ATM network is in principle capable of supporting variable rate video coding, in which video

signals are efficiently transmitted with constant picture quality by varying the coding bit-rate according to the input video characteristics. However, there are many barriers to be overcome in the use of ATM for television transmission. Cell losses, cell delay and cell delay jitter inherent in the ATM put specific requirements on the coding for television transmission. Therefore, studies on HDTV digital transmission, including the examination of the suitability of ATM networks for television transmission, should be conducted while maintaining close contact with CCITT. ATM transmission of video signals is discussed in Report 1240.

3.2 Bit-rate reduction coding of HDTV

A bit-rate reduction codec for HDTV was developed. This can transmit 1125 line/60 Hz HDTV signals at 97.728 Mbit/s, which is the fourth level bit rate in the Japanese digital hierarchy. This codec is also applicable to the H4 channel rate (about 135 Mbit/s) coding for broadband ISDN. This codec employs adaptive prefiltering for noise reduction, extrapolative and interpolative coding with adaptive intrafield/interframe prediction, and variable word length coding [Yashima and Sawada, 1987; Sawada and Yashima, 1988; Yashima and Sawada, 1989; CCIR, 1986-90e].

Aimed at the realization of practical compact codec hardware, an intrafield DPCM codec was also developed, which employs line-offset 1/2 sub-sampling, a noise shaping filter and a folded quantizer. This codec is designed for transmitting 1125 line/60 Hz HDTV signals via INTELSAT satellite at 120 Mbit/s, or via optical fibre cables at 140 Mbit/s (fourth level of the digital hierarchy), including robust doubly encoded Reed-Solomon codes as forward error correction against transmission errors [Matsumoto, S. *et al.*, 1988; CCIR, 1986-90g].

Since digital processing is employed in the MUSE and HE-TCM systems mentioned in section 2, a digital output with 16 MHz sampling and 8 bits/sample can be used, and it could be digitally transmitted at a bit rate of about 135 Mbit/s. The bit rate might be reduced to about 90 Mbit/s if DPCM is applied. A MUSE DPCM codec has been implemented and coding experiments were carried out [CCIR, 1986-90h].

In the above-mentioned coding systems, HDTV signal bandwidths used are about 20 MHz for luminance (Y) and about 7 MHz for colour difference (C). For future very high quality HDTV transmission systems which maintain the original production picture quality as far as possible, an HDTV signal whose Y bandwidth is 30 MHz will be used. Therefore, studies on bit-rate reduction coding schemes and digital transmission systems for such broader bandwidth HDTV signals also should be encouraged [CCIR, 1986-90f].

Document [CCIR, 1986-90i] describes some preliminary results obtained using the DCT algorithm. It is concerned with HDTV signals with parameter values as proposed in Document [CCIR, 1986-90j], employing progressive scanning and resulting in a bit rate of 2304 Mbit/s. It refers to the intensive work carried out during the present study period aimed at the specification of a 34/45 Mbit/s contribution codec for conventional digital signals conforming to Recommendation 601. It states that the promising results obtained have led several organizations to apply such bit rate reduction techniques to HDTV signals, the target being HDTV contribution codecs at 140 Mbit/s. It points out that simple extrapolation of these results is not appropriate since a) a higher compression ratio is required, b) progressive scanning yields a richer information content, and c) because of the high data rate, specific algorithm restraints must be taken into account in order to make the algorithms implementable in compact hardware.

The extra studies required to adapt the conventional codecs to HDTV are currently being addressed by some European cooperative projects. Simulations are being performed based on the DCT coding algorithm proposed by IWP CMTT/2. Whilst some further work has to be carried out, Document [CCIR, 1986-90i] states that extension of the algorithm has yielded promising results for both interlaced and progressive scan HDTV formats. Its conclusion is that 140 Mbit/s can be considered a possible bit rate for the transmission of HDTV in both formats, enabling all the benefits of progressive scanning to be retained in transmission.

Based on the DCT algorithm a prototype of a codec suitable for HDTV is being implemented in Italy and Spain and the development schedule is reported in Document [1986-90k].

Document [CCIR, 1986-90l] describes a source coding algorithm based on DPCM plus noise shaping for HDTV contribution applications using 565 Mbit/s transmission rate. Assuming a sampling frequency of 72 MHz for the luminance and 36 MHz for the two colour-difference signals, a total of 1152 Mbit/s will have to be reduced by about a factor of two; DPCM with 2D-prediction (fixed positive coefficients $1/2$, $1/4$, $1/4$) and fixed quantization (5 bits for luminance, 4 bits for colour-difference) is sufficient for the required high picture quality. For a compact codec realization dedicated 1.5 μ m-CMOS chips are being implemented.

It is necessary to specify a method allowing digital transmission of HD-MAC signals mentioned in Section 2. This subject is studied within the European EUREKA 95 project. The following two coding proposals were realized as prototypes and tested with HD-MAC signals.

The first one is described in Document [CCIR, 1986-90m]. The technique employs a hybrid of 8-bit PCM and 5-bit DPCM with a folded quantizer, to reduce the bit rate of HD-MAC multiplex to the H4 transmission level of about 140 Mbit/s. The method has a simple implementation, is rugged in the presence of transmission errors and can convey HD-MAC signals while still in their scrambled form.

An alternative coding approach is given in [CCIR, 1986-90n]. This approach de-multiplexes the HD-MAC/packet signal into sound and video data and codes the video and sound/data information separately. For the video signal non-adaptive hybrid DPCM, combining Van Bual techniques with a folded quantizer, is used. To reduce further the sensitivity to transmission errors, the 7-bit HDPCM code words are replaced by the 8-bit PCM input code words every ninth sample. The multiplexing of the coded HD-MAC/packet signal is chosen for transmission such that the complete signal will fit into the video channel of the TV container which is described in Document [CCIR, 1986-90o].

First test results in comparing both proposals show that the first one could be used for secondary distribution and the second one for primary and secondary distribution.

4. 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 and Table II.

4.1 Examples of analogue transmission

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].

Several experimental radio transmissions of HDTV using the MUSE signal were carried out. These included three tandem satellite transmission, satellite transmission for live broadcasting, and simultaneous distributive transmission with a communication satellite and terrestrial radio links. From these results, it was confirmed that the analogue radio transmission systems which have the bandwidth required for the MUSE signal are applicable to transmission of HDTV programmes [CCIR, 1986-90p].

MUSE signal analogue transmission was also demonstrated via an optical fibre line. The SWFM-IM (square wave frequency modulation - intensity modulation) technique was employed in this transmission [CCIR, 1986-90p].

4.2 Examples of digital transmission

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

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

97.728 Mbit/s HDTV transmission was demonstrated using a bit-rate reduction codec described in Section 3.2 and optical fibre transmission lines. The HDTV baseband signals used were component signals of Y, P_R and P_B with bandwidths of 20 MHz and 7 MHz, respectively [Yashima and Sawada, 1989; CCIR, 1986-90e].

International field trials at rates of 120 Mbit/s and 140 Mbit/s have been carried out between the United States and Japan through the INTELSAT 72 MHz bandwidth transponder and it was confirmed that using the DPCM codec described in Section 3.2 the system gives good picture quality even at a channel bit error ratio of 10^{-4} [Matsumoto, S. *et al.*, 1989].

Digital HD-MAC signal transmissions were demonstrated through optical fibre. In this demonstration bit-rate reduction codecs described in Section 3.2 were employed [CCIR, 1986-90h].

Table I Examples of HDTV analogue transmission systems

Transmission system	Radio transmission			Optical fibre transmission	
	Terrestrial	Satellite			
Application	Field Pick-Ups	Feeder link Distribution		Distribution	
Transmission signal format	TCI	MUSE	HD-MAC	MUSE	TCM
Baseband signal bandwidth	20 MHz	8 MHz	12 MHz	8 MHz	20 MHz
Bandwidth after decoding	Y :	(1)	(1)	(1)	
	C ₁ :	20 MHz	20 MHz	20 MHz	20 MHz
	C ₂ :	7 MHz	7 MHz	7 MHz	5 MHz
		7 MHz	11.5 MHz	7 MHz	5 MHz
Transmission bandwidth	110 MHz	27 MHz	27 MHz	9 MHz	20 MHz
Modulation	FM	FM	FM	SWFM-IM	Direct-IM
Wave-length/ carrier frequency	38 GHz	(2)	(2)	1.3 μ m	0.89 μ m
		14 GHz	18 GHz		
Transmission distance	32 km	--	--	20 km	2km

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

(2) These figures show the carrier frequencies used in broadcasting satellite feeder links.

Table II Examples of HDTV digital transmission systems

Transmission system	Optical fibre transmission			Satellite transmission
Application	Trunk-line/distribution-line			International transmission
Transmission signal format	TCM	Component Y/P _R /P _B	HD-MAC	Component Y/C _w /C _N
Baseband signal bandwidth	20 MHz	20/7/7 MHz	12 MHz	20/7/5.5 MHz
Bandwidth Y : after C ₁ : decoding C ₂ :	20 MHz 5 MHz 5 MHz	20 MHz 7 MHz 7 MHz	23 MHz ⁽¹⁾ 11.5 MHz 11.5 MHz	20 MHz 7 MHz 5.5 MHz
Transmission bit-rate	397.20 Mbit/s	97.728 Mbit/s	140 Mbit/s 155 Mbit/s	120 Mbit/s 140 Mbit/s
Sampling frequency	44.1 MHz	Y : 44.55 MHz P _R : 14.85 MHz P _B : 14.85 MHz	20.25 MHz	Y : 48.6 MHz C _w : 16.2 MHz C _N : 16.2 MHz
Coding method	PCM 8 bit/sample	Intrafield/interframe adaptive prediction, Variable length coding	Hybrid PCM and DPCM, HDPCM	Intrafield DPCM, 1/2 subsampling
Modulation	PCM-IM	PCM-IM	-	4φ and 8φ PSK
Wave-length / RF bandwidth	1.3 μm	1.3 μm	-	72 MHz

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

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.

KISHIMOTO, R., SAKURAI, N. and ISHIKURA, A. [February, 1987] - Bit-rate reduction in the transmission of high-definition television signals, *SMPTE JOURNAL*, Vol. 96, No.2.

KISHIMOTO, R. and SAWADA, K. [November, 1986] - High-efficiency coding technologies for HDTV signals, Workshop on Coding of HDTV, L'Aquila (Italy).

NINOMIYA, et al. [1987] - Concept of MUSE system and its protocol, NHK Labs., Note, No. 345.

YASHIMA, Y. and SAWADA, K. [June, 1987] - A highly efficient coding method for HDTV signals, *Proceedings of ICC' 1987*, 5.1.

MELWIG, R. [1986] - Après demain, la TVHD, *L'Echo des Recherches*, No. 126.

ARRAGON et al. [November 1986] - HD-MAC, une approche compatible vers la TVHD, Workshop on coding of HDTV, L'Aquila (Italy).

BOYER, R. and MELWIG, R. - Approche évolutive et compatible de la TVHD en Europe, Colloque HDTV '87, Ottawa.

SAWADA, K. and YASHIMA, Y. [March, 1988] - High efficiency coding of HDTV signals at the Broadband ISDN H4 rate, 2nd International Workshop on Signal Processing of HDTV, L'Aquila (Italy).

YASHIMA, Y. and SAWADA, K. [September, 1989] - 100Mb/s coding and transmission of HDTV signals, 3rd International Workshop on HDTV, Turin (Italy).

MATSUMOTO, S. et al. [March, 1988] - 120/140 Mbps intrafield DPCM system for digital transmission of HDTV programs, 2nd International Workshop on Signal Processing of HDTV, L'Aquila (Italy).

MATSUMOTO, S. et al., [August-September, 1989] - 120/140 Mbit/s compact HDTV codec and its performance, 3rd International Workshop on HDTV, Turin (Italy).

CCIR Documents

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

[1986-90]: a. CMTT/40 (Japan); b. CMTT/35 (11/62, 10-11S/28) (Japan); c. CMTT/52 (France); d. 11/82 (United Kingdom); e. CMTT/165 (11/327) (Japan); f. CMTT/168 (Japan); g. CMTT/166 (11/328) (Japan); h. 11/579 (Japan); i. IWP CMTT/2-91 (Thomson-CSF); j. IWP 11/7-219 (Belgium et al.); k. IWP CMTT/2-102 (Italy-Spain); l. IWP CMTT/2-94 (Germany (Federal Republic of)); m. IWP CMTT/2-70 (United Kingdom); n. IWP CMTT/2-89(Rev.) (Germany (Federal Republic of)); o. IWP CMTT/2-90 (Germany (Federal Republic of)); p. CMTT/169 (Japan).

BIBLIOGRAPHY

KISHIMOTO, R. et al. [June, 1989] - Generation interval distribution characteristics of packetized variable rate video coding data streams in an ATM network, *IEEE Journal of selected areas in communications* Vol. 7, No 5.

TURNER, J.S. [March, 1986] - New directions in communications, *Proc. Int. Seminar on Digital Comm.*, Zurich.

ANNEX

This figure shows possible examples of formats for analogue component and composite signals of HDTV. It also shows estimated bit rates for the digital transmission of HDTV signals. All figures have been derived for an HDTV system proposed in Study Group 11, having, 1250-line, 60-field/second, 2-to-1 interlace and 16-to-9 aspect ratio [CCIR, 1986-90a, b].

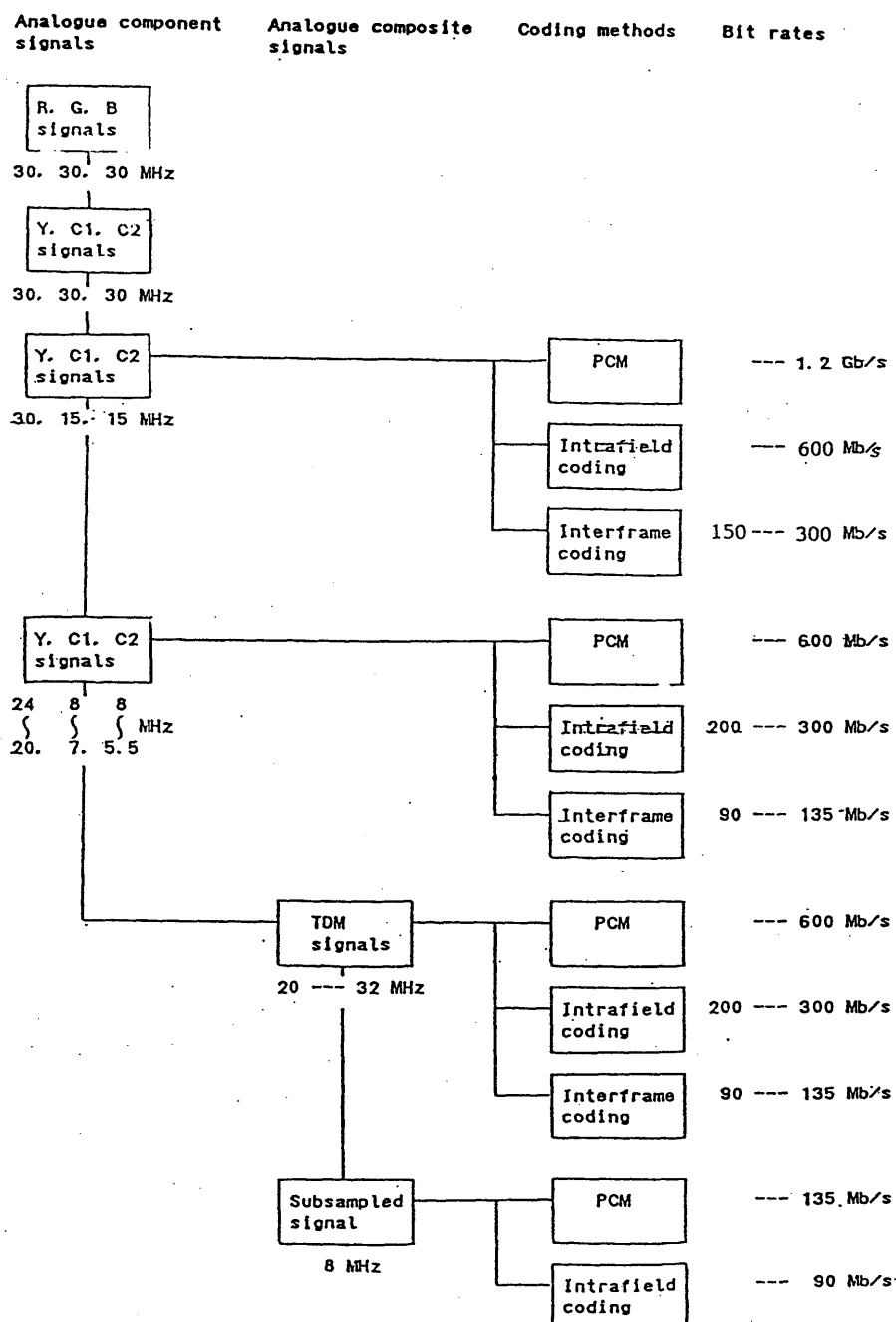


Figure 1 Signal formats and coding methods for the 1125/60 HDTV system

CCIR Documents

[1986-90]: a. CMTT/168 (Japan); b. CMTT/170 (Japan).

REPORT 646-4*

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

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

(1974-1978-1982-1986-1990)

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. The coding of colour television signals in component form (Recommendation 601) has important implications for the CMTT which include the requirement for more than one transmission standard. 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.

Recommendation 604 has set out the general principles for all-digital transmission of television signals. It has been stressed in this Recommendation 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 and that component coding should be used.

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. Information given in Report 1235 — on 30-34 and 45 Mbit/s systems and in Report 1234 — on 68 and 140 Mbit/s systems should be considered as a possible basis for these future new Recommendations.

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

Recommendation 658 for mixed analogue-and-digital transmission of analogue composite television signals has been approved. It 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. An annex to this Recommendation gives a specification for a digital section of a mixed link. Adherence to the parameter values in this annex will ensure that mixed links respect the principle adopted in Decision 18 that systems employing digital or mixed digital and analogue methods should have overall subjective performances not worse than the equivalent analogue systems. Most administrations consider that, although necessary at present, the need for this new Recommendation will diminish as time goes on.

The discussions leading to this Recommendation 658 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.

Very rapid developments in the past few years have directed attention to the need for a group of Recommendations to deal with the transmission of the following types of signals:

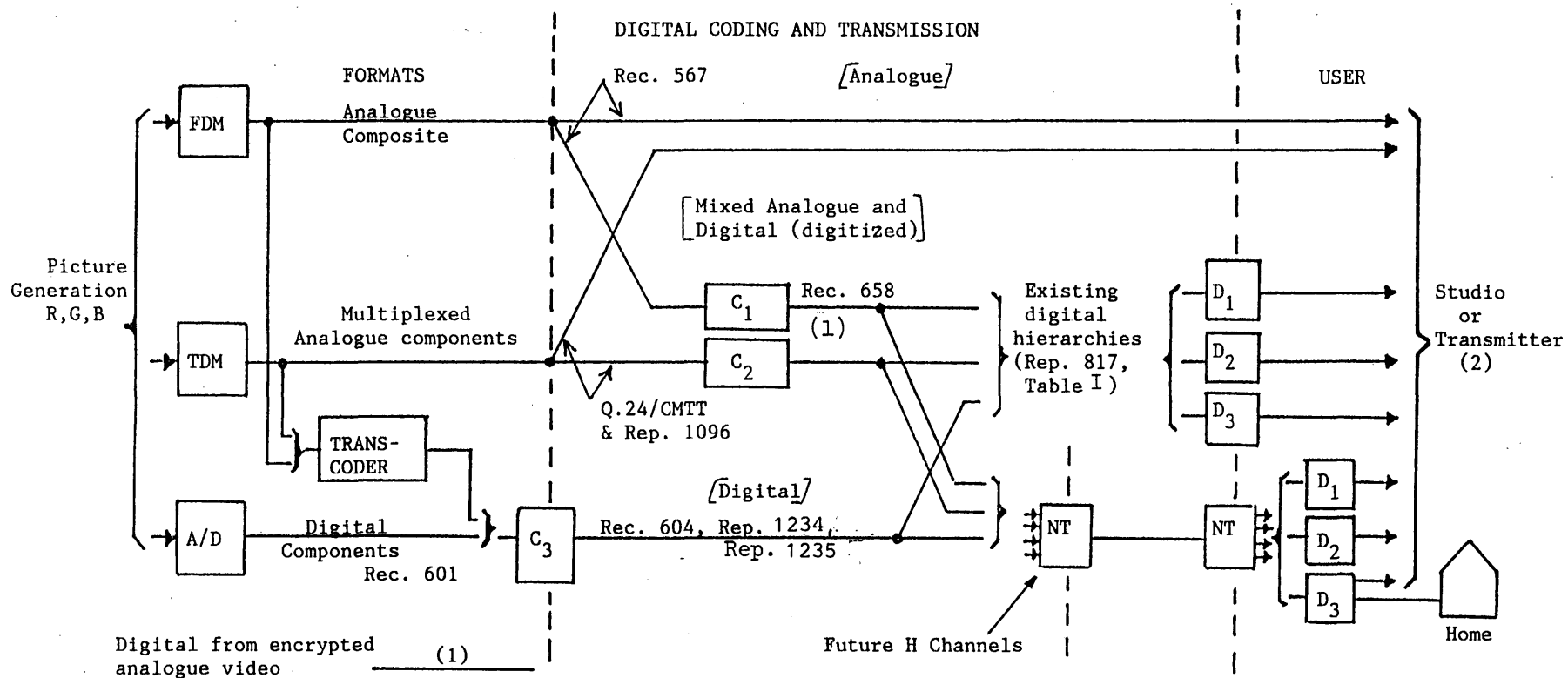
- analogue composite signals;
- multiplexed analogue component signals;
- digital component signals;
- encrypted analogue signals;
- encrypted digital signals.

[CCIR, 1986-90a] discusses video transmission coding options, which are summarized diagrammatically in Fig. 1.

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 encryption methods, which may comprehensively modify the statistics of the analogue signal prior to digital coding, may influence bit rate reduction algorithms relying on the exploitation of signal statistics.



NT = Transparent Network Termination

C = Digital Coder) Suffix 1 for analogue composite

D = Digital Decoder) 2 for multiplexed analogue components

3 for digital components (with bit rate reduction, if applied)

Note 1 - Anticipated Recommendations identified in Report 646

Note 2 - Input to studios and transmitters may be in analogue composite, multiplexed analogue components or digital form

FIGURE 1 - Video coding and transmission options

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	Some extra details could be added to Annex 1
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. 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.

Information on users' requirements for the signals to be transmitted is given in Report 1211.

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 Encryption data

Future transmission systems are likely in some cases to require the signals to be conveyed in scrambled form, and accompanied by an encrypted stream of data that will enable authorized users to perform the appropriate descrambling process. [CCIR, 1986-90b] presents some preliminary considerations

of these matters. Various applications are considered and it is concluded that all of them should be met by providing a signalling data capacity of about 6 kbit/s. The document also considers the effect of video signal scrambling on the processes involved in coding and multiplexing the signal. Scrambling and bit rate reduction are conflicting processes since scrambling seeks to randomize the signal whereas bit rate reduction processes attempt to exploit correlation. The conclusion is that scrambling cannot be added as an afterthought, but must be considered at the design stage. Care is necessary in deciding where in the signal chain scrambling should be carried out.

[CCIR, 1986-90c] adds the following additional factors to be taken into account:

- the scrambling of the sound and, possibly, other signals, should be independent of the scrambling of the vision signal (but the algorithms and key arrangements could be the same). This implies that some extra bits must be available for managing the scrambling of these signals. It also implies that it may be better to scramble the sound, etc., before multiplexing it with the vision signal.
- Cell loss in an ATM system may have a greater effect on a scrambled signal than on an unscrambled signal.
- Further study is required to find the possible repercussions of cascading systems, especially those with different bit-rates (for example, 140 Mbit/s and 34 Mbit/s), in each of which the signals are scrambled.
- Study is also required to find the effects of cascading scrambling for encryption and scrambling for transmission purposes.

2.6 *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, 1978-82a] 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, 1978-82b] 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, 1978-82c] 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, 1982-86b] 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 that takes account of the different digital transmission hierarchies or H channel rates and interfaces for ISDN recommended by the CCITT. However, consideration of the future needs of broadcasters should not be constrained to the use of only H channel rates based on existing hierarchical levels.

Due to economic factors (e.g., reduced cost of circuits), special circumstances (e.g., multiple commentaries or languages associated with a single event) or new services (e.g., high-definition television) broadcasters might have a requirement for a network termination which can support data rates in excess of a single H channel.

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, 1978-82d, 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 currently lies in coding methods which will lead to transmission bit-rates which can be accommodated within the existing 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.

The coding standards adopted for studio applications is described in Recommendation 601. The main features of this standard (usually known as "4:2:2" standard) include component coding, a spatially static sampling pattern having 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, 1978-82d].

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 1978-82 i], 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, NTSC or SECAM signals usually involves:

- sampling the signal at a frequency above the Nyquist limit. This frequency may be an integer multiple of the colour sub-carrier frequency (f_{sc}), typically $3 f_{sc}$ or $4 f_{sc}$, or of the line frequency or it may be synchronous with the transmission frame clock frequency;
- quantizing the signal into PCM form with 8 or 9 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 1978-82j]. 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, 1978-82d], 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. [Lange, 1987] gives the results of tests on six cascaded codecs working with 9-bit quantization. The measurements were done with PAL, SECAM and NTSC signals. These codecs have been used since 1985 in the television networks of the Federal Republic of Germany.

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.

Another practical system, designed for the transmission of any type of composite video signal on 139.264 Mbit/s digital paths, is described in [CCIR, 1986-90d]. The system uses a 17 MHz sampling frequency synchronous with the transmission frame clock frequency, and linear quantizing with 8 bits per sample. It is indicated that the specifications of Recommendation 567 are still satisfied when three such codecs are used in tandem, and that with line error rates better than 1×10^{-7} no impairment is perceptible.

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, 1978-82k] 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, 1978-82k].

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, 1978-82] 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/bits intrafield DPCM coding system for electronic news-gathering applications is described in [Rossi, 1980].

It is claimed in [CCIR, 1978-82m] 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, 1978-82n] 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 $2f_{sc}$ sampling frequency, sub-sampled from $4f_{sc}$ 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 $3f_{sc}$ 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, 1978-82c] 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, 1978-82p]. 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* convertors are unrelated to the choice of sampling-frequency (provided that this is at the Nyquist criterion or above), whether or not the latter is simply related to the sub-carrier frequency.

4.3 *"Digital-to-digital" conversion methods*

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

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

Several digital-to-digital converters for System M/NTSC are outlined in [CCIR, 1974-78k], which draws attention to the fact that, using only a modest amount of digital circuitry, conversion of the sampling rate of $3 f_{sc}$ to $4 f_{sc}$ is possible. It is further claimed that a completely reversible $3 f_{sc} - 4 f_{sc} - 3 f_{sc}$ 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.

A transitional period will occur during which analogue composite television signals will be transmitted through a circuit comprising analogue and digital sections in tandem. It should be noted that:

- a) the mixed analogue-digital form is only a transitional one, and
- b) establishing a unique system of quality specification for the mixed form at this time may prove an obstacle in establishing a similar system for the all-digital form.

For the transitional period, it is preferred to retain specifications for the all-analogue form as long as possible, and not to introduce new specifications for only the mixed form.

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.

[CCIR, 1986-90e] recognizes that conventional test signals are inadequate for testing codec performance, particularly when bit-rate reduction is used. Therefore, six test signals are described which have been used for assessment of impairments introduced by the digital parts of System M-NTSC codecs.

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, 1978-82q] 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, 1978-82r] 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.

5.4 Distortions in mixed analogue and digital transmission

Up to the present, picture quality specification problems have been considered individually for analogue or digital systems. Taking into account the psychological independence of picture quality degradation phenomena observed on a TV monitor, this idea is also applicable to mixed systems. Namely, they can be classified into one of the following three groups from the viewpoint of psychological independence:

- (1) impairments caused only by the analogue section,
- (2) impairments caused only by the digital section,
- (3) impairments caused by both analogue and digital sections (which might be independent factors in each individual system).

Impairments belonging to group (1) or (2) will be dealt with as independent factors. A method for estimating the overall picture quality in this case is given in Report 1205. The estimation method is applicable when certain mutually independent psychological factors exist in the same instance.

On the other hand, in group (3), where picture quality degradation phenomena from both sections are so similar that they cannot be regarded as independent, it is necessary to find a new way to allocate picture quality degradation to both the analogue and digital sections before applying the above mentioned estimation equation.

An example of the investigation results in such a case is reported in [Inoue, 1987]. In the reference, a combination of random noise from the analogue system and granular noise from the intraframe DPCM system was investigated to show that it is possible to replace a physical measure in the analogue system with a corrected value based on visual sensitivity differences.

6. Mixed analogue and digital circuits

6.1 Hypothetical reference circuit (HRC)

Since mixed analogue-and-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. The hypothetical reference circuit defined in Recommendation 567 is divided into three equal sections by intermediate video points. Each of these sections could be replaced by a digital section; therefore, the HRC for a mixed link could contain up to three digital sections. Since the distortion introduced by a digital section (apart from transmission errors) is due entirely to the ADC and DAC, it follows that any number of digitally-connected links of any length may be considered as making up one such digital section. Similarly, a digital section whose performance is equivalent to one-third of the HRC may have a worse performance than that of a shorter analogue section.

6.2 Requirements for mixed circuits

Analogue vision circuits are used to carry a wide variety of signal formats, for example:

- 625-line PAL)
- 625-line SECAM) in clear or
- 525-line NTSC) scrambled form and possibly with teletext signals
- MAC signals)
- Test signals, either alone or as insertion test signals on any of the above video formats
- Digital sound signals as sound-in-synchs
- Digital sound signals in a continuous stream or PCM-F1 format.

It is important for operational flexibility that mixed circuits should be able to carry this variety of signal formats without restriction. In practice this will mean that the digital sections of mixed circuits must use linear coding without bitrate reduction, since bitrate reduction algorithms rely on redundancy in the video signals being coded and are therefore specific to particular video formats. Scrambled video signals, whose redundancy has been removed by the scrambling process, and signals whose structure is essentially random, such as teletext or digital sound signals, will be seriously impaired by any video bitrate reduction system.

Another problem arising from the variety of video formats in use is that of clamping. Analogue video circuits rarely contain clamps, whereas in mixed circuits there will be a clamp in every ADC. These clamps will have to be carefully specified to avoid the build-up of streaky noise that would otherwise occur. Clamps will have to detect the different video formats listed above and clamp accordingly; signals without blanking intervals (such as continuous digital sound bitstreams) must not be clamped, but must be coupled to the ADC in such a way that their mean level is close to the centre of the conversion range.

New forms of impairment will arise on mixed circuits; furthermore, measurement methods and test signals specified for all-analogue circuits may not be suitable for testing digital sections. These subjects are discussed in §§ 4.2.1 and 5.

REFERENCES

- BOYER, R. [1978] Un système de transmission numérique. Deuxième partie: Réalisation de codeurs et décodeurs MCD pour un système à 52 Mb/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 TSUKKERMANN, 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 farbfernsehsignal (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.
- INOUE, M. [May 1987] - The proposed method for noise specifications to mixed analogue-digital video transmission systems, J.ITEJ, Volume 41, No. 5 (in Japanese).
- 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.
- LANGE, H. [1987] - Digital TV signal transmission over optical fibre cables in local broadcast and CATV trunk networks. 15th International TV Symposium, Montreux/CATV Sessions, 316-333. Also published in Cable Television Engineering, Vol. 14, No. 3, June 1988, 537-543.

- 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. 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 sredsiv 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. 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).
- [1978-82]: a. CMTT/267 (France); b. CMTT/52 (Japan); c. CMTT/85 (Germany (Federal Republic of)); d. CMTT/18 (United Kingdom); e. CMTT/86 (Germany (Federal Republic of)); f. CMTT/286 (EBU); g. CMTT/13 (EBU); h. CMTT/296 (Canada); i. CMTT/8 (EBU); j. CMTT/59 (Japan); k. CMTT/230 (Japan); l. CMTT/45 (Japan); m. CMTT/47 (Japan); n. CMTT/229 (Japan); o. CMTT/261 (Germany (Federal Republic of)); p. CMTT/31 (United Kingdom); q. CMTT/236 (Japan); r. 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).
- [1986-90]: a. CMTT/68 (Australia); b. CMTT/47 (IWP CMTT/2); c. CMTT/191 (EBU); d. CMTT/53 (France); e. CMTT/88 (United States).

BIBLIOGRAPHY

- GUIONNET, J. and SALLIO, P. [1978] Système expérimental de télévision numérique à 52 Meb/s. Évaluation subjective de l'effet des erreurs de transmission. *Radiodif.-Télév.*, Vol. 4/5, 54, 1-7.
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REPORT 1235*

DIGITAL TRANSMISSION OF COMPONENT-CODED TELEVISION
SIGNALS AT 30-34 MBIT/S AND 45 MBIT/S*

(1990)

1. Introduction

This report concerns systems to be used for 30-34 Mbit/s and 45 Mbit/s transmission of signals conforming to CCIR Recommendation 601. This report is based on the studies of CCIR IWP CMTT/2 and CMTT.

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

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

A low bit rate system in the region 30-45 Mbit/s would be expected to be suited to distribution applications and may be suited to some contribution applications.

2. Digital bit rate for transmission

In Recommendation I.121, the CCITT has specified the H21 channel rate to be 32768 kbit/s. The format will also be relevant for the 34 Mbit/s level of the hierarchy in CCITT Recommendation G.702.

The H22 channel rate has not yet been precisely specified; it will be in the approximate range of 43-45 Mbit/s. The CMTT recognizes that the system specification for this channel should also be suitable for operation at 44736 kbit/s for interworking in accordance with CCITT Recommendation G.802.

(Note: Some parts of this Report could not be accepted by some Administrations. The revisions proposed by those Administrations (which were not accepted by other Administrations) are given in Annex IV.)

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

3. User requirements for H2 channels

3.1 CCIR IWP 11/7 has proposed preliminary user requirements for low bit rate systems in the region 30-45 Mbit/s [CCIR 1986-90a]. These are as follows:

i) Basic quality requirements - quality, as measured using the double stimulus continuous quality scale method, to be not more than 12% less than that of the 4:2:2 reference. For contribution, quality is to be tested if possible using two codecs in tandem interfaced at the 4:2:2 level. For distribution, quality is to be tested using one codec only.

ii) Downstream processed quality requirements - quality, as measured using the double stimulus continuous quality scale method, to be not more than 18% less than that of equivalently processed 4:2:2 material. The downstream processing operations to be considered are: colour-matte, slow motion, and manipulation of picture geometry.

iii) Failure characteristics requirements - for error performance, impairment not exceeding one grade, as measured using the double stimulus impairment scale method, for random errors at 1×10^{-4} or for burst errors of as many as 30 consecutive bits. For recovery following a 50 ms break in data, recovery time should not exceed 160 ms.

iv) Synchronization and delay requirements - the synchronization behaviour of the decoder in the presence of errors must be defined. Errors affect the picture impairment directly but also affect the synchronization of the decoder and, thus, may cause picture impairments additional to those assessed above. The decoded output signal must be synchronized to its associated studio centre and this means that errors or momentary losses of input signal must not cause disturbances to this synchronization.

The difference in the delay through the decoder before and after such disturbances must not exceed an amount of time which should be small. This amount is for further study but a value of ± 20 microseconds is suggested. Document [CCIR 1986-90b] draws attention to this requirement; furthermore it suggests that the decoder buffer store could also provide a basis for synchronizing the decoder with the studio centre, thus avoiding the need for additional synchronizing equipment. The latter suggestion is a matter for further study.

The question of quantifying an acceptable level of codec complexity is quite difficult, but it is expected that a 30 Mbit/s codec could be several times more complex than a 140 Mbit/s codec and still be acceptable because of the saving in link bit rate.

CCIR IWP 11/7 has suggested certain principles which may be useful for system selection at 34 Mbit/s, and these are given in [§ 8 of Report 1211].

The reference system which is outlined in section 4.1 of this report is based on 4:2:2 - 3:1:1 conversion at the codec input. Sub-sampling and its effects on quality have been the subject of a number of contributions to IWP CMTT/2 and [CCIR 1986-90c].

Some administrations have the opinion that when sampling frequencies smaller than those of the 4:2:2 signals are used, as is the case for 3:1:1 systems, the horizontal resolution of the pictures is inevitably reduced, giving an initial and irreversible reduction of the available quality of the pictures to be then coded [CCIR 1986-90d], and they therefore believe that standardization should await more complete studies of alternative systems. A number of other administrations believe that a 3:1:1 system would lead to an optimum overall quality.

The EBU opinion is that codecs should be defined more precisely to indicate not only the interface standard but also the essential features of the internal standards. As an example, the codec given in [CCIR 1986-90e] should be labelled as a 3:1:1 codec with a 4:2:2 interface. Having made this point however, the EBU wishes to reaffirm its view that the latter codec is a well designed 3:1:1 codec.

The EBU believes that IWP CMTT/2 should encourage the development of a standard for 4:2:2 codec with a 4:2:2 interface within the present study period of the CCIR. If this proves not to be possible, then it could be useful to define a separate standard, for example a "3:1:1 codec with 4:2:2 interface" for use by those who have a need for such codecs now, but without prejudice to the definition of a full 4:2:2 codec.

The EBU thus believes that IWP CMTT/2 should encourage the study of a range of techniques at least until one technique can be shown to approach closely the full 4:2:2 standard performance [CCIR 1986-90f].

In [CCIR 1986-90g] the EBU develops further the performance objectives given in section 3 above. The main points suggested by the EBU are as follows:

- a) a preference for 1536 kbit/s for audio and 384 kbit/s for auxiliary data;
- b) a relative sound-vision delay of no more than ± 2 ms per codec;
- c) a basic quality loss equivalent to 0.5 grade for one or two codecs in tandem;
- d) a quality loss due to random bit errors no more than the equivalent of 1 grade relative to 4:2:2 signals at BER $10 \exp^{-4}$ with up to 30-bit burst length;
- e) a system recovery time of less than 160 ms;
- f) a quality difference after processing (colour-matte, $\pm 15\%$ expansion and compression, 10:1 slow motion) of less than the equivalent of 0.75 grade compared to 4:2:2.

The above criteria were developed in the light of the assessment methods which will be used by IWP 11/7.

The OIRT has made similar proposals for the objectives a), b), c) and f) above [CCIR 1986-90h].

In [CCIR 1986-90i] the RTVE also proposes that 384 kbit/s should be allocated to auxiliary data; including provision for error correction, and that the residual error rate for this auxiliary data should be arranged (by FEC) to be always better than $10 \exp -5$. The auxiliary data could include teletext data and ancillary data.

In [CCIR 1986-90j] the Chairman of JIWP 10-11/5 proposes that an auxiliary data allocation comparable with the above should be made, and also stresses the importance of adequate FEC for the data signals.

3.2 User requirements for non-video signals

3.2.1 Sound

For purposes of programme distribution, it would be reasonable to allow for a transmission capacity of 960 kbit/s for sound (corresponding to one high-quality stereo pair) [CCIR 1986-90a].

Considering that other standards may produce a higher bit rate (for instance contribution links with a 48 kHz sampling frequency) and that sound channels may be delivered by other equipments, the codec should offer the possibility to transmit a plesiochronous 2 Mbit/s sound channel [CCIR 1986-90k].

Document [CCIR 1986-90l] requires that, in addition to the video, the codecs should offer a 2048 kbit/s transparent channel. For distribution applications, this will allow the codecs to be configured to meet different demands. 2048 kbit/s is enough for digital stereo sound with distribution quality (NICAM 728), the accompanying mono sound, teletext signals according to CCIR and also address data for conditional access. For international transmission, the sound codec proposed by IWP CMTT/4 should be used. This will allow transmission of studio quality sound according to CCIR Recommendation 647. It also offers two 48 kbit/s channels and two strongly protected 1 kbit/s channels for user data.

3.2.2 Teletext data

Considering the existing teletext systems, the maximum capacity required corresponds to a gross bit rate of 378 kbit/s. A 384 kbit/s channel for teletext transmission would seem to meet the requirements [CCIR 1986-90k].

3.2.3 Ancillary data

Provision should be made to carry ancillary data, pending clarification of user needs. IWP 11/7 is studying this Question.

3.3 Additional factors which may influence the selection of codecs

3.3.1 Intellectual property rights (IPR)

It is essential to ensure that licences for any patents, or other IPR, embodied in equipment that would have to be used in order to conform to a proposed Recommendation on BRR codec characteristics, will be available under reasonable conditions.

Owners of such patents should submit a statement to this effect, a standard form for which is attached as Annex 3 to this Document. This standard form has been prepared in the CCIR Secretariat, based on CCITT experience and taking into account legal advice from experts in the World Intellectual Property Organization (WIPO), and from the Legal Adviser of the ITU.

3.3.2 Overall operational costs

These costs cover purchase, installation, operation, maintenance and the cost of transmission. Intellectual property rights might also involve a significant element. All these costs should of course be minimized.

Transmission costs need to be included because it may, for example, be preferable to operate at higher bit rates for some applications where the transmission costs would be offset by the lower cost of the codecs, using simpler BRR algorithms.

How the relative costs of the various contending methods would be compared is a question which remains for study.

3.3.3 Complexity

Degrees of complexity would generally be reflected in codec costs. However, when comparing codecs, other aspects being equal, the less complex codec would be preferred because in the case of a relatively complex codec there might be doubts as to reliability and to ease of maintenance in service. These are characteristics otherwise generally difficult to quantify in the absence of data accumulated in operational service.

3.3.4 System extensibility

There will be many applications requiring bit rate reduction techniques at different channel bit rates such as conventional TV and HDTV contribution circuits, primary and, later on, secondary distribution, digital video tape recording. It would be advantageous to have a common basic algorithm set to realize such codecs if it results in a decrease in manufacturing costs of such equipment. Document [CCIR 1986-90m] suggests that distribution codecs can be implemented by simplifying the contribution codec or running it at a different bit rate. Document [CCIR 1986-90n] presents preliminary subjective assessments of the same algorithm simulated by computer at different bit rates.

3.3.5 Compatibility

It is desirable for BRR methods proposed for different applications and for 625/50 and 525/60 systems to be compatible so as to facilitate conversion between the signals, especially for the international exchange of programmes.

3.3.6 Future developments

As well as the 4:4:4 level and a proposed level for ENG, there are new areas of application to be studied, including enhanced television, high definition television and, tentatively, digital television broadcasting by cable or optical fibre networks. BRR methods which have characteristics adaptable in respect of performance, costs, compatibility, etc., to these new developments are to be preferred.

4. General specifications of the systems

This section contains brief specifications of coding systems studied by IWP CMTT/2. Further details of these may be found in the appendices to this report.

For a codec specification to form part of a proposed draft Recommendation it must contain sufficient detail that the minimum performance objectives defined by the user requirements can be guaranteed at the decoder output. A consequence of this is that not only must the transmission format be defined, but also that any parts of the encoder and decoder processing which influence decoded quality must be adequately specified and toleranced. Within these constraints it is important to retain the flexibility for future codec improvement while avoiding the need for formal CCIR assessment of these equipments. These issues are discussed in [CCIR 1986-90o].

4.1 Predictive coding system

The IWP CMTT/2 Expert Group first defined a specification for a 3:1:1 codec, with 4:2:2 interfaces, for distribution applications at transmission rates of 30-34 Mbit/s. It combined the best features of the practical systems described in [CCIR 1986-90e and p], and some improvements based on other contributions were added.

The Expert Group believed at that time that this codec would be the best compromise likely to achieve the quality objectives stated in [CCIR 1986-90q].

Subsequent subjective tests showed that 4:2:2 sampling would give somewhat better picture-quality. Moreover, following the meeting of IWP CMTT/2, together with IWP 11/7, in May 1987, it was the general opinion that a 4:2:2 codec would be more suitable as an international standard for systems based on CCIR Recommendation 601.

The practical problems of implementing a 3:1:1 codec and a 4:2:2 codec were therefore reappraised and compared. It was concluded that a 4:2:2 codec would need a more sophisticated and complex coding algorithm and would operate at a higher clock frequency. On the other hand, however, the complex digital filters from 4:2:2 to 3:1:1 and vice versa would not be needed. Taking account of the rapid advances in technology, IWP CMTT/2 is confident that, by the time the standardization process has been completed the difference in the practical difficulties of implementation between the two types of codec would not be significant.

Major changes compared to the 3:1:1 predictive coding system are as follows:

- a) no sub-sampling is used. All samples in the active picture area, corresponding to a 4:2:2 sampling structure, are coded;
- b) motion compensation is improved;
- c) conditional replenishment for Y component is applied.

In addition, further considerations from viewpoints of specification enhancements, hardware implementation and hardware simplification have been done in the following points [CCIR, 1986-90r]:

- a) clarification and precise definition of the fundamental parameters of the motion compensation function;
- b) a change in the motion vector coding technique from differential vector to absolute vector processing;
- c) revisions of the predictive technique and quantizer for the fallback mode;
- d) to improve variable length coding performance, the minimum word length of binary code has been changed from two bits to one bit.

Based on these considerations, the specification of the predictive coding system has been completed. The outline is described in [CCIR 1986-90s].

Through discussion to complete the specification, it is confirmed that proposals for 45 Mbps systems complying with CCITT Recommendation G.802 and suitable for the future H22 channel not yet specified by CCITT Study Group XVIII and preferably compatible with the 30-34 Mbps system are requested.

Hardware realization

Based on the enhanced specifications, hardware implementation was carried out by two groups, one in Europe and one in Japan.

The parameters used are given below:

1. European group (625-line system at 34 Mbps) prototype scheduled be completed by the end of February 1989;
2. Japan (525-line system at both 32 Mbps and 45 Mbps and a 625-line system at 34 Mbps) prototype scheduled be completed by the end of February of 1989.

After the completion of these prototype systems, a compatibility test between the two under 625-line at 34 Mbps conditions was planned.

4.2 Hybrid predictive/transform coding system

It is now generally agreed that the future reference system will need 4:2:2 sampling pattern to be able to cover both distribution and contribution applications and provide maximum transparency to CCIR Recommendation 601.

The DCT Expert Group believes that this goal may be achieved by applying DCT source coding techniques [CCIR, 1986-90d, f, t, and u].

This system is described in [CCIR 1986-90s].

It is based on a hybrid predictive/transform coding scheme involving intra-field, inter-field and motion compensated inter-frame modes with use of adaptive quantization of transform (DCT) coefficients and VLC coding. Two options for VLC are described in Appendix II of [CCIR 1986-90s]. Only option a (Annexes 4-a and 7-a of [CCIR 1986-90s]) was tested.

For option b, a new version of video framing was proposed in Document [CCIR 1986-90v] during the June-July 1989 IWP CMTT/2 meeting, but has not yet been studied by the DCT Expert Group. It is described in [CCIR, 1986-90s].

TABLE I
Specification for predictive coding system

Video input/output	Standard	525-line or 625-line digital video in component form.
	Coding	4:2:2 level of Rec. 601
	Interface	Bit-parallel or bit-serial in accord with Rec. 656
Signal preprocessing	Horizontal	Full digital active line of 720 samples luminance (Y) and 360 samples each of colour diff. (C _B , C _R)
	Vertical	<u>525</u> 248 lines per field (Note1) Field 1 - Lines 16 to 263 Field 2 - Lines 278 to 525 <u>625</u> 288 lines per field Field 1 - Lines 23 to 310 Field 2 - Lines 336 to 623
	Sub-sampling	None
Coding algorithm	Motion compensation	Blockwise Motion Compensation. (see Annex 1) (Appendix I in [CCIR, 1986-90s])
	Prediction	Three predictors, intrafield and interfield and motion-compensated interframe predictors, are used. (see Annex 2) (Appendix I in [CCIR, 1986-90s])
	Adaptive control of predictors	The predictor selected is signalled by side information and is based on a one dimensional block consisting of 12 luminance samples and 6 samples of each colour difference signal.
	Conditional replenishment	Conditional replenishment for 12 Y pel blocks are used. (see Annex 4) (Appendix I in [CCIR, 1986-90s])
	Quantizer	Eight quantizers for each component are available. (see Annex 3) (Appendix I in [CCIR, 1986-90s])
	Adaptive control of quantizers	One of the eight quantizers is selected according to the buffer memory occupancy. Quantizer selection is made every eight lines.
	Variable length coding	Variable length code for coded video signals (word length: 1 - 12 bits) (see Annex 4) (Appendix I in [CCIR 1986-90s])
Sound		The bit rate for sound transmission will vary from zero to 1536 Kbit/s. This corresponds to up to 4 sound channels, each coded with 384 kbit/s (eg. acc. to Rec. 560). The number of sound channels is signaled to the receiver (see Channel Framing).
Video data error protection		(1) <u>FEC</u> : The (239, 255) BCH double error correction code (see Annex 6) (Appendix I in [CCIR, 1986-90s]) (2) <u>Periodic refreshing</u> : Periodic refresh with intrafield predictor need be used for one field period only. During the next field interfield predictor is also allowed.
Buffer memory capacity		1.572864 Mbit
Video framing		This comprises sync code bits, side information for coding control, the coded video signals and associated ancillary data. (see Annex 5) (Appendix I in [CCIR, 1986-90s])
Channel Framing		Frame structure are proposed for 31488, 34368 and 32064 Kbit/s channels. See Annex 7 in [CCIR 1986-90s] for details.

Note 1: Only 244 lines per field are processed in the decoder.

TABLE II

Specification of hybrid predicting
transform coding system

Video input/output	Standard	525-lines or 625-lines digital video in component form.
	Coding	4:2:2 level of Rec 601.
	interface	Bit-parallel or bit-serial in accord with Rec 656.
Signal preprocessing	Horizontal	Full digital active line of 720 samples for luminance (Y) and 360 samples for each colour difference (Cr,Cb).
	Vertical	525 lines : 248 lines per field Field 1 : lines 16 to 263 Field 2 : lines 278 to 525 625 lines : 288 lines per field Field 1 : lines 23 to 310 Field 2 : lines 336 to 623
	Subsampling	none
Coding	Modes	3 modes (intra-field, inter-field and motion compensated inter-frame) are used. The 3 following processings are applied either on 8x8 intra-field blocks (intra field mode) or on differential blocks obtained by difference between the current 8x8 intra-field block and a reference block taken in the previous field (inter-field mode) or in the field with same parity in the previous frame (inter-frame mode). (See Annex 2) (Appendix II in [CCIR, 1986-90s])
	DCT	Discrete Cosine Transform applied on rectangular blocks of 8 lines of 8 samples for the three components Y, Cr, Cb. (See Annex 1) (Appendix II in [CCIR, 1986-90s])
	Quantization	A different quantization characteristic is used for each coefficient. Its parameters are adapted to the buffer occupancy, the criticality of the block and the type of the block (luminance/chrominance). The shape of the characteristic (non-linear/quasi-linear) can depend on the block prediction mode (intra/inter). (See Annex 3) (Appendix II in [CCIR, 1986-90s])



	Variable length Coding	VLC's are used to encode the non-zero quantized DCT coefficients and run-lengths of zeros according to preset scanning patterns. The VLC's used depend on the prediction mode and on the component to be coded. (See Annexes 4-a and 4-b) (Appendix II in [CCIR, 1986-90s])
	Prediction of the block	For each block processed according to inter field mode, the reference block is determined with pixels of the previous field with an interpolation processing which is described in Annexe 5. For each block processed according to inter-frame mode, the reference block is taken in the previous frame. Its position is derived from the position of the current block by application of a displacement vector. (See Annex 5) (Appendix II in [CCIR, 1986-90s])
	Motion Compensation	A semi-global motion compensation (9 vectors per field) is used. Each quad-block (two adjacent 8x8 blocks of Y and the two copositioned Cr and Cb blocks) may also be assigned a single vector belonging to a set of 31 vectors, (palette mode). (See Annex 6) (Appendix II in [CCIR, 1986-90s])
Buffer memory capacity		1 Mbit
Video Framing		It includes synchronization words and coded video signals with specific error protection for some important parameters. (See Annexes 7-a and 7-b) (Appendix II in [CCIR 1986-90s])
Video data error protection		-A specific protection is used for some coded parameters (parity bit or majority logic : see Annexe 7-a and 7-b) (Appendix II in [CCIR, 1986-90s]) -Forward error correction is globally applied on video data (See Annex 8) (Appendix II in [CCIR, 1986-90s])
Channel Framing		Frame structures are proposed for H2 channels and G702 third hierarchical levels (See Annex 9) (Appendix II in [CCIR, 1986-90s])

4.3 Items requiring further study

IWP CMTT/2 has identified the following items that will require definition and clarification before any reference system can be completely finalized.

4.3.1 Sound

Provision is made to include sound data in the channel multiplex.

In the predictive coding system, it is assumed to be in accordance with CCIR Recommendation 660 which defines a rate of 384 kbit/s for each sound channel inclusive of FEC coding (if used). The system can accommodate up to four such sound channels, with the precise number user defined, for TV programme exchange at a bit rate up to 32 Mbit/s. In the 34 Mbit/s hierarchy the number of sound channels can be increased by using the existing 2048 kbit/sec transparent channel.

In the hybrid predictive/transform system, a 1920 kbit/s bit rate or a 2048 kbit/s channel has been reserved for sound data. This will allow to accommodate either up to 5 sound channels in accordance with CCIR Recommendation 660, or a digital contribution-quality stereo sound programme which is presently studied by CMTT.

The specific needs of broadcasters and other users require specification.

In [CCIR, 1986-90w] the EBU describes that for Region 1, a 2048 kbit/s bit stream will be necessary to carry the audio and ancillary data signals alongside the video signal within the 34 Mbit/s bit stream, which allows transmission of up to five sound channels in accordance with CCIR Recommendation 660.

4.3.2 Teletext and ancillary data

Provision is made in the predictive coding system to carry teletext and ancillary data at a rate up to 300 kbit/s, while in the hybrid predictive/transform coding system up to 408 kbit/s or 280 kbit/s have been reserved for that purpose.

The bit rate which is needed, as well as the necessary processing to interface this data stream to the Recommendation 656 interface is currently undefined, pending clarification of user needs. Proposals for teletext and ancillary data are made in section 3 of the report. IWP 11/7 is studying this question and IWP 10-11/5 is studying the related data broadcasting concerns [CCIR 1986-90j].

4.3.3 Error correction

Both systems use the same BCH (255, 239) forward error correction method.

However, the interleaving ratio of the hybrid predictive/transform coding system is twice as long as for the predictive coding system resulting in different capabilities of correcting long (more than 16 bits) error bursts.

Using radio and satellite links for the transmission, the capability of the proposed codecs to correct error bursts may not be sufficient [CCIR 1986-90x].

As very little information about the length of the error bursts is available, this point requires further study. This may result in the necessity of the use of more effective error correcting code and/or higher interleaving factor.

As both codecs may be equipped with the same error correction system the hybrid predictive transform model provider has accepted, in order to have a fair comparison between both coding algorithms, to change over the present design of the error correction hardware to an interleaving factor of 8.

4.3.4 Transmission multiplexing

The precise transmission multiplex is still the subject of study in IWPs CMTT/2 and CMTT/3.

Taking into account the latest information from CCITT Study Group XVIII, the bit rates which will be available from the proposed H2 channels are 32768 kbit/s for the H21 channel and around 44 Mbit/s for the H22 channel.

The predictive coding system which has been studied before the recent agreement about H21 bit rate with CCITT does not take into account that information.

The hybrid predictive/transform coding system specifications propose a multiplexing scheme for the agreed H21 ISDN channel as well as a frame structure to insert H21 ISDN channel in the third hierarchical level at 34 Mbit/s either synchronously or plesiochronously, using a positive, null, negative justification.

Because of error sensitivity of the encoded video signals the demultiplexer frame-alignment algorithm should be particularly error resilient. Such a technique resilient to multiple errors and justification errors on higher order multiplex is described in [CCIR 1986-90y].

Both systems use different transmission multiplexing schemes in the tests, causing a quite different performance in recovery from gross disturbances. In view of this the results of recovery-time tests must be carefully interpreted.

4.3.5 *Parameter optimization by hardware evaluation*

In the modification of the 34/45 Mbit/s Draft Recommendation several parameters have been modified in order to improve the performance of the codec. However in [CCIR, 1986-90z] an opinion is expressed that some parameter optimization could be further performed based on a hardware evaluation that would provide an extra improvement without a significant hardware change.

These parameters are :

- Visibility threshold matrices [CCIR, 1986-90aa] ;
- Allowed block mode choices within a super-block ;
- Accuracy (number of bits) of DCT coefficients, particularly for DC;
- Criticality.

5. Coding performance

5.1 Evaluation

5.1.1 Algorithm evaluation

Documents [CCIR 1986-90 a and ab] present a number of quality criteria, evaluation methods, and picture material which are to be used for the evaluation of reference systems (or alternative methods).

A fair comparison of the different systems should be made by using the same methods and same picture material considering the three aspects presented in § 3.1 (picture quality requirements, failure characteristic requirements and downstream processing).

To be a future Recommendation any system should at least meet the requirements of § 3.1.

5.2 Test results of existing systems

IWP 11/7 undertook to test the coding methods employed in existing prototype codecs operating at 34-45 Mbit/s on material conforming at source to the 4:2:2 specification. The tests evaluated performance in respect of basic quality, downstream-processed quality (colour-matte and slow motion), and failure characteristics (response to random and burst errors and to momentary interruptions).

Three motion-compensated, inter-frame codecs identified by IWP CMTT/2 were tested:

- a DCT codec operating at 34 Mbit/s and 50 Hz;
- a DCT codec operating at 45 Mbit/s and 60 Hz; and
- a DPCM codec operating at 45 Mbit/s and 60 Hz.

An additional intra-field DCT codec operating at 34 Mbit/s and 50 Hz, described in [CCIR 1986-90 ac] was included in the tests for reference purposes. The results of the tests were examined in the light of the user requirements given in Report 1211 and [CCIR 1986-90.ad]. It should be noted that some of these user requirements were based on preliminary information.

The results of the tests were as follows:

1. Basic quality - with low/medium activity sequences, both 45 Mbit/s codecs met the user requirements. With low/medium activity sequences, the 34 Mbit/s DCT codec did not meet the requirements although this probably reflects a momentary loss of synchronization when a test sequence was passed through the codec. With a high activity sequence, the 34 Mbit/s DCT codec did not meet the requirements (high activity sequences were not available for tests at 45 Mbit/s).
2. Downstream processed quality - all codecs tested met the user requirements for colour-matte and slow motion.
3. Failure characteristics - none of the codecs tested met the user requirements for response to random and burst transmission errors or for recovery from a momentary interruption of data.

Fuller details of the tests and results are given in [CCIR 1986-90ae, af and ag].

6. Further to the tests undertaken by IWP 11/7, an improved FEC has been implemented and tested on the DCT candidate codec operating at 34 Mbit/s. This FEC was a BCH [511, 475] having a redundancy of 7.6%, close to that of the BCH

(255, 239, 6.7%) used in the test of IWP 11/7. Subjective assessment of the codec behaviour with the improved FEC has shown that the requirements for transmission errors was met in the new conditions [CCIR 1986-90ah].

In [CCIR 1986-90ai] the EBU discusses the potential performance of the CMTT/2 codec in the light of assessments of the candidate codec from which the design is drawn. The document concludes that, basic picture quality requirements is likely to meet EBU requirements, the failure characteristics are likely to be acceptable with the improved FEC system, and recovery time is likely to be useable. The quality achieved after processing suggests that the new codec may be suitable for certain contribution applications. In anticipation of agreement of the full specification in an appropriate timescale, the EBU is recommending the adoption of the new system to its members.

7. Variable-length coding and video framing

Following the tests described in § 5.2 an Expert Group was formed to specify a compromise solution [CCIR, 1986-90aj] in order to establish a single world-wide standard. This compromise system is based on DCT coding with blockwise motion compensated prediction. As result of this work a draft new Recommendation has been established [CCIR, 1986-90s]. A few main parameters have not been agreed : three candidate systems are reported in Annexes VI and VII of [CCIR, 1986-90ak].

The coding efficiencies of the three systems have been compared and the results are reported in [CCIR, 1986-90al, am and an].

7.1 System A

System A is based on the organisation of the video information into stripes of blocks. Each stripe of blocks contains all the information (fixed-length code and variable-length code) relative to 8 video lines. Synchronization is possible every stripe of blocks. Only one channel is needed to transmit the coded video. Video framing and transmission multiplex framing are independent.

Inside the stripes of blocks ACVLC, (arithmetically calculated variable length code) is used. Up to 16 different sets of variable-length words are available, adapted to the statistics of the various events to be encoded, which enables very high coding efficiency.

[CCIR, 1986-90ao] gives the development status of the hardware meeting the specifications of system A (at 34 Mbit/s) and of its extension to HDTV (at 140 Mbit/s).

[CCIR, 1986-90ap, aq, ar and as] provide a complete specification of the transmission multiplex including sound, teletext and analogue test line transmission for insertion into the G.751 and G.752 frame structures. In addition, [CCIR, 1986-90at] proposes a method for compatible insertion of some or all of the signals into G.751 (or G.752) and SDH (G.707 to G.709) frames.

7.2 System B

During the CMTT/2 meeting of March 1990 a real hardware implementation of this system, not including motion compensation, was successfully demonstrated indicating its efficiency and good performance in presence of errors [CCIR, 1986-90au]. Additional information concerning this system is given in [CCIR, 1986-90z, aa, av, aw, and ax].

Due to its particular structure it is claimed that it has the following advantages:

- The B2-code has proved to be very efficient, resistant to errors, and allows an easy synchronization recovery in case of high error rates [CCIR, 1986-90aw, ay, ax, az and ba].
- The packet structure for video framing using fixed-length packets gives great robustness for a very low overhead/information ratio [CCIR 1986-90bb and bc].
- The final bit stream can be adapted to any form of synchronous or asynchronous channel multiplexing.
- There is an easy extensibility to HDTV (already checked) due to its video framing and packet structure [CCIR, 1986-90au].

7.3 System C

System C uses universal variable-length coding (UVLC) and video framing (UVF) and is described in [CCIR, 1986-90bd]. The word universal refers to the fact that the variable-length coding is designed in order to be independent of the picture statistics and that the video framing is open to a wide range of coding algorithms and of video formats.

The UVLC operates on sequences of DCT coefficients of the same order taken in the blocks belonging to a full stripe of 8 lines. It exhibits interesting properties in relation with compatible coding schemes [CCIR, 1986-90be]. The structure of the UVF is synchronized on a stripe basis; it is adapted to fixed and variable bit rate coding schemes [CCIR, 1986-90bf].

It is claimed that this system provides a superior coding efficiency at the expense of a moderate hardware complexity [CCIR, 1986-90al and be].

Further studies to achieve a unique coding system for the transmission rates 32-45 Mbit/s will be carried out in Interim Working Party CMTT/2.

8. System extensibility

There will be many applications requiring bit rate reduction techniques at different channel bit rates such as conventional TV and HDTV contribution circuits, primary and, later on, secondary distribution, digital video tape recording. It would be advantageous to have a common basic algorithm set to realize such codecs if it results in decreased manufacturing costs of such equipment. [CCIR 1986-90 m] suggests that distribution codecs can be implemented by simplifying the contribution codec or running it at a different bit rate. [CCIR 1986-90n] presents preliminary subjective assessments of the same algorithm simulated by computer at different bit rates.

REFERENCES

CCIR Documents

[1986-90]: a. 11/58 (IWP 11/7); b. IWP CMTT/2-88 (United Kingdom); c. CMTT/59 (Italy); d. CMTT/11 (France); e. IWP CMTT/2-3 (Germany, (Fed. Rep.of)); f. CMTT/83 (EBU); g. IWP CMTT/2-63 (EBU); h. IWP 11/7-205 (OIRT); i. IWP CMTT/2-61 (RTVE); j. IWP CMTT/2-78 (JIWP 10-11/5); k. CMTT/14 (France); l. IWP CMTT/2-93 (Sweden); m. IWP CMTT/2-79 (France); n. IWP CMTT/2-76 (Italy); o. IWP CMTT/2-92 (United Kingdom); p. IWP CMTT/2-1 (Japan); q. IWP CMTT/2-18 (IWP CMTT/2); r. IWP CMTT/2-67 (Expert Group); s. CMTT/199 (IWP CMTT/2); t. CMTT/57 (Italy); u. CMTT/93 (Spain); v. IWP CMTT/2-104 (Italy and Spain); w. CMTT/242 (EBU); x. IWP CMTT/2-72 (EUTELSAT); y. CMTT/16 (France); z. CMTT/2-161 (Spain/Italy); aa. CMTT/2-141 (Spain/Italy); ab. 11/57 (IWP 11/7); ac. CMTT/58 (Italy); ad. 11/501 (11/7); ae. CMTT/2-69R2 (AHG-BCT); af. CMTT/2-97 (AHG/BCT); ag. CMTT/2-108 (IWP 11/7); ah. CMTT/252 (France); ai. CMTT/254 (EBU); aj. CMTT/2-202 (IWP CMTT/2); ak. CMTT/2-172 (France); al. CMTT/2-134 (Belgium); am. CMTT/2-155 (Italy/Spain); an. CMTT/2-163 (Thomson-CSF); ao. CMTT/2-137 (Thomson-CSF); ap. CMTT/2-144 (France); aq. CMTT/2-145 (France); ar. CMTT/2-146 (France); as. CMTT/2-147 (France); at. CMTT/2-164 (UK); au. CMTT/2-160 (Italy/Spain); av. CMTT/2-148 (Spain); aw. CMTT/2-150 (Spain); ax. CMTT/2-156 (Italy/Spain); ay. CMTT/2-151 (Spain); az. CMTT/2-155 (Italy/Spain); ba. CMTT/2-159 (Italy/Spain); bb. CMTT/2-157 (Italy/Spain); bc. CMTT/2-158 (Italy/Spain); bd. CMTT/2-131 (Belgium); be. CMTT/2-132 (Belgium); bf. CMTT/2-133 (Belgium).

ANNEX I

VARIABLE LENGTH CODING AND VIDEO FRAMING FOR SYSTEM A

1 - GENERALITIES1.1 - Structure of a VLC word

A VLC word includes k groups of bits, which lengths are $l_1, l_2, \dots, l_k$

$l_1 \quad l_2 \quad l_3 \quad \dots \quad l_k$
 └───┘ └───┘ └───┘ └───┘ └───┘

All the bits belonging to the first k-1 groups are set to one. The last group of bits is the only one that must contain at least one zero.

The arithmetic rules to encode an integer value e are the following ones :

. $e \geq 0$

- if $e < S_1 = 2^{l_1} - 1$ then e is coded with l_1 bits
- if $S_1 \leq e < S_1 + 2^{l_2} - 1$ then the codeword is constituted of two information :
 - * the prefix made of l_1 bits equal to "1"
 - * the suffix which is the $e - S_1$ value coded with l_2 bits.
- and more generally if $S_i \leq e < S_{i+1} = S_i + 2^{l_{i+1}} - 1$
 - * the prefix is filled with $l_1 + l_2 + \dots + l_i$ bits equal to "1"
 - * the suffix represents the value $e - S_i$ coded with l_{i+1} bits.

. $e \geq 1$

The rules are the same except that the encoded value is e-1.

. Positive and negative values including the zero value

The sign bit is transmitted first

"0" for positive values

"1" for negative values

The absolute value AV is VLC encoded for positive numbers and AV-1 for negative numbers.

. Non zero positive and negative values

The sign bit is transmitted first and AV-1 is VLC encoded after.

1.2 - The maximum length of a VLC is limited :

$$\sum_{i=1}^k l_k \leq 31 \text{ (resp. 32) if a sign bit is needed (resp. if not)}$$

1.3 - The maximum value for k is 10

A VLC tree is completely defined by a set of 8 parameters $l_1, l_2, \dots, l_8$. Each of them requires 4 bits ($1 \leq l_k \leq 10$)

1.4 - The number of VLC trees is fixed to 16

The full description of the different VLC's used requires $16 \times 8 \times 4 = 512$ bits (a description with less bits is for further studies).

1.5 - VLC is used to encode different parameters needed for the description of a block :

- number of transmitted coefficients
- relative addresses
- absolute values of the coefficients
- motion vectors

for each parameter, a VLC tree has to be identified (see paragraph 5).

1.6 - VLC can also be used for quasi Fixed Length Coding (QFLC).

A QFLC word can be easily generated if the probability distribution is rather flat. For a N bit word, the (QFLC) VLC parameters will be $l_1 = N$ and $l_2 = 2$. The $2^N - 1$ first values will be FLC coded with N bits and the last one will be encoded with N+2 bits. QFLC is a particular case of VLC.

2 - BLOCK DESCRIPTION IN THE DCT DOMAIN

2.1 - Scanning paths

The 8x8 DCT blocks are scanned according to paths shown in figure 1. A different path is used for luminance and chrominance. Tables in figure 1 show how the 2-D DCT block is converted in sequential 1-D block. Entries in the table give the location of the $(i,j)^{th}$ DCT coefficient in the sequential 1-D block.

Figure 1 : Scanning Paths

luminance

0	2	6	12	20	28	36	44
1	5	11	19	27	35	43	51
3	7	13	21	29	37	45	52
4	10	18	26	34	42	50	57
8	14	22	30	38	46	53	58
9	17	25	33	41	49	56	61
15	23	31	39	47	54	59	62
16	24	32	40	48	55	60	63

chrominance

0	2	3	9	10	20	21	35
1	4	8	11	19	22	34	36
5	7	12	18	23	33	37	48
6	13	17	24	32	38	47	49
14	16	25	31	39	46	50	57
15	26	30	40	45	51	56	58
27	29	41	44	52	55	59	62
28	42	43	53	54	60	61	63

2.2 VLC description of an 8x8 block

A block (consisting of 64 cosine coefficients) is described with the help of three kinds of information :

- number of transmitted coefficients
- the relative addresses calculated according to the appropriate scanning path.
- the transmitted coefficient amplitudes.

The data framing is the following one :

$N [A] * [VLC]$

Where :

N : the number of transmitted coefficients

$[A]$: the relative addresses of the transmitted coefficients

$[VLC]$: VLC's of the transmitted coefficients

* Note 1 : when N exceeds 32, the $[A]$ informations are omitted.

* Note 2 : with reference to note 1, when N is lower or equal to 32, the transmitted coefficients are only non zero coefficients.

* Note 3 : with reference to note 1, when N exceeds 32, the transmitted coefficients are zero and non zero coefficients.

3 - DESCRIPTION OF OTHER PARAMETERS

One can take some advantage of encoding the motion vectors with a variable length code. In order to have a zero centered probability distribution, the motion vectors should be transmitted according to a differential mode. The prediction is the previous vector when it exists and zero when it does not exist (at the beginning of a stripe or when the intra field or inter field modes are the preceding block).

4 - VLC ENCODING MODES

<u>Parameter</u>	<u>Codeword</u>
Number of coefficients N	$(N)_{VLC}$
Relative address A	$(A)_{VLC}$
Non zero positive coefficient	$\text{Sign}, (AV-1)_{VLC}$
Positive coefficient (including zero)	$\text{Sign}, (AV)_{VLC}$
Negative coefficient	$\text{Sign}, (AV-1)_{VLC}$
Positive motion information (including zero)	$\text{Sign}, (AV)_{VLC}$
Negative motion information	$\text{Sign}, (AV-1)_{VLC}$

5 - VLC PARAMETERS FOR THE 34/45 MBIT/S CONTRIBUTION CODEC

16 sets of VLC parameters are predefined for this application. One set defines a VLC tree.

Tree number	l_1	l_2	l_3	l_4	l_5	l_6	l_7	l_8
1	1	1	1	1	2	4	7	8
2	1	2	2	2	3	4	7	9
3	2	2	2	2	3	3	7	9
4	2	3	3	3	4	7	9	-
5	2	4	5	5	6	9	-	-
6	3	4	4	4	7	9	-	-
7	4	3	3	7	9	-	-	-
8	4	4	4	7	9	-	-	-
9	4	5	5	7	9	-	-	-
10	5	4	4	7	9	-	-	-
11	5	5	5	7	9	-	-	-
12	5	6	6	5	9	-	-	-
13	6	6	5	9	-	-	-	-
14	6	7	9	-	-	-	-	-
15	8	9	-	-	-	-	-	-
16								

Motion informationMx
MyVLC tree No. 6
VLC tree No. 4Luminance intra mode

- number of luminance coefficients VLC tree n° 10
- relative addresses VLC tree n° 1
- coefficients (depending on the location)

VLC tree number :

	1	2	3	4	5	6	7	8
1	15	14	12	6	4	1	1	1
2	14	9	6	4	1	1	1	1
3	12	7	4	3	1	1	1	1
4	11	6	3	2	1	1	1	1
5	9	3	2	1	1	1	1	1
6	6	3	1	1	1	1	1	1
7	4	1	1	1	1	1	1	1
8	2	1	1	1	1	1	1	1

Luminance, inter mode

- number of luminance coefficients VLC tree n° 10
- relative addresses VLC tree n° 1
- coefficients (depending on the location)

VLC tree number :

	1	2	3	4	5	6	7	8
1	4	6	6	4	1	1	1	1
2	9	4	4	1	1	1	1	1
3	9	4	2	1	1	1	1	1
4	6	4	1	1	1	1	1	1
5	4	1	1	1	1	1	1	1
6	4	1	1	1	1	1	1	1
7	2	1	1	1	1	1	1	1
8	2	1	1	1	1	1	1	1

Chrominance, intra mode

- Number of chrominance coefficients VLC tree n° 7
- Relative addresses VLC tree n° 1
- Coefficients (depending on the location)

VLC tree number :

	1	2	3	4	5	6	7	8
1	14	5	4	1	1	1	1	1
2	4	1	1	1	1	1	1	1
3	1	1	1	1	1	1	1	1
4	1	1	1	1	1	1	1	1
5	1	1	1	1	1	1	1	1
6	1	1	1	1	1	1	1	1
7	1	1	1	1	1	1	1	1
8	1	1	1	1	1	1	1	1

Chrominance, inter mode

- number of chrominance coefficients VLC tree n° 7
- relative addresses VLC tree n° 1
- coefficients (depending on the location)

VLC tree number :

	1	2	3	4	5	6	7	8
1	1	1	1	1	1	1	1	1
2	1	1	1	1	1	1	1	1
3	1	1	1	1	1	1	1	1
4	1	1	1	1	1	1	1	1
5	1	1	1	1	1	1	1	1
6	1	1	1	1	1	1	1	1
7	1	1	1	1	1	1	1	1
8	1	1	1	1	1	1	1	1

6. VIDEO FRAMING

6.1--GENERAL STRUCTURE

FSW	00	FS	PI	ST	VF	VBR	BO
FSW	01	FS	PI	ST	VF	VBR	BO
FSW	10	FS	PI	ST	VF	VBR	BO

$$[SSW \ SN_1 \ BO_1 \ TFY_1 \ TFC_1 \ [MB_1^1] \dots [MB_1^{N_2}] \ CRC_1]$$

$$[SSW \ SN_M \ BO_M \ TFY_M \ TFC_M \ [MB_M^1] \dots [MB_M^{N_2}] \ CRC_M]$$

6.2--DESCRIPTION OF MACROBLOCK [MB]

$$[MB] = MI \ CT \ MV_x \ MV_y$$

N_{Y1}	$[A_{Y1}]^*$	$[VLC_{Y1}]$
N_{CB}	$[A_{CB}]^*$	$[VLC_{CB}]$
N_{Y2}	$[A_{Y2}]^*$	$[VLC_{Y2}]$
N_{CR}	$[A_{CR}]^*$	$[VLC_{CR}]$

Where : $[A_{Y1}] = A_{Y1}^1, \dots, A_{Y1}^{N_{Y1}}$

Relative addresses of the transmitted coefficients of the Y_1 block.

$[A_{Y2}] = A_{Y2}^1, \dots, A_{Y2}^{N_{Y2}}$

Relative addresses of the transmitted coefficients of the Y_2 block.

$[A_{CB}] = A_{CB}^1, \dots, A_{CB}^{N_{CB}}$

Relative addresses of the transmitted coefficients of the CB block.

$[A_{CR}] = A_{CR}^1, \dots, A_{CR}^{N_{CR}}$

Relative addresses of the transmitted coefficients of the CR block.

$[VLC_{Y1}] = VLC_{Y1}^1, \dots, VLC_{Y1}^{N_{Y1}}$

VLC's of the transmitted coefficients of the Y_1 block.

$[VLC_{Y2}] = VLC_{Y2}^1, \dots, VLC_{Y2}^{N_{Y2}}$

VLC's of the transmitted coefficients of the Y_2 block.

$[VLC_{CB}] = VLC_{CB}^1, \dots, VLC_{CB}^{N_{CB}}$ VLC's of the transmitted coefficients of the CB block.

$[VLC_{CR}] = VLC_{CR}^1, \dots, VLC_{CR}^{N_{CR}}$ VLC's of the transmitted coefficients of the CR block.

- * Note 1 : When NY_1 (respect NY_2 , N_{CR} , N_{CB}) exceeds 32 the $[A_{y1}]$ informations (respect $[A_{y2}]$, $[A_{CR}]$, $[A_{CB}]$) are omitted.
- * Note 2 : With reference to Note 1, when NY_1 (respect NY_2 , N_{CR} , N_{CB}) is lower or equal to 32, the transmitted coefficients are only non zero coefficients.
- * Note 3 : With reference to Note 1, when NY_1 (respect NY_2 , N_{CR} , N_{CB}) exceeds 32, the transmitted coefficients are zero and non-zero coefficients.

6.3 - NOTATIONS

FSW	: Field synchronization word (47'1'+0')	48 bits
FS	: Field status	1 bit
	0 = even 1 = odd	
PI	: Phase information (this field could be used to transmit the field phase information when the system is interfaced with PAL studios for instance)	12 bits
ST	: System type 0 = 50 Hz 1 = 60 Hz	1 bit
VF	: Video format 0000 = 4:2:2 (other formats for further studies)	4 bits
VBR	: Video bit rate 0000 = 34 Mbit/s (other video bit rates for further studies)	4 bits
BO	: Buffer occupancy Indicates the buffer occupancy at the coder at the end of the previous active field or stripe (only the 16 MSB's transmitted)	16 bits
SSW	: Stripe Synchronisation word ('0' + 46'1' + '0')	48 bits
SN	: Stripe number (from 0 to M-1)	N1 bits

TFY	: Transmission factor for luminance (from 0 to 175)	8 bits
TFC	: Transmission factor for colour differences (from 0 to 175)	8 bits
CRC	: Cyclic Redundancy Code to be applied to all bits of the encoded stripe excluding SSW Polynomial : $1 + x^2 + x^{15} + x^{16}$	16 bits
MI	: Mode identification 00 Intra field 01 Inter field 10 Inter frame 11 For further application (for instance global motion vector or inter-frame mode without motion compensation)	2 bits
CT	: Criticality (from 0 to 3)	2 bits
MV_x^*	: Motion prediction error in the horizontal direction (only for the inter-frame mode)	VLC
MV_y^*	: Motion prediction error in the vertical direction	VLC

* Predictive encoding of the motion vectors is performed along a strip of blocks. MV_x and M_y represent the prediction error on the horizontal and vertical components of the motion vector. The prediction is the motion vector of the macro-block located immediately before (on the left) of the current macro-block. The prediction for the first macro-block of a strip of blocks or for the macro-block following an intra-field or inter-field encoded macro-block is 0 for both coordinates of the motion vector.

N_{y1}	: number of transmitted coefficients of the Y1 block	VLC
N_{y2}	: number of transmitted coefficients of the Y2 block	VLC
N_{CR}	: number of transmitted coefficients of the CR block	VLC
N_{CS}	: number of transmitted coefficients of the CB block	VLC
A_{y1}	: relative addresses of the transmitted coefficients of the Y ₁ block	VLC
A_{y2}	: relative addresses of the transmitted coefficients of the Y ₂ block	VLC
A_{CR}	: relative addresses of the transmitted coefficients of the CR block	VLC

A_{CB}	: relative addresses of the transmitted coefficients of the CB block	VLC
VLC_{Y1}	: Variable length description of the transmitted coefficients of the Y1 block	VLC
VLC_{Y2}	: Variable length description of the transmitted coefficients of the Y2 block	VLC
VLC_{CR}	: Variable length description of the transmitted coefficients of the CR block	VLC
VLC_{CB}	: Variable length description of the transmitted coefficients of the CB block	VLC

NOTA : Data are transmitted with MSB first.

6.4 - PARAMETER VALUES FOR THE 4:2:2 34/45 MBIT/S CONTRIBUTION CODEC

$N1 = 6$ (50 Hz and 60 Hz)

$N2 = 45$ (50 Hz and 60 Hz)

$M = 36$ (50 Hz)
 $= 31$ (60 Hz)

ANNEX II

Part A. - VARIABLE LENGTH CODING AND VIDEO FRAMING FOR SYSTEM B

Non-zero DCT coefficients are quantized and VLC coded. Runs of zeros, according to the scanning path shown in Figure 1 are coded as run-lengths.

0	2	6	12	20	28	36	44
1	5	11	19	27	35	43	51
3	7	13	21	29	37	45	52
4	10	18	26	34	42	50	57
8	14	22	30	38	46	53	58
9	17	25	33	41	49	56	61
15	23	31	39	47	54	59	62
16	24	32	40	48	55	60	63

scanning path for luminance

0	2	3	9	10	20	21	35
1	4	8	11	19	22	34	36
5	7	12	18	23	33	37	48
6	13	17	24	32	38	47	49
14	16	25	31	39	46	50	57
15	26	30	40	45	51	56	58
27	29	41	44	52	55	59	62
28	42	43	53	54	60	61	63

scanning path for chrominance

Figure 1

The words at the output of the variable length coder have the following structure:

$$\begin{array}{l} |1 X_i| \dots |1 X_i| |0 X_0|, \quad i=0..8, \quad X_i \in \{0,1\} \\ |1 X_8| \dots |1 X_1| |1 X_0|, \quad X_i \in \{0,1\} \end{array}$$

where | | corresponds to a pair of bits.

The first bit is the *continuation bit* and if 0 the present pair is the last one except for the word having length 18; the second bit of each pair, and both the bits of the last pair if the word length is 18, is the *information bit* and can assume the value 0 or 1. The length of the available words can vary from 2 to 18, as indicated in Table 1, and the words having less than 18 end with a $|0\ X_0|$ pair.

Table 1

i	word length (bit)	word structure	No of words
0	2	0 X_0	2
1	4	1 X_1 0 X_0	4
2	6	.	8
3	8	.	16
4	10	.	32
5	12	.	64
6	14	.	128
7	16	1 X_7 0 X_0	256
8	18	1 X_8 $X_9 X_0$	1024

The total number of words is 1534. 66 of them are used to code the run length of zeros, the special words EOB_0 and EOB_1 , indicating the end of each block, and the word NULL (used in case of underflow). The remaining 1468 words can be used to code the quantized levels from -734 to +734.

Two assignment tables for the values of X (quantized levels) lower than 17 and greater than -17 and for the run lengths up to 28 zeros are used. In Table 2 the value of luminance or chrominance coefficients assigned to the codewords differ in 20 cases, thus two different assignment tables are required to specify the luminance or the chrominance coefficients. Two other tables are available, but not defined. The assignments of EOB_0 and EOB_1 are fixed for all the tables.

The assignments for run lengths of zeros from 29 up to 63 and for the codeword NULL are shown in Table 3.

Table 2

Assignment for luminance (Y) and chrominance (C) coefficients

codeword	Y	C	codeword	Y	C
0 1	1		0 0	-1	
1 1 0 0	2		1 0 0 1	-2	
1 1 0 1	0*1		1 0 0 0	0*2	
1 1 1 0 0 0	0*3		1 0 1 1 0 1	0*4	
1 1 1 0 0 1	3	0*5	1 0 1 1 0 0	-3	0*6
1 1 1 1 0 0	4	0*7	1 0 1 0 0 1	-4	0*8
1 1 1 1 0 1	EOB ₁		1 0 1 0 0 0	EOB ₀	
1 1 1 0 1 0 0 0	0*5	3	1 0 1 1 1 1 0 1	0*6	-3
1 1 1 0 1 0 0 1	0*7	4	1 0 1 1 1 1 0 0	0*8	-4
1 1 1 0 1 1 0 0		0*9	1 0 1 1 1 0 0 1		0*10
1 1 1 0 1 1 0 1		0*11	1 0 1 1 1 0 0 0		0*12
1 1 1 1 1 0 0 0		5	1 0 1 0 1 1 0 1		-5
1 1 1 1 1 0 0 1	6	0*13	1 0 1 0 1 1 0 0	-6	0*14
1 1 1 1 1 1 0 0	7	0*15	1 0 1 0 1 0 0 1	-7	0*16
1 1 1 1 1 1 0 1	8	0*17	1 0 1 0 1 0 0 0	-8	0*18
1 1 1 0 1 0 1 0 0 0	0*13	6	1 0 1 1 1 1 1 1 0 1	0*14	-6
1 1 1 0 1 0 1 0 0 1	0*15	7	1 0 1 1 1 1 1 1 0 0	0*16	-7
1 1 1 0 1 0 1 1 0 0	0*17	8	1 0 1 1 1 1 1 0 0 1	0*18	-8
1 1 1 0 1 0 1 1 0 1		0*19	1 0 1 1 1 1 1 0 0 0		0*20
1 1 1 0 1 1 1 0 0 0		0*21	1 0 1 1 1 0 1 1 0 1		0*22
1 1 1 0 1 1 1 0 0 1		0*23	1 0 1 1 1 0 1 1 0 0		0*24
1 1 1 0 1 1 1 1 0 0		0*25	1 0 1 1 1 0 1 0 0 1		0*26
1 1 1 0 1 1 1 1 0 1		0*27	1 0 1 1 1 0 1 0 0 0		0*28
1 1 1 1 1 0 1 0 0 0		9	1 0 1 0 1 1 1 1 0 1		-9
1 1 1 1 1 0 1 0 0 1		10	1 0 1 0 1 1 1 1 0 0		-10
1 1 1 1 1 0 1 1 0 0		11	1 0 1 0 1 1 1 0 0 1		-11
1 1 1 1 1 0 1 1 0 1		12	1 0 1 0 1 1 1 0 0 0		-12
1 1 1 1 1 1 1 0 0 0		13	1 0 1 0 1 0 1 1 0 1		-13
1 1 1 1 1 1 1 0 0 1		14	1 0 1 0 1 0 1 1 0 0		-14
1 1 1 1 1 1 1 1 0 0		15	1 0 1 0 1 0 1 0 0 1		-15
1 1 1 1 1 1 1 1 0 1		16	1 0 1 0 1 0 1 0 0 0		-16

Note: The information bits for the positive values and the corresponding negative values are complemented. The information bits for the odd run lengths of zeros and the corresponding even ones are complemented. The information bits for EOB₀ and EOB₁ are complemented.

Table 3
Assignments for run length of zeros

codeword	run of zeros	codeword	run of zeros
1 1 1 0 1 0 1 0 1 0 0 0	0*29	1 0 1 1 1 1 1 1 1 0 0 1	0*30
1 1 1 0 1 0 1 0 1 0 0 1	0*31	1 0 1 1 1 1 1 1 1 0 1 1	0*32
1 1 1 0 1 0 1 0 1 1 0 0	0*33	1 0 1 1 1 1 1 1 0 0 0 1	0*34
1 1 1 0 1 0 1 0 1 1 0 1	0*35	1 0 1 1 1 1 1 0 0 0 1 1	0*36
1 1 1 0 1 0 1 1 1 0 0 0	0*37	1 0 1 1 1 1 0 0 1 1 0 0	0*38
1 1 1 0 1 0 1 1 1 0 0 1	0*39	1 0 1 1 1 1 0 0 1 1 0 1	0*40
1 1 1 0 1 0 1 1 1 1 0 0	0*41	1 0 1 1 1 1 0 0 1 0 0 0	0*42
1 1 1 0 1 0 1 1 1 1 0 1	0*43	1 0 1 1 1 1 0 0 1 0 0 1	0*44
1 1 1 0 1 1 1 0 1 0 0 0	0*45	1 0 1 1 1 0 1 1 1 1 0 0	0*46
1 1 1 0 1 1 1 0 1 0 0 1	0*47	1 0 1 1 1 0 1 1 1 1 0 1	0*48
1 1 1 0 1 1 1 0 1 1 0 0	0*49	1 0 1 1 1 0 1 1 1 0 0 0	0*50
1 1 1 0 1 1 1 0 1 1 0 1	0*51	1 0 1 1 1 0 1 1 0 0 0 1	0*52
1 1 1 0 1 1 1 1 1 0 0 0	0*53	1 0 1 1 1 0 1 0 1 1 0 0	0*54
1 1 1 0 1 1 1 1 1 0 0 1	0*55	1 0 1 1 1 0 1 0 1 1 0 1	0*56
1 1 1 0 1 1 1 1 1 1 0 0	0*57	1 0 1 1 1 0 1 0 1 0 0 0	0*58
1 1 1 0 1 1 1 1 1 1 0 1	0*59	1 0 1 1 1 0 1 0 1 0 0 1	0*60
1 1 1 1 1 0 1 0 1 0 0 0	0*61	1 0 1 0 1 1 1 1 1 1 0 0	0*62
1 1 1 1 1 0 1 0 1 0 0 1	0*63	1 0 1 0 1 1 1 1 1 1 0 1	NULL

The values X greater than 16 are coded as follows:

- an offset +33 is added
- a pair having the information bit equal to each significant bit is generated.

Examples:

$$X=23 \Rightarrow 23+33=56 = (000000\ 111000)_2 =$$

$$(|1\ 1|1\ 1|1\ 1|1\ 0|1\ 0|0\ 0|)_{\text{VLC}}$$

$$(\quad X_5\ X_4\ X_3\ X_2\ X_1\ X_0)_{\text{VLC}}$$

$$X=403 \Rightarrow 403+33=436 = (000\ 110110100)_2 =$$

$$(|1\ 1|1\ 1|1\ 0|1\ 1|1\ 1|1\ 0|1\ 1|1\ 0|0\ 0|)_{\text{VLC}}$$

$$(\quad X_8\ X_7\ X_6\ X_5\ X_4\ X_3\ X_2\ X_1\ X_0)_{\text{VLC}}$$

X_9

$$X=734 \Rightarrow 734+33=767 = (000\ 101111111)_2 =$$

$$(|1\ 0|1\ 1|1\ 1|1\ 1|1\ 1|1\ 1|1\ 1|1\ 1|)_{\text{VLC}}$$

$$(\quad X_8\ X_7\ X_6\ X_5\ X_4\ X_3\ X_2\ X_1\ X_0)_{\text{VLC}}$$

X_9

The values X lower than -16 are coded as follows:

- an offset -34 is added
- a pair having the information bit equal to each significant bit is generated.

Examples:

$X=-23 \Rightarrow -23-34=-57 = (111111\ 000111)_2 =$
 $(\begin{smallmatrix} 1 & 0 & 1 & 0 & 1 & 0 & 1 & 1 & 1 & 1 & 0 & 1 \end{smallmatrix})_{vLc}$
 $(\begin{smallmatrix} X_5 & X_4 & X_3 & X_2 & X_1 & X_0 \end{smallmatrix})_{vLc}$

$X=-403 \Rightarrow -403-34=-437 = (111\ 001001011)_2 =$
 $(\begin{smallmatrix} 1 & 0 & 1 & 0 & 1 & 1 & 1 & 0 & 1 & 0 & 1 & 1 & 0 & 1 \end{smallmatrix})_{vLc}$
 $(\begin{smallmatrix} X_8 & X_7 & X_6 & X_5 & X_4 & X_3 & X_2 & X_1 & X_0 \end{smallmatrix})_{vLc}$
 X_9

$X=-734 \Rightarrow -734-34=-768 = (111\ 0100000000)_2 =$
 $(\begin{smallmatrix} 1 & 1 & 1 & 0 & 1 & 0 & 1 & 0 & 1 & 0 & 1 & 0 & 1 & 0 & 0 & 0 \end{smallmatrix})_{vLc}$
 $(\begin{smallmatrix} X_8 & X_7 & X_6 & X_5 & X_4 & X_3 & X_2 & X_1 & X_0 \end{smallmatrix})_{vLc}$
 X_9

The transmission of the last run length of zeros in a block is avoided, by anticipating the EOB. The decoder assumes that the last coefficients in a block are all zeros, when less than 64 coefficients are decoded.

If there are one or more coefficients +1 between two runs of zeros, or between a run of zeros and the EOB, one of them is not transmitted and the decoder reinsert it.

Examples:

coefficients:	-2	0	0	0	+1	+1	0	0	+2	0	0	0	0	EOB
codewords:	1001	111000	01	01	1000	1100	101101	111101						
transmitted:	1001	111000	01		1000	1100		111101						

coefficients:	-2	0	0	0	0	0	0	0	0	0	+1	EOB
codewords:	1001				11101100				01	111101		
transmitted:	1001				11101100					111101		

In the first example a +1 is not transmitted because there are two +1 between two runs of zeros and a run of 4 zeros is not transmitted because the EOB is anticipated.

In the second example a +1 is not transmitted because it is between a run of 9 zeros and the EOB.

The absolute values of the difference components dx and dy between the motion vector and the prediction value are coded assigning the 82 codewords following the scanning path of Figure 2.

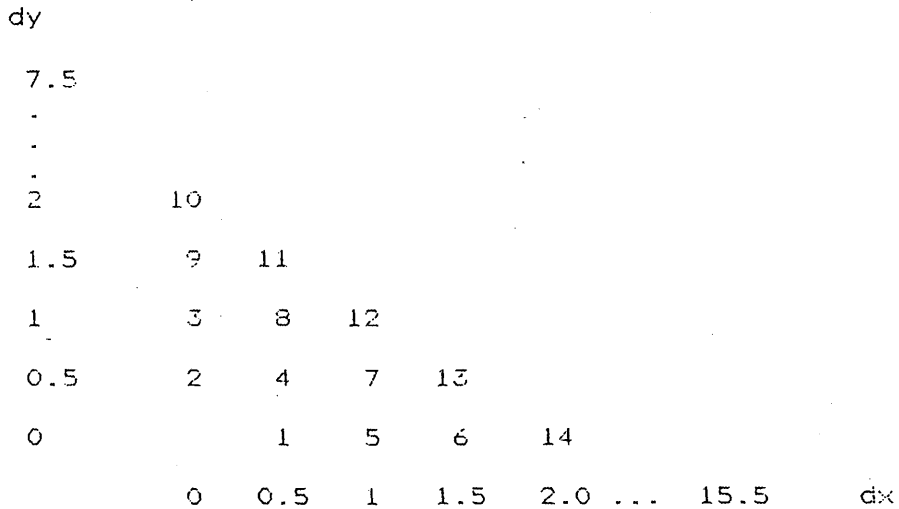


Figure 2

if $dx > 15.5$ or $dy > 7.5$ the macroblock is coded as inter-frame without motion compensation.

For simplicity, only the first part of the scanning path is indicated in the Figure. The codeword assignment is in Table 4.

Table 4

path order	dx	dy	codeword
1	0.5	0	0 0
2	0	0.5	0 1
3	0	1	1 0 0 0
4	0.5	0.5	1 0 0 1
5	1	0	1 1 0 0
6	1.5	0	1 1 0 1
7	1	0.5	1 0 1 0 0 1
8	0.5	1	1 0 1 1 0 0
9	0	1.5	1 0 1 1 0 1
10	0	2	1 1 1 0 0 0
11	0.5	1.5	1 1 1 0 0 1
12	1	1	1 1 1 1 0 0
13	1.5	0.5	1 0 1 0 1 0 0 0
14	2	0	1 0 1 0 1 0 0 1
...			...

The codeword reserved for EOB0 and EOB1 are not assigned to motion vector differences.

Part B. - VIDEO FRAMING AND FRAMING STRUCTURE FOR VLC SCHEME 2

1. General structure

The transmission frame structure is based on packets. All the information, video, audio or other data, are organised in packets of fixed length.

Each packet is constituted by 4080 bits organised as 255 words of 16 bits. Figure 1 of Annex 8 of Appendix II of [CCIR, 1986-90a] shows the basic organisation of a packet for video data.

Every two packets 64 bits (8 bytes) are inserted for framing purposes. The first three bytes constitute the *Frame alignment* word which is 0000 1111 0110 1001 1011 0100. The following byte is the *Multiframe* word and it specifies the assignment and use of the following two packets. Three following bytes carry the information to allow the recovering at the decoder end of the video signal frequency from the transmission frequency. Finally the last byte is a *Status* byte used for transmission maintenance.

For each packet, the first word is the *header* and the 16 bits have the following functions:

service identifier (SI, 3 bits): identifies the service (video, audio, other data).

source address (SA, 4 bits): identifies, within a service, the source of data.

continuity index (CI, 3 bits): each packet produced by a source is identified by this index which is incremented from 000 or 001 to 111 in order to allow the revelation of packet losses.

complete/incomplete flag (CF, 1 bit): a packet can be complete (CF=1) and all its capacity is used for useful data, or it can be incomplete (CF=0), the last words are dummy bits, e.g. NULL words (see Annexe 4-B) of Appendix II of [CCIR, 1986-90a].

error protection (EP, 5 bits): the previous 11 bits of information are protected by a Hamming code (15,11) plus parity.

The structure of the remaining 254 words of a packet is service depending, i.e. for each service it is specified their organisation and function. In this Annexe the organisation of the packets for the video information is described.

2. Terminology

Each video *field* is made of 288 active lines for the 625-line standard and 248 lines for the 525-line standard.

Each line contains 720 pixels. There are 720 Y samples (luminance), 360 C_B samples and 360 C_R samples (chrominance).

The lines are organised in *stripes* constituted by 8 lines, 36 stripes form a field in the case of 625-line standard and 31 stripes form a field in the case of the 525-line standard.

The samples are organised as 8·8 sample *blocks*.

Each stripe contains 90 Y blocks, 45 C_B and 45 C_R blocks.

A group of a first Y block, a C_B block, a second Y block and a C_R block is called macro-*block*. Each stripe contains 45 macro-blocks.

The 64 DCT coefficients belonging to a block are scaled and coded with a Variable Length Code (VLC) and the run lengths of zeros are coded as single VLC words (see Annex 4-B of Appendix II of [CCIR, 1986-90a]). After the last non-zero coefficient an End-of-Block word (EOB) is sent. Two different EOB words are available: EOB₀ and EOB₁.

For each block a word of bits is generated. It indicates which mode, intra-field, inter-field or inter-frame is used for the macro-block, one of four criticality factors is signalled. When all the blocks are in inter-frame mode and the motion compensation is applied, then the information on the vector components are indicated.

For each stripe two bytes are generated, they contain the transmission factors for luminance and chrominance used for the stripe.

For each video frame a set of information is produced, indicating, for example, the video system characteristics (field frequency, number of lines per field, etc.)

This information is carried by two different kinds of video packets: the *V-type packets* contain the VLC data and the *F-type packets* contain the fixed-length data.

Note: Data are transmitted with MSB first.

3. Video packet structure

3.1 The header

The video packets are characterised by the following header (see figure 1):

service identifier: SI = 000 (video)

source address:

for the 4:2:2 format

SA = 0000 for the V-type packets

SA = 0001 for the F-type packets

the other configurations are reserved, but not assigned.

continuity index: the first packet, for each source, in video frame is characterised by CI = 000, all the others use the CI from 001 to 111, cyclically incremented.

complete/incomplete flag: the last V-type packet in a video frame is normally incomplete (CF=0).

3.2 Error protection code for the packet

Both for V-type and F-type packets the data are organised as 16 words of 255 bits. For each word the information bits are 239 and the remaining 16 bits are used for the error correction (see Annex 8 of Appendix II of [CCIR, 1986-90a]).

3.3 V-type packets

The structure of a V-type packet is shown in figure 1.

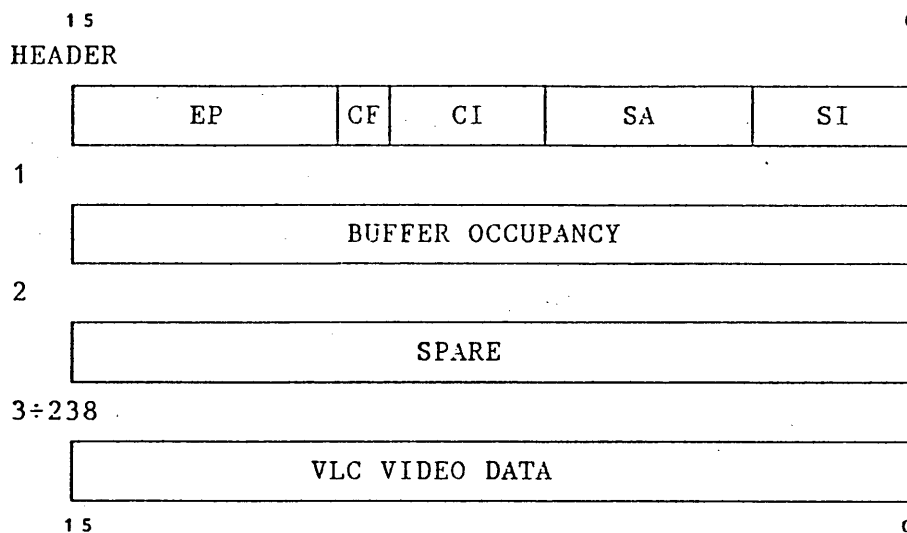


Figure 1 - V-type packets structure.

The second 16-bit word, following the header, contains the buffer occupancy information as number of packets in the coder buffer. A capacity up to $2^{16} \cdot 4080$ bits can be specified.

The third 16-bit word is reserved for future uses.

The remaining 236 words contain the VLC video data.

The video framing inside the V-type packets is recovered on the basis of the sequence of EOBs.

The EOB₀ and EOB₁ words are generated following these patterns:

0000 1111 0101 1001
or
1111 0000 1010 0110

where 0 corresponds to EOB₀ e 1 to EOB₁.

The choice of generating the first pattern of 16 EOBs or the second one is made on the basis of a predetermined sequence. This sequence is given by a the following polynomial:

$$g = x^{11} + x^9 + 1$$

the length of such sequence is up to $2^{11}-1$, the initial state of the pseudo-random generator is loaded with the first block of each video frame. If the output of this generator is 0, then the 16 EOBs are as in the first pattern, if the output is 1, then the second pattern is generated.

Each block in the frame is therefore identified by the unique position of its EOB inside the pseudo-random sequence.

3.4 F-type packets

In the case of the 525 and 625-line formats 6 F-type packets are generated for each frame. The continuity index is from 0 to 5.

In the case of 625-line format a field is formed by 36 stripes and the packets of F-type contain the information of fixed length relevant to the frame and the group of 12 macroblocks.

For each group of 12 macroblocks the following data are organised as a group of 5 16-bit words:

- twelve 5-bit words contain the Mode Information (MI, i.e. intra, inter and information on the motion compensation vector).
- one 14-bit word for the CRC protection of the VLC data of the 12 macroblock group (VCRC) with the following generator polynomial:

$$g(x) = \quad \quad \quad \text{(to be defined)}$$
- one 6-bit word for the CRC protection of the FLC data of the 12 macroblock group (FCRC) with the following generator polynomial:

$$g(x) = x^6 + x^5 + 1$$

The 16-bit words of the F-type packets are assigned as follows for the 625-line format:

- 0 header
- 1 ancillary data
- 2..13 transmission factors
- 14..238 45 groups of 12 macroblocks

In the case of the 525-line format, a field is formed by 31 stripes, F-type packets with continuity index 0 and 3 contain data relevant to the first 11 stripes in a field, F-type packets with continuity index 1,2,4 and 5 contain data relevant to 10 stripes each.

For each group of 9 macroblocks the following data are organised as a group of 4 16-bit words:

- nine 5-bit words contain the Mode Information (MI, i.e. intra, inter and information on the motion compensation vector).
- one 13-bit word for the CRC protection of the VLC data of the 9 macroblock group (VCRC) with the following generator polynomial:

$$g(x) = \quad \quad \quad \text{(to be defined)}$$
- one 6-bit word for the CRC protection of the FLC data of the 9 macroblock group (FCRC) with the following generator polynomial:

$$g(x) = g_6 + x^5 + 1$$

The 16-bit words of the F-type packets are assigned as follows for the 525-line format:

0	header
1	ancillary data
2..11	transmission factors
12..231	55 groups of 9 macroblocks
232..238	reserved for future uses

Ancillary data (word 1 of the packet):

ST System type (3 bits)
 000 = 625-line/50 Hz field frequency
 001 = 525-line/60 Hz field frequency
 010...111 = reserved for future uses

FS Field Status (1 bit)
 0 even field
 1 odd field

the remaining 12 bits are reserved for future uses.

For each macroblock a 5-bit word (MI) contains the following information:

MSB				LSB
m2	m1	m0	c1	c0

where c are the criticality bits as defined in Annex 5 CCIR, 1986-90a and m are used as follows:

m2	m1	m0	
0	0	0	intra-field mode
0	0	1	inter-field mode
0	1	0	inter-frame mode without motion compensation
0	1	1	inter-frame mode with motion compensation (motion vector difference = 0)
1	0	0	inter-frame with motion vector sign dx + dy +
1	0	1	inter-frame with motion vector sign dx + dy -
1	1	0	inter-frame with motion vector sign dx - dy +
1	1	1	inter-frame with motion vector sign dx - dy -

Each macroblock in inter-frame mode can be no-motion compensated or motion compensated. In this last case the difference between the actual vector and the prediction (obtained from the previous macroblock in the stripe) is computed. For the first macroblock in a stripe the prediction vector is the null vector. If the difference is zero, this is signalled by MI data. If the difference is not zero, the signs of the two difference components dx and dy are signalled in the MI data and the absolute values of dx and dy is coded according to the scanning path and codeword assignment given in Annex 6 CCIR, 1986-90a.

The VLC coded data are transmitted with the following order:

[dx,dy] [VLCy1] EOBi [VLCcb] EOBi+1 [VLCy2] EOBi+2 [VLCcr] EOBi+3

where:

dx,dy is a VLC codeword associated to absolute values of dx and dy
 VLC are the codewords associated to DCT coefficients and
 run lengths of zeros for the components Y, Cr and Cb.
 EOB are the End of Block words.

CCIR Documents

[1986-90]: a. CMTT/199.

ANNEX III

Statement on patent policy related
to [SUBJECT XYZ]

[COMPANY ABC] owns patents covering equipment for [Subject XYZ] which
 could be key elements in the application of a CCIR recommendation.

[COMPANY ABC] recognizing the advantage of international standardization for
 both users and manufacturers

DOES NOT WISH to restrict the application of a CCIR recommendation on
 [SUBJECT XYZ] and therefore

FORMALLY STATES that this company is willing to conclude licence agreements
 on REASONABLE CONDITIONS without any UNFAIR DISCRIMINATION with
 all manufacturers who wish to use such patents for implementing
 CCIR recommendations in hardware.

[COMPANY ABC] accepts in addition that cases of alleged violation of this
 statement can be brought to the attention of the Director, CCIR
 who shall then inform participants in the work of CCIR.

Signature

ANNEX IV

Note: Some parts of Report 1235 could not be accepted by some Administrations. The revisions proposed by those Administrations (which were not accepted by other Administrations) are given in this Annex:

"The debate was inevitably highly technical but very interesting and at the end, of those administrations who were prepared to express an opinion, there was a preference by 5 to 2 for the adaptive VLC scheme with a conventional video multiplexing structure. The degree of preference was insufficient for the preferred system to become the single system to be specified in Recommendation 723, and the VLC/video multiplex remains "under study". Both systems have been incorporated in Report 1235, together with a programme of work to enable the optimum system to be agreed using measurements of signal statistics and computer simulations of the complete coding schemes. This should enable the optimum system to be agreed much quicker than would be the case if tests with hardware meeting the full specification were to be required.

When opinions were sought as to whether the preferred system should be designated the "reference" system, the preference for such a proposal was 10 to 3."

In Section 7, under System B, modify sub-item 2.4) to read as follows:

"2.4) An important criterion to evaluate the VLC is the number of bits produced on both data-bases. A difference of the order of 10% is considered significant (see: System A - VLC item 2)."

In Section 7, under System B, insert between existing items 3 and 4, the following new items:

- "4) Other criteria to evaluate the coding performance are:
- sensitivity of the proposed VLC to transmission errors (evaluated as subjective impairment versus error rate after error correction by FEC)
 - recovery time after sync loss
- 5) Detailed and comprehensive information on the following items must be provided before the Plenary Assembly:
- criticality and visibility threshold matrix.
 - suitability of the video framing for application on other transport media, such as ATM, optical fibre links, satellite channels, characterized by different error rates and statistical error distributions
 - complexity of the hardware and evaluation of the technology required
 - extensibility of the video multiplexing to HDTV and to different video formats
 - adaptability to different transmission rates."

Renumber the subsequent items accordingly.

REPORT 1234*

DIGITAL TRANSMISSION OF COMPONENT-CODED TELEVISION
SIGNALS AT BIT RATES NEAR 68 MBIT/S AND 140 MBIT/S

(1990)

1. INTRODUCTION

This report considers transmission systems for vision and associated sound and data signals using circuits having data rates near 68 Mbit/s and 140 Mbit/s, corresponding approximately to access levels of the future B-ISDN currently under study by CCITT.

These rates can be conveniently related to the hierarchical levels defined in CCITT Recommendations G.702 and G.802. Sections of the report address also the quality objectives and methods of assessment for transmission systems under both normal and failure conditions.

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 distribute television programmes when no further post-production processing is expected.

Section 2 of the report provides some consideration of the basic principles of transmission systems. Section 3 considers systems for operation at a data rate near 68 Mbit/s and Section 4 systems near 140 Mbit/s.

2. General Considerations2.1 Introduction

The transmission of television signals at about 68 Mbit/s or 140 Mbit/s is subject to similar quality objectives and technology limitations. The difference between systems at these data rates is likely to be in the level of complexity required. This section of the report addresses the concerns common to both data rates.

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

2.2 Bit Rates for Transmission

The present digital hierarchy provides access to bit-rates around 34 Mbit/s and 140 Mbit/s. For the future ISDN, it is envisaged that the user will have access to so-called "H channels"*. For television, the relevant bit-rates are

H2: around 33 Mbit/s or around 44 Mbit/s.

The broadband channel rate for the H₂₁ broadband channel will be 32768 kbit/s [CCIR 1986-90a]. The channel rate for the H₂₂ broadband channel has not yet been defined.

H4: The latest information from CCITT Study Group XVIII indicates that in the case of the future broadband ISDN, the total bit rate available to the user will be in the range of 132 000 to 138 240 kbit/s. The final figure will provide an integer multiple of 64 kbit/s. Considering that there is a trend towards establishing fully synchronous digital networks in many parts of the world and there are some operational advantages in using asynchronous multiplexing for sound and data within the video signal, the use of a 64 kbit/s multiplexing structure may be recommended, as such a system will provide a greater flexibility in either synchronous or asynchronous operation.

The CCITT has stated that, within an H channel, the user has complete freedom in formatting the data.

2.3 Multiplexing

The total available bit rate has to contain not only the video and its associated error protection data, but also the sound (and its associated data), test signals, teletext or other data broadcast services [CCIR 1986-90 b]. Error protection data appropriate for the latter services must also be included to achieve an adequate level of quality and transparency.

The multiplex requirements of data rates near 68 Mbit/s are still under discussion but at data rates near 140 Mbit/s the EBU has indicated that, for 625 line television long distance transmissions, its general requirements would be:-

- Video, coded at approximately 6 bits/sample for the active picture only (plus necessary video framing) 125 Mbit/s
- Error protection for video, based on BER of 10⁻³ for a period of greater than 10 seconds (corresponding to complete failure of the digital network) (Note 1) 8 Mbit/s
- Ancillary data lines: 3 per field and 8 bits/sample 2 Mbit/s
- Sound/data/ancillary signal multiplex (including error-protection) 4 Mbit/s

* See _____ Report 1241.

Note 1 - Proposals for other error-rate objectives are being considered.

The EBU has made the following proposals [CCIR, 1986-90c] for the appropriate bit-rate budgets. In the synchronous approach the use of a data rate corresponding to a standard ISDN level for the sound and data package seems to be in accordance with a general ISDN point of view. In that case, a 64 kbit/s structure can be used, which may be useful in data formatting.

In the asynchronous approach, a 2048 kbit/s package using the Recommendations G.704-G.732 frame also offers a 64 kbit/s structure.

As the proposed H4 access level has a bit-rate smaller than that available for the fourth hierarchical level, only one sound/data package can be included. We are thus led to the following proposals:

a) Synchronous multiplex

As the proposed H4 access level of 135 168 kbit/s equals 2112 x 64 kbit/s, it is convenient to use a 2112 bits frame length (1 bit per frame = 64 kbit/s), giving the bit allocation per frame below:

Framing, signalling and data (teletext)	: 16 bits (1 024 kbit/s)
Sound and programme-related data	: 30 bits (1 920 kbit/s)
Video (Note 2)	: 2 066 bits (132 224 kbit/s)

Frame length	: 2 112 bits (135 168 kbit/s)

b) Asynchronous multiplex

The use of the same frame length seems to be attractive, as such a frame can be synchronized in the ISDN network.

As the 2048 kbit/s sound and data package occupies exactly 32 bits in the frame, a positive, null, negative justification has to be used.

A frame structure derived from Rec. G.754 can be used:

Framing, signalling and data (teletext)	: 16 bits (1 024 kbit/s)
Justification control bits (2 x 3)	: 6 bits (384 kbit/s)
Justified slots (2 x 1)	: 2 bits (128 kbit/s)
Sound and data (including slots for negative justification)	: 32 bits (2 048 kbit/s)
Video (Note 2) (including slots for negative justification)	: 2056 bits (131 584 kbit/s)

Frame length	: 2 112 bits (135 168 kbit/s)

Such a frame structure may be preferred to that in a), because it is more flexible and offers the possibility of providing either synchronous or asynchronous operation.

Note 2 - Including error protection and ancillary data. Precise proposals for the division of the bit rate between these various functions for transmission according to G.751 and G.707 are given in the Annex to Recommendation 721.

Audio capacity and ancillary data requirements may vary according to the needs of users.

Therefore, the frame structure that is to be used in the transmission of digital television signals at 140 Mbit/s must allow such flexibility.

To accommodate this requirement, the EBU considers that for Region 1, the useful data capacity of two 2048 kbit/s bit streams would be appropriate to carry the audio and ancillary data signals alongside the video signal within the 140 Mbit/s bit stream [CCIR 1986-90d].

2.4 Performance objectives

A detailed description of the user requirements for digital television transmission including system objectives for contribution purposes, the assessment methods for bit rate reduced systems, the performance in the presence of digital errors, the target system requirements, the requirements for sound signals in a digital multiplex which includes television sound, guidelines for the bit rate budget at bit rates around 140 Mbit/s, encryption and the relative timing of sound and vision is given in Report 1211.

There are three main areas in which the performance of the transmission system needs to be defined:

2.4.1 The Basic Picture Quality of the System

At bit rates of around 140 Mbit/s, and possibly also at 68 Mbit/s, it should be possible to achieve a picture quality closely approaching that of the input 4:2:2 - sampled signal. Thus at both of these bit rates it is reasonable to specify that an impairment grade of 4.5 should be achieved after 3 transmission codecs with 4:2:2 interfaces in tandem.

2.4.2 Downstream Processing Capability

These bit rates are likely to be used on contribution links, in which case processes such as the following may be used at the end of the link:

- colour matte
- picture expansion and compression
- slow motion

For such downstream processing it is considered appropriate that an output signal impairment grading of 4.25 should be achieved, referred to the signal produced when using the original 4:2:2 signal as an input to the processor.

2.4.3 Performance in the Presence of Transmission Errors

Codecs designed for television transmission are normally equipped with error protection circuits. These will generally cope with infrequently occurring errors, but as the error rate increases the operation of decoding circuits will be progressively impaired. Studies are in hand to determine the random error rates for which an error performance test will yield a mean impairment grade of 4.5.

If a prolonged burst of errors occurs, it is possible for the operation of the receiving terminal to be so disrupted that the effect is similar to that of removing the input signal altogether. It is necessary to determine how rapidly the system will recover from such an event, and new Report 1211 gives suggestions as to how this aspect of performance might be defined.

2.5 Quality Assessment

IWP 11/4 is studying the choice of methodology for bit-rate reduction systems. The objective is that results, derived independently in a number of laboratories throughout the world, can be fairly compared. Report 1211 suggests test material for the evaluation of 30-33 Mbit/s codecs (which is believed also suitable for assessing codecs at high bit rates). The initial list consists of still pictures, moving picture sequences, moving colour-matte foregrounds and colour-matte backgrounds. A reduced set of pictures is provided for the assessment of characteristics in the presence of errors. Downstream processes, such as colour matting or colour keying, are critically sensitive to impairments. For the moment, these processes are difficult to specify objectively, therefore, in these circumstances, picture quality evaluations can only be carried out by means of a specific reference process, such as an analogue RGB chroma keyer.

2.6 Error Processing

The failure characteristic is an important facet of bit-rate reduction system performance. At the error rates and with the distribution of errors likely to be encountered in practice, the decoded pictures should have acceptable quality, both when viewed directly and when subjected to further ("downstream") processing such as picture expansion and colour matte. The effects of short duration errors should not be extended due to the need for synchronization. In this way the error management systems for the video, audio and data could be independent. The degree of protection required for the data is heavily dependent on the application (eg., data used for system control might require powerful protection). It should be noted that very short duration defects are less tolerable in sound than in the picture. In the design of schemes to be used when the performance of the link deteriorates, the switching to the protection link should not cause a disturbance in the video, sound or data signal. Interruptions lasting several seconds or more resulting from system recovery from gross errors which cause a complete loss of signal to the decoder are intolerable (Report 1211).

2.7 Implementation Considerations

In the design of a codec for the transmission of television pictures and sound, a number of conflicting factors must be considered. The optimisation must relate quality objectives, network capability and technological constraints to arrive at a solution that is acceptable from the viewpoints of performance, complexity and cost. Report 1211 states that in general, the encoding method selected should achieve a balance between performance and complexity for the anticipated application. The complexity of the codec should be such that it can be implemented with available, or nearly available, technologies at a cost reasonably related to the transmission costs for the intended application.

An example of a design that meets these criteria is reported in [CCIR 1986-90e] that describes the implementation of a DPCM codec. This codec can be achieved with reasonable effort by the use of integrated circuits of well known component families operating at high processing speed (AS-TTL, ECL). But, instead of cascading arithmetic, logic and memory elements in such a conventional way to produce a DPCM loop with sufficiently low delay time, an alternative method of building up prediction codecs with relatively inexpensive hardware is based on the use of look-up tables with an appropriate address width to store the results of all possible coding states. By forming this address from both the PCM input value (eg. 8 bits) and the PCM prediction value (eg. 8 bits), the output code word (eg. 6 bits) and the prediction value (eg. 8 bits) for the next picture element are predetermined definitely. In principle, the contents of the look-up tables used as such an arrangement can be chosen completely arbitrarily thus giving a high degree of freedom in optimising the coding quality of DPCM codecs.

A number of additional practical considerations in the selection of a codec algorithm are summarized by the Chairman of IWP 11/7 in [CCIR 1986-90f]. These include:

- the cost and availability of licensing arrangements for any possible intellectual property rights;
- the overall cost of operation including both the codec and the related transmission facilities;
- the effects of complexity on reliability and maintainability;
- flexibility to adapt to other transmission rates or video signal formats;
- compatibility in both 525/60 and 625/50 formats for international programme exchange;
- future development for other applications such as HDTV or secondary distribution.

It is noted that responsibility for these considerations is not clearly defined in Decision 18-5.

2.8 Intellectual property rights (IPR)

It is essential to ensure that licences for any patents, or other IPR, embodied in equipment that would have to be used in order to conform to a proposed Recommendation on BRR codec characteristics, will be freely available.

Owners of such patents should submit a statement to this effect for which a standard form has been prepared in the CCIR Secretariat, based on CCITT experience and taking into account legal advice from experts in the World Intellectual Property Organization (WIPO), and from the Legal Advisor of the ITU [CCIR 1986-90g].

3. Systems for use at a bit rate of approximately 68 Mbit/s

Document [CCIR, 1986-90h] proposes a coding process for use at around 68 Mbit/s based on DPCM with multi-element fixed prediction and adaptive quantization. This system, the details of which are under study, is summarized in Table I, column A. The subsequent information in column B, DPCM with adaptive prediction, and column C, discrete cosine transformation (DCT), provides other alternatives.

It is hoped that these systems will give a performance comparable with that obtainable near 140 Mbit/s, but with increased coder complexity. The outputs of two such codecs could be multiplexed together for transmission at 139 264 kbit/s; such codecs could also be used in the B-ISDN, where the use of ATM will in principle allow the transmission of signals of any bit rate.

4. Systems for use at a bit rate of approximately 140 Mbit/s

Transmission systems at about 140 Mbit/s are of special interest for point-to-point connections (inter-studio links) where the main system requirements are broadcast picture quality and the preservation of 4:2:2 downstream processing capability which can be fulfilled at a moderate level of

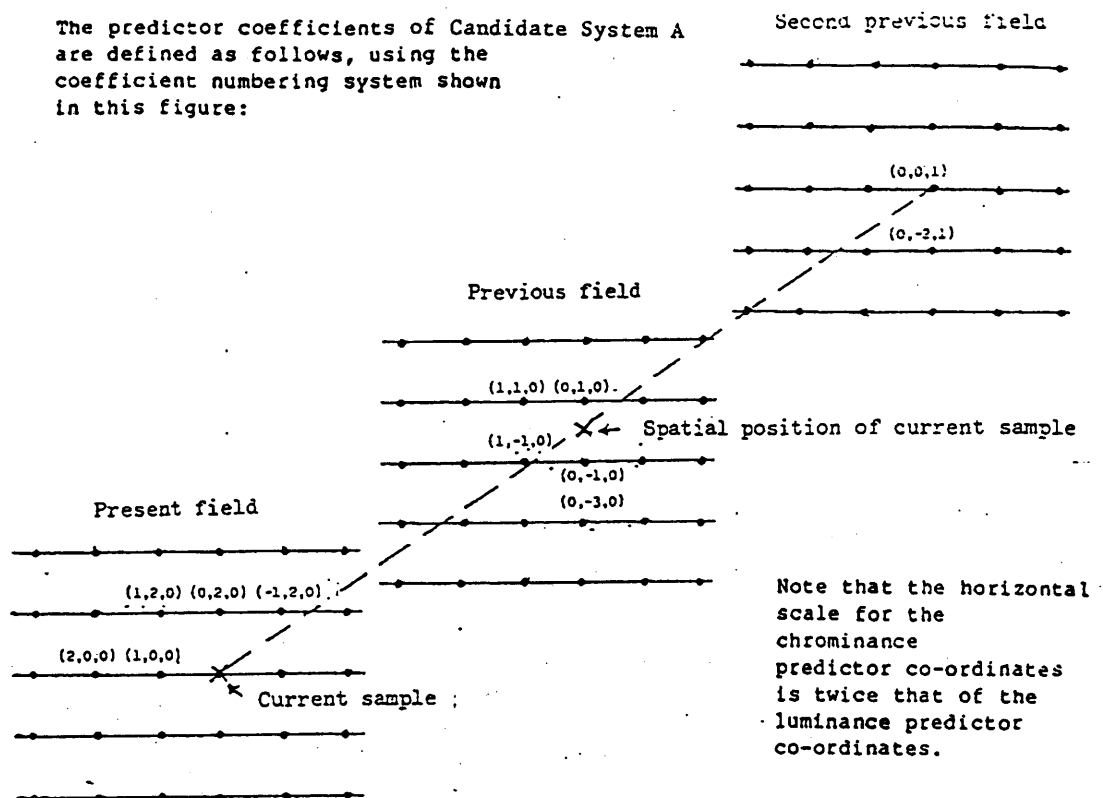
codec complexity. Different techniques - PCM, DPCM and hybrid DPCM have been applied to the coding of component signals at about 140 Mbit/s, but only coding schemes maintaining the spatial and temporal resolution of the 4:2:2 signal according to Recommendation 601 might fully meet the performance requirements of contribution links with respect to downstream processability.

Recommendation 721 ——— derived from the reference system has been adopted by CMTT. This Recommendation includes two different approaches to build up an appropriate multiplex for the transmission of component-coded video signals for contribution-quality applications.

TABLE I - Candidate systems for use at approximately 63 Mbit/s

SYSTEM	A	B	C
Pre-processing	None	None	None
Blanking Delete			
H	Yes	Yes	Yes
V	Yes (data normally found here carried in separate part of multiplex)	Yes	Yes
Subsampling			
H	None	None	None
V	None	None	None
Coding Scheme	Non-adaptive DPCM Multi-element fixed prediction, as defined in Fig. 1, possibly with quantising error feedback	Adaptive DPCM	Transform - DCT
Variable-Length Coding	Buffer occupancy controls switching between up to 8 quantizers for each component. About 60% of bit rate allocated to luminance samples, 40% for colour difference samples. Buffer occupancy and quantiser selection signalled to decoder	Yes	Yes
Post Processing	None	None	None
Error Performance	Under study; thought that 7% bit-rate overhead will allow acceptable performance at channel error rate of about 5×10^{-5}	Not known	Not known

The predictor coefficients of Candidate System A are defined as follows, using the coefficient numbering system shown in this figure:



Luminance Predictor

Coefficient Number		Value
1	0 0	1.000
2	0 0	-0.437
3	0 0	0.235
0	2 0	0.327
1	2 0	-0.195
0	-1 0	0.201
1	-1 0	-0.158
0	1 0	0.172
1	1 0	-0.172
0	0 1	0.460
1	0 1	-0.497
2	0 1	0.277
3	0 1	-0.149
0	2 1	-0.291
1	2 1	0.198

Chrominance Predictor

Coefficient Number		Value
1	0 0	0.500
-1	2 0	0.194
0	2 0	0.272
0	0 1	0.655
1	0 1	-0.317
-1	2 1	-0.137
0	2 1	-0.191

FIGURE 1 - Details of Predictors for Candidate System A near 68 Mbit/s.

5. Results of the evaluations by IWP 11/7 of the proposed 140 Mbit/s codecs

IWP 11/7 made a series of subjective evaluations of the 140 Mbit/s coding algorithm described in draft new Recommendation AM/CMTT. IWP 11/7 also compared the results achieved with the user requirements for such codecs that have been agreed by CCIR Study Group 11 and CMTT (CCIR 1986 - 90 p).

The evaluations of basic picture quality after three codecs in tandem showed that the codec is virtually transparent to the source quality for all the test sequences used at both 4H and 6H. The codec thus meets the user requirements in this area.

For the evaluations of downstream processing capacity, there were some problems with the test tape, which made it difficult to interpret all the results. For the results of the correctly recorded sequences, the additional impairment in the processed picture, due to the codec, was always within the limits specified in the user requirements (about three quarters of a grade).

Concerning the failure characteristic, one of the codecs used a RS FEC (codec 1) and the other a BCH FEC (codec 2). The performance of the codec at 6H with the BCH FEC is given below in Figure 2. This meets the user requirements (Impairment Grade 4 at BER of 10^{-4}). The codec with the RS FEC nearly met the requirements. It should be noted, however, that the RS FEC will not be included in the specification, rather an RS-based product code is proposed.

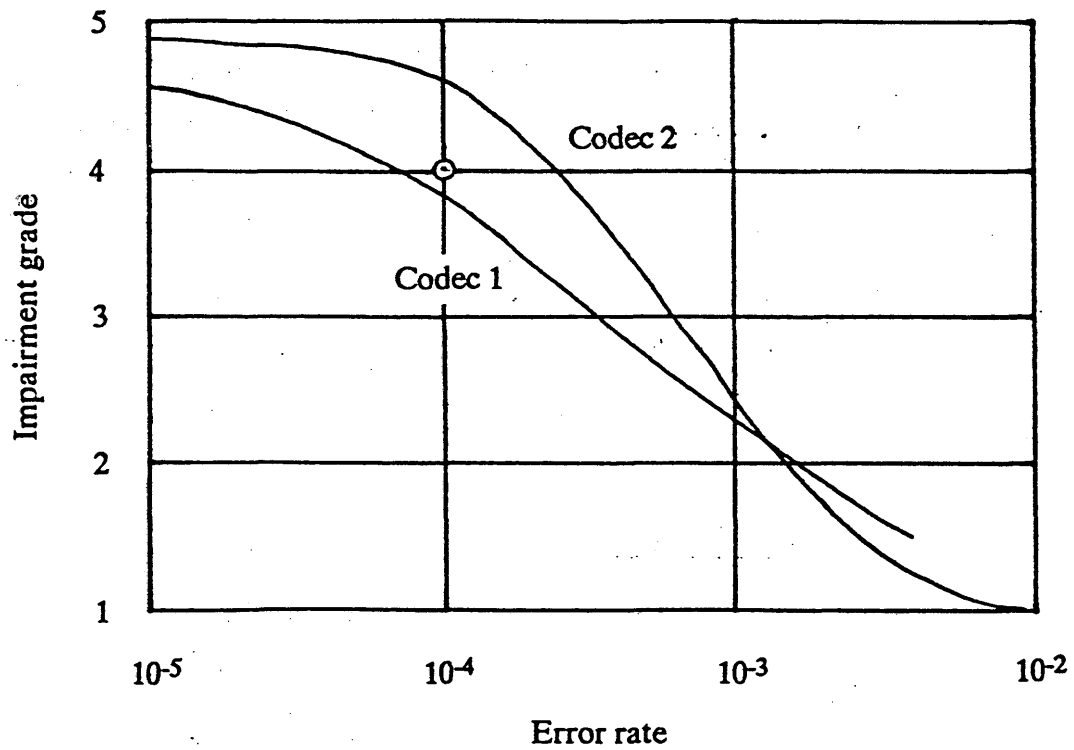


Figure 2 - Codec performance at 6 H

The codecs tested did not meet the user requirements in respect of recovery time. This is considered to be a lower priority requirement, and not sufficient to withhold endorsement. Nevertheless, Administrations should study ways of improving this.

6. CONCLUSIONS

At a bit rate of approximately 140 Mbit/s, the quality objectives for both contribution and distribution links could be met by the reference system as described — in — Recommendation 721.

At a bit rate of approximately 68 Mbit/s it is possible that the performance objectives will be met using the techniques of DPCM and variable length coding. Other techniques are also under investigation and tests to verify the performance and complexity of each approach will be required before a firm recommendation can be made.

A need also exists to align the bit rates of the codecs with the as yet undefined ISDN access levels, while retaining the necessary flexibility to interconnect with the current hierarchy of transmission bit rates. Firm definition of the multiplex formats must await these definitions.

REFERENCES

CCIR Documents

[1986-90]: a. CMTT/161 (IWP CMTT/3); b. IWP CMTT/2-78 (JIWP 10-11/5); c. IWP CMTT/2-40 (EBU); d. CMTT/243 (EBU); e. IWP CMTT/2-42 (Germany (Federal Republic of)); f. IWP CMTT/2-77 (IWP 11/7); g. IWP CMTT/2-105 (IWP CMTT/2); h. IWP CMTT/2-35 (United Kingdom); j. IWP CMTT/2-90 (Germany (Federal Republic of)); k. IWP CMTT/2-75(Rev.) (Germany (Federal Republic of)); l. IWP CMTT/2-89(Rev.) (Germany (Federal Republic of)); m. IWP CMTT/2-100 (France); n. CMTT/239 (CCITT Study Group XVIII); o. CMTT/253 (France); p. IWP CMTT/2-128 (IWP 11/7).

BIBLIOGRAPHY

BOSTELMANN, G. [1974] - A Simple High Quality DPCM Codec. Nachrichtentechn. Z., Vol. 27 No. 3, pp. 115-117.

KUMMEROW, Th. [May, 1981] - Der Einfluss von Übertragungsfehlern bei der DPCM-codierten Übertragung von Farbfernsehsignalen (The Influence of Transmission Errors on Digital-Encoded Colour Television Signals). Heinrich-Hertz-Institut Berlin, Technischer Bericht Nr. 214, (ISSN 0172-8873)

VAN BUUL, M.C.W. [1978] - Hybrid D-PCM, a combination of PCM and DPCM. IEEE Trans. Comm., Vol. COM-26 No. 3, pp. 362-368.

Annex I.

LIAISON STATEMENT TO CCITT STUDY GROUP XVIII

CMTT informs CCITT Study Group XVIII that it has adopted Recommendation 721—on the digital coding of component television signals for transmission at bit rates at about 140 Mbit/s.

The multiplex structure has still to be agreed but two proposals are under study; they both aim at transport of the entire signal multiplex within the 139 Mbit/s level of the digital hierarchy (Recommendation G.702).

CMTT is working on the assumption, that the total bit rate for both proposals could be carried as payload in an SDH transmission system (Recommendation G.707).

In case of an ATM based network, a rate of 135.168 Mbit/s is assumed to be available as stated in CMTT Document CMTT/162, Annex II, (Liaison statement to CCITT Study Group XVIII, Interregnum Meeting of ISDN Experts, San Diego).

It must be emphasized, however, that acceptance of this lower bit rate would imply a reduction in the amount of the sound/data/FEC information that can be transmitted.

REPORT 1240*

TELEVISION TRANSMISSION IN AN ATM-BASED NETWORK

(1990)

1. Introduction

Broadband aspects of the ISDN (B-ISDN) are being studied by CCITT Study Group XVIII for a future customer-switched digital network. The proposal of Study Group XVIII for B-ISDN as described in Recommendation I.121 is that the target transfer mode is the Asynchronous Transfer Mode (ATM), in which the data is transmitted in a series of fixed size blocks called cells. Packet-switched networks already exist for the transmission of digital data for non-real time services (for example, the exchange of information between computer databases). In this instance, if a packet is corrupted or lost, the receiving terminal can request that the particular packet be retransmitted. Recommendation I.121 however envisages that the B-ISDN will carry all the telecommunications services provided in the future including real-time services such as telephony, videoconferencing, videophony as well as television and sound contribution and distribution services. For these real-time services, if a cell is corrupted or lost, retransmission of cells is not possible and so degradation of the signal may occur.

The main advantages claimed for ATM is that the network and switches are no longer bit-rate and service specific; in the B-ISDN all services (including future new, and as yet unspecified services) are expected to be carried, and a common user-network interface will exist for all services. Many of the important parameters of B-ISDN have still to be specified. However, an ATM-based network will introduce some effects not experienced in synchronous networks, such as cell delay jitter and occasional cell loss.

This report describes some of the advantages and disadvantages of television transmission in an ATM-based network, and indicates areas where further study is required. Some matters requiring the attention of the CCITT are given in the annex.

2. ATM for television transmission

An ATM-based network will, in principle, provide the user with whatever bit rate is required (within the constraints of the interface and the network), so that broadcasters, for example, could decide on the optimum picture quality/transmission engineering compromise for contribution, distribution and ENG applications. Additionally, new television services at different bit rates

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

could be transmitted over the network through the same user-network interface. With continuing improvements in picture coding algorithms, and with advances in technology allowing more complex algorithms to be implemented, service providers could, in the future, offer either an improved quality of service at the same average bit rate, or the same quality of service at a lower average bit rate.

Many of the more efficient picture coding schemes in use today employ:

- conditional replenishment - where only those parts of the picture that have changed are coded and transmitted; and
- statistical coding - where the shortest codewords are allocated to the most commonly occurring values.

Both techniques result in the data in the video encoder being generated at a variable rate. When the codec is part of a fixed bit rate transmission network, a buffer (which must not be allowed to overflow) is used in the encoder to match the variable data rate generated to the fixed rate of the network. If the buffer is in danger of overflowing, the amount of data being generated is deliberately decreased by, for example, selecting a coarser quantizer until the buffer level is again within its specified range. Such a coding scheme can be considered as a fixed bit rate/variable quality scheme.

An ATM-based network will in principle have the flexibility to provide additional transmission capacity when required, and could allow the development of a variable bit rate/constant quality coding scheme.

Two illustrations of this flexibility are:

- a maximum bit rate is guaranteed for a particular application, which may not necessarily use all its permanent resource allocation. An example of this might be a contribution encoder to which a resource of 34 Mbit/s is allocated, but which uses a part, only, of this bit rate for some of the less critical picture sequences;
- a minimum bit rate is guaranteed to provide a picture quality which is considered acceptable for critical picture sequences. Network loading allowing, the (e.g. distribution) encoder will operate above this threshold. This does imply that the encoder will constantly receive information from the network on the bit rate that is available to it.

Further work is needed to confirm the above hypotheses.

3. Performance considerations of ATM for television transmission

3.1 Cell loss

In an ATM-based network it is possible for cells to be lost, although this will occur with probabilities of the order of 10^{-9} to 10^{-8} . This would be acceptable for many services other than real-time video services. As an example, [CCIR 1986-90a], for a transmitted bit rate of 34 Mbit/s and a cell length of 48 octets, the mean time between two cell losses for a cell loss probability of 10^{-8} is 18 minutes, which is significant, and it is no more than three hours for a cell loss probability of 10^{-9} .

There are three factors which contribute to cell loss:

- i) header errors due to transmission errors, and consequent cell discard by error checking to minimize misdelivery;
- ii) overflows or underflows of the smoothing buffer in the adaptation layer, due to large variations in queueing delay;
- iii) buffer overflows in the ATM layer due to instantaneous cell traffic overload.

In [CCIR 1986-90b] two possible approaches to dealing with cell loss are described:

One deals with a two-channel transmission scheme in which all the information which is essential for the reconstitution of the picture is transmitted with a high "Quality of Service" in one channel, and the remaining information is transmitted with a lower "Quality of Service" in the second channel which, due to network loading, may be subject to cell loss.

The second approach is to ensure that an error corrector is employed whose design would accommodate occasional burst errors lasting for the duration of a data cell. CCITT Study Group XVIII have agreed a cell size of 53 octets, comprising 48 bytes of information plus five header bytes. A burst error corrector for a cell this size would be much more complex than current designs which can typically correct erroneous data bursts up to 32 bits in length using bit interleaving techniques. The overall signal delay introduced by such processing could cause further problems.

Another example of a cell loss compensation scheme using a Reed-Solomon random error corrector with bit interleaving is described in [CCIR 1986-90c]. It is suggested that essentially cell loss free transmission will result from such a correction scheme when the original cell loss probability of the network is 10^{-8} .

Further study of cell loss compensation is required.

3.2 Cell insertion

Transmission errors affecting the cell header can result in the intrusion of extraneous cells. This should be a much rarer event than that of cell loss. Cell insertion could be detected by the use of a continuity index in the information field of each cell.

3.3 Cell delay jitter

As the network consists of a series of queues, the cell routing delay between the transmitter and the receiver is variable. The jitter that results can be compensated for by using a receiving FIFO (first in first out) buffer in the decoder. To ensure adequate reception buffer dimensioning it is essential to characterize the maximum network-induced jitter that might be expected. The orders of magnitude currently indicated (from 0.1 to 1 ms) result in acceptable storage capacity requirements.

3.4 Synchronization difficulties

The variable cell-routing delay in an ATM-based network is responsible also for problems relating to inter-service synchronization. For instance, the lag tolerances (20 ms lead, 40 ms lag) between sound and video within the same service will be respected only if the network routing delay is less than the lower limit (i.e. less than 20 ms). The use of the same physical path for both service components would minimize their temporal divergence at the receiver. A further refinement would be the introduction of synchronizing markers in the bitstreams, especially of sound services.

Two options for the synchronization of transmitter and receiver could be envisaged:

In the first, a high precision (e.g. to within one part in 10^9) network clock drives local clocks at both the transmitter and receiver. This typifies the plesiochronous operation found in synchronous networks, and the same methods (e.g. justification) may be applied to transfer clock information.

In the second, the concept of a network clock is absent. Nevertheless, the transmitter has to send timing information to synchronize the decoder, and this is achieved by emitting cells with the necessary timing data. Various methods are used, timing cells either being transmitted concurrent with the encoder's timing markers (at the beginning of rows of n lines, for example) or at regular intervals in which case a phase locked loop is used to synchronize the decoder.

Solutions not using timing markers may also be contemplated, in which an image store operating in synchronizer mode in conjunction with a bigger receiving buffer can act as a compressed image synchronizer.

3.5 End-to-end transfer delay

It is understood from liaison statements from CCITT Study Group XVIII that the end-to-end transfer delay of an ATM-based network will be greater than the delay of a synchronous network.

CCIR Documents

REFERENCES

[1986-90]: a. CMTT/214 (France); b. CMTT/182 (EBU); c. CMTT/171 (Japan).

BIBLIOGRAPHY

DEVAULT, M., COCHENNEC, J.Y. and SERVEL, M. [December 1988] - The prelude ATD experiment: Assessments and future prospects. IEEE Journal on selected areas in communications, Vol. 6, No. 9.

KISHIMOTO, R. et al. [18 June 1989] - Generation interval distribution characteristics of packetized variable rate video coding data streams in an ATM network. IEEE Journal, Vol.7, No.5.

TURNER, J.S. [March 1986] - New directions in communications or, which way to the information age? Proc. Int. Seminar on Digital Comm., Zurich.

VERBIEST, W. and DUPONCHEEL, M. [January 1987] - Video coding in an ATD environment. GSLB Seminar on Broadband switching, Albufeira.

ANNEX

Matters relating to ATM television transmission
that the CMTT wishes to draw to the attention of the CCITT

In accordance with information previously received from CCITT Study Group XVIII, broadcasters have been developing fixed bit rate codecs for television transmission and during the 1986-1990 study period, IWP CMTT/2 has been attempting to standardize codecs for the user rates equivalent to the H2 and H4 access levels. (Some of the basic performance characteristics are given in ——— Recommendations 721 and 723.)

It is likely that such codecs will be in widespread use by the time that ATM techniques are fully implemented in networks. CMTT therefore assumes that the standardized fixed bit rate codecs will be suitable for use on ATM based networks. This may entail the use of specific terminal adapters incorporating cell loss detection and, possibly, correction.

It may be that advantages can be gained by designing codecs specifically for transmission in an ATM environment. These codecs will then take full account of the characteristics of the ATM-based network. The design of these codecs would require that answers to the following questions be available:

- 1) what parameters have to be negotiated between the user and the network during call set-up? It is understood that these might include mean bit rate, peak bit rate (over what time interval?) and Quality of Service parameters;
- 2) what information (if any) can be specified by the user to tell the network how to treat individual cells forming part of a single call? It is noted for example that some proposals for VBR (Variable Bit Rate) coding would require the network to recognize and act on cell Quality of Service parameters specified by the user;
- 3) what information (if any) is fed back from the network to the user during the call? It is noted for example that some proposals for VBR coding would require feedback of information on congestion in the network to adapt the coding to make best use of the bit rate available;
- 4) what is the maximum cell loss probability (if this is not specified at call set-up);
- 5) what are the characteristics of cell arrival time jitter;
- 6) what provision is made for the end-to-end transfer of timing information? For reconstruction of the video signal a clock with jitter less than 1 ns is required (see Report 967); it may be difficult to regenerate such an accurate clock from the mean cell arrival rate;
- 7) what will be the basis of charging in an ATM-based network? It must be understood that the object of bit rate reduction is to minimize transmission cost. In a network using a specified range of fixed bit rates it may be assumed that cost will be monotonically related to bit rate. But in an ATM based network this may no longer be true; parameters such as signal burstiness (peak-to-mean ratio) and priority specified could have as great an influence on charging as mean bit rate?



REPORT 1239

STANDARDS FOR DIGITAL SECONDARY DISTRIBUTION SYSTEMS

(1990)

1. Introduction

Secondary distribution must be understood as the delivery of television programmes to consumers. Particular attention must be paid to the digital delivery of television signals over future Broadband Integrated Services Digital Networks (B-ISDN) which will be used for both communicative and distributive services including a whole range of visual services from videotelephony to television distribution according to future high definition standards.

Historically, it was the realization of this trend which caused the CMTT, at its Interim Meeting in 1987, to propose a change in its terms of reference, so that its perceived responsibility for studying digital delivery of television programming especially over optical fibre networks would be unequivocally resolved. The CCITT formally approved the proposed change at its IXth Plenary Meeting held in Melbourne, October 1988, subject to certain provisos [CCIR, 1986-90a].

To ensure that these studies are aligned with those on the broadband ISDN, the Plenary Assembly decided that CMTT should undertake its projected video coding studies for secondary distribution with the objective of consistency with the broadband ISDN being specified in CCITT Study Group XVIII and with coding for videotelephony in CCITT Study Group XV. Consistency with the video coding for broadcasting services should also be considered.

Different picture transmission requirements have hitherto led to the consideration of different bit rate reduction algorithms for each application and each transmitting channel. For instance, CCITT Study Group XV, IWP CMTT/2 and the ISO have proposed often similar but not identical algorithms for videotelephony, television contribution and distribution and still picture transmission respectively.

Greater compatibility between bit rate reduction algorithms will be required for the secondary distribution of pictures (mainly, but not exclusively in the broadband ISDN).

2. Interconnection requirements

There is already an increasing need for interconnection between services and many administrations require the new HDTV service to be compatible with the television service, i.e., for a television receiver to be connectable to an HDTV source as cheaply as possible. This need is likely to develop with the arrival of new visual services in the user's premises (still pictures,

videotelephony, etc.). Some terminals will even be of the "multiservice" type and will permit the reception and simultaneous display (by windowing) of several pictures (e.g., videophone conference picture insert on a television or HDTV screen). Because it uses ATM technology, the broadband ISDN will in addition permit the interconnection of terminals originally designed to operate at different bit rates; in other words, terminals (and bit rate reduction algorithms) will no longer be tied strictly to a single channel bit rate.

The need for interconnection imposes certain constraints on the picture sampling structure. In particular, practical consideration should, as has been proposed for HDTV, be given to a hierarchy of resolution standards based principally on progressive scanning. According to this proposal, the HDTV image has twice as many lines and points per line as a conventional television image. It is even possible to extend this hierarchy to lower resolutions and consider a 1/4 TV format. Such a system makes sense only if television and HDTV standards have the same picture frequency.

Several constraints, sometimes contradictory, have to be taken into account to design a transmission system for secondary distribution. They mainly concern:

- Bit rates

It is important to consider reduced bit rates to allow the distribution of numerous channels to the end users.

- Complexity

The coding methods for secondary distribution should result in low cost decoders to allow an economic introduction of the service.

- Picture quality

Users must be satisfied with the quality of the decoded signal. Precise criteria must be defined; nevertheless, it seems clear that the systems intended for secondary distribution of digital television must provide a better quality than the existing analogue composite systems.

3. Compatibility

In order to meet the need for interconnection between services using different picture formats, secondary distribution standards must ensure:

- upward compatibility (i.e., to enable a higher-resolution receiver to decode a signal transmitted with a lower resolution),
- downward compatibility (i.e., to enable a lower-resolution decoder to decode a higher-resolution signal as cheaply as possible).

Since the complexity of a decoder depends on the number of pixels to be processed during a frame period, downward compatibility is obviously more difficult to achieve if it is not taken into account in the compression algorithm.

Coding methods for secondary distribution of digital television should be defined with the objective of conformity with the features of the B-ISDN developed in CCITT Study Group XVIII.

They should be designed with the objective of the greatest compatibility with primary distribution and contribution codecs and also with the codecs defined for other communicative applications, as defined in CCITT Study Group XV.

4. Systems for secondary distribution of digital television

Document [CCIR 1986-90b] presents the main features of four compression systems based on the Discrete Cosine Transform (DCT) intended for secondary distribution applications at bit rates around 15 and 30 Mbit/s. One of these systems is a simplification to the intra-field mode of the hybrid DCT coding algorithm described in Document [CCIR 1986-90c]. Subjective evaluations were carried out to compare these four systems using a double stimulus method with graphic scale, including a high anchor point (4:2:2 reference) and a low anchor point (PAL coded-decoded sequences). The results of the tests show that the four algorithms provide a visual picture quality better than PAL and sometimes very close to 4:2:2, and therefore may fulfil the picture quality requirements for secondary distribution.

Document [CCIR 1986-90d] reports the results of subjective assessments of the hybrid DCT algorithm, based on computer simulations at different bit rates and from them the conclusion can be drawn that the performance at 10-15 Mbit/s is suitable for secondary distribution.

A system based on purely intra-field DCT has been studied by CMTT/2 and its specifications are in [CCIR 1986-90e]. This system has not been proposed for contribution purposes, but a codec at 34 Mbit/s has been compared with the existing prototype codecs during the tests undertaken by IWP 11/7. The results of the tests are in [CCIR 1986-90f]. The basic quality was satisfactory for most of the test sequences, the performance in the presence of errors was good and the system recovery time was satisfactory.

5. Codecs and display devices

Conformity of video coding studies for secondary distribution by CCITT Study Group XV and CMTT with the B-ISDN studies of CCITT Study Group XVIII will allow the advantages available through a multiservices network to be extended to the end user by minimizing the number of video terminals needed to access a range of interactive and distributive video and still image based services. The objective is to achieve the highest level of service integration through minimizing the number of coding techniques used across a wide range of video services and maximizing commonality of display devices.

To achieve this objective there will be a requirement for video terminals to be able to present video and still image material from a range of services other than that of their primary application, but within the limits of the display resolution. Thus, for example, a videotelephone terminal should be able to access and present (at the quality limit imposed by its low resolution display) a video signal originating as high resolution TV quality. The use of layered coding may be necessary to allow a terminal to extract only that part of the coded video signal that it is capable of representing. This layered coding structure means that information describing the different levels of image resolution are separately transmitted in a way that permits their selective

reception and reconstruction at a decoder. The coding methods may also need to be matched to the characteristics of ATM, the cell-based information transfer mode of B-ISDN.

Use of a common display device goes some way toward rationalization of a user's terminal needs for access to multiple video services. However, when this is combined with a single common decoder utilizing a layered coding structure, the objective of maximizing the commonality between interactive and distributive services can best be realized. An important issue however, is where and to what extent is conversion between different video formats to be performed. The question of whether this function should be performed at the source or in the display or a combination of both, requires further study.

Another important issue requiring further study is the definition of an interface between the codec (or other video sources) and the display, including the fundamental issue of whether the interface should be analogue or digital.

Note should be taken of the report of CCITT Study Group XVIII on Question 22/XVIII, "Broadband - ISDN influence on principles for video coding", from its June 1989 meeting [CCIR, 1986-90g].

Note should also be taken of the liaison statement from CCITT Study Group XVIII to IWP CMTT/1 coming from the June 1989 meeting, covering the role relationships between CMTT, CCIR Study Group 11 and CCITT Study Groups I, XV and XVIII for video coding studies and proposed amendments to CMTT Decision 18-5 to account for these role relationships [CCIR, 1986-90h].

6. Conclusion

Given the increasing need for compatibility between various visual services which will be eventually provided over future digital networks for distribution to the home, it will be necessary to ensure that coding algorithms have the following characteristics:

- i) simple but compatible coding algorithms in order to minimize the costs of terminal equipment;
- ii) an image quality which is equivalent to, or better than that presently available at the home;
- iii) compatibility between individual services and with services derived from primary distribution systems;
- iv) use of low bit rates to offer the widest possible range of services to the home.

REFERENCES

CCIR Documents

[1986-90]: a. CMTT/164 (CCIR); b. CMTT/217 (France); c. IWP CMTT/2-66 (France); d. IWP CMTT/2-76 (France); e. IWP CMTT/2-58 (France); f. IWP CMTT/2-97 (France); g. IWP CMTT/2-109 (CCITT Study Group XVIII); h. CMTT/221 (CCITT Study Group XVIII).

REPORT 1241*

LIAISON BETWEEN THE CMTT AND OTHER STUDY GROUPS OF THE CCITT
ON MATTERS RELATED TO DIGITAL TRANSMISSION

(1990)

1. Introduction

The CMTT previously liaised with the other Study Groups of the CCITT only by means of Special Rapporteurs. These drew the attention of the CCITT to the relevant Questions and information drawn from the texts of the CMTT Recommendations and Reports.

Because of the speed with which the CCITT has been endeavouring to reach agreement on the parameters of digital transmission networks and, in particular, the broadband ISDN, the CMTT set up in 1985 (see Decision 68) an Interim Working Party (IWP CMTT/3) charged with the task of formulating and communicating to the CCITT all the necessary information and opinions, during the period between the full meetings of the CMTT.

On its part, the CCITT issues liaison statements responding to CMTT comments and providing information of particular interest to the CMTT. There are also Special Rapporteurs.

2. Basic requirements of CMTT

In order to put the CMTT comments into a correct context, an outline of the general objectives and constraints of the CMTT interests were explained to the CCITT.

For example, the importance of financial aspects has been emphasized, and it has been stated that the total transmission cost for a broadcaster using a digital network must be kept within the range of costs at present applicable to analogue networks.

In considering the future requirements for television, the CMTT attention is concentrated on the needs of 525-line and 625-line systems using component-coding based on CCIR Recommendation 601. However, account will also have to be taken of the future need to transmit high-definition television signals.

The possible need to ensure privacy of a digital transmission may introduce additional factors.

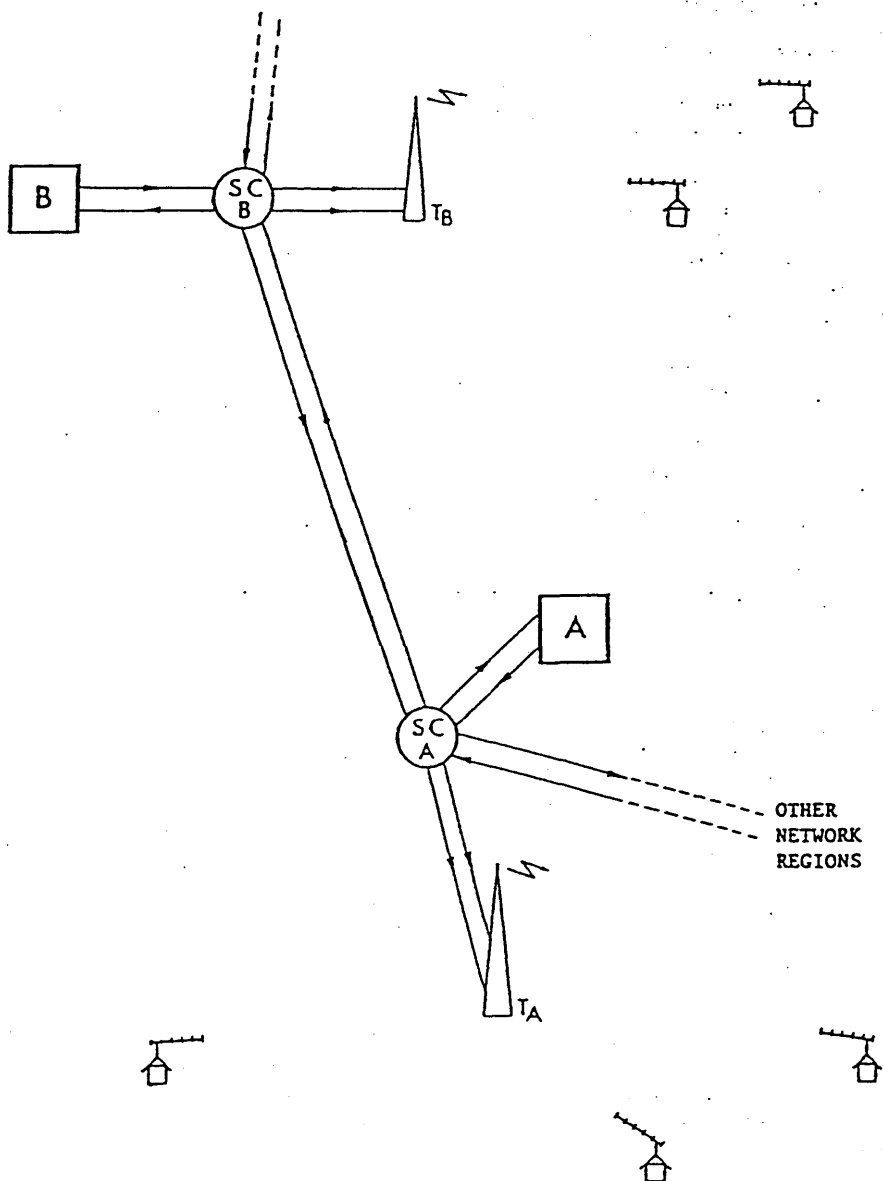
The transparency of the transmission channel is considered to be important, and bit-sequence integrity must be maintained, independently of the clock-frequency accuracy or the structure of the transmitted digital signals.

* This Report replaces Reports 817 and AA/CMTT.

It was considered useful to explain to the CCITT the distinction between "contribution" and "distribution" requirements.

The concepts of "contribution" and "distribution" relate to the way in which broadcasters operate at the present time, and it is useful to review this situation.

Figure 1 illustrates a typical broadcaster's network:



A and B : Studios

SC: SWITCHING (ROUTING) CENTRE

T: TERRESTRIAL TRANSMITTER

FIGURE 1 - The traditional network

Two studio centres, A and B, are connected via network links to switching and routing centres and to other studios. They are also connected to their respective transmitters, TA and TB.

The television signal chain from studio to transmitter is based at present upon the composite signal formats. The developments towards digital technology have begun to alter this situation, and it is now possible to envisage different signal formats for the various processes involved in programme production and distribution.

The distinction between production and distribution is one of the fundamental differences between network types and the purposes which networks have to fulfil. From the production viewpoint, whether the programme is a news bulletin, a sports event or a high-prestige drama, the complete programme is made from segments, often recorded on tape or film at different times and places. Sometimes programmes are "live", being brought to the studio for presentation to the viewer as an outside broadcast. These uses are all described by the term "contribution". It is an essential feature of a contribution that the signal is not yet intended for the viewer and does not necessarily represent a complete programme, because it may be subjected to editing and other post production - so called "down-stream" - processing at the studio. It is ready for viewing by the consumer only when the programme is completed; the process of conveying this signal to the viewer is known as "distribution".

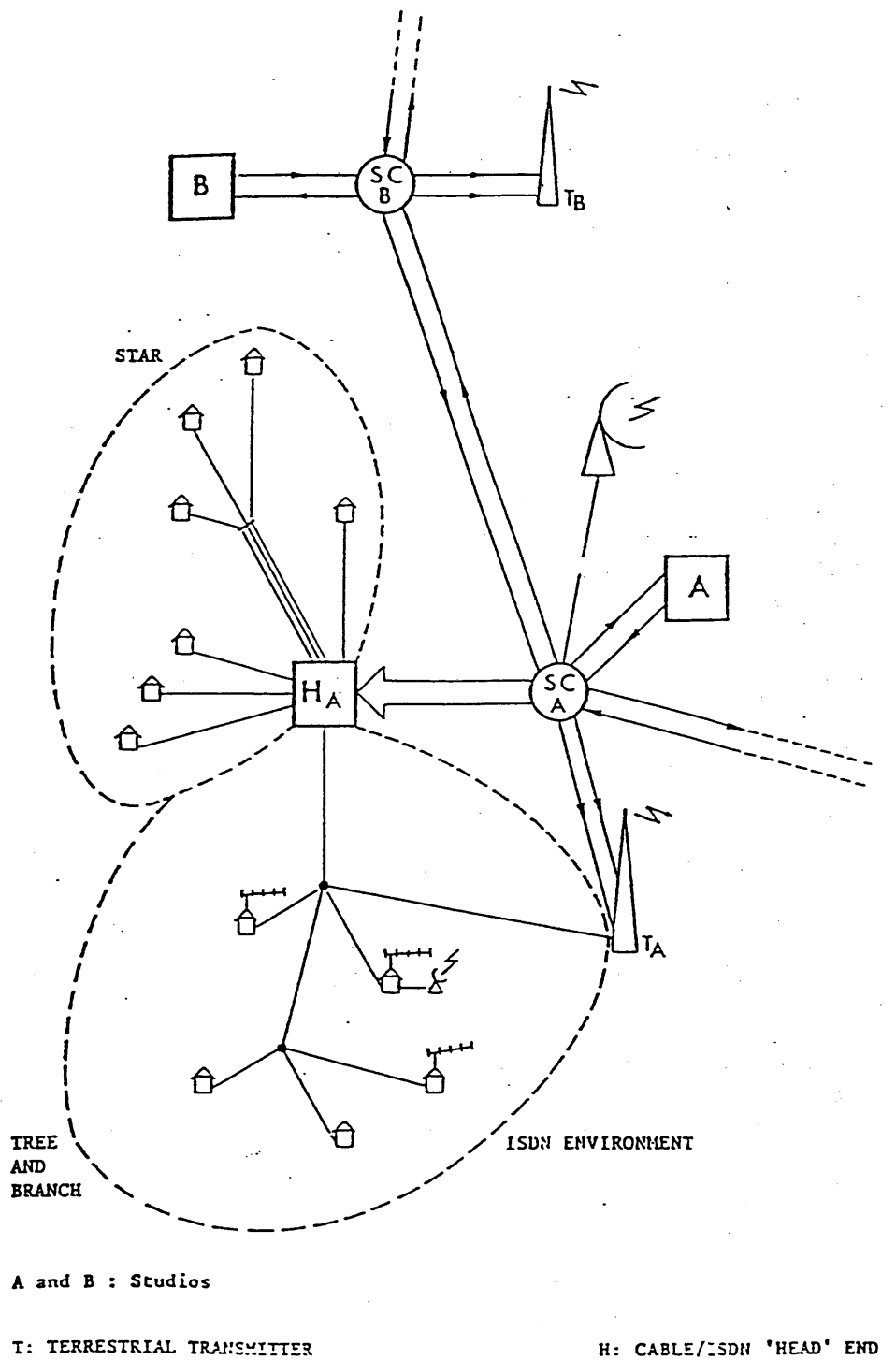
Sometimes (for example, in the case of sports events or live broadcasts) the distinction between distribution and contribution is small, but at other times (for example, in the case of documentaries and drama) the distinction is clear.

A given signal is labelled mainly by referring to the purpose which this signal is fulfilling at the time of transmission.

In some network arrangements - usually those with a great deal of re-routing and made up of numerous input and output modes - it is impossible to make the distinction without knowing the source and destination of the signal and the purpose which it is intended to fulfil.

In future digital transmission, the potential for improved operational flexibility is such that it is worthwhile to consider providing for these different cases. It may, however, prove that economic considerations will force a substantial degree of standards commonality and compatibility between contribution and distribution modes, to avoid having to operate two separate networks.

Figure 2 illustrates a development from Figure 1. Here, in addition to the development of traditional, hierarchical, broadband point-to-point networks, there is also a system carrying a wide variety of services, both unidirectional (information sources, television, etc.) and bi-directional (interactive services such as telephone, visual telephony, home banking, home shopping, etc.). The integration of television services into this scheme requires that television coding standards be adapted to the peculiarities of this newer network.

FIGURE 2 - ISDN in point-to-point network context

The bit rates and signal-formatting proposals for the broadband ISDN are different from the traditional ones; thus, at the local or wide-area network central offices (see block H in Figure 2), there will be some interfacing between the two systems. For many services this may not be significant, but for television it may be very significant.

The development of an HDTV transmission system has become very important, since HDTV may be applied to various fields, such as video telecommunication services, in the next generation of TV broadcasting systems, and possibly in direct satellite broadcasting systems.

Bit rate reduction techniques applied to HDTV may result in transmission bit rates of 100 to 130 Mbit/s for distribution. A bit rate higher than the H_4 channel will, however, be needed for HDTV contribution services.

One important factor to be taken into account by the broadcaster is the proposed use of the asynchronous transfer mode (ATM). The subject of digital television signals in an ATM-based network is discussed in Report 1240.

3. Position in CCITT

The Recommendations regarding the network aspects of digital transmission appear in the CCITT "Blue Book" issued after the IXth Plenary Assembly in Melbourne (November 1988). For the current plesiochronous network, the digital network is designed to function in a hierarchical fashion. The signals which pass between equipments have been standardized in two distinct ways:

- a) the form of the signals and their characteristics are specified in CCITT Recommendation G.703, and
- b) the structure of the signals and the way they are combined in the multiplex is defined in the G.750-Series of CCITT Recommendations.

In order for broadcasters' signals to enter the network in the standard way, they must conform to Recommendation G.703 as a minimum. The signal structure, however, does not have to conform to Recommendation G.751, because it is not a telephony assembly which would need dis-assembling and re-assembling. However, if the broadcaster wishes to avoid having non-standard, and thus expensive, equipment in PTT premises in order to de-multiplex signals which are transmitted over links in multiplex and to arrange for routing and switching, then some consideration must be given to the signal structures defined in Recommendation G.751.

For the future ISDN, there is a Series-I set of Recommendations [CCIR, 1986-90a]. One of these (Recommendation I.110) gives the structure and a list of the I-Series Recommendations. Particular attention should be paid to Recommendations I.112 and I.113, which contain the vocabulary used in connection with the ISDN. It is suggested that the CMTT should endeavour to use the same vocabulary, in order to facilitate communication with the CCITT and to encourage a common understanding of the various issues.

The Recommendation of the most direct importance for the CMTT is Recommendation I.121 - Broadband aspects of ISDN. The contents may be briefly summarized as follows:

The ISDN is intended to support a wide range of audio, video and data applications in the same network. It will support both switched and non-switched connections. These connections will support both circuit-mode and packet-mode services.

The target transfer mode is the Asynchronous Transfer Mode (ATM). This will influence the standardization of digital hierarchies and multiplexing structures, switching and interfaces for broadband signals. It will offer a flexible transfer capability common to all services. However, in the early stages of evolution, some interim user-network arrangements (e.g., combinations of Synchronous Transfer Mode (STM) and ATM) may be adopted in certain countries.

The broadband services are classified, and each class is defined. Examples are also given.

The user-network interface will be required to support a mixture of simultaneous services. In particular, the interface must be able to accommodate at least an H4 user rate (or a mixture of services having an aggregate bit-rate up to H4), plus some narrow-band services and signalling.

The basic architectural model is indicated, and examples of physical reference configurations and user-network interfaces are given.

A protocol model for ATM is described, with particular attention being paid to the adaptation layer, which supports higher-layer functions of the user and control planes, and also supports connections between ATM and non-ATM interfaces.

The important characteristics of ATM include the header functions, header format, header size, and information field size.

The broadband channel rates are currently defined as follows:

- 1) H_{21} broadband channel: the bit rate of this broadband channel will be 32768 kbit/s;
- 2) H_{22} broadband channel: the bit rate of this broadband channel will be:
 - in the approximate range of 43 to 45 Mbit/s;
 - an integer multiple of 64 kbit/s;
 - not greater than the payload of existing third level asynchronous transmission systems of the 1.5 Mbit/s based hierarchy.

Consistent with these three requirements, one objective is to maximize the bit rate of the H_{22} broadband channel.

- 3) H_4 broadband channel: the bit rate of this broadband channel will be:
 - in the range of 132 to 138.240 Mbit/s;
 - an integer multiple of 64 kbit/s.

When subsequently defining the exact bit rate, the following factors will be taken into account:

- the ATM basis of the 150 Mbit/s user-network interface;
- the possible need, during an interim period, to use STM techniques to carry the bit stream of this channel in transmission systems based on the existing and the new digital hierarchy;

- the possible need to support a television signal multiplex as specified by the CMTT.

The final specification of H_{22} and H_4 broadband channel rates should be such that:

$$4 \cdot H_{21} \text{ rate} \leq H_4 \text{ rate}$$

$$3 \cdot H_{22} \text{ rate} \leq H_4 \text{ rate.}$$

Additional broadband channels may be defined if necessary, but now that ATM has been adopted as the target transfer mode for the broadband ISDN, the future of the H channels has been called into question [CCIR, 1986-90b].

Some structural, physical and functional characteristics of broadband user-network interfaces are given.

Two bit rates will be standardized for the interfaces - one at approximately 150 Mbit/s and the other at approximately 600 Mbit/s.

The interface need not be symmetrical.

Each interface must be capable of supporting broadband services as well as services based on 64 kbit/s.

Various network aspects are reviewed.

Annex 1 to the Recommendation gives examples of possible services, their applications and some possible attribute values describing the main characteristics of the services.

Annex 2 of the Recommendation gives some definitions of possible broadband service families and their attribute values.

Both of these annexes contain material relating to television and sound services.

A new synchronous digital hierarchy (SDH) has also been developed but it is not yet completely clear how this new hierarchy will affect users of broadband digital circuits.

Three new Recommendations deal with the concept and the specifications. These are:

- G.707 - Synchronous Digital Hierarchy Bit Rates
- G.708 - Network Node Interface for the Synchronous Digital Hierarchy
- G.709 - Synchronous Multiplexing Structure

These new Recommendations are significant to CMTT since they affect the transport of broadband signals through the digital network and ISDN.

The SDH offers advantages of simplified multiplexing/demultiplexing, direct access to lower speed tributaries without demultiplexing the entire high speed signals, enhanced operations and maintenance capabilities and easy growth to higher bit rates in step with the evolution of transmission technology.

Recommendation G.707 provides the rational and specifies the bit rates for the synchronous digital hierarchy. Currently two bit rates are specified:

- Level 1 155 520 kbit/s
- Level 4 622 080 kbit/s.

Key candidate for a higher level is level 16 at 2 488 320 kbit/s.

These bit rates are the transmission rates at the Network Node Interface (NNI) and include overhead capacity. The potentially useable payload for broadband signals is approximately 3% lower.

Recommendation G.708 specifies the basic multiplexing principle and multiplexing elements, the frame structure, interconnection of networks, and the definition and description of overhead types.

The multiplexing element most important to the work of the CMTT is termed "container". It is defined as the unit of payload capacity which is sized to carry any of the levels currently defined in Recommendation G.702, and may also provide capacity for carriage of broadband signals which are not yet defined.

Although the multiplexing elements currently defined could theoretically carry a slightly higher payload than given by the existing hierarchy bit rates (G.702), it has been decided to limit the payload to the levels of G.702 in order to allow interworking with existing networks.

On the other hand, the synchronous multiplexing scheme does allow for the transport of broadband signals in excess of the G.702 bit rates through the process of concatenation of certain multiplexing elements. Thus, for example, broadband signals at $n \times 6.3$ Mbit/s ($n=1$ to 7) could be readily transported.

Recommendation G.709 specifies the detailed mapping formats and multiplexing method, the pointer mechanism for synchronisation and concatenation, and path overhead.

Currently detailed mappings are specified for the following tributary bit rates:

- 1544, 6312, 44 736 kbit/s and
- 2048, 8448, 34 368, 139 164 kbit/s.

4. Bit-rate capacity requirements

4.1 Sound

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.

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.

Methods of coding and multiplexing sound signals of digital studio quality for transmission at the H₁ channel rates are given in Recommendation 724.

More detailed information about the digital transmission of sound-programme signals is given in Report 647.

4.2 Television

4.2.1 H₂ channel

As indicated in section 3, the H₂₁ channel bit-rate has been specified. However, the H₂₂ channel bit-rate has not yet been exactly defined.

4.2.2 H₄ channel

This channel (120 - 140 Mbit/s) appears to satisfy the broadcasters' requirements for contribution, as well as for distribution; it might even satisfy the requirements of other bearer services, such as the distribution of enhanced television signals.

However, H₄ has the advantage of not being feasible for existing satellite systems.

The exact bit rates for H channels have not yet been agreed, the requirements of contribution services imply the need for a minimum capacity for H₄, below which compromises to the quality of the service should be expected.

This minimum bit rate was proposed by IWP CMTT/3 to be 135.168 Mbit/s although a higher value (139 - 264 Mbit/s) is preferred.

Moreover, for the H₄ channel, the broadcasters require the maximum capacity available as well as asynchronous inclusion of this payload into the ISDN hierarchy, in order to maintain independence from the network in terms of synchronization.

In [CCIR, 1986-90c], the CMTT has again drawn the attention of the CCITT to the problems caused by the lack of a Recommendation for an exact bit rate for the H₄ channel; in a liaison statement to CCITT Study Group XVIII (see Annex III of Report 1234) the need to maintain at least the minimum bit rate already requested is emphasized.

4.2.3 Access above the H₄ channel rate

The normal television multiplex will contain a certain capacity for sound and data channels. This capacity will correspond to the normal contribution or distribution requirements. However, there may be particular circumstances where additional capacity is required. For example, when transmitting signals from a large event such as the Olympic Games, it may be desirable to have an additional thirty commentary channels associated with the normal television multiplex. Another example might be the special need of a particular country to transmit sound in several different languages. A further example is the possible requirement to support the full uncompressed bit rate

used in the studio (216 Mbit/s) or a higher rate for a multiplex of video, sound and data. In cases such as these, there might be a need to combine an H channel with a number of B channels, if it proves uneconomic or impractical to use a higher level H channel. However, for the H₄ channel rate the option of using an H channel higher than H₄ does not appear, at present, to be available.

Also, in the foreseeable future, there may be a need to transport HDTV signals. At present, the bit rate required for either contribution or distribution is not known. However, it is almost certain that, for contribution, a bit rate higher than that of the H₄ channel will be needed.

Advice has been sought from the CCITT as to whether services requiring capacity in excess of the H₄ channel rate can be supported using currently-proposed interfaces at about 150 Mbit/s.

5. Interworking and multiplexing

In their studies of coding and multiplexing schemes to transmit television signals at bit rates corresponding to the various levels of the ISDN, CCIR IWP CMTT/2 and IWP CMTT/3 have posed a number of questions for CCITT Study Group XVIII.

Recommendation G.802 specifies the interworking between networks based on different digital hierarchies and speech encoding laws for the transport of 64 kbit/s digital information, including 7 kHz audio. Therefore, the Recommendation does not cover, as originally anticipated, interworking of 384 kbit/s digital sound-programme channels.

5.1 Transmission at 45 Mbit/s

In discussing the possible advantages and disadvantages of using a bit rate around 45 Mbit/s for transmitting digital television signals, it is not yet known whether there would be significant advantages in adopting such a bit rate instead of one around 34 Mbit/s in Region 1; in North America, it is likely to be used.

In Region 1, therefore, carriers and broadcasters may be faced with the problem of handling 4/5 Mbit/s television signals coming from North America, and perhaps also being generated in Region 1. This gives rise to the following questions:

- a) will it be possible to have access at this level in the ISDN, effectively and economically?
- b) will it be possible to establish non-switched point-to-point connections at this level?

5.2 Use of transmission structure

Regarding arrangements for multiplexing video, sound and associated data signals for various levels of the digital hierarchy CCITT Study Group XVIII has been requested to indicate the guidelines that should be used in forming such multiplexes.

Specific questions that need consideration include:

- a) is it necessary that the frame length and structure already defined for telephony be used (e.g., Recommendation G.751 in the case of the fourth level of the hierarchy)?

- b) what factors must be taken into account by the broadcasters regarding the possibility of interworking between the proposed H_{21} , H_{22} and H_{23} bit rates? Similarly, what constraints will there be in transferring the information between three H_2 channels and one H_4 channel?
- c) what is required to allow the retention of circuit-related functions such as circuit protection, alarms, BER monitoring, etc.

6. Performance requirements

The subject of transmission impairments and methods of protection for digital television signals is treated in Report 967. For sound signals, reference should be made to Report 648.

6.1 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 I.

TABLE I – *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 I, 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.

The "Blue Book" version of CCITT Recommendation G.821 gives preliminary guidelines for the assessment of higher bit rate systems. The formulas given there can be used to obtain an estimate of the high bit rate system performance in terms of the 64 kbit/s parameters cited in Recommendation G.821. However, caution is advised when planning to use the formulas for certain types of bursty error distribution.

Further study is being undertaken in CCITT Study Group XVIII.

The CMTT has pointed out that Recommendation G.821 does not address the requirements relating to "burst errors". The length and frequency of the bursts must be defined. [CCIR, 1986-90c] gives the CCITT information about the assumptions currently being made by the CMTT for the purpose of testing codecs.

The CCITT has stated [CCIR, 1986-90d] that:

Study Group XVIII is undertaking a study aimed at producing a Recommendation concerning network performance at high bit rates before the end of the current period (1992). In so doing, it was agreed that the Recommendation:

- could be expressed in terms of the probability of meeting particular thresholds;
- would take account of burstiness (duration/frequency/intensity);
- would be based on the concept of the errored time interval;
- would aim at having a single integration period if this proved possible;
- would start from the premise of "equivalent quality" to G.821;
- would consider the requirements of high bit rate services which could be more demanding;
- should take account of the appropriate CCIR Study Groups 4 and 9 and CCITT Study Group XV Recommendations.

It was also noted that one possible method satisfying CMTT's requirement for network performance information might include a specification based on a short integration period whilst retaining a 1-second interval for practical performance assessment purposes.

CMTT is requested to provide information on:

- the length, frequency and intensity of bursts that can be tolerated;
- any other objectives that are considered to be appropriate.

6.2 Slips and delays

It is assumed that automatic routing relates to switching which must be performed to establish a digital connection. In particular, the acceptable values for slips and delays should be defined.

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.).

6.3 *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.

In addition, there are two supplements: one gives guidelines concerning the measurement of jitter and wander, and the other deals with jitter and wander accumulation in digital networks.

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.

6.4 Cell loss

In an ATM-based network, cell loss is an additional source of impairment. The CMTT assumes that, for high-quality sound programme or television services, the cell loss ratio would be about 10^{-7} or 10^{-8} . The CCITT has been asked [CCIR, 1986-90c] to confirm the validity of this assumption, and to indicate whether or not there would be a sequence number in the information field in each cell, to detect cell loss.

7. Failure characteristics

Report 1211 contains some comments regarding failure conditions for a television multiplex. It states that interruptions of any part of the multiplex signal for several seconds or more are intolerable, and protection modes would be required.

For burst errors of short duration the synchronizing systems should be protected. By maintaining synchronization during the interruption, the error management systems for the video, audio and data could be independent.

For interruptions of the order of 50 ms, the decoder circuits will have to relock. It is suggested that the relocking sequence should be complete within 160 ms.

8. Attributes values

As part of its liaison activities, IWP CMTT/3 made some detailed comments on a CCITT status report on the broadband ISDN. In particular, some amendments and additions were proposed concerning tables of service attribute values for television and sound programme transmission. These were included in the relevant CCITT Recommendations.

9. Performance methodology for ATM

In defining the performance requirements in an ATM environment, it is important to clarify the respective roles of the CCITT and the CMTT. With this in mind, it has been agreed that the layered methodology in Figure 3 will be applied.

Further clarification is required about the application of the methodology to the ATM adaptation layer, which is not yet clearly defined. A liaison statement to CCITT Study Group XVIII (see Annex I) contains some proposals on this subject.

10. Conclusions

The activities of IWP CMTT/3 together with those of the Special Rapporteurs, have proved to be effective in drawing the attention of the CCITT to the particular requirements of concern to the CMTT regarding the digital transmission of television and sound signals.

Further work is, however required, especially with regard to the use of an ATM-based network.

Close liaison between the CCITT and CMTT must continue.

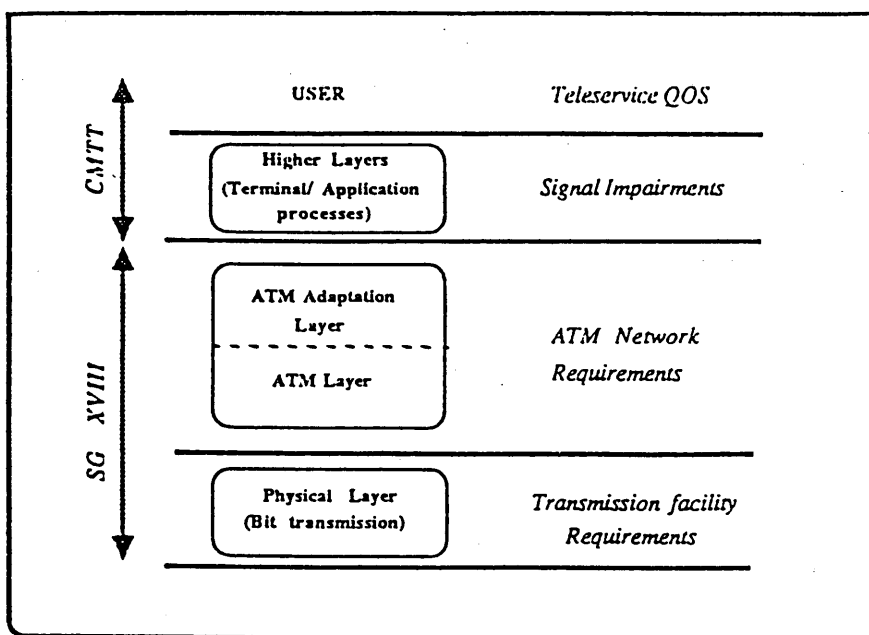


Figure 3. A Layered performance methodology for ATM

REFERENCES

CCIR Documents

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

[1986-90]: a. CMTT/161 (IWP CMTT/3); b. CMTT/163 (Director, CCIR);
c. CMTT/162 (+ Addendum 1) (Chairman, IWP CMTT/3); d. CMTT/222 (CCITT Study Group XVIII).

ANNEX I

Liaison statement to CCITT Study Group XVIII

Further to the previous liaison statement from the CMTT to CCITT Study Group XVIII [CCIR, 1986-90] on an approach to an ATM performance specification, the CMTT has discussed possible implementation scenarios for current and future coding technologies with regard to transmission in an ATM-based network. The purpose was to obtain a better understanding regarding the division of responsibilities and the preparation of Recommendations between the CMTT and CCITT Study Group XVIII.

The current view of the CMTT is presented in the attached. Fig. 4 and 5. CCITT Study Group XVIII is requested to confirm this view and/or provide comments.

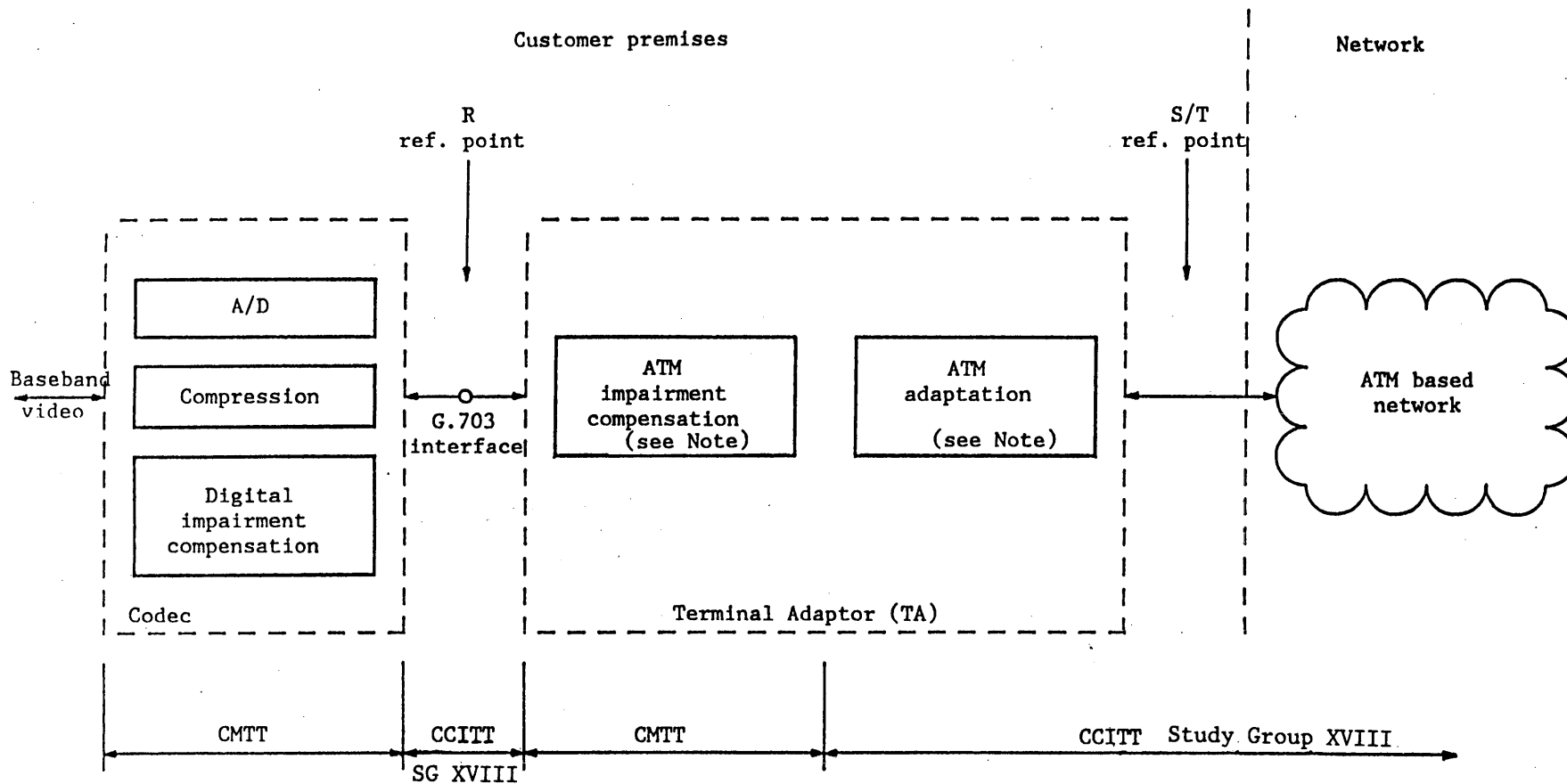


FIGURE 4

Fixed bit rate codec in ATM based network

Note - Typical TA functions

ATM impairment compensation for:

- cell loss;
- cell transfer delay jitter;

ATM adaptation:

- cell assembly/disassembly;
- header insertion;
- cell loss detection;
- delay smoothing.

REFERENCES

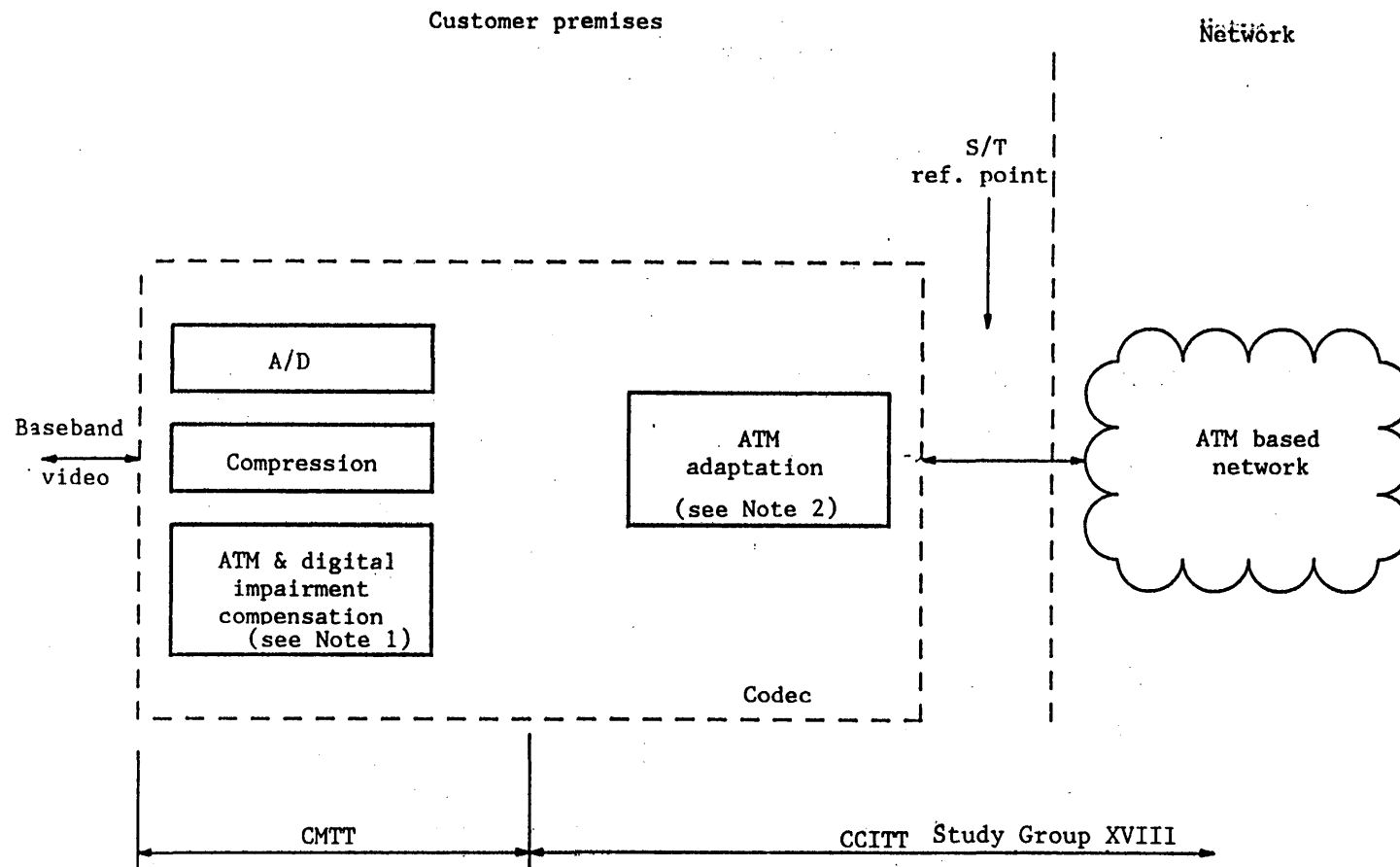


FIGURE 5

ATM optimized codec

Note 1: ATM impairment compensation for:

- cell loss;
- cell transfer delay jitter;
- information errors;

Note 2: ATM adaptation:

- cell assembly/disassembly;
- header insertion;
- cell loss detection;
- delay smoothing.

REPORT 967-2*

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

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

(1982-1986-1990)

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.
- in the case of a system using the Asynchronous Transfer Mode (ATM), those that are characteristic of ATM such as a loss of cells, etc.

These — 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. They will also depend on the mode of transmission (based on packets or on continuous information flow).

In a packet-based system the signal, even if unimpaired, may be subject to a troublesome end-to-end delay.

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.

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;

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

- the maximum acceptable bit error ratio should be less for television than for telephony and defined in an appropriate form for television;
- error-correction techniques will be useful when transmitting television signals after bit-rate reduction.
- Single mode optical fibre transmission is likely to predominate because of its cost advantages over other forms of transmission.

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 signal may be 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].

Low frequency jitter and wander are not so easily overcome by jitter removal circuits. Their effects in television transmission systems, with embedded multiplexing functions, in particular the effects of low-frequency waiting-time jitter on television signals, require further investigation.

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. Component video signals are much less susceptible to jitter than composite PAL signals. On real pictures sinusoidal jitter at critical frequencies (small offsets from multiples of line or field frequency) becomes perceptible at about 5.5 ns p-p and white noise jitter becomes perceptible at about 2 ns r.m.s. [CCIR, 1986-90a]. 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.

The permissible amount of jitter at a digital network interface (e.g. at the input to a video decoder) is given in CCITT Recommendation G.823, in terms of unit intervals over a range of bit rates from 64 kbit/s to 139.264 Mbit/s. The CCITT has also proposed a more stringent output jitter limit where the signal is controlled by an autonomous clock, [CCIR, 1986-90b]. in accordance with CCITT Recommendation G.823.

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.

However, if concentrated error bursts produce consistently poor BER measured over smaller integration times, then there is an increased probability of loss of multiplex framing synchronization (depending on the particular strategy used), resulting in severe subjective impairment.

[CCIR, 1986-90c] describes an example of bit-error burst compensation method for television signals. This compensation method is carried out using a random error correcting code, Reed-Solomon code RS(40, 36) and an interleaved structure at the customer's terminal. When the bit error burst probability before compensation is 10^{-6} , the bit error burst probability after compensation is 10^{-14} . Furthermore, when the bit error burst probability before compensation is 10^{-8} , the decoded sequence can be regarded as free of error bursts.

It is considered that any target for error performance for broadcast quality television services must take into account both error impairments to the viewed picture information, and the possibility of loss of multiplex framing synchronization.

One such appropriate target is for the BER to be maintained better than 10^{-4} measured over an integration period of 10 ms (centisecond). This utilizes the lowest integration period presently achievable by state of the art measuring equipment. It is estimated that the BER and integration period will provide sufficient sensitivity to be applicable for multiplex framing considerations, while still maintaining a meaningful measure of errors in the viewed picture information.

Alternatively, two performance parameters may be derived. An appropriate performance parameter for impairment to the picture might be a BER worse than 10^{-6} (or possibly 10^{-4} if the use of FEC is specified) for an integration period of up to five seconds; while an appropriate parameter for the failure of multiplex framing might be a BER worse than 10^{-2} over a period of 500 μ s. It is noted that the actual BER at which failure of multiplex framing occurs depends on the particular strategy used; the latter figure quoted here was found in a practical high bit rate video codec to be the point at which framing failure was likely to occur [CCIR, 1982-86d].

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.

EUTELSAT has reported in [CCIR, 1986-90d] that, for transmitting via its satellite system a 34 Mbit/s digital television signal, the satellite capacity can be more efficiently utilized by using a modem incorporating forward error correction and scrambling. In this case, the combination of a Viterbi decoder and a self-synchronizing scrambler/descrambler may result, for some conditions, in bursts of errors of more than 30 bits in length.

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 radio transmission 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.

It is considered, however, that the acceptable level of circuit availability, for digital television connections, should be no worse than that experienced at present on existing analogue television connections.

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.

When making subjective tests to assess the effects of bit errors, account has to be taken of both the degree of impairment caused by an isolated error-occurrence and also the number of times in a given time interval that the impairment occurs. This matter is discussed in [CCIR 1986-90e].

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.

4.4.1 Results of practical studies

[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 $9 \cdot 10^{-6}$.

It is confirmed by field tests via an INTELSAT satellite [Murakami, *et al.*, 1986] using a motion compensated interframe, interfield and intrafield adaptive coding method [Matsumoto *et al.*, 1986] that the (239, 255) BCH code is usable for multi-channel digital TV transmission by the PSK/FDMA system.

[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.

[CCIR, 1982-90f] includes preliminary proposals for assessment material.

[CCIR, 1986-90g] describes the performance of an INTELSAT link carrying two 30 Mbit/s digital television signals, and of a 280 Mbit/s submarine optical-fibre cable system. The tests indicate that the error ratio for either type of transmission would not be worse than 10^{-6} . Using a (255, 239) BCH code, such an error-ratio can be corrected so that the effective error-rate falls to 1 bit in 12 days.

The document also gives the results of a field trial with an experimental 34 Mbit/s system based on 2-dimensional DPCM. The signals were transmitted via an INTELSAT transponder. A (275, 261) FIRE code was used to correct random errors. This code could correct an error burst of 20 bits.

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. Under certain circumstances an uncontrolled slip can be avoided by discarding or repeating a complete transmission frame, thus avoiding loss of alignment in downstream equipment. This would then constitute a controlled slip. The CCITT has stated that the philosophy of its Recommendations is that timing alignment is absorbed by controlled slips.

In digital television transmission a bit slip may——— 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. However, if the transmission system and video codec recover from the loss of alignment rapidly enough, then the slip may result in an error burst rather than a service interruption.

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. For 64 kbit/s connections, CCITT Recommendation G.822 gives the controlled slip rate as not more than 1 slip in 70 days, due purely to plesiochronous operation, per plesiochronous interexchange link. The CCITT has stated that a target of five slips in twenty-four hours is an equivalent rate at the primary bit rate.

However, CCITT Recommendation G.822 also allows higher slip rates due to other mechanisms.

5.3 Effects of cell loss

In an ATM-based network cells may be lost, and this could result in severe bit-slips.

6. Evaluation of systems failure characteristics

[CCIR, 1982-90f] proposes that an assessment of system failure characteristics should include the following.

6.1 Forward error correction disabled

Subjective tests should be conducted in which a number of picture sources are impaired by errors. The bit error ratios used in the test should cover the range of interest, for example 10^{-3} to 10^{-9} and the results presented as a curve of picture grade versus bit error ratio. [CCIR, 1986-90h] discusses error characteristics.

6.2 Forward error correction enabled

The above procedure should be repeated as in § 6.1 to obtain a graph of comparative results.

6.3 System recovery time

a) Assessment of system recovery time is the time taken for the system to recover following a complete loss of signal to the decoder. The recovery time can be measured as the number of fields delay that are required between the connection of signal to the decoder, and switching the picture monitor input from a grey level signal (or a suitably delayed non-processed signal) to the decoder output signal, such that no picture disturbance can be observed.

b) A similar procedure should be adopted to assess the recovery time associated with bit-slips, as might occur following, for example, a non-sync cut.

Report 1211 considers aspects of system failure characteristics.

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 interframe, interfield and intrafield adaptive prediction coding – component coding based on CCIR Recommendation 604 (in Japanese). *J. Inst. Telev. Eng.* (Japan), Vol. 39, 10.
- MATSUMOTO S., MURAKAMI H. and YAMAMOTO H., [May 1986] - "Coding performance of motion-compensated interframe, interfield and intrafield adaptive prediction coding for composite and component TV signals", SMPTE Journal, Vol. 95, No. 5.
- 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.
- MURAKAMI H., SATO G., MATSUMOTO S., HIRABAYASHI T., YAMAMOTO H., DONALD W.W. and BACHMANN E., [May 1986] - "Field test results of multi-channel digital TV transmission at 15 and 30 Mbit/s through INTELSAT satellite in PSK/FDMA system", 7th International Conference on Digital Satellite Communications.
- 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)).
- [1986-90]: a. CMTT/219 (United Kingdom); b. CMTT/74 (Study Group XVII, CCITT); c. CMTT/172 (Japan); d. IWP CMTT/2-72 (EUTELSAT); e. IWP CMTT/2-74 (Japan); f. IWP CMTT/2-30 (EBU); g. IWP CMTT/2-43 (IWP CMTT/2, Expert Group); h. IWP 11/7-134 (EBU).
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SECTION CMTT B: METHODS OF OPERATION AND ASSESSMENT OF PERFORMANCE OF
TELEVISION TRANSMISSIONS

REPORT 314-7

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

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

(1966-1970-1974-1978-1982-1986-1990)

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.

The addition of an energy dispersal signal to the television waveform during transmission may result in modification of some performance measurement results. For further information, see Report 1094.

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 line 334 (and in some cases to line 21 only) 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.

In the United Kingdom, the BBC is changing to the use of a pair of alternating insertion test signals using one line only in each field, which will replace the use of two lines in each field for test signals. The new national signals will provide substantially the same video signal monitoring facilities as the original test signals and will additionally provide some control data capacity and a new audio signal monitoring facility. [CCIR 1986-90a, b].

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).

[1986-90]: a. CMTT/28 (United Kingdom); b. CMTT/238 (United Kingdom).

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.

HOLDER, J.E. and SPICER, C.R. [1987] - A new insertion test signal with audio monitoring facilities. Third International Conference on Television Measurements, Montreux, 1987, IERE Publ. No. 74, p.77.

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

SPICER, C.R. and HUBBARD, R.P. [1988]. A New Insertion Test Signal Incorporating Audio Monitoring And Data. International Broadcasting Convention, Brighton 1988.

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(!)

(!) 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 1094-1*

**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)

(1986-1990)

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 such that each of its points of inflection occurs within the field blanking period. 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.

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. Such a clamping circuit may not be perfect and the larger the dispersal waveform voltage, the more likely it is that clamping will introduce additional distortions.

3. Problems of automatic measurement

Test signals are placed in specified lines in the vertical blanking interval and are repetitively evaluated using automatic measuring equipment. The relevant CCIR texts are identified in the introductory paragraph of Recommendation 569.

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

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

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.

In the USSR, studies have shown that measurements of parameters of TV channels using a dispersal signal which is not synchronous with the TV signal should be carried out by digital measuring instruments, and that additional averaging of the measurement data should be performed [CCIR 1986-90].

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/20 (United Kingdom); b. CMTT/70 (United States of America).
 [1982-86]: CMTT/212 (Germany (Federal Republic of)).
 [1986-90]: CMTT/55 (USSR).

BIBLIOGRAPHY

VILENCHIK, L.S., KRIVOSHEEV, M.I., LOKSHIN, B.A., SHAVDIYA, YU.D., [1986] - "Izmerenie differentsialnykh iskazhenii v kanale izobrazheniya sistem sputnikovogo televideniya, ispolzuyushchikh signal dispersii" (Measurement of differential distortions in the picture channel of satellite TV systems using a dispersal signal), "Elektrosvyaz", No. 5.

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): $2\ \mu\text{s}$ and $64\ \mu\text{s}$;
- 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 televiziya*, 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 vidéo fréquence. 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).

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).

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

REPORT 496-5*

**CIRCUITS FOR HIGH-QUALITY MONOPHONIC AND
STEREOPHONIC TRANSMISSIONS****

(Questions 17/CMTT and 19/CMTT, Study Programme 19B/CMTT)

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

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.
- 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 difference between the two series of values relate to the phase difference between channels *A* and *B*.

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 Parameters for digital transmission of sound programme signals can be found in Recommendations on this subject and Report 647 and Report 1236.

TABLE I

Limiting values of parameters of high quality circuits
for monophonic and stereophonic transmission

Parameter	OIRT Recommendation 67	Recommendation 505-3
1. Level stability (dB)	± 1 (long term)	± 0.5 (daily)
2. Amplitude/frequency response (dB) Reference frequency: 1 kHz		
0.040 to 0.125 kHz	+0.5 to -2	+0.5 to -2
0.125 to 10 kHz	+0.5 to -0.5	+0.5 to -0.5
10 to 14 kHz	+0.5 to -2	+0.5 to -2
14 to 15 kHz	+0.5 to -2	+0.5 to -3

* 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 (continued)

Parameter	OIRT Recommendation 67	Recommendation 505	
3. Group delay time difference (ms)			
0.040 kHz	50	55	
0.075 kHz		24	
14 kHz		8	
15 kHz	8	12	
4. Noise		analogue	digital
Idle channel noise (dBqOps) ¹⁾	-42	-42	-51
Programme-modulated noise (dBqOps) ¹⁾		-30	-39
Unweighted noise (dBmOs) ²⁾	-46		
Impulsive noise (dB) ³⁾			
5. Single tone interference measured selectively (dBmOs) ⁴⁾	Not taken into consideration	$\leq -(73+\Psi)$	
6. Non-linear distortion ⁷⁾		THD	2nd or 3rd Harmonic
6.1 Harmonic distortion (dBmOs)			
0.040 to 0.125 kHz		-31	-34
0.125 to 2.0 kHz	-40	-37	-40
2.0 to 4.0 kHz	-40	-40	-43
6.2 Intermodulation (dBmOs)			
2nd order with 5.6 + 7.2 kHz			-43
3rd order with 4.2 + 6.8 kHz			-43
7. Intelligible cross-talk ratio (dB) ⁵⁾			
0.040 kHz	78	50	
0.040 to 0.050 kHz		oblique segments	
0.050 to 5.0 kHz		74	
5.0 to 15.0 kHz		oblique segments	
15.0 kHz		60	
8. Amplitude linearity (dB) ⁶⁾			
1.0 kHz		± 0.5	
0.040 to 15.0 kHz	± 0.5		
9. Difference in gain between A and B-channels (dB)			
0.040 to 0.125 kHz	2.0	1.5	
0.125 to 10.0 kHz	1.0	0.8	
10.0 to 14.0 kHz	2.0	1.5	
14.0 to 15.0 kHz	2.0	3.0	

TABLE I (continued)

Parameter	OIRT Recommendation 67	Recommendation 505
10. Phase difference between A and B-channels (degrees)		
0.040 kHz	20°	30°
0.040 to 0.200 kHz	oblique segments	oblique segments
0.2 to 4.0 kHz	10°	15°
4.0 to 14.0 kHz	} oblique segments	oblique segments
14.0 kHz		30°
14.0 to 15.0 kHz		oblique segments
15.0 kHz	20°	40°

Notes

- 1) The measurement to be made with an instrument conforming to Recommendation 468.
- 2) The unweighted measurement is made with an instrument with the band limitation 22 Hz to 22 kHz (see Recommendation 468 Annex II) the value given is an r.m.s. value. OIRT expresses the value in dB uOs.
- 3) The method of measurement is discussed in point 2.2.3 of this report. The tolerances will have to be studied.
- 4) (old note (5) on page 161, Volume XII).
- 5) See Recommendation 571 and Report 497.
- 6) The purpose of this parameter is to take into account the effect on the signal of the use of companders in links. The method of measurement is the following:

When an input signal is stepped from -6 dBmOs to +6 dBmOs or vice versa the output level shall change accordingly by 12 ± 0.5 dB.
- 7) The CMTT considers it necessary to measure non-linear distortion for frequencies above 5 kHz. See section 2.4.

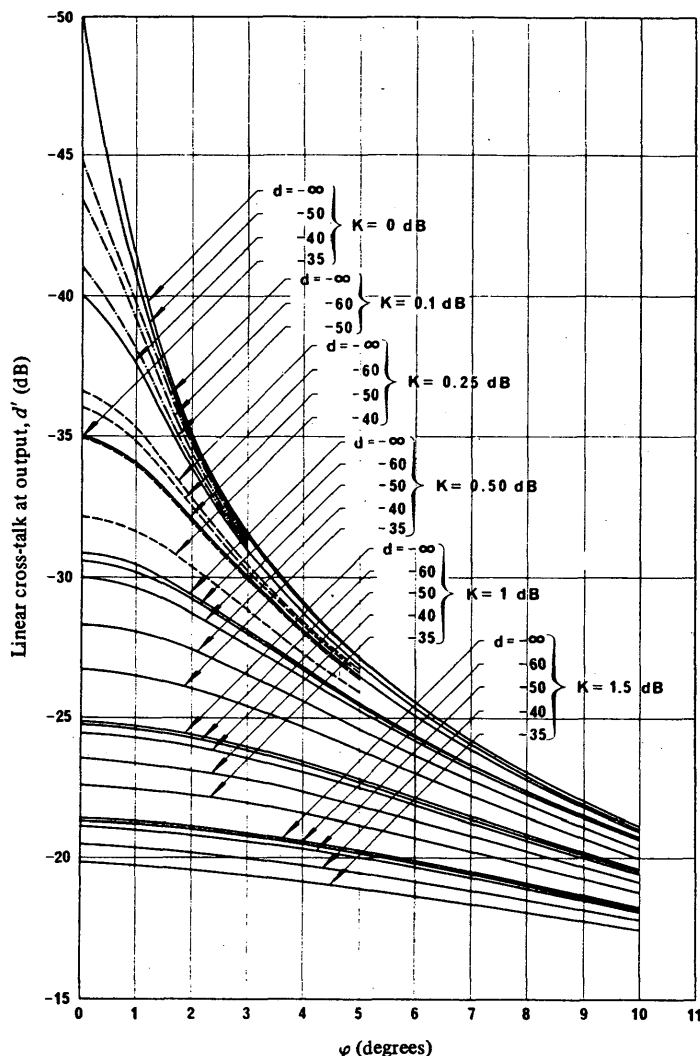


FIGURE 1 – Admissible cross-talk at the input of a transmitter-to-receiver chain

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;
- (d) unintelligible cross-talk (speech);
- (e) intelligible cross-talk (speech in German);
- (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);
- (o) data transmission signals.

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. This instrument is specified in respect of weighting network, the quasi-peak indicating instrument and the filter used for unweighted measurements. Noise measurements made with the instrument are expressed in dBqOps and dBqOs. Measurements made using either the weighting network or the filter with a true r.m.s. indicating instrument are expressed in dBmOps or dBmOs respectively. These readings may differ from those taken using the Recommendation 468 instrument due to the response of the quasi-peak instrument to noise signals which may be steady state and/or impulsive.

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 to measure a Gaussian noise signal. Similar effects may be observed when the quasi-peak instrument conforming to Recommendation 468 is used.

2.2.1 *Comparison of measurements (see Note) of Gaussian noise*

The CMTT has decided to apply the — 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 — weighting characteristic taking — into account the effects that various pre/de-emphasis characteristics have — 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.

Table II gives the calculated results of making weighted and unweighted noise measurements using the — weighting network upon sound-programme circuits of 5 kHz, 10 kHz and 15 kHz bandwidth (see note) 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 — weighting network — defined as the difference in readings between the weighted and unweighted noise measurement, 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), (c) and (d) 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.

Note - 5 and 10 kHz bandwidths refer to older type circuits. For new circuits only 7 and 15 kHz bandwidth are recommended.

TABLE II

Spectral distribution of noise from system	Upper frequency limit of the circuit under consideration (kHz)	Level of noise measured at a 0 dBm point at the output of the circuit			
		Unweighted measurement (dBm)	Weighted measurements		
			Weighted measurement (dBm)	Improvement by de-emphasis (dB)	Weighting factor of network (4)-(3) (dB)
(1)	(2)	(3)	(4)	(5)	(6)
White noise	5	(a) 0	+ 7.9	--	+7.9
		(b) - 1.3	+ 1.0	+6.9	+2.3
		(c) -2.0	+4.5	+3.4	+6.5
		(d) -3.1	+2.5	+5.4	+5.6
	10	(a) 0	+ 9.9	--	+9.9
		(b) - 3.3	+ 1.0	+8.9	+4.8
		(c) -4.0	+4.1	+5.8	+8.1
		(d) -5.4	+1.7	+8.2	+7.1
	15	(a) 0	+ 8.5	--	+8.5
		(b) - 5.2	- 0.5	+9.0	+4.7
		(c) -5.4	+2.5	+6.0	+7.9
		(d) -7.0	+0.1	+8.4	+7.1
Triangular noise	5	(a) 0	+ 10.1	--	+10.1
		(b) - 7.2	+ 1.9	+8.2	+9.1
		(c) -3.5	+6.0	+4.1	+9.5
		(d) -5.6	+3.7	+6.4	+9.3
	10	(a) 0	+ 10.9	--	+10.9
		(b) - 9.9	+ 0.7	+10.2	+10.6
		(c) -7.4	+3.4	+7.5	+10.8
		(d) -10.1	+0.5	+10.4	+10.6
	15	(a) 0	+ 6.6	--	+6.6
		(b) - 10.8	- 3.7	+10.3	+7.1
		(c) -10.2	-1.4	+8.0	+8.8
		(d) -13.2	-4.3	+10.9	+8.9

Notes.

(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 dBm point. (This applies to any value shown in the rows marked (b));

Note - When de-emphasis in accordance with CCITT Recommendation J.17 is employed with a 6.5 dB gain at 800 Hz, the noise levels in rows marked (b) should be increased by 5 dB. Conversion factors shown in columns (5) and (6) remain unchanged.

(c) De-emphasis 50 μ s is employed as defined in Recommendation 450; (This applies to any value shown in the rows marked (c));

(d) De-emphasis 75 μ s is employed as defined in Recommendation 450; (This applies to any value shown in the row marked (d));

Note - The individual values in column (5) show the improvement in weighted signal noise ratio obtained by the use of the emphasis and de-emphasis.

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

f (kHz)	Insertion-loss (dB)
0	18.75
0.05	18.70
0.2	18.06
0.4	16.48
0.8	13.10
2	6.98
4	3.10
6.4	1.49
8	1.01
10	0.68
∞	0

The measured pre-emphasis and de-emphasis curves should not depart by more than ± 0.25 dB from the theoretical curves when the measured levels at 800 Hz are made to coincide with the theoretical levels.

Note. — The formula given above defines only the "insertion-loss/frequency" characteristic. The level at which the modulated programme signal is given to the group link is different for the various types of sound-programme equipments.

In some cases pre/de-emphasis 50 μ s or 75 μ s as defined in Recommendation 450 is used on sound-programme circuits e.g., frequency-division multiplex of sound and vision signals transmitted by microwave links or at FM sound broadcasting.

2.2.3 Measurement of impulsive noise

Recommendation 468 is commonly used to measure noise including impulsive noise.

In [CCIR, 1986-90b] it is proposed to use a tone-compensated measurement of impulsive noise to determine a "pulse weighted peak value $\hat{U}_{IW}$ ", which is defined as follows:

$$\hat{U}_{IW/dB} = 20 \cdot \lg \frac{\hat{U}}{U_0} + 20 \cdot \lg (k_1 \cdot k_2 \cdot k_3)$$

where k_1 , k_2 and k_3 are the objective rating factors for the pulse duration, pulse shape and pulse repetition frequency which result from predetermined subjective evaluation tables.

The formation of the value $\hat{U}_{IW}$ can only be made with a computer-assisted method of measurement. In doing so, the arriving pulse is first stored and analysed. The pulse amplitude $\hat{U}$ and the pulse duration t_i can directly be measured, likewise the duration until the arrival of the next pulse. The pulse shape can be determined by using the spectral analysis by means of FFT analysers or (simpler) by comparison with pre-given standard pulse shapes. From the measured values for t_i , $u(t)$ and f_0 the factors k_1 , k_2 and k_3 are determined using the subjective evaluation tables, and finally the value $\hat{U}_{IW}$ is determined therefrom.

When evaluating the quality of transmission paths, this value $\hat{U}_{IW}$ may serve as a measure for the important parameter "impulsive noise". In respect to its admissible value, however, further studies are required.

[Weinauge, N. 1988] and [Schaffner, H. 1988] contain more information about this measurement method.

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. Regarding the latter method, further developments and results have become available [CCIR, 1986-90a] and [CCIR, 1986-90c] and these are detailed in Annex V to Recommendation 644-1 — (Düsseldorf, 1990). 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

SCHAFFNER, H. [November 1988] - Untersuchungen zur subjektiven Bewertung und objektiven Messung impulsartiger Störungen im Hörrundfunk-Tonkanal (Investigations of subjective assessment and objective measurement of impulsive noise in sound broadcasting channels). Paper presented at the eighth meeting of "Broadcasting" of ITG, Mainz, Federal Republic of Germany. (See proceedings).

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.

WEINAUGE, N. [1988] - Beitrag zur subjektiven Bewertung und objektiven Messung impulsartiger Störungen im Hörrundfunk-Tonkanal (Contribution to subjective assessment and objective measurement of impulsive noise in sound broadcasting channels) Dissertation at the College of Traffic and Telecommunication Dresden, 1988.

CCIR Documents

[1966-69]: CMTT/165 (Italy).

[1970-74]: CMTT/184 (OIRT).

[1978-82]: CMTT/239 (United Kingdom).

[1986-90]: a. CMTT/85(10/99) (Australia); b. CMTT/180 (German Democratic Republic); c. CMTT/248 (Australia).

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)

[1986-90] CMTT/232 (Yugoslavia).

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):
 $-15 + 10 \log 12 \approx -4 \text{ dBm0}$
- conventional load:
 $-1 + 4 \log 12 = +3.3 \text{ dBm0}$
- peak level exceeded with a probability of 10^{-5} times:
 $+19 \text{ dBm0}$.

To be able to verify the compatibility of the sound-programme channel load with these values, the Administration of the Federal Republic of Germany invited the "Institut für Rundfunktechnik" in Hamburg to perform fundamental investigations concerning the amplitude statistics of broadcasting signals for the following practical cases:

- A: unchanged sound signal;
- B: sound signal changed by pre-emphasis network according to CCITT Recommendation J.17 (see also [CCIR, 1970-74a]) 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.

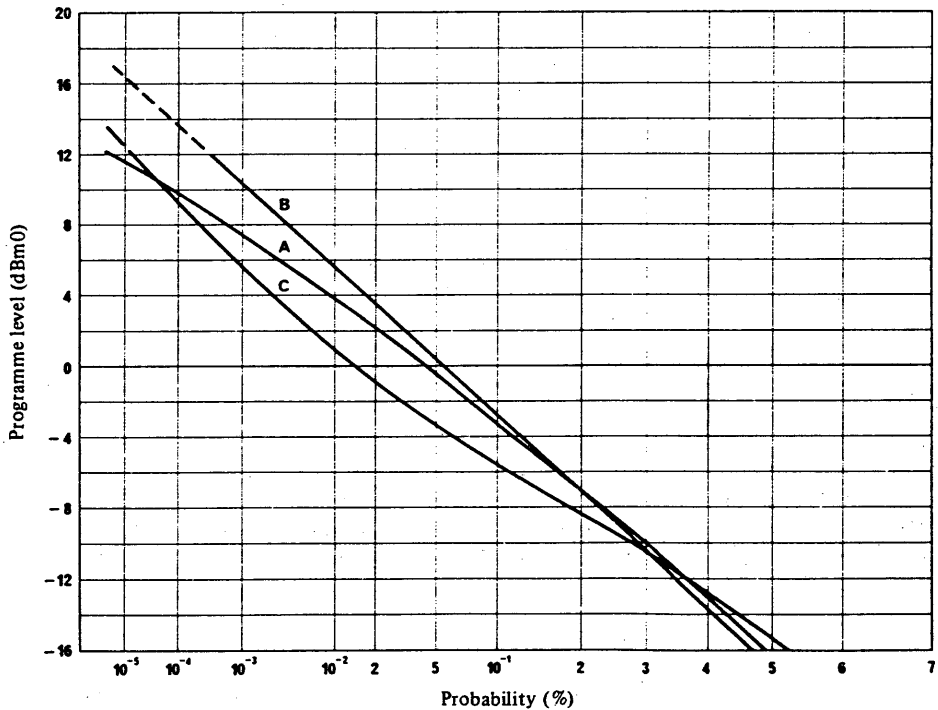


FIGURE 1 – Cumulative distribution
Cases A, B and C: Distribution of the programme level in 100% of the time
Curves A: unchanged signal
B: changed by pre-emphasis 1.5 dB/0.8 kHz
C: changed by pre-emphasis 6.5 dB/0.8 kHz and compressor

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].

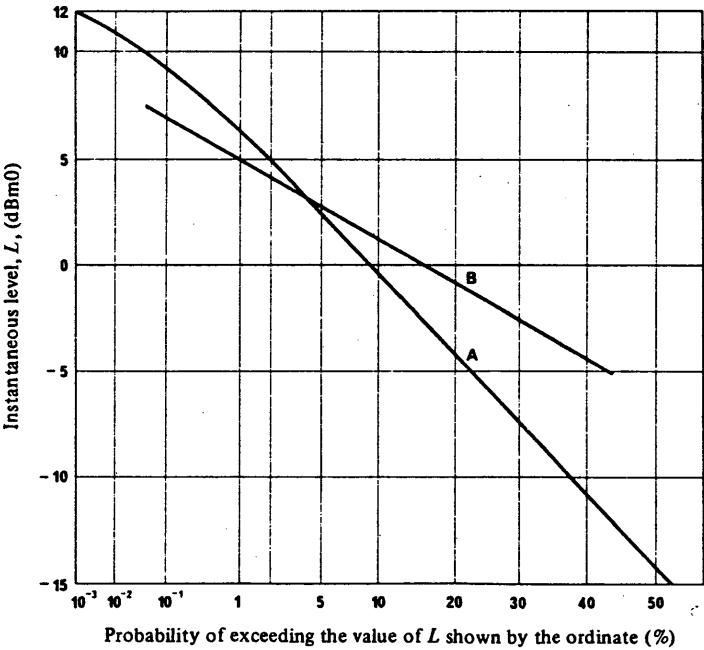


FIGURE 2 – Distribution of instantaneous level, L , expressed in equivalent sine-wave powers of the signal in a programme channel at audio frequencies (Curve A)

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 $\frac{1}{2}$ 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.

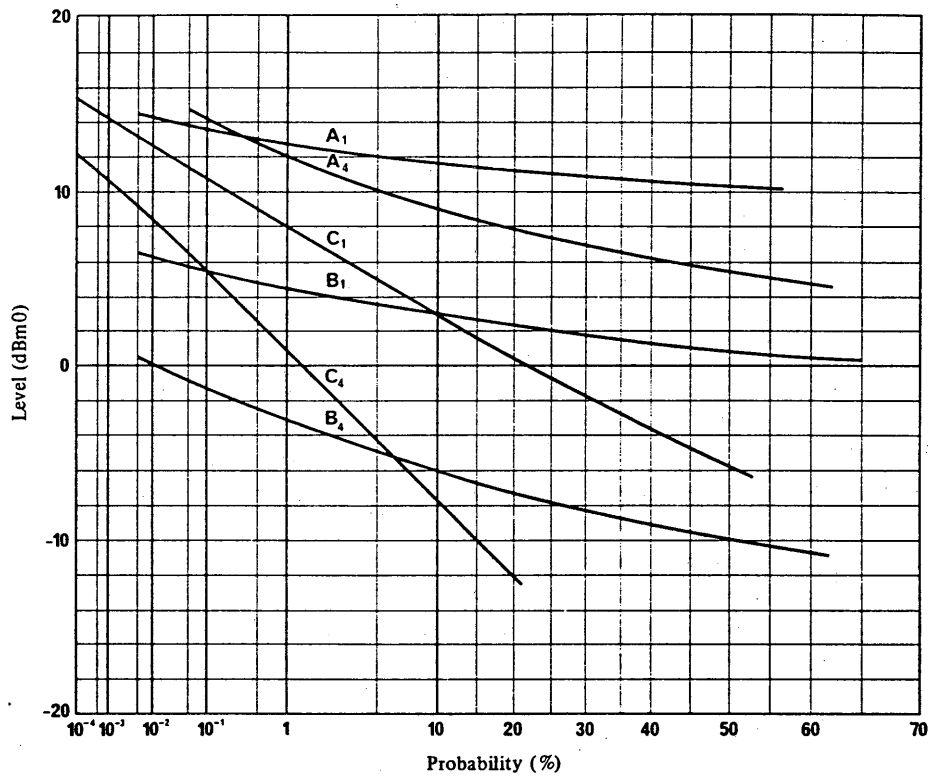


FIGURE 3 – Characteristics of unprocessed signals

- 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

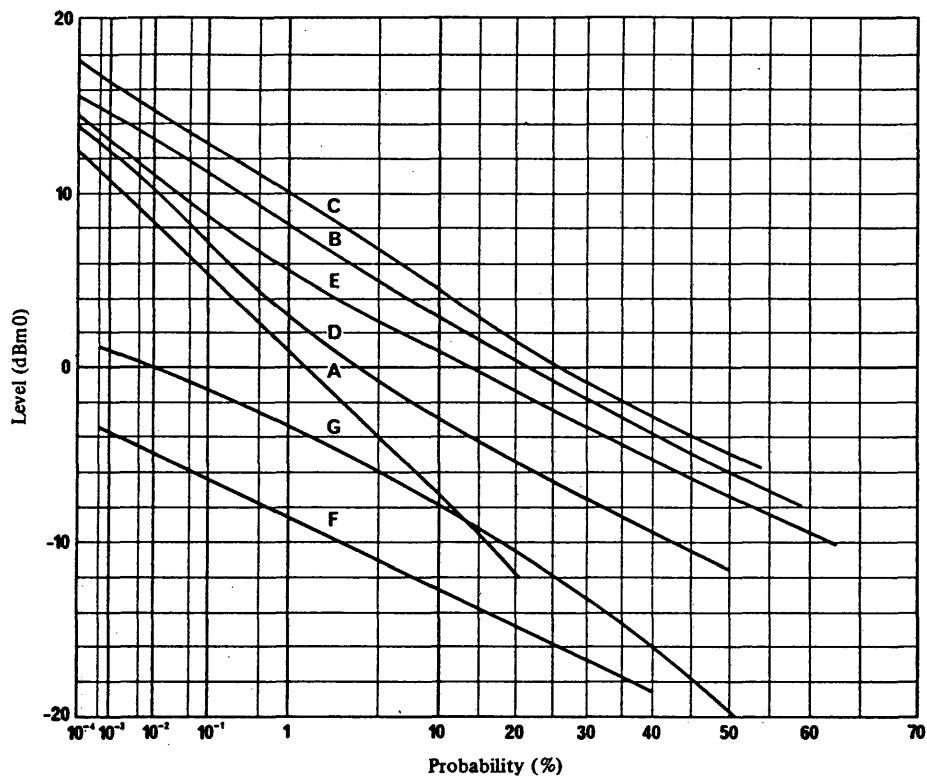


FIGURE 4 – *Instantaneous powers*

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)

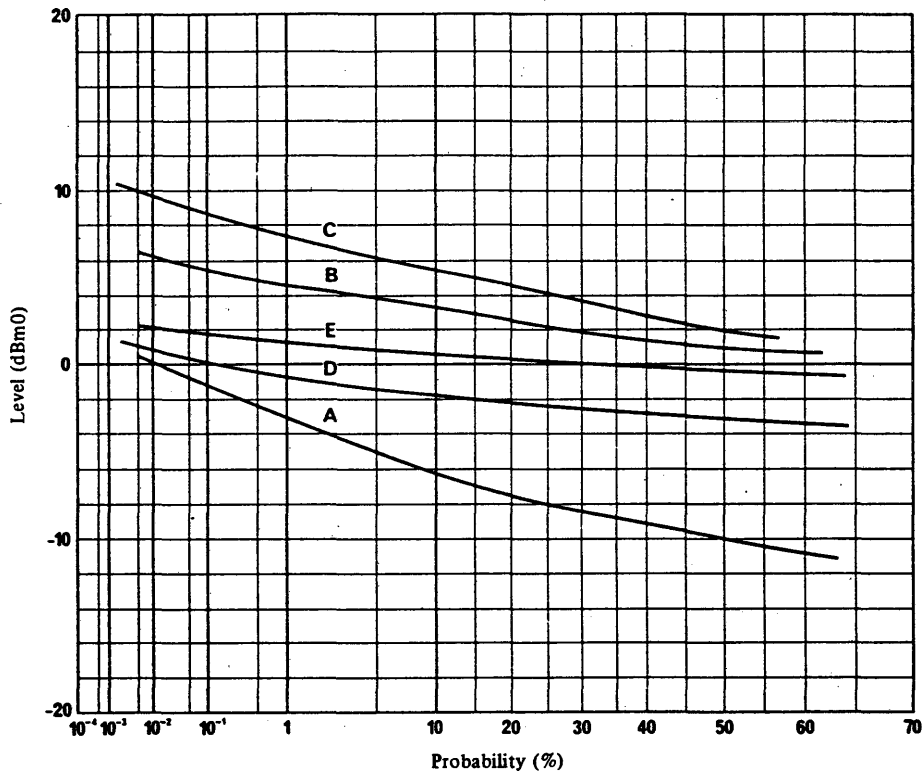


FIGURE 5 - 1-Minute mean powers

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

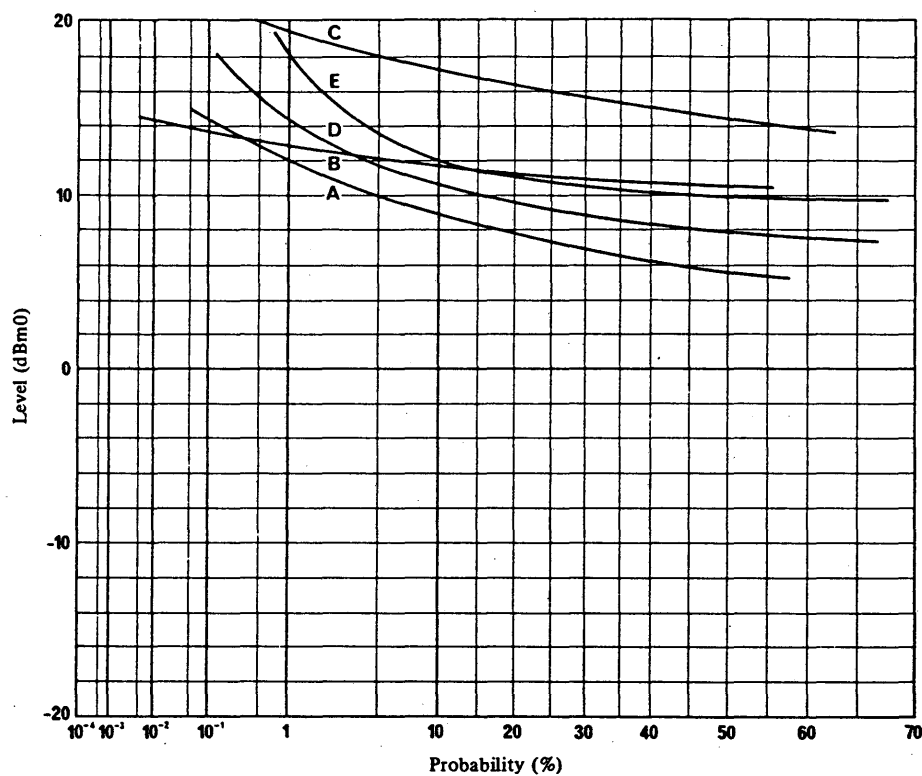


FIGURE 6 – Peak powers occurring within each 1-minute period

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

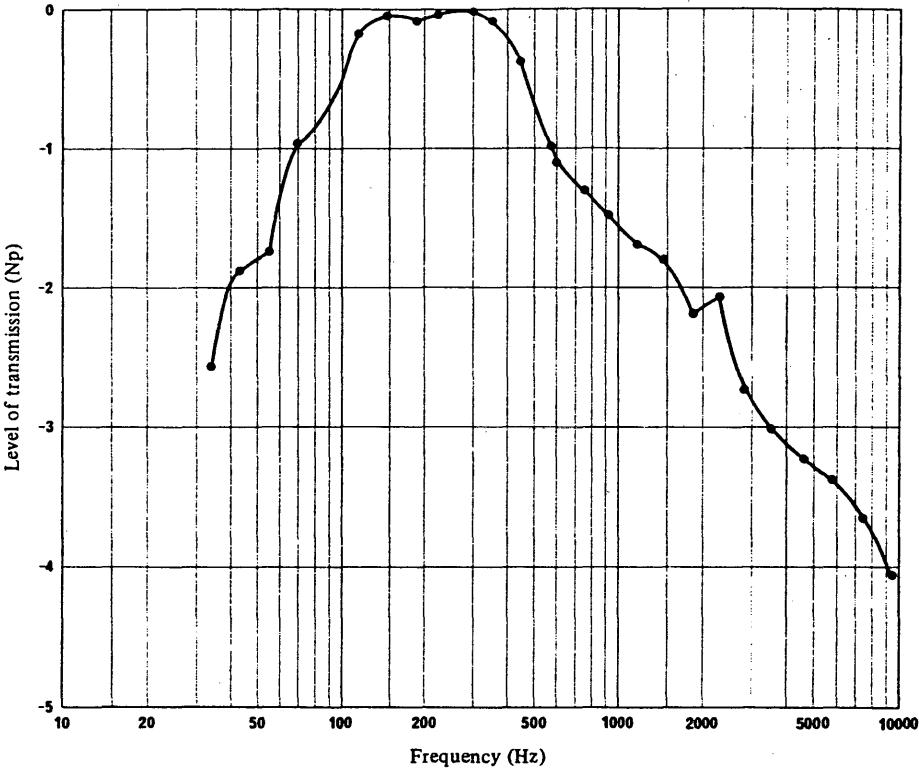


FIGURE 7 – Averaged spectral characteristics of level of transmission of male speech

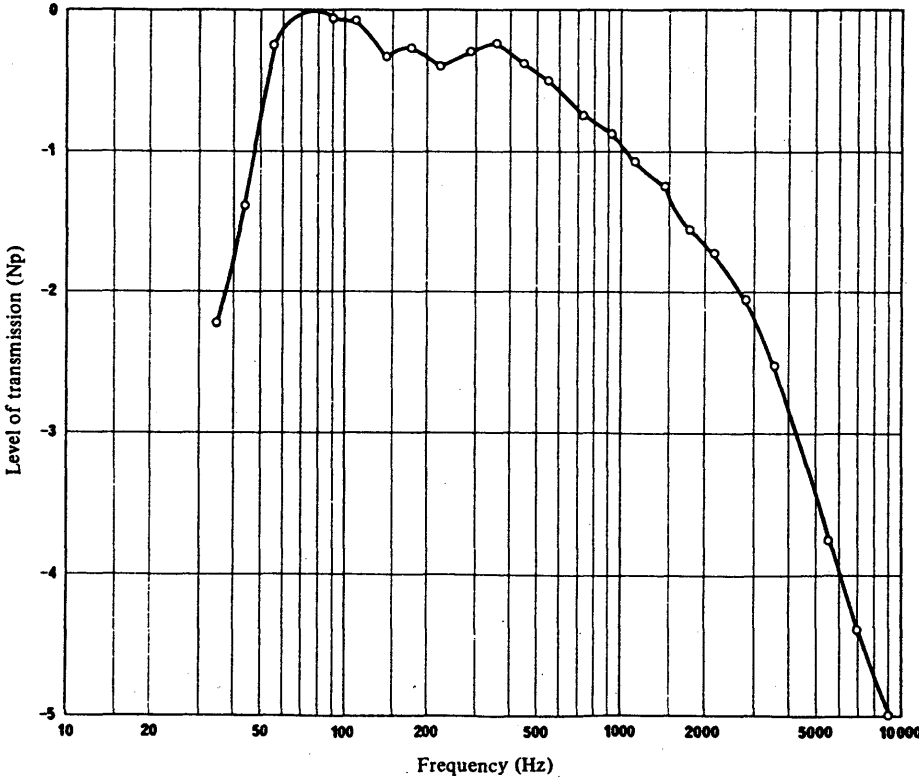


FIGURE 8 – Averaged spectral characteristics of level of transmission of a symphony orchestra

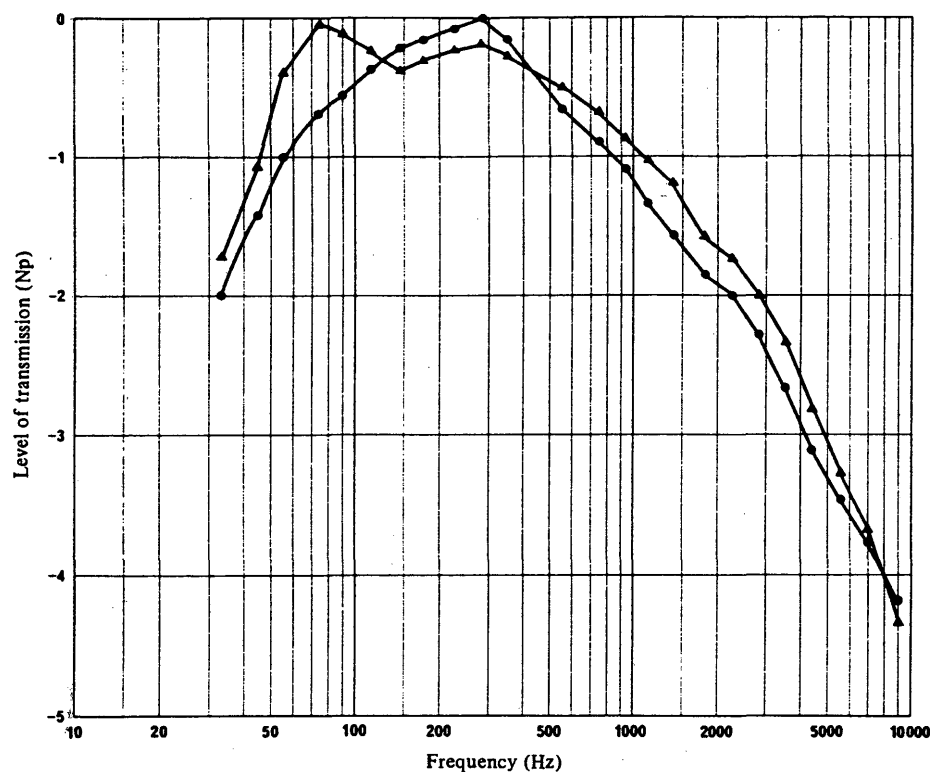


FIGURE 9 – Averaged spectral characteristics of level of transmission of daily broadcasting programme, taking account of the incidence of transmissions with different characteristics

—●— Programme No. 1

—●— Programme No. 2

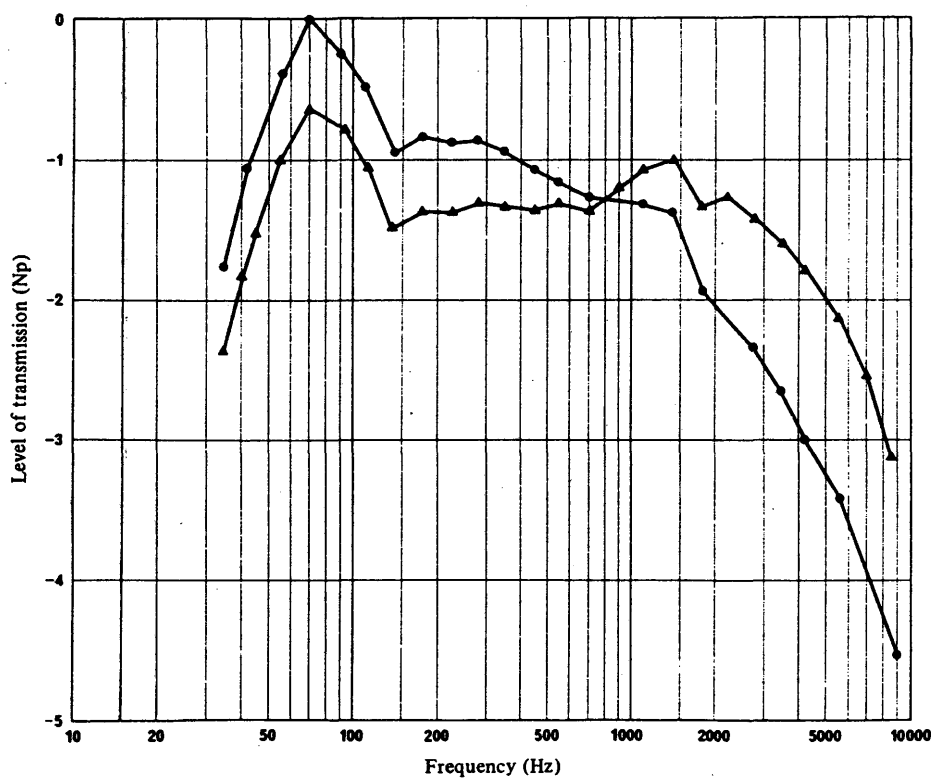


FIGURE 10 – Averaged spectral characteristics of level transmission of a light orchestra with the pre-emphasis circuit switched off and with it switched on

—●— with pre-emphasis circuit switched off

—▲— with pre-emphasis circuit switched on

3.2 Figures 11 and 12 show the spectral average power distribution for some different programme items recorded on magnetic tape [CCIR, 1966-69d].

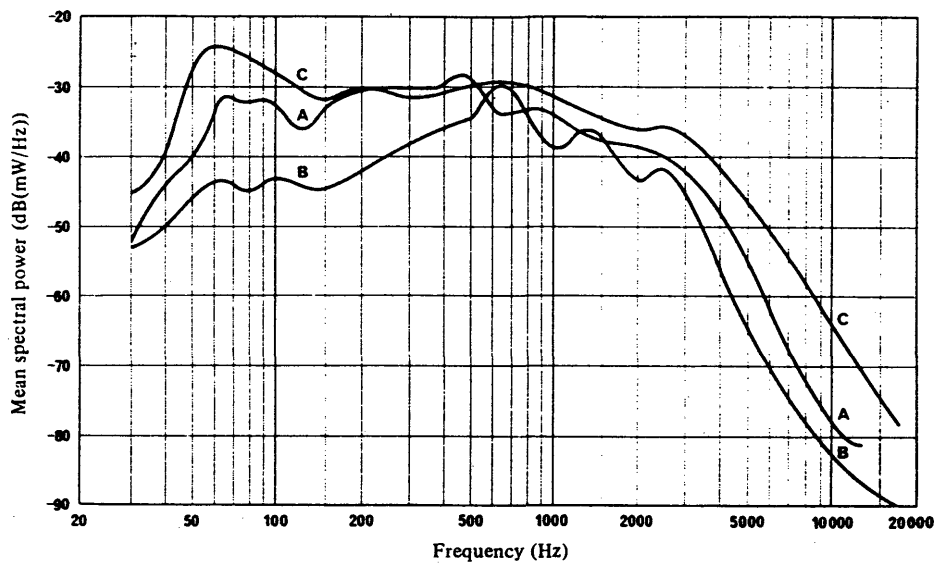


FIGURE 11 – Mean spectral power of sound signals

Curve A: classical music

Curve B: opera

Curve C: modern music

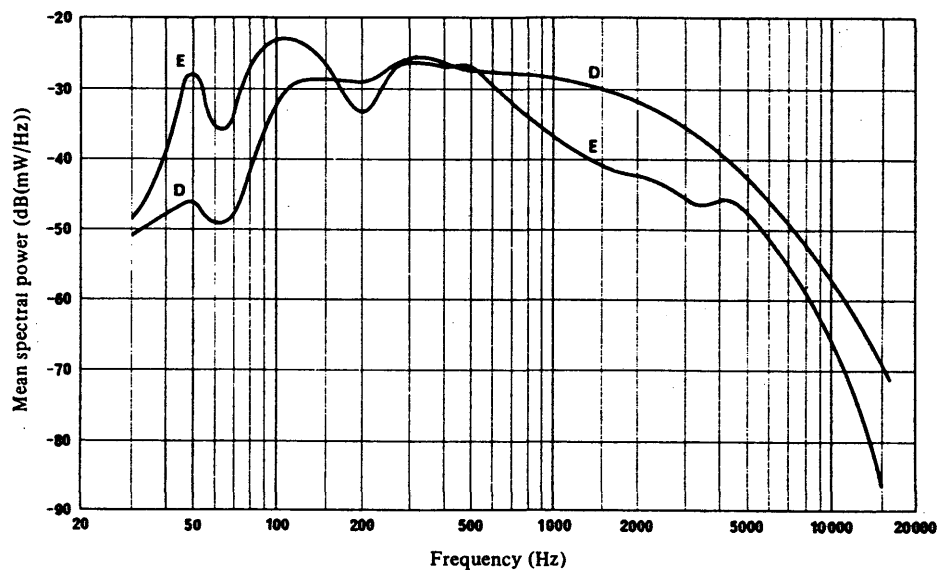


FIGURE 12 – Mean spectral power of sound signals

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 mean 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.

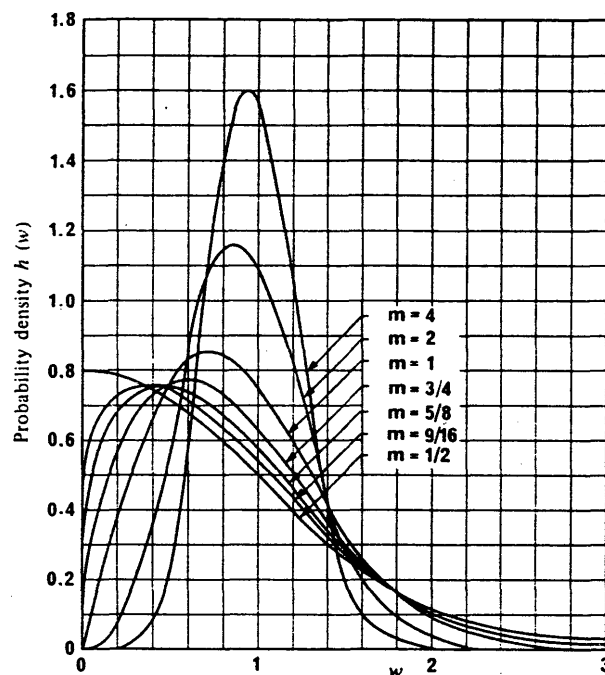


FIGURE 13 - The probability density function of the M-distribution

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.

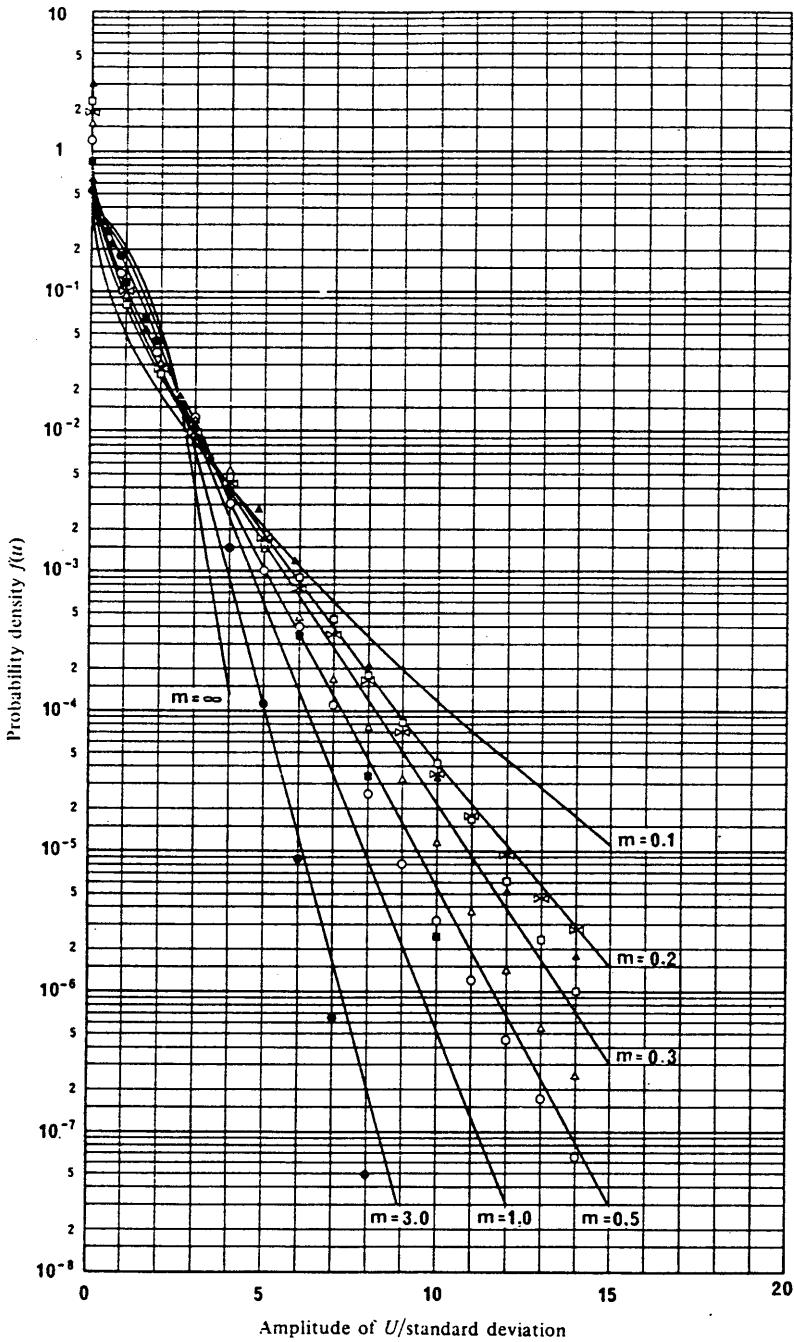


FIGURE 14 – The probability density functions of the $U(t)$ signal and sound-program signals

- 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].

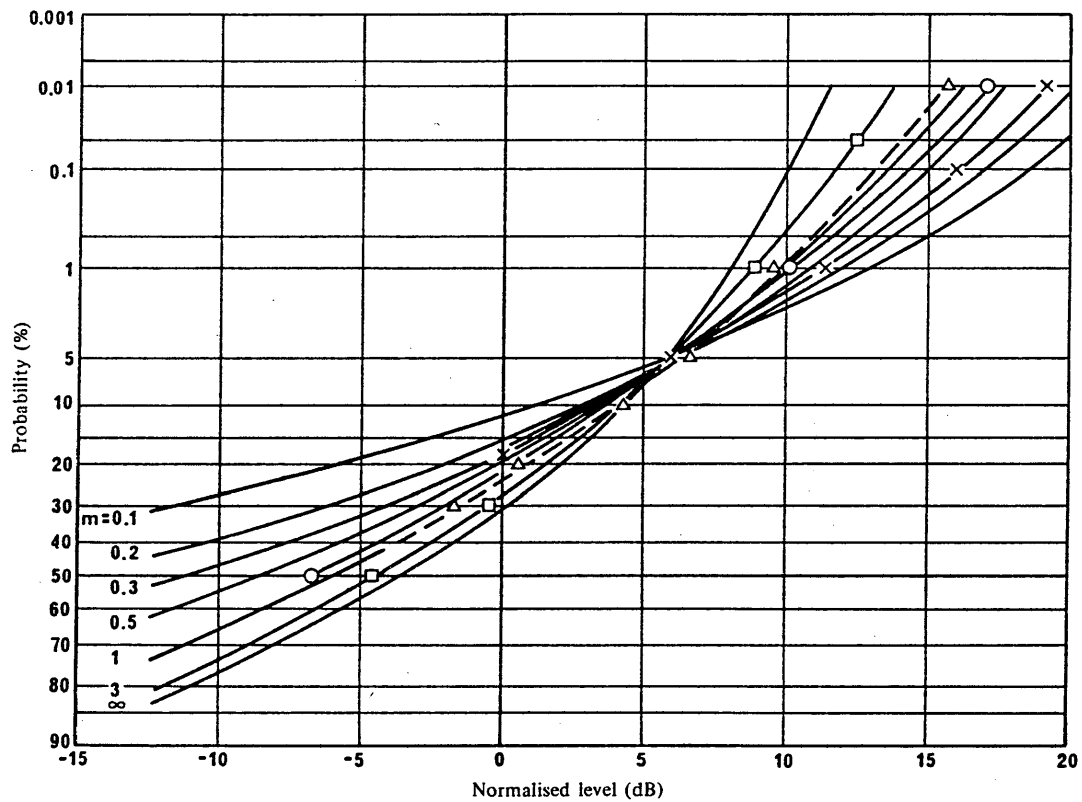


FIGURE 15 – Cumulative distribution function and data plots for instantaneous amplitude (full wave) or power

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

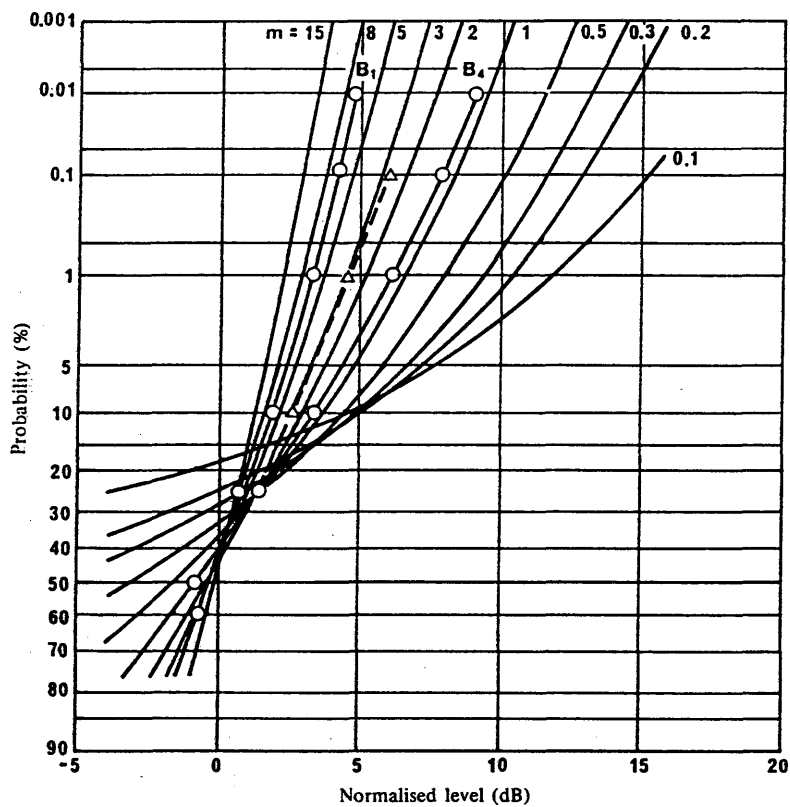


FIGURE 16 - Cumulative distribution function and data plots
for 1 min mean power

Data plotted were taken from:

- O : curves B_1 and B_4 of Fig. 3
- Δ : Tables III-A and IV

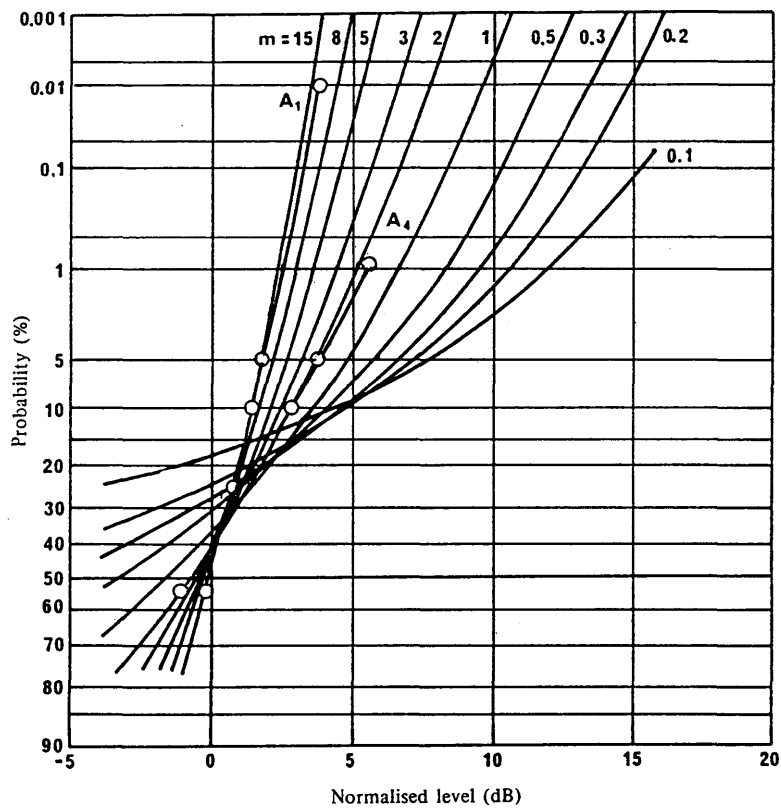


FIGURE 17 – Cumulative distribution function and data plots
for 1 min peak power

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

$$V(t) = \sum_{i=1}^n C_i \cos(\omega_i t + \varphi_i) \quad (6)$$

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.

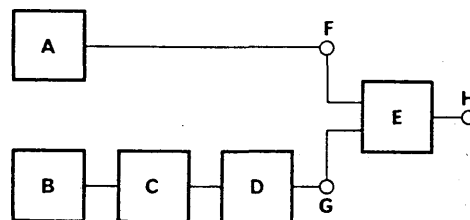


FIGURE 18 – A schematic block diagram for generating signals $U(t)$, $V(t)$, and $W(t)$

- 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.

333 *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.

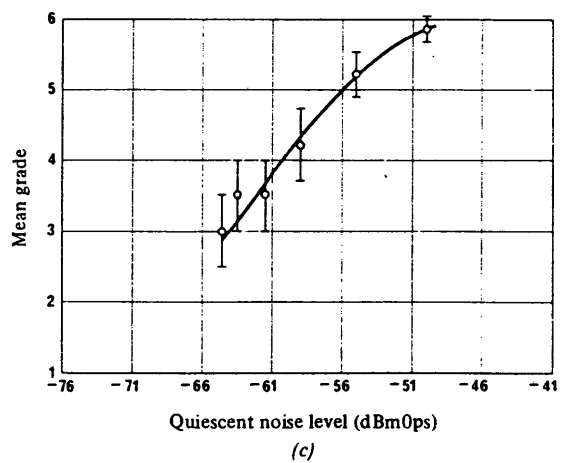
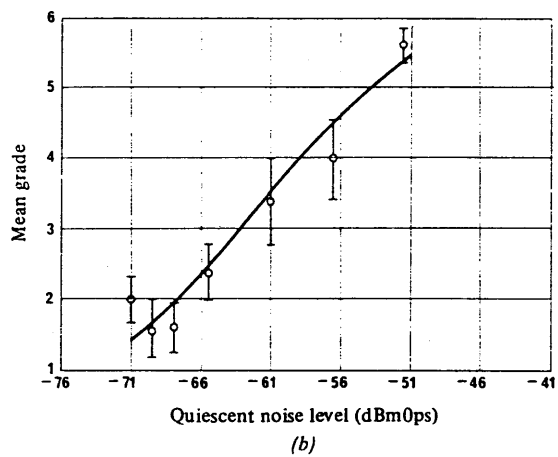
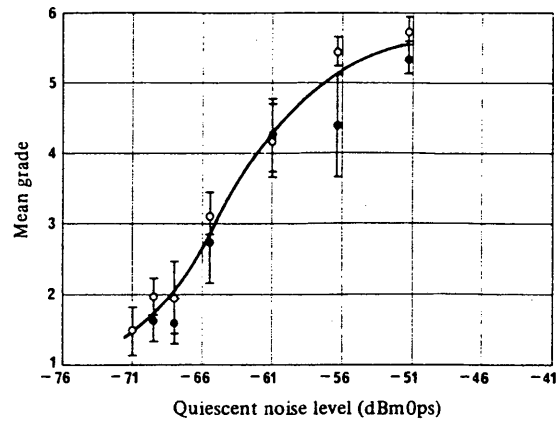


FIGURE 1 – Impairment versus quiescent noise level: simulated white noise on bearer circuit

○ standard deviation of mean ($\pm$)

(a) one modem: experienced observers

(b) one modem: inexperienced observers

(c) four modems in cascade: experienced observers

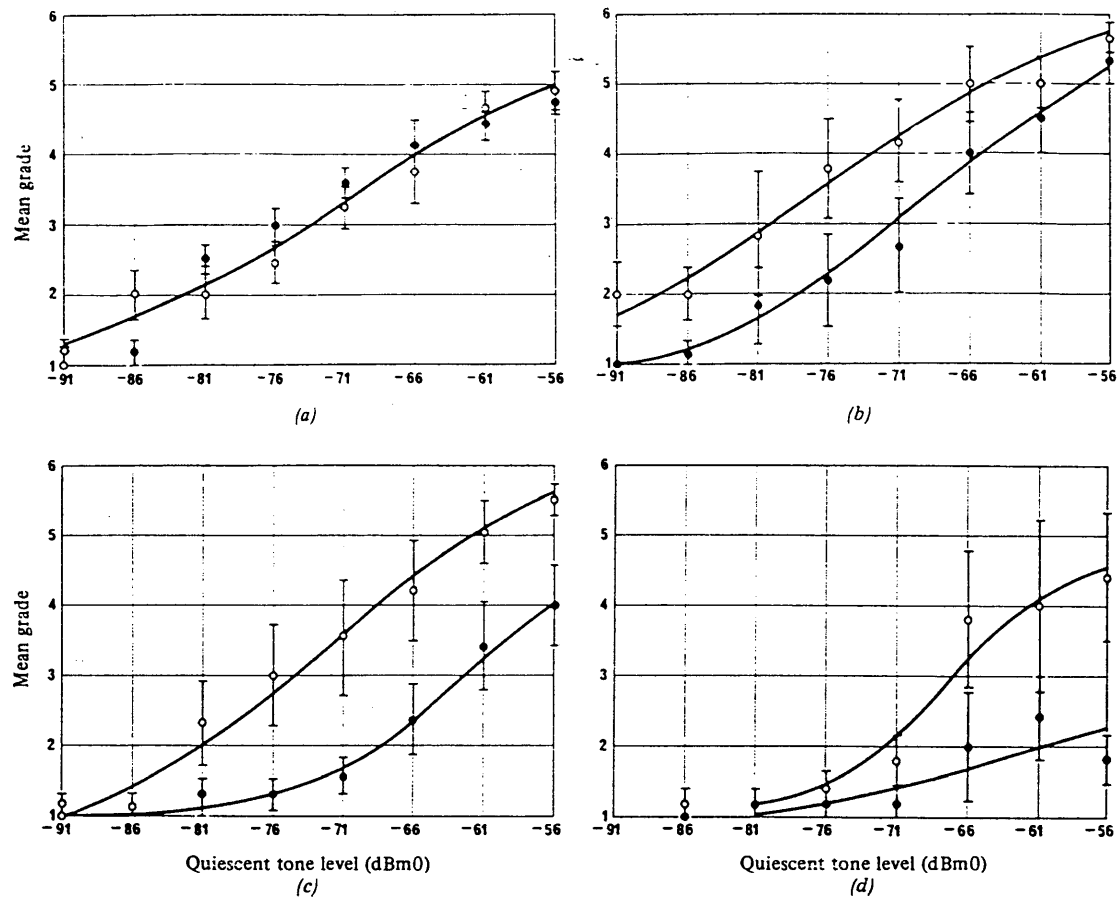


FIGURE 2 - Impairment versus quiescent tone level: simulated carrier leak

○ standard deviation of mean (±)
○ 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).

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.

REPORT 647-4 *

DIGITAL TRANSMISSION OF SOUND-PROGRAMME SIGNALS**

(Question 18/CMTT)

(1974-1978-1982-1986-1990)

1. Introduction

This Report gives information about access to 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. Access to digital networks

CCITT Study Group XVIII has developed a number of Recommendations relating to the Integrated Services Digital Network (ISDN). In Report 1241, information on the work of CCITT Study Groups XV and XVIII regarding digital transmission systems including ISDN is available.

2.1 *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, 1978-82a, b and c] or part of separate dedicated networks [CCIR, 1978-82c, d and e].

The concept of "drop-insert" facilities given in § 2.3 is appropriate to the above application.

* 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.

2.2 Network access

Examples of network access envisaged for sound-programme signals [CCIR, 1982-86a] 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, J.43 and J.44);
- 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.

————— 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).

Recommendation 724 ———— specifies codec parameters for transmission of digital studio quality sound signals over H₁ channels. The digital interface between the studio equipment and the codec is in accordance with Recommendation 647.

Recommendation 718 specifies coding at 480 kbit/s per mono channel and multiplex structures for access to the H₁₂ level and to the primary level 2048 kbit/s.

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, 1982-86b]. For this interface, a coding method is also specified [CCIR, 1982-86c].

[CCIR, 1982-86d] 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, 1982-86c] will be studied.

2.3 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;
- existing Recommendation G.802 - Interworking between Networks based on Different Digital Hierarchies and Speech Encoding Laws - was significantly amended to include specifications for interworking arrangements [CCIR, 1986-90a]. Currently the ~~revised~~ Recommendation deals with transport of 64 kbit/s digital information, including 7 kHz audio coded signals.
- 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.

2.4 Aspects of multiplexing

The technique of multiplexing high quality sound channels should be such that no bits are lost.

There are two possible realistic solutions:

2.4.1 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.

2.4.2 A technique has been described [CCIR, 1978-82f; CCIR, 1982-86e] 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, 1978-82a, b, c, e, f, g, h, i, j, k, l, m, n, o, p, q, r, s, t, and u; CCIR, 1982-86f, g, h and i]. 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, 1978-82c].

Experience [CCIR, 1986-90b] has shown that the quantizing noise has a spectrum distribution which is approximately uniform. Therefore, the weighted noise values to be expected with sine wave tones can be estimated using the formula:

$$n_n = n_o - 20 \log 2^x + K_1 + K_2 - K_3 - K_4$$

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 ($10 \log 12/8$).

K_4 : applies to instantaneous coding and takes into account the effect of varying maximum sample error magnitudes during the measurement integration period with sine wave test signals. The value of K_4 is 0 dB when the value of "x" is set for the maximum resolution, 1.2 dB when "x" is one bit less than maximum resolution and 1.4 dB when "x" is two or more bits less than maximum resolution.

Note 1 - When "x" is less than that applicable for maximum resolution the above formula produces a significant error when the test level is 3 dB or more below the top level of the segment for which "x" is set.

Note 2 - Experience with coding method B5 indicates that the cumulative value of the preemphasis plus weighting (K_1 , system B5) and the quasi-peak factor (K_2) is from 4 dB to 6.5 dB when results are compared to r.m.s. measurements without emphasis and weighting.

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	0.05-7 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⁾	0.05-7 ⁽³⁾ CCITT Rec. J.17 6.5 dB/ 0.8 kHz ⁽⁵⁾
Overload point ⁽⁶⁾ (dBm0s)	+12		+12	+12	+12	+12	+12	+15 ⁽⁷⁾	+15
Sampling frequency (kHz)	32		32	32	16	32	32	32	16
Companding law	5 ranges		5 ranges	5/3 ranges ⁽⁸⁾	5/3 ranges ⁽⁸⁾	13 segments	7 segments	11 segments	11 segments
Bit rate reduction (bits)	14/10.1 (12)		14/10.1 (12)	14/9.5	14/9.5	14/10	13/11	14/11	14/11
Finest resolution (bits/sample)	14		14	14	14	14	13	14	14
Corresponding noise (dBq0ps)	-65		-65	-65	-65	-66	-55	-62	-60
Coarsest resolution at overload point/ f_0 (bits/sample)	10		10	10	10	8	10	9	9
Corresponding noise (dBq0ps)	-41		-41 (13)	-41	-39	-30	-37	-33	-30
Resolution at +9 dBm0s/60 Hz (bits/sample)	12		12	12	12	10	10	12	12
Corresponding noise (dBq0ps)	-55		-55	-55	-53	-42	-37	-50	-48
Source coding (kbit/s)	323		323 ⁽⁹⁾	308	308	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, 1978-82n, o, p, v, w]		UK [CCIR, 1978-82c]	USSR [CCIR, 1982-86j, k]	USSR [CCIR, 1982-86k]	Italy [CCIR, 1970-74d; 1978-82g, k]	Japan [CCIR, 1978-82s, v, w]	Switzerland, Federal Republic of Germany, the Netherlands, Denmark, Norway, Canada, USA [CCIR, 1978-82a, h, i, j, m, r, t, u, v, w, x, 1982-86l, 1]	Federal Republic of Germany, the Netherlands, [CCIR, 1978-82a, h, i, j, u]

 f_0 : zero loss frequency of pre-emphasis.

Notes to Table 1:

- (¹) Performance characteristics for analogue 15 kHz type sound-programme circuits are given in Recommendation 505 and the proposals in Table 1 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, 1978-82p].
- (³) Performance characteristics for analogue 7 kHz type sound-programme circuits are given in Recommendation 503 and the proposals in Table 1 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, 1978-82c].
- (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.
- (12) Including scale factor bits.
- (13) This figure requires correction for the effects of varying level and varying bit resolution on the quantization noise in a manner similar to that described in section 3 for the factor "K₄". This work should be carried out in the next study period.

The weighted noise levels calculated using the above-mentioned formula exclusive of K_4 and considering the proposed range of sample word resolutions and the two proposed overload levels, restricted to 15 kHz bandwidth, are listed in Table II. For instantaneous coding the figures should be corrected with K_4 .

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 ₀ = +12 dBm0s			n ₀ = +15 dBm0s
		Rec. J.17 de-emphasis with 6.5 dB gain at 0.8 kHz	B1 de-emphasis	Without de-emphasis	Rec. J.17 de-emphasis with 6.5 dB gain at 0.8 kHz
14	84	-65	-66	-61	-62
13	78	-59	-60	-55	-56
12	72	-53	-54	-49	-50
11	66	-47	-48	-43	-44
10	60	-41	-42	-37	-38
9	54	-35	-36	-31	-32
8	48	-29	-30	-25	-26

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	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		-57	-51	-45	-39	-33			
B6	-60		-54	-48	-42	-36	-30			

(1) [CCIR, 1982-86 k].

(2) μ law coding and no pre-emphasis are used in this system.

(3) The values given are based on the theoretical constants for emphasis, weighting and r.m.s./quasi peak conversion given for the noise expression in section 3. The noise produced by real systems may be different (see Note 2, section 3).

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 I 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, 1978-82y]. 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, 1978-82z]. Thus the choice of 32 kHz as the sampling frequency appears to be satisfactory [CCIR, 1970-74e, 1978-82v]. 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, 1982-86m].

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, 1978-82s].

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, 1978-82aa; 1982-86n]. 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, 1982-86h] 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, 1982-86o].

The information communicated by CCITT Study Group XVIII on wideband speech coding [CCIR, 1982-86p] 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, 1982-86q] 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, 1982-86r and s] 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, 1978-821], 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, 1978-821], 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 pré-/de-emphasis using 14 bits for the maximum resolution. A formula to calculate idle channel noise is given in [CCIR, 1978-82k]. The value to be applied to codecs which do not use 14 bits for maximum resolution or have no pré-/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 Programme modulated noise

- Under study.

Note. – One administration [CCIR, 1978-82i] 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.

A self-tracking comb filter has been manufactured in the United Kingdom [CCIR, 1986-90c] which may be used with a standard noise measuring set with weighting to CCIR Recommendation 468 for measurement of programme modulated noise. This filter removes the test tone and its harmonically related distortion components, automatically tracking the test signal over the frequency range 1.9 to 2.1 kHz. The notch bandwidth at f_0 is approximately ± 50 Hz at -3 dB and approximately ± 2 Hz at -60 dB. The minimum notch attenuation is as follows:

f_0	$2f_0$ and $3f_0$	$4f_0$ and $5f_0$	$6f_0$ and $7f_0$
60 dB	50 dB	40 dB	30 dB

4.1.9 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 50 dB for the combination of the input and output filters of a single codec. There are two examples of out-of-band input filter as shown in Figure 1 [CCIR, 1986-90d] and Figure 2. This specification of these filters should take into account the presence of any preemphasis.

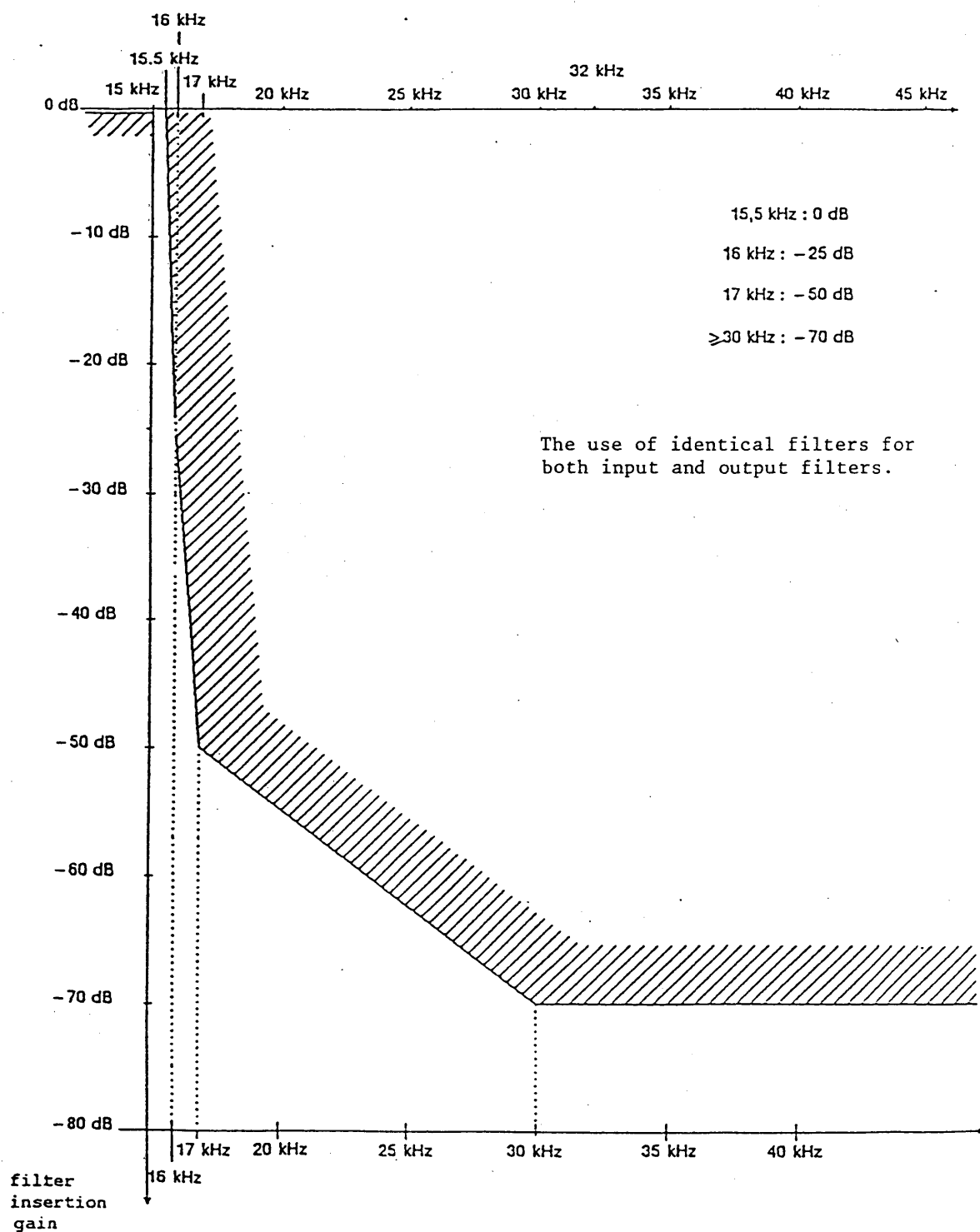


FIGURE 1

Example 1 for out-of-band input and output filters

4.1.10 *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 50 dB for the combination of the input and output filters of a single codec. There are two examples of out-of-band output filter as shown in Figure 1[CCIR, 1986-90d] and Figure 2. These filters should be specified in conjunction with the effect of $(\sin x)/x$ correction.

More work is required to finalize the specification.

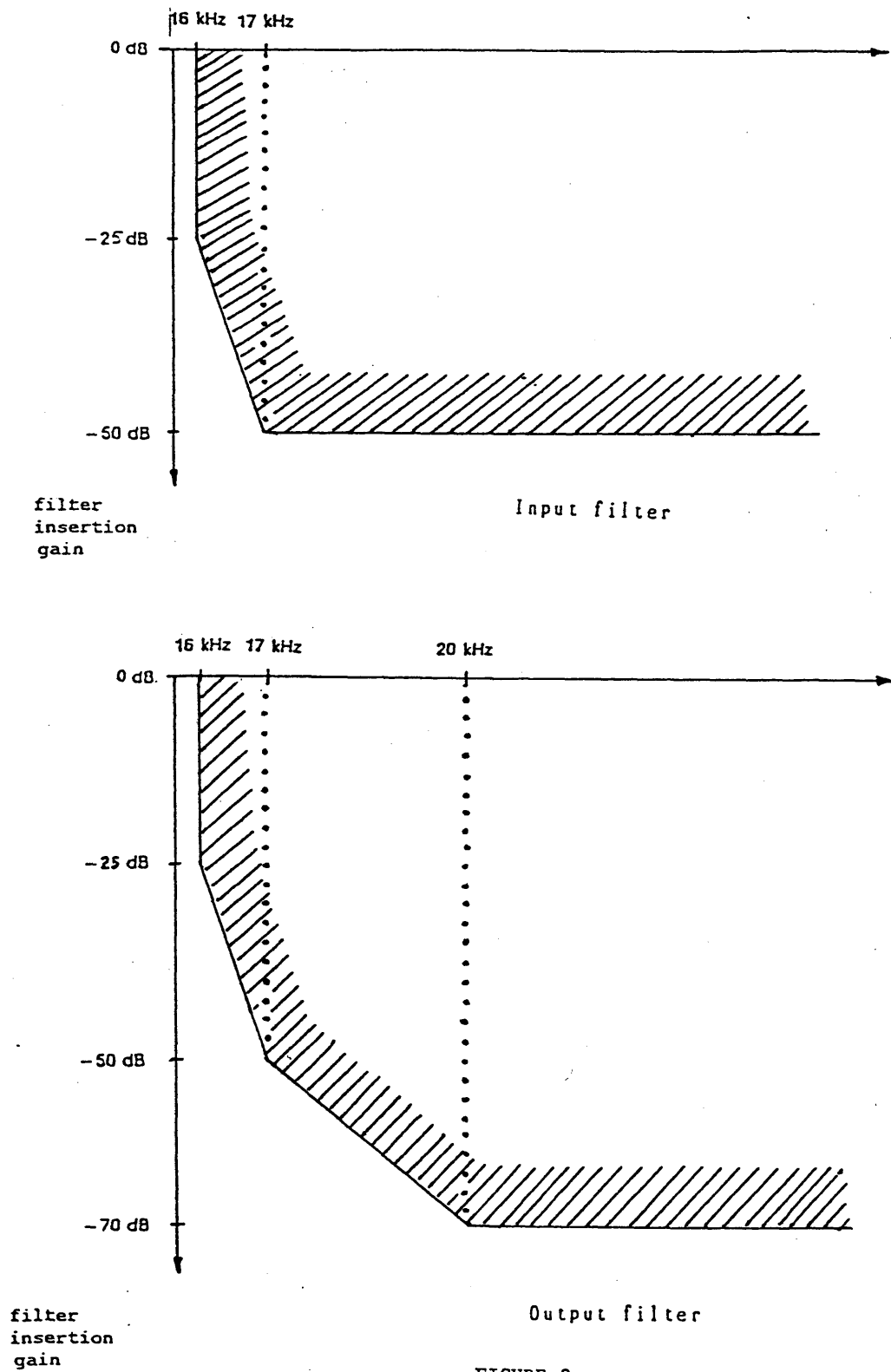


FIGURE 2

Example 2 for out-of-band input and output filters

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	<i>X</i>
0.125 to 7.5	<i>Y</i>

Note. – The values for *X* and *Y* are under study.

4.1.11.2 *Spurious in-band signals*

- With a sine-wave signal in the frequency range 0.04 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, 1978-82m], 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, 1978-821], 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, 1978-821], 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 I 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.

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)

[1978-82]: a. CMTT/82 (Germany (Federal Republic of)); b. CMTT/92 (France); c. CMTT/240 (United Kingdom); d. CMTT/90 (France); e. CMTT/91 (France); f. CMTT/231 (Japan); g. CMTT/271 (Italy); h. CMTT/234 (Netherlands); i. CMTT/251 (Germany (Federal Republic of)); j. CMTT/258 (Germany (Federal Republic of)); k. CMTT/275 (Italy); l. CMTT/276 (Canada); m. CMTT/297 (Denmark); n. CMTT/93 (France); o. CMTT/112 (France); p. CMTT/277 (France); q. CMTT/22 (United Kingdom); r. CMTT/6 (Norway); s. CMTT/51 (Japan); t. CMTT/80 (Germany (Federal Republic of)); u. CMTT/81 (Germany (Federal Republic of)); v. CMTT/35 (France); w. CMTT/110 (CCITT Study Group XVIII); x. CMTT/143 (USA); y. CMTT/247 (EBU); z. CMTT/10 (EBU); aa. CMTT/239 (United Kingdom).

[1982-86]: a. CMTT/62 (CCITT Study Group XVIII); b. CMTT/214 (Germany (Federal Republic of)); c. CMTT/202 (Germany (Federal Republic of)); d. CMTT/190 (Sweden); e. CMTT/34 (EBU); f. CMTT/48 (France); g. CMTT/12 (USA); h. CMTT/41 (United Kingdom); i. CMTT/46 (Canada); j. CMTT/10 (CCITT Study Group XV); k. CMTT/186 (USSR); l. CMTT/220 (Canada); m. CMTT/55 (EBU); n. CMTT/35 (10/26) (United Kingdom); o. CMTT/42 (United Kingdom); p. CMTT/194 (CCITT); q. CMTT/226 (Switzerland); r. CMTT/3 (Germany (Federal Republic of)); s. CMTT/4 (Germany (Federal Republic of)).

[1986-90]: a. CMTT/77 (CCITT Study Group XVIII); b. CMTT/192 (Canada); c. CMTT/187 (United Kingdom); d. CMTT/8 (France).

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.

ANNEX I

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, 1974-78a]

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, 1978-82a]

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, 1978-82b]

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, 1982-86a]

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, 1982-86b]

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

[1974-78]: a. CMTT/265 (Italy).

[1978-82]: a. CMTT/51 (Japan); b. CMTT/237 (Germany (Federal Republic of)).

[1982-86]: a. CMTT/223 (Canada); b. CMTT/241 (Canada).

REPORT 1095-1

GUIDELINES REGARDING TRANSMISSION OF DIGITAL SOUND-PROGRAMME
SIGNALS OF DIGITAL STUDIO QUALITY

(Question 18/CMTT, Study Programme 18A/CMTT)

(1986-1990)

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 a studio complex is specified. This interface has a gross bit rate of 3072 kbit/s at 48 kHz sampling frequency and is capable of carrying a stereophonic signal or up to two monophonic signals.

This report gives some guidance for the choice of transmission format for contribution and primary distribution. In Annex I coding and transmission formats, even those with a quality lower than digital studio quality, recommended by the CCIR are listed.

2. Sampling frequency

Although CCIR Recommendation 606 specifies a sampling frequency of 32 kHz for transmission of high-quality digital sound signals, a sampling frequency of 48 kHz will be used in digital studios (see Recommendation 646). Conversion of sampling frequency leads to a reduction of signal-to-noise ratio and erodes the system bandwidth; it also introduces a significant delay.

For contribution of sound-programme signals of digital studio quality the format according to ~~Recommendation 724~~ can be used.

For distribution or contribution where it may be acceptable to convert the sampling frequency from 48 to 32 kHz (e.g. where it is not likely that post-production will take place), one or the other of the systems described in ~~Recommendation 718~~ can be used.

3. Dynamic range

A dynamic range equivalent to at least 16 bits per sample is specified in Recommendation 646. The present trend seems to be in favour of a higher dynamic range, to reduce operational problems. As a minimum the transmission system must provide a dynamic range equivalent to 16 bits per sample and if possible should be extendable to 18 bits per sample or more.

4. Error protection

CCIR Report 648 suggests that a bit error ratio of 10^{-9} or better is needed if the effect of errors on high-quality digital sound signals is to be negligible. It is unlikely that digital links (even on optical fibre) will satisfy this requirement all the time so some form of error protection will be essential.

Other network impairments such as error bursts and controlled slips must also be taken into account.

5. Transmission delay

The operational effects of transmission delay are discussed in [CCIR, 1986-90a]. This document shows that in certain applications it is important that the coding/decoding delay should be as short as possible. Tolerable limits for delay are being studied by Study Group 10 (see Report 1200).

6. Ancillary data

CCIR Recommendation 647 recommends that the digital audio interface should provide a total bit rate of 2×48 kbit/s per stereophonic audio channel for the transmission of ancillary data (48 kbit/s for channel status data and 48 kbit/s for users data).

These figures seem to represent a maximum. In fact some form of processing may be envisaged to reduce the bit rate to be transmitted.

7. ISDN compatibility

The transmission system should be compatible with the user-network interfaces and transmission channels being specified by CCITT Study Group XVIII for the evolving ISDN. For high quality digital sound transmission the H0 and H1 levels have been suggested as being suitable.

Although CCIR Recommendation 660 specifies methods for transmission of a high-quality monophonic sound signal in the H0 channel (384 kbit/s), these methods use a sampling frequency of 32 kHz and a dynamic range equivalent to 14 bit/sample. Therefore, they do not meet the requirements of § 2 and § 3 above. As long as a suitable algorithm for transmission in 384 kbit/s channels is not specified, the only possibility within the ISDN lies in the use of level H1.

At the H1 level two channels are specified:

H11 - 1536 kbit/s in the 1544 kbit/s hierarchy;

H12 - 1920 kbit/s in the 2048 kbit/s hierarchy.

REFERENCES

CCIR Documents

[1986-90]: a. CMTT/20 (United Kingdom).

ANNEX I

TABLE I

Types of digital transmission formats
for sound-programme signals

CCIR Recommendation	Maximum resolution (bits)	Sampling frequency (kHz)	Channel rate kbit/s)
724 Note (1)	20	48	1920/1536 for a stereo channel
718 Note(2)	16	32	480/496
660 Note (3)	14	32	384
719 Note (3)	14	32	320

Notes to the table

1. This type is well suited for contribution applications. Post-processing of the transmitted signal is possible without restriction, because samples are carried transparently from studio to studio.

2. This type of transmission link can be applied for distribution. The system operating at 496 Kbit/s (so-called DS1) is used as a studio-to-earth station link for direct satellite broadcasting of sound programmes (Digital Satellite Radio, DSR). It is also in operation in Germany (Federal Republic of) for realizing a contribution network. The same coding and multiplexing technique is applied within the EBU, as an interim solution, for Euroradio, the exchange of sound-programme signals in satellite.

3. These systems are foreseen for use on mixed analogue-and-digital circuits.

BIBLIOGRAPHY

CCIR Documents

[1982-86]: a. CMTT/202 (Germany (Federal Republic of)); b. CMTT/214 (Germany (Federal Republic of)).

[1986-90]: a. CMTT/205 (Germany (Federal Republic of)).

REPORT 1236*

**DIGITAL TRANSMISSION OF MEDIUM- AND HIGH-QUALITY
SOUND SIGNALS AT LOW BIT RATES**

(Question 18/CMTT, Study Programmes 18A/CMTT, 18G/CMTT)

(1990)

1. Introduction

This Report gives details of new audio-frequency signal coding and compression methods much more efficient than those used at 384 kbit/s which are standardized in CCIR Recommendation 660 and used in CCITT Recommendations J.41 and J.42. Annex I contains information on subjective quality assessment of signals coded by some of these new methods of reduced bit-rate coding.

2. Reduced bit rate coding algorithms

Psycho-acoustic laws may be exploited to reduce the bit rate without causing any annoying subjective impairment.

The main categories of coding which may be contemplated in order to obtain a significant reduction in the bit rate are:

- predictive coding;
- sub-band coding;
- transform coding.

2.1 Predictive coding

This type of coding harnesses short-term linear prediction techniques, whereby the difference between the real signal and an estimated value is transmitted. By using efficient predictors, the differences and hence the bit rate can be reduced. The bit rates obtained are between 192 kbit/s and 224 kbit/s, i.e. a reduction by a ratio of 2 with respect to coding in accordance with Recommendation 660.

2.2 Sub-band coding

With this type of coding, the initial band is split up into a number of sub-bands by means of a battery of filters such as, for example, quadrature mirror filters (QMF). The signal of each sub-band is coded after sub-sampling by

* This Report should be brought to the attention of CCIR Study Group 10. It is also of interest to CCITT Study Groups XV and XVIII.

means of adaptive quantizers and/or adaptive predictors specific to each spectral zone. For high-quality sound channels (40 Hz to at least 15 kHz), configurations of 2, 3, 4 or up to 24 sub-bands or more (Theile *et al.*, 1987) may be used to obtain bit rates of the order of 128 kbit/s or lower, i.e. a reduction by a ratio of 3 or more.

2.3 Transform coding

The principle of this type of coding consists in converting a block of consecutive samples to the spectral domain (e.g. Fourier transform or cosine transform). The spectral components are then quantized in order of size, allowing for the properties of frequency masking and aural sensitivity. A dynamic allocation of the bits required for the coding of each component makes it possible to optimize the bit rate at values of the order of 128 kbit/s or lower, representing a reduction by a ratio of 3 or more.

3. Coding algorithms

3.1 Predictive coding

In the United Kingdom, the BBC is studying a predictive near-instantaneous companding technique [CCIR, 1986-90a] for the bit rate reduction of digital sound signals. Starting with sound signals sampled at 32 kHz and converted into 16-bit uniformly-coded samples, the predictive NICAM technique compresses the samples into 5-bit words for transmission. At the receiver, an inverse operation to reconstruct the signal is performed.

3.2 Sub-band coding

The sub-band coding methods proposed for the transmission of medium- and high-quality sound signals at bit rates below 192 kbit/s per monophonic channel are shown in Table I.

TABLE I
Proposed SB coding algorithms

Algorithm		SB-ADPCM					SB-coding MUSICAM
		A1	A2	A3	A4 ⁷⁾	A5	A6
Bit rate (kbit/s)		2 x 64 ⁸⁾	1 x 64 ⁸⁾	96-192 ⁸⁾	104-128 ⁸⁾	1 x 64	64, 96, 128, 192 ⁸⁾
Nominal bandwidth (kHz)		0.04-15	0.04-7	0.04-15/20	0.04-15	0.05-7	0.04-20
Sampling frequency (kHz)		32	32	48	32	16	48
Number of uniform A/D converter bits		16	16	16	16	14	16-20
F i B l a t e r	Number of Sub-bands (SB) frequency range (kHz)	4 0-2, 2-4, 4-8, 8-16	4 ¹⁾ 0-2, 2-4, 4-8, 8-16	4 ⁶⁾ 0-6, 6-12, 12-18 18-24	4 0-4, 4-8, 8-12, 12-16	2 0-4, 4-8	32 (polyphase network) 750 Hz each band
	Number of taps	32	32	32	49	24	-
	processing delay (ms)	6.75 ⁴⁾	6.75 ⁴⁾	2 ⁴⁾	3 ⁴⁾	4	24 ⁴⁾
Pre- dictor	Number of poles	2	2	2	2	2	-
	Number of zeros	6	6	6	16 in SB1-2 8 in SB3-4	6	
	coeff. up-dating	gradient method	gradient method	gradient method	gradient method	gradient method	
Quan- tizer	Total number of bits/ Number of predictor bits	³⁾ */5-*/4-*/3-*/2 in SB1-2-3-4	³⁾ 6/5-4/4-3/3-2/2 in SB1-2-3-4	³⁾ 8/4-4/2-2/2-2/2 in SB1-2-3-4	³⁾ 7/5-4/3-3/2-2/2 in SB 1-2-3-4	⁵⁾ 6/4-2/2 in SB1-2	9)
	2) leakage constant B	127/128	127/128	127/128	127/128	127/128	
Reference		Switzerland [CCIR, 1986-90b, e]	Switzerland [CCIR, 1986-90b]	Japan [CCIR, 1986-90c]	France [CCIR, 1986-90d]	CCITT G.722	France [CCIR, 1986-90f] FRG [CCIR, 1986-90g]

Notes to the table:

- 1) The input to the decoder of SB4 is made equal to 0 and the third QMF-stage in the decoder is replaced by another filter of the same order but of a lower cut-off frequency to reduce aliasing distortion. Six bits are permanently allocated to SB1, but only five are used for prediction.
- 2) Embedded 7-segment quantizer in SB1, -2 and -3; 2-bit Laplace quantizer in SB4 is embedded in the 7-segment one.
- 3) The allocation of the embedded bits is calculated for blocks of eight input samples and it is based on the logarithmic scaling factors and the instantaneous values of the difference signals (quantized with two segment bits).

Two 8-bit flags are transmitted at the rate of 250 bit/s/channel to synchronize the two 64 kbit/s channels and to identify the correct octet sequences in the receiver.
- 4) In the digital part only.
- 5) Embedded 60- and 4-level Laplace quantizers in the low- and high-sub-bands respectively.
- 6) In the 15 kHz transmission mode, the input to the decoder of SB4 is made equal to 0.
- 7) Adaptive noise shaping is used in sub-band 1 to increase the subjective quality and automatic overload control avoids any saturation of predictors and quantizers.
- 8) The different bit rates can be handled by the same decoder.
- 9) Quantizing is applied to blocks of 8 ms duration (12 normalized samples per sub-band) by taking into account the masking threshold for each sub-band.

4. Transmission characteristics

The performance of digital transmission systems should be no worse than the performance characteristics specified in Recommendations 503 and 505.

5. Programmable coding

To provide contribution, transmission and distribution links meeting different requirements, it would be desirable to specify a type of variable configuration codec at $n \times 64$ kbit/s of reasonable complexity. It should be capable, without any change in hardware configuration, of processing a wide variety of sound-programme signals to ensure optimum utilization of available transmission rates. Available bit rates must be multiples of 64 kbit/s in order to retain the cross-connect possibilities and other insert and drop functions in the telecommunication networks, including the ISDN.

This type of codec should be capable of meeting the following requirements:

- sound-programme transmission between digital studios on high quality contribution circuits with 20 kHz bandwidth (n = 3 or 6);
- high-quality sound-programme transmission on basic and distribution networks with 15 kHz bandwidths (n = 2 or 4);
- transmission of outside broadcasts on 7 kHz circuits (n = 1);
- transmission of audio signals on the ISDN (2B + D) for audio-visual services (n = 1...2);
- sound-programme transmission over the ISDN;
- data transmission.

6. Subjective assessment of quality

Annex I outlines the results of subjective tests.

7. Contribution circuits at 384 kbit/s

Annex II discusses the future requirements for digital contribution circuits.

REFERENCES

THEILE, G., LINK, M. and STOLL, G. [1987] - Low bit rate coding of high-quality audio signals. 82nd AES Convention, Preprint No. 2432.

CCIR Documents

[1986-90]: a. CMTT/19 (United Kingdom); b. CMTT/184 (Switzerland); c. CMTT/173 (Japan); d. CMTT/245 (France); e. CMTT/185 (Switzerland); f. CMTT/213 (France); g. CMTT/194 (Federal Republic of Germany).

BIBLIOGRAPHY

CHARBONNIER, A. and PETIT, J.P. [1988] - Sub-band ADPCM coding for high-quality audio signals. ICASSP 88, New York.

ANNEX I

SUBJECTIVE ASSESSMENT OF THE QUALITY OF SOUND SIGNALS CODED USING REDUCED BIT RATE CODING ALGORITHMS

1. Introduction

On the basis of tests carried out by simulation or on laboratory models, a number of administrations (France, United Kingdom, Switzerland,

Federal Republic of Germany and Japan) and CCITT Study Group XVIII have conducted subjective assessments of signals coded using relatively complex algorithms to achieve a significant reduction in the bit rate. Subjective assessment of sound quality was conducted in line with CCIR Recommendation 562. It is not yet possible to make a direct comparison between the various sets of results since the test conditions are not the same. In this connection, the requirement for test material representative of high-quality sound has now been met by the use of the EBU-SQAM compact disk by some administrations.

2. Subjective test results

2.1 Switzerland [CCIR, 1986-90a, e]

Tests were carried out by the Swiss PTT during 1987. The first test was based on computer simulation of the algorithm in the 2 x 64 and 1 x 64 kbit/s mode (columns A1 and A2 in Table I) using the seven-grade comparison scale in the case of:

- a) 384 kbit/s coded (Recommendation 660, method a) versus 2 x 64 kbit/s SB-ADPCM for male/female speech, piano, trumpet, gong, orchestra (electronic), glockenspiel and violin;
- b) 2 x 64 versus 1 x 64 kbit/s mode of the SB-ADPCM algorithm for female speech, trumpet, orchestra (classical music) and vocal music;
- c) 2 x 64 kbit/s codec at a bit error rate (BER) of $BER = 0$ versus $BER = 10^{-4}$ for male speech and piano.

The results indicate no significant difference between the 384 and 2 x 64 kbit/s codecs. On the seven-grade scale, all signals considered, averages of 0.11 and 0.03 were obtained respectively by non-expert and expert listeners, favouring the 384 kbit/s coder. Very good results were obtained even at 1 x 64 kbit/s. Furthermore, no special data protection techniques are required for transmission error rates of up to at least 10^{-4} .

The results under a) and b) have also been confirmed by tests carried out on the hardware model.

Further formal tests [CCIR, 1986-90e] have been carried out on hardware models according to CCIR Recommendation 562-2 using the seven-grade comparison scale for the case of 512 kbit/s PCM vs. 2 x 64 kbit/s SB-ADPCM with dynamic bit-allocation in all four sub-bands using extracts from the EBU-SQAM compact disk. All signals considered, an average of 0.005 was obtained indicating practically no difference between these two coding methods.

2.2 CCITT COM XVIII/8 [CCIR, 1986-90b]

During 1985 and 1986, COM XVIII/8 carried out mean opinion score (MOS) and paired comparison tests for speech and a limited number of music signals to evaluate the codec according to CCITT Recommendation G.722 (column A5, Table I of this document). Good results were obtained also for music signals in the 64 kbit/s mode of operation.

2.3 France [CCIR, 1986-90c, i]

A study carried out by CNET shows that one of the possible coding methods uses a three sub-band configuration with an allocation of 7, 4 and

2 bit/s sample respectively. Informal listening tests have indicated a coding quality which would be acceptable for the transmission of high-quality sound.

Later tests have used a four sub-band configuration (column A4 of Table I) with an allocation of 7, 4, 3 and 2 bit/s sample respectively.

2.4 United Kingdom [CCIR, 1986-90d]

In the United Kingdom, the BBC is currently studying a predictive NICAM technique for sound signal bit rate reduction. Preliminary results indicate that transmission of high-quality sound signals can be achieved with as little as 5 bit/s per sample, making it possible to transmit a stereophonic signal in an HO channel.

2.5 Japan [CCIR, 1986-90f]

A study carried out in Japan shows that the proposed algorithm (column A3 in Table I) has sound signal quality almost equal to the original at the 15 kHz and 20 kHz transmission modes. It also shows that an adaptive bit allocation scheme makes it possible to further improve the quality of coded sound signals.

2.6 France, Federal Republic of Germany [CCIR, 1986-90g, h]

Informal listening tests carried out by the CCETT in France and the IRT in the Federal Republic of Germany showed different quality levels for the sub-band coding method, MUSICAM.

a) 64 kbit/s

The subjective quality is close to that of compact disk for most of the test material - it is considered adequate for high quality commentary channels.

b) 96 or 128 kbit/s

The subjective quality is identical to that of compact disk for any critical test material. It is considered entirely sufficient for advanced radio-sound programme emission systems planned for terrestrial and satellite broadcasting for mobile reception.

c) 192 kbit/s

The subjective quality is identical to that of studio signals so that critical test material has a "headroom" for post-processing such as equalizing, editing and cascading (multiple coding and decoding). The quality is thus sufficient even for contribution links. This application is discussed in Annex II.

REFERENCES

CCIR Documents

[1986-90]: a. CMTT/63 (Switzerland); b. CMTT/77 (CMTT, SG XVIII); c. CMTT/7 (France); d. CMTT/19 (United Kingdom); e. CMTT/255 (Switzerland); f. CMTT/173 (Japan); g. CMTT/213 (France); h. CMTT/194 (Federal Republic of Germany); i. CMTT/245 (France).

BIBLIOGRAPHY

CCIR Document

[1982-86]: CMTT/194 (CCITT).

ANNEX II

TRANSMISSION OF DIGITAL STEREOPHONIC SOUND-PROGRAMME SIGNALS OF
DIGITAL STUDIO QUALITY ON CONTRIBUTION CIRCUITS AT 384 KBIT/S

Highly sophisticated modern source coding schemes give the possibility to realize contribution links for carrying stereophonic programmes on channels at 384 kbit/s. As long as ISDN-HO channels are not available, the network should be accessed according to CCITT Recommendation G.737.

Telecommunication organizations should offer world-wide the possibility to feed the coded digital studio-quality sound-programme signals by multiplexing into the digital networks and to route these 384 kbit/s signals to the destination.

In future when ISDN is commonly used and the HO access level is available world-wide, it will be easy to adopt interfaces to the ISDN. Then it can be envisaged that e.g. North America and Europe could exchange studio-quality programme signals without the necessity for further processing.

Another very attractive feature for broadcasters and audio engineering companies can be realized in the ISDN-world: the possibility for the user to determine the signal routing by himself according to the operational requirements (switching by dialling).

A list of user requirements for the digital transmission of sound-programme signals of studio quality is presented in CCIR Report 1095. These requirements also have been discussed in context with the sub-band coding method (columns A1, A2 and A6 in Table I) [CCIR, 1986-90a, b].

REFERENCES

CCIR Documents

[1986-90]: a. CMTT/185 (Switzerland); b. CMTT/206 (Federal Republic of Germany).

REPORT 648-4*

DIGITAL SOUND-PROGRAMME TRANSMISSION IMPAIRMENTS AND METHODS OF PROTECTION AGAINST THEM*

(Study Programmes 18A/CMTT and 18D/CMTT)

(1974-1978-1982-1986-1990)

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.

Information on error performance objectives and slip rate objectives is to be found in Report 1241.

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;	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	detection of 6 non-zero syndromes within 2 blocks of 13 syndromes each, causes the 13 samples of the former block to be muted	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. Information on jitter characteristics of digital transmission systems is given in Report 1241. 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.

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.

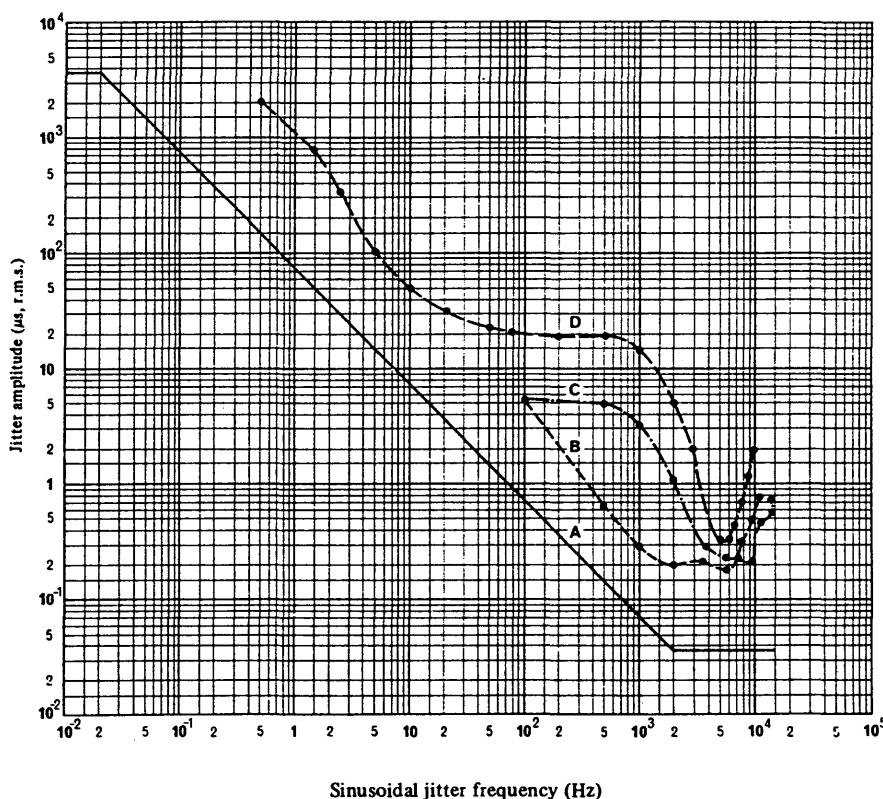


FIGURE 1 – Proposed maximum allowable r.m.s. amplitude of sinusoidal timing jitter in high-quality sound signals

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])

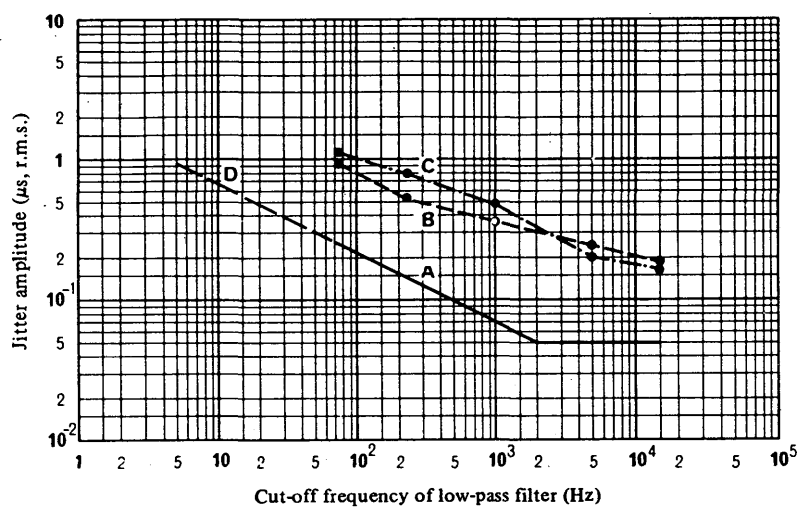


FIGURE 2 – Proposed maximum allowable r.m.s. amplitude of random timing jitter in high-quality sound signals

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).

ANNEX I

ANSWER TO THE LIAISON STATEMENT FROM
CCITT STUDY GROUP XVIII ON SLIPS ON
DIGITAL SOUND PROGRAMME CHANNELS (H1 CHANNEL)

(Liaison statement to CCITT Study Group XVIII, Ref. Q7, SG XVIII)

CMTT takes note of the objective indicated in Table I/G.822.

CMTT is of the opinion that five controlled slips per 24 hours may be acceptable for systems which have been specifically designed to resist controlled slips. However in existing systems, slips will in general introduce significant impairments (i.e. clicks), sometimes more severe than caused by long bursts of errors. A precise indication of the required performance would require further studies.

Table I/G.822 is based on worst-case configurations. Information on typical performance of practical systems is needed.

SECTION CMTT D: METHODS OF OPERATION AND ASSESSMENT OF PERFORMANCE OF SOUND-PROGRAMME TRANSMISSION CHANNELS

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.

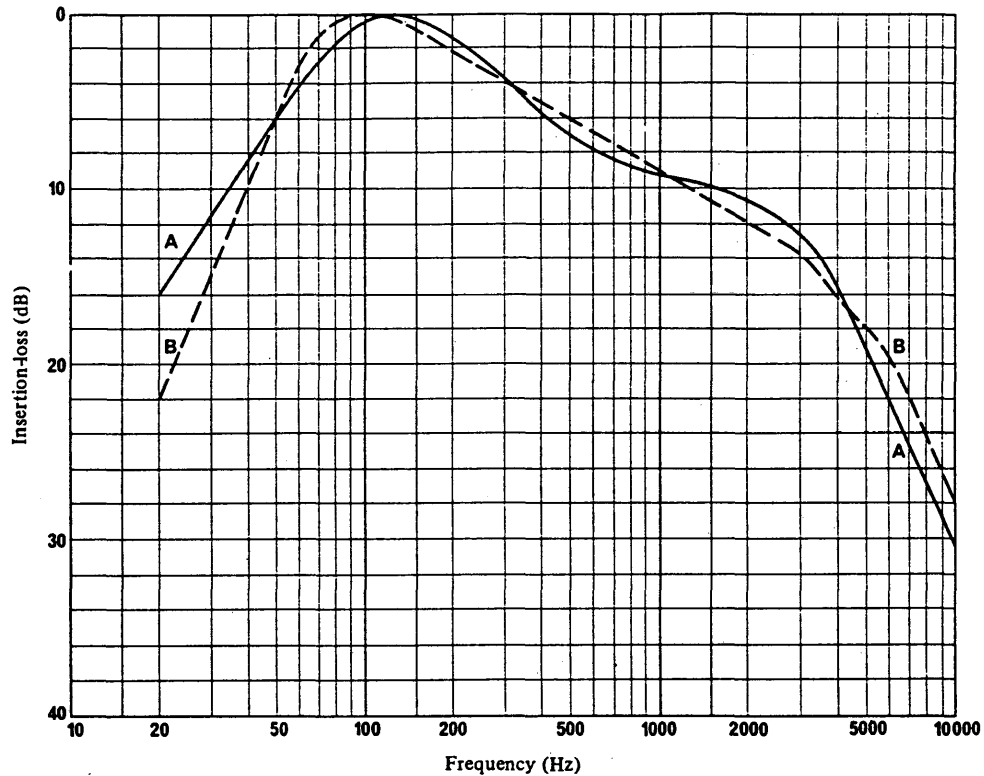
Note. — In Recommendation 559, for the objective measurement of radio-frequency protection in LF, MF and HF broadcasting, a similar test signal based on filter-shape white noise is recommended. In this case not only the insertion loss of the shaping filter is specified but also the frequency spectrum 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

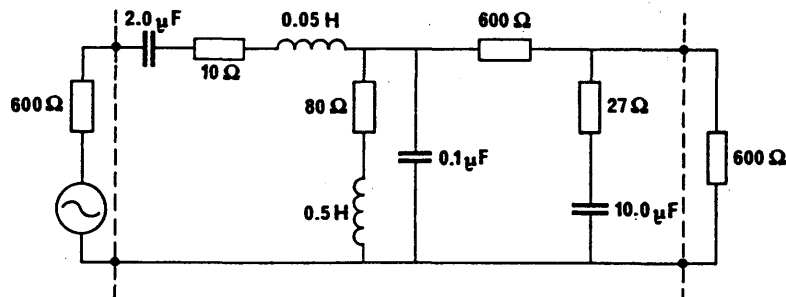
Note 1. — The integrated loss of the network complying with Table I has been calculated as 14.4 dB for white noise in the band 0 to 15 kHz.

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.

REPORT 820-2*

**RELATIVE VALUES OF SOUND-PROGRAMME SIGNAL LEVELS
ESTABLISHED WITH THE VU METER AND WITH A
PEAK-PROGRAMME METER**

(Study Programme 17A/CMTT, Kyoto, 1978)

(1978-1986-1990)

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.)

Where the impedance is not specified, the corresponding power level cannot be calculated and the terms dBm, dBm0 and dBm0s are inappropriate.

In these cases the term dBu, dBu0 and dBu0s should be used to refer to the absolute voltage level (see Recommendation 574 Appendix 1 (3.2)) at the measuring point. When measuring across a 600 Ω resistive load, the levels in dBu and dBm are identical.

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 dBrs 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).
 [1986-90]: CMTT/65 (Australia)

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.

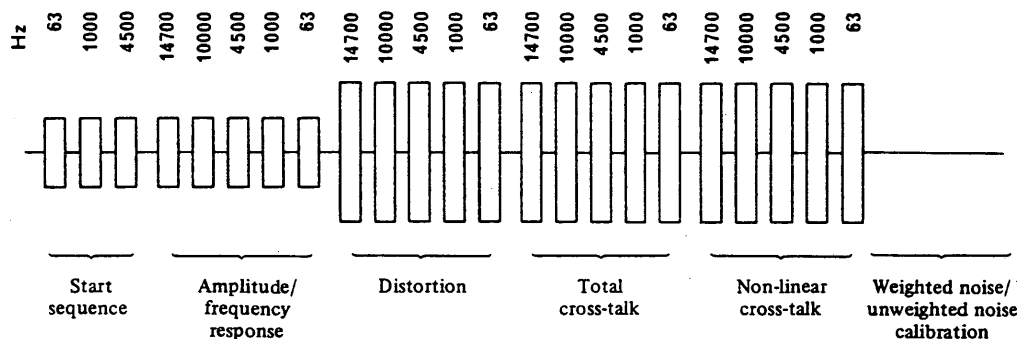


FIGURE 1

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.

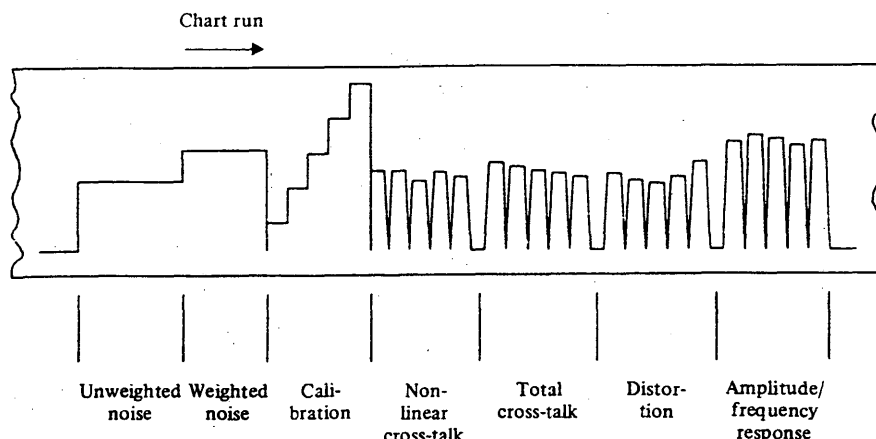


FIGURE 2

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

REPORT 412-4*

TRANSMISSION TIME DIFFERENCES BETWEEN THE SOUND
AND VISION COMPONENTS OF A TELEVISION SIGNAL

(Study Programme 21A/CMTT)

(1966-1970-1974-1978-1990)

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.

These may be categorized into two parts:

- non-synchronism of the sound and vision component (lip-sync)
- non-synchronism between the different sound components.

2. Conclusions based on observer reaction to sound and vision delay difference

Studies of observer reaction to non-simultaneous presentation of pictures and associated sound were reported in [CCIR, 1963a and b; CCIR, 1966-69; CCIR, 1986-90a and b]. 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.

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

2.2 Quantitative conclusions

The quantitative effects of sound/vision time differences are outlined in CCIR Report 1081. This report indicates that for a detectable difference a figure of 40 ms is given when the sound is delayed with reference to the vision. For sound advance, a "detectable" figure of 20 ms is quoted.

Recommendation 717 provides tolerances for the quantitative effects of sound/vision time differences.

3. Effects of time delay between sound components

The time differences between the sound signals making up the complete sound programme can cause disturbing effects which are as follows:

3.1 Starting gun effect

Poor acoustic isolation of commentary equipment can cause this effect because the commentator's microphone picks up the ambient sound. This is fed via the commentator's microphone and reaches the destination before the same sound carried on the international sound circuit is received, for example because of the use of different routes. Unless the organization delays the commentary sound, the viewer/listener will hear the starting pistol twice.

3.2 Echo effect

This effect occurs when there is an acoustic or electrical coupling (e.g. when using a studio loudspeaker instead of headphones) between the go and return directions of the circuit delayed in time.

3.3 Drain-pipe effect

In cases where the commentary and international sound are synchronous and coherent, this effect is due to a slight frequency shift of the commentary channel and occurs when the two signals are mixed by the receiving organization. This effect increases as pick-up at the commentary booth increases, leading to unwanted transmission of international sound via the commentator's microphone.

3.4 Self-echo effect

The effects of delay have also become increasingly important in radio operations where difficulties have arisen when the presenter's own speech is fed back in the cue-programme feed. It has been reported [CCIR, 1986-90c] that delays as small as 13 ms can be troublesome even to experienced broadcasters.

The tolerable differences for these effects are for further study.

4. The problem of division of time differences among the links of a built-up connection

The time differences are distributed throughout the connection and are 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. The available information on which a practical division may be based is being studied under Question 35/11.

With the advances in technology, the different types of transmission, interface, and processing equipment will give different values of time delay. In some cases significant differences between the components will arise because of multiple choices in routing and signal processing methods.

If the sound and vision components of a television signal or the sound components of the complete programme signal are transmitted along the same route, and by the same methods, there will be no perceptible time difference between the signals. Where such common routing is not feasible time differences may occur.

5. Typical time delay of sound and vision signals on various circuits

In a circuit of 2500 km (see Note 1), 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: (see Note 2)
 - 6.4 kHz bandwidth: 52.0 ms;
 - 10 kHz bandwidth: 40.0 ms;
 - 15 kHz bandwidth: 25.0 ms.
- digital codecs (for further study)
- optical fibres (for further study)

Note 1 - 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.

Note 2. - Contributions are sought from Administrations as to whether the continued inclusion of these categories of circuits is necessary.



6. Further work

The use by broadcasters of point-to-point or point-to-multipoint connections is increasing particularly for itinerant (once-only) transmissions. It is recognized that the problem of sound advance or delay is not a threshold effect and in reality, there is an increasing perception by the observer of the problem as the time-delay difference increases. However for the purposes of the CMTT, it would be convenient to specify delay limits above which corrective action is required.

Contributions are invited from broadcasters and administrations with regard to the following:

- the minimum time-difference which is generally considered practicable to achieve in digital and analogue connections;
- new experiments which introduce conditions beyond those previously studied;
- the use of the five-point subjective quality scale as described in CCIR Recommendation 562 which may be an appropriate study method.

REFERENCES

CCIR Documents

[1963-66]: a. CMTT/1 (Canada); b. CMTT/12 (United States of America).

[1966-69]: CMTT/55 (Federal Republic of Germany).

[1986-90]: a. CMTT/4 (EBU); b. CMTT/25 (United Kingdom); c. CMTT/20 (United Kingdom).

REPORT 488-5*

**TRANSMISSION OF SOUND AND/OR VISION SIGNALS BY TIME-DIVISION MULTIPLEX
OR FREQUENCY-DIVISION MULTIPLEX**

(Study Programmes 21A/CMTT and 21B/CMTT)

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

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 I — 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) 525-line (NTSC)	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 65 dB } channel 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 Koimoto, 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.

[CCIR, 1986-1990] reports on the implementation in Denmark of a digital stereo sound broadcasting system. For transmission on analogue links a digital sound in sync (DSIS) system is developed and used to carry the 728 kbit/s NICAM stereo sound signal.

For information about the NICAM 728 system see CCIR Report 795.

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.

[CCIR, 1986-1990] reports on the implementation in Denmark of a new TV programme channel with a digital stereo sound broadcasting system. For the digital transmission of the TV-signal a 140 Mbit/s codec is used in which a 728 kbit/s channel is inserted to carry the 728 kbit/s NICAM stereo sound signal.

For information about the NICAM 728 system see CCIR Report 795.

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).

[1986-1990] CMTT/260 (Denmark).

REPORT 969-1*

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)

(1982-1990)

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.

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

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].

Two contributions from the Federal Republic of Germany [CCIR, 1986-90a, b] describe a set of parameters for automatic measurement of teletext signals. This information has not been included in this report but has been fully transferred into ——— Recommendation 720.

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).

[1986-90]: a. CMTT/34 (Germany (Federal Republic of)); b. CMTT/207 (Germany (Federal Republic of)).

REPORT 970*

CHARACTERISTICS OF VIDEO SIGNALS

Limitation of out-of-band spectral components

(Study Programme 21D/CMTT)

(1982)

The spectra of video signals in general extend beyond the nominal video bandwidth. This part of the spectrum may affect other signals transmitted on the same analogue transmission path above or below the television signal. Such signals are, e.g. frequency modulated sub-carriers for sound programme transmission, 2048 Mbit/s digital signals on radio-relay systems and telephony signals or other television signals on cable systems.

For the multiplexing of signals combining filters are used, the design of which depends on the out-of-band spectra of the signals. The design of these filters must take into account interference between the multiplexed signals on the one hand and undue distortion of the television signal on the other hand. As in general the responsibilities for the signal source and the transmission path are with different authorities, optimum design can hardly be achieved. Even a well designed transmission line might give unsatisfactory results, if new test signals, signals incorporated in the television signal to provide new services, or new processing methods of the video signal are introduced after installation of the equipment. It would be of great help to designers if a limitation of the out-of-band spectral components could be specified.

Practical tests with actual multiplex designs have shown that these are capable of handling actual television programme signals [CCIR, 1978-82]. It remains to ensure that signals applied for test purposes will not impose conditions which are unnecessarily onerous, in particular in respect of the characteristics of the conventional loading signal, as defined in Recommendation 570 and further treated in Report 643.

It is therefore proposed in [CCIR, 1978-82], that for any television signal which contains completely or partially electronically synthesized components and for any other inserted signal liable to be applied to the video input terminal of multiplex transmission equipment, the power in any 4 kHz band in the frequency range above 1.2 times of the nominal video bandwidth shall be at least 50 dB below that of a sine-wave with the same peak-to-peak amplitude as the nominal video signal (as defined in Recommendation 567, § B.2.4) at a point whose choice rests with the signal generating authority.

Further study of the implications of this limitation is required.

REFERENCES

CCIR Documents

[1978-82]: CMTT/63 (LM Ericsson).

* 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.

REPORT 823-2

**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)

(1978-1986-1990)

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].

In the USSR one or more lines in each field are allocated for the transmission of information for various users at a rate of up to 9600 bit/s [CCIR, 1986-1990].

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).

[1986-1990] CMTT/188 (USSR).

REPORT 1096-1*

TRANSMISSION OF TELEVISION SIGNALS USING
MULTIPLEXED ANALOGUE COMPONENTS

(Question 24/CMTT)

(1986-1990)

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. This type of signal is very convenient for contribution circuits when the television signal is delivered in a parallel or serial component form, as in the case of digital studio signals or some electronic news gathering signals.

Various systems of this kind have also been developed for satellite broadcasting, e.g. B-MAC, C-MAC, D2-MAC, D-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.

2. Use of MAC signals over contribution circuits

When a television signal is delivered in a parallel or serial component form, as in the case of digital studio signals or some electronic news gathering signals, the use of MAC signals on analogue circuits is very convenient because it avoids the initial signal impairment due to the use of composite encoders-decoders, on top of the limitation arising from the fact that composite signals are of lower quality than the delivered ones.

When determining the MAC signal waveform to be used, line and field blanking duration as well as compression ratios for luminance and for chrominance are important parameters to be defined. Sound and data transmission must also be considered.

2.1 Compression ratios in MAC signals

The bandwidth and noise of the transmission systems will determine the compression ratios to be applied.

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

2.1.1 Bandwidth

A study carried out in France [CCIR, 1986-90a] comparing different compression ratios for luminance and chrominance (4-2-2, 4-1-1, 4-2-0, 3-1-0 MAC) showed that, although the 4-2-2 MAC format was preferable because of its direct compatibility with the 4-2-2 digital standard when the transmission channel bandwidth is greater than 10 MHz, the 4-2-0 MAC format is more convenient to use when the channel bandwidth is 5 to 6 MHz. Moreover, this study showed that the 4-2-0 MAC format allowed advantage to be taken of improving the quality when the channel bandwidth was extended up to 10 MHz.

2.1.2 Noise

A study conducted in France [CCIR, 1986-90a] showed that, as the noise varies as the cube of the compression ratios when frequency modulation is used, (e.g. in microwave links and satellite transmission) the 4-2-0 MAC format is less affected by noise than the 4-2-2 MAC format.

2.2 Line and field blanking

Some transmission equipments exploit the line and field blanking portions of the existing composite signal waveforms. It would thus be premature to introduce any fundamental change in respect of blanking. It would however seem feasible to shorten line blanking signals slightly, whilst retaining the front blanking porch and nominal synchronization pulse, by reducing only the back blanking porch. This porch, which is chiefly used by alignment systems, may be shortened by around 2 μ s without affecting the operation of most equipments.

It is also particularly important to have a precise time reference on reception which is used for timing the time-decompression of the MAC signal components. A precautionary measure would be to use a burst on the back porch to provide a more accurate time reference than the line synchronization edges.

2.3 Sound and data transmission

If one imposes the constraints of maintaining the same line and field blanking signals as those used for the usual composite signals and of limiting the transmitted MAC signal bandwidth so that it is no greater than that of the usual composite signals, transmission of sound and data does not pose any particular problem with existing equipments. The bandwidth of the components after transmission and decompression of the MAC signal is inversely proportional to the compression ratios used. If it is desired to increase the bandwidth of the components by increasing the bandwidth of the transmitted MAC signal, there may be some difficulties, in particular with video-sound frequency multiplex systems in which the sound is transmitted on a relatively low sub-carrier, ruling out any extension of the video band.

2.4 Results of transmission tests

MAC 3-1-0 and MAC 4-2-0 type signals have been laboratory tested in France by T.D.F. simulating cable, radio-relay and satellite links. Objective and subjective tests were carried out [CCIR, 1986-90a].

The objective tests confirmed the theoretical results as regards noise and bandwidth.

The subjective tests conducted reveal a significant overall improvement in picture quality for MAC coded pictures as against composite pictures. In particular, the absence of any chrominance-luminance intermodulation problem was noted, as well as an improvement in the passband for luminance and more particularly for chrominance with MAC 4-2-0. The results obtained for a transmission channel bandwidth of 6 MHz demonstrated that, except in unfavourable signal-to-noise conditions which occur for less than 0.1% of any month on the hypothetical reference circuit according to Recommendation 567 (weighted S/N of 45 dB), definite benefits are obtained by using MAC 4-2-0 signals, which overall prove more effective than composite signals under normal signal-to-noise conditions occurring during more than 99% of any given month (weighted S/N of 53 dB).

Further field tests have been conducted over paths of the EBU network comprising land and satellite links using MAC 4:2:0-type signals (known as "T-MAC") with a waveform as illustrated in Figure 1. They confirmed the results obtained earlier using simulation methods [CCIR, 1986-90b].

[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.

2.5 Conclusion

Theoretical studies, simulated and field tests have demonstrated the value of MAC 4:2:0-type signals for transmitting enhanced quality television signals over existing networks. The use of "T-MAC" for transmission over existing analogue networks requires that associated sound signals are transmitted via sound-in-sync or by a sound sub-carrier.

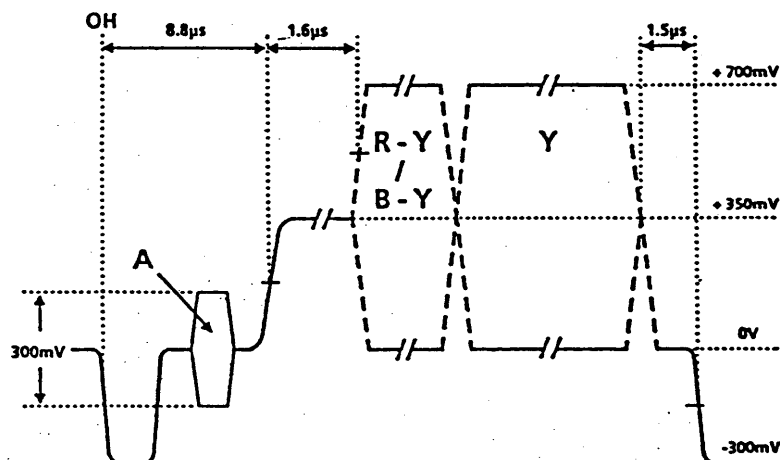


FIGURE 1

Representation of the "T-MAC" waveform

A: 4.5 MHz burst

3. Results of transmission tests

3.1 Results of transmission tests with DBS 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).

Tests described in [CCIR, 1986-90c] have been carried out in France with the TELECOM 1B transmission satellite with the D2-MAC/packet system. It was demonstrated that there were no difficulties in transmitting a television signal using the D2-MAC/packet standard over a long distance transmission circuit with equipment designed for composite signal transmission and comprising some radio-relay link sections and a telecommunication satellite circuit with minimum modifications to the modulator/demodulator.

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 DBS MAC signals. These new clamps should also be capable of operating with composite signals, preferably without manual intervention.

3.2 Results of transmission tests on digital links

It is necessary to specify a method allowing digital transmission of MAC-signals mentioned in section 2. Such studies are done within the European Eureka-95 project. The following two coding proposals were realized as prototypes and tested with MAC signals. The first one is described in Document [CCIR 1986-90d]. The technique employs a hybrid of 8-bit PCM and 5-bit DPCM with a folded quantizer, to reduce the bit rate of D/D2-MAC multiplex to the H4 transmission level of about 140 Mbit/s. The method has a simple implementation, is rugged in the presence of transmission errors and can convey HD-MAC signals while still in their scrambled form. Such a codec was demonstrated at IBC '88 in Brighton, United Kingdom, where digital D/D2-MAC signals were transmitted over 2 km of optical fibre.

An alternative coding approach in the Eureka-95 project is given in [CCIR 1986-90e]. This approach de-multiplexes the MAC/packet signal into sound and vision data and codes the vision and sound/data information separately. For the vision signal non-adaptive hybrid DPCM, combining Van Buul techniques with a folded quantizer, is used. To reduce further the sensitivity to transmission errors, the 7-bit HDPCM code words are replaced by the 8-bit PCM input code words every ninth sample. For the sound/data packets, binary (1-bit) transmission of the decoded duo-binary information is used. The multiplexing of the coded MAC/packet signal is chosen for transmission such that the complete signal will fit into the video channel of the TV container which is described in Document [CCIR 1986-90f].

First test results in comparing both proposals show that the first one could be used for secondary distribution and the second one for primary and secondary distribution. Details are shown in Table I.

3.3 Transmission of MAC signals over secondary distribution circuits

The D2 MAC/packet signal may be broadcast by existing radio-relay networks, or distributed over e.g. cable networks. It is therefore subject to a certain number of disturbances inherent in the broadcasting or distribution channel: noise, linear distortions, echoes.

These echo and distortion phenomena make the signal more sensitive to other disturbances such as noise and parasitic effects.

In D2 MAC/packet, the sound(s) associated with the image signal are transmitted by the data signal, as is all the digital information such as messages relating to conditional access.

[CCIR 1986-90g] gives information on a study carried out in France of an equalization system for the D2 component of the D2-MAC signal intended for secondary distribution. The preliminary results obtained with an experimental prototype show the efficiency of such a system in the presence of linear distortion, echoes and noise.

TABLE I

Examples of digital D/D2-MAC transmission systems

Transmission system	139.264 Mbit/s systems (CMI interface) 155.52 Mbit/s systems (SDH)	
Application	Trunk-line/Distribution-line	
Transmission signal format	D/D2-MAC	D/D2-MAC
Net bit rate	131.625 Mbit/s	124.750 Mbit/s
Sampling frequency	20.25 MHz	20.25 MHz
Coding method	Hybrid 8-bit PCM and 5-bit DPCM on alternative samples of multiplex	7-bit HDPCM for the video signal Binary (1 bit) transmission of decoded duobinary data
Prediction	One dimensional interpolative, based on previous and following PCM samples	One dimensional, based on the previous sample in the same horizontal line

4. 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 2 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.

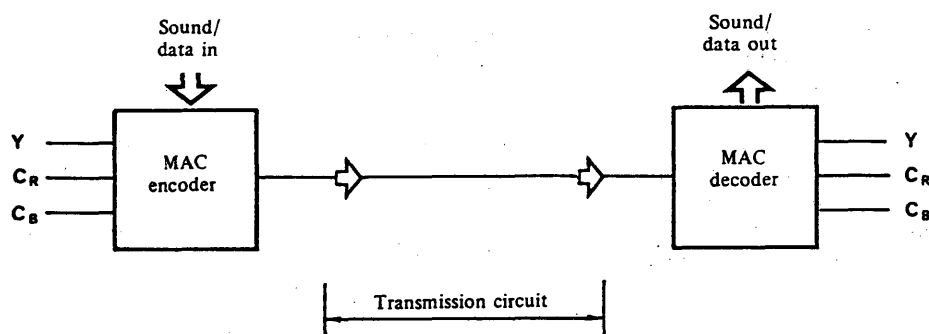


FIGURE 2 - Transmission of MAC signals

4.1 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. 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.

More recently, [Alard, 1987] has proposed new waveforms which are intended for automatic measurements of MAC-packet signals. These waveforms take into account the sampling frequency of the MAC-packet systems. Their spectrum is restricted to 8.5 MHz (-6 dB).

Annex I details test signals proposed by France [CCIR, 1986-90h] and the Federal Republic of Germany. It could form the basis for a new Recommendation, which should be envisaged for the end of the 1990-1994 study period.

Annex II describes some specific measurement techniques proposed by France [CCIR, 1986-90j] and the Federal Republic of Germany [CCIR, 1986-90k].

Weighted measurements of continuous random noise using the unified weighting network described in Recommendation 567 are inappropriate for compressed MAC signals.

The United Kingdom [CCIR, 1986-90l] has proposed for MAC systems with a luminance compression ratio of 1.5, that wideband random noise should be measured in a bandwidth of 8.4 MHz and using a constant impedance noise weighting network with a time constant of 90 ns. Such a weighting network is based partly on the assumption that with the trend towards larger picture displays and with improved picture quality available from the MAC family of television standards, future subjective tests will more commonly employ a viewing distance of four times (rather than six times as at present) the picture height.

The Federal Republic of Germany [CCIR, 1986-90m] has proposed to use the unified weighting network, scaled according to the 3:2 compression ratio as used in most MAC systems, as the common weighting network for all MAC systems. Harm [1989] points out that this filter gives the same results as would be obtained in decompressed form with the unified weighting filter, described in Recommendation 567, takes care of noise carried by the more compressed colour difference signals, and takes care of a possibly higher noise sensitivity at high frequencies with HD-MAC signals using the same networks as designed for current MAC signals. Annex III describes this filter in detail.

4.2 Sound/data

[CCIR, 1986-90h] gives some information about investigations carried out in France by the CCETT. It shows that bit error ratio measurement can be replaced for D and D2-MAC/packet signal by a measurement of the violation rate of the duobinary code. This allows for accurate on-line measurement.

Furthermore, it is suggested that the data signal quality is evaluated by adding a Gaussian noise to the signal received and plotting the bit error ratio versus the noise level. For a given bit error ratio, the difference in dB between the measured noise level and the theoretical level produces by definition the "equivalent impairment".

REFERENCES

ALARD, M. - Automatic measurement methods for the MAC/packet family. Third International Conference on Television Measurements, Montreux 18-20 June 1987, IERE Pub. No. 74, pp. 168-179.

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.

Harm, H. [1989]: Proposal for a Noise Weighting Filter for Component Signals.

Rundfunktech. Mitt. 33 (1989) H. 1, S. 13 - 17 and

EBU Review-Technical, Revue de l'UER-Technique No. 233, February 1989, pp 17 - 22

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).

[1986-90]: a. CMTT/2 (France); b. CMTT/218 (France); c. CMTT/17 (France); d. IWP CMTT/2-70 (United Kingdom); e. IWP CMTT/2-89(Rev.) (Fed. Rep. of Germany); f. IWP CMTT/2-90 (Fed. Rep. of Germany); g. CMTT/211 (France); h. CMTT/3 (France); j. CMTT/190 (France); k. CMTT/208 (Fed. Rep. of Germany); l. CMTT/186 (United Kingdom); m. CMTT/209 (Fed. Rep. of Germany)

BIBLIOGRAPHY

FARJAUDON, T., SEUTIN, R., DESMARS, M., CAZIVASSILIO, D., GELLY, A.
[April-June 1988] - "T-MAC, un système de transmission à qualité améliorée" -
Revue Radiodiffusion-Télévision No. 101, 9-13.

ANNEX I

Test signals for MAC/packet systems1. General remarks

1.1 Three insertion test signals are proposed, primarily for automatic measurement. In addition three optional waveforms may be used either as test line or as full field signals for monitoring purposes.

1.2 As far as possible essential signal elements have been allocated to the luminance period of the line (samples 590 to 1286). This will enable these signals to be used also for testing MAC coders and decoders.

1.3 Only in a restricted sample range between 245 and 1277 do the proposed signals vary from the 0 mV level. This allows blanking and reinsertion of the test signals at appropriate points of the transmission chain (e.g. between terrestrial and satellite section).

1.4 The spectral content of all waveforms is restricted to 8.5 MHz (-6 dB).

1.5 High frequency signal amplitudes are restricted to ± 250 mV to avoid non-linear distortion and to allow conversion into an AM-VSB-MAC-system with Nyquist filtering at the transmitting end.

2. Definition of elementary waveforms2.1 Basic definitions

T: MAC-sampling period ≈ 49.38 ns

k: MAC sample number (see Report 1073 and CCIR special publication "Specification of transmission systems for the broadcasting satellite service")

2.2 Transition

A transition is a signal defined for a duration $4T$ according to the shaping:

0.000 - 0.114 - 0.500 - 0.886 - 1.000

(Hamming window integral on $4T$)*

2.3 Pulse

A pulse is a signal defined for a duration $6T$ according to the shaping:

0.000 - 0.130 - 0.630 - 1.000 - 0.630 - 0.130 - 0.000

(Blackman window on $6T$)**

* Hamming: $y(t) = 0.54 + 0.46 \cos \pi t/2T$

** Blackman: $y(t) = 0.42 + 0.50 \cos \pi t/3T + 0.08 \cos 2 \pi t/3T$

2.4 Ramp

A ramp is a signal defined for a duration nT in the equation:

$$k = 0 \text{ to } n: y_k = k/n \quad \text{for a rising ramp}$$

$$y_k = 1 - k/n \text{ for a falling ramp}$$

2.5 Complex wobbleulation

A complex wobbleulation is made up of two signals defined for a duration 512T according to equations:

Real part:

$$k = 0 \text{ to } 512: y_k = \left(\cos \frac{\pi(k-256)^2}{512} \right) W(k)$$

Imaginary part:

$$k = 0 \text{ to } 512: y_k = \left(\sin \frac{\pi(k-256)^2}{512} \right) W(k)$$

where $W(k)$ is a window provisionally defined as:

$$k = 0 \text{ to } 28: W(k) = 0$$

$$k = 28 \text{ to } 53: W(k) = \sin^2 \frac{\pi(k-28)}{50}$$

$$k = 53 \text{ to } 459: W(k) = 1$$

$$k = 459 \text{ to } 484: W(k) = \sin^2 \frac{\pi(484-k)}{50}$$

$$k = 484 \text{ to } 512: W(k) = 0$$

The complex wobbleulation signals are transmitted in a 4-frame sequence in positive and negative (inverted) polarity as follows:

even frame: real part not inverted
 odd frame: imaginary part not inverted
 even frame: real part inverted
 odd frame: imaginary part inverted
 even frame: real part not inverted
 etc.

2.6 Modulated pulse

A pulse modulated at a frequency f (MHz) is a signal defined for a duration 81T according to the equation:

$$k = 0 \text{ to } 81: y_k = \cos^2 \frac{4\pi f k}{81} \sin^2 \frac{\pi k}{81}$$

2.7 Burst

A burst, modulated at a frequency f (MHz) is a signal defined for a duration $81T$ in the equation:

$$k = 0 \text{ to } k_1: \quad y_k = \sin \frac{8\pi f k}{81} \cdot \left[\frac{1}{2\pi} \left(\frac{2\pi k}{k_1} - \sin \frac{2\pi k}{k_1} \right) \right]^*$$

$$k = k_1 \text{ to } 81 - k_1: \quad y_k = \sin \frac{8\pi f k}{81}$$

$$k = 81 - k_1 \text{ to } 81: \quad y_k = \sin \frac{8\pi f k}{81} \cdot \left[\frac{1}{2\pi} \left(\frac{2\pi(81 - k)}{k_1} - \sin \frac{2\pi(81 - k)}{k_1} \right) \right]**$$

$$f = 1 \text{ to } 6: \quad k_1 = 15$$

$$f = 7: \quad k_1 = 25$$

$$f = 8: \quad y_k = \sin \frac{8\pi f k}{81} \cdot \sin^2 \frac{\pi \cdot k}{81}; \quad k = 0 \text{ to } 81$$

3. Test signal description

3.1 Test signal No. 1, see Figures 3a and 3b, Tables IIA and IIB

Test signal No. 1 is proposed as a mandatory signal and allocated to line 312. It is intended for automatic measurement and composed of a bipolar bar signal with inverse polarity in even and odd frames. Positive and negative Blackman pulses are contained in the even frame signal only.

The first part of the signal ($k = 225$ to 612) is provisionally set to 0 mV. It may be used in the future for insertion of other test signals in the time domain (sin x/x signals for example).

3.2 Test signal No. 2, see Figures 4a, 4b and Table III

Test signal No. 2 is proposed as a mandatory signal and allocated to line 623. It is intended for automatic measurement of noise and non-linear distortion. It comprises a rising ramp (even frames) and a falling ramp (odd frames). This allows separation of linear distortions (e.g. tilt) from non-linear ones.

3.3 Test signal No. 3, see Figures 5a, 5b and Table IV

Test signal No. 3 is proposed as a mandatory signal and allocated to line 624. The first part of this line is already defined in the MAC/packet standards. The second part of this line contains a complex wobble.

* Alternatively the expression in square brackets could be replaced by $\sin^2 \frac{\pi k}{k_1}$. Optimization requires further study.

** Alternatively the expression in square brackets could be replaced by $\sin^2 \frac{\pi(81 - k)}{k_1}$. Optimization requires further study.

3.4 Test signal No. 4 (optional), see Figure 6 and Table V

This optional signal is intended for evaluation of linear distortions on a waveform monitor. This signal consists of a bipolar pulse and bar signal and eight modulated pulses (1 to 8 MHz) of 500 mV amplitude. It may be used also with full amplitude (1000 mV), if non-linear distortions are unlikely to occur.

If used as a test line signal it should be inserted on line 311.

3.5 Test signal No. 5 (optional), see Figure 7 and Table VI

This optional signal comprises an 8-riser staircase waveform and is intended for evaluation of non-linear distortions using a waveform monitor.

If used as a test line signal it should be inserted on line 1.

3.6 Test signal No. 6 (optional), see Figure 8 and Table VII

This optional signal is intended for display of the amplitude frequency response on a waveform monitor and is composed of eight multiburst signals (1 to 8 MHz) of 500 mV amplitude preceded by a reference bar. It may be used also with full amplitude (1000 mV), if non-linear distortions are unlikely to occur.

If used as a test line signal it should be inserted on line 313.

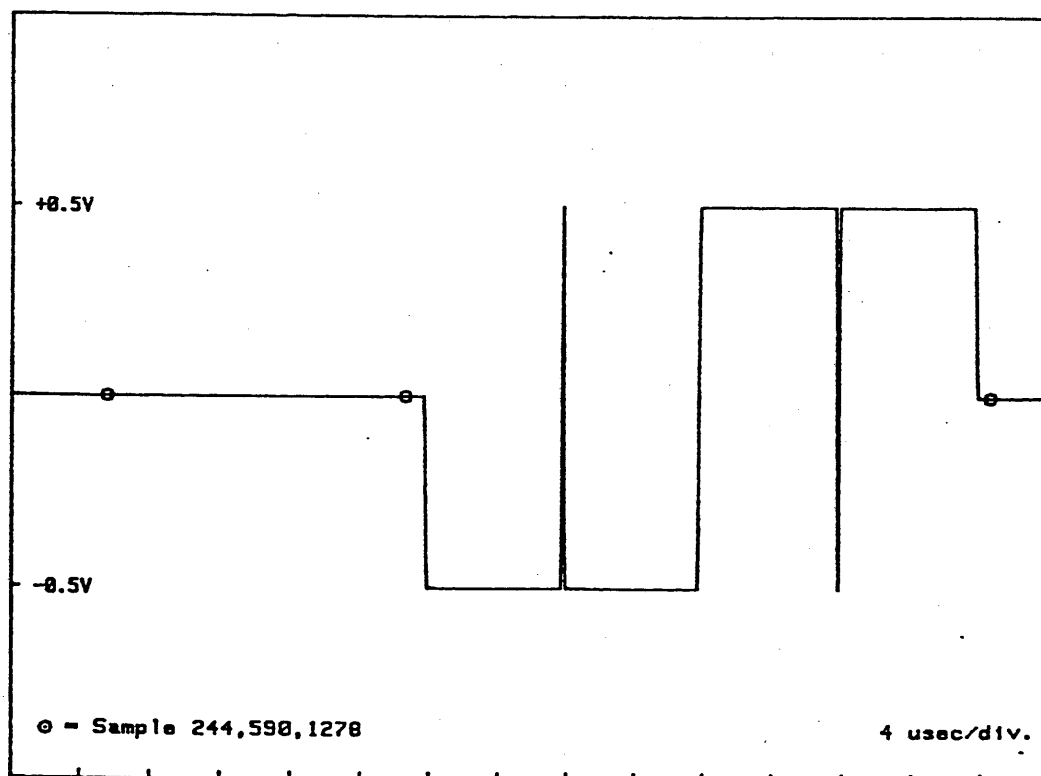


FIGURE 3a

Test signal 1a, even frame

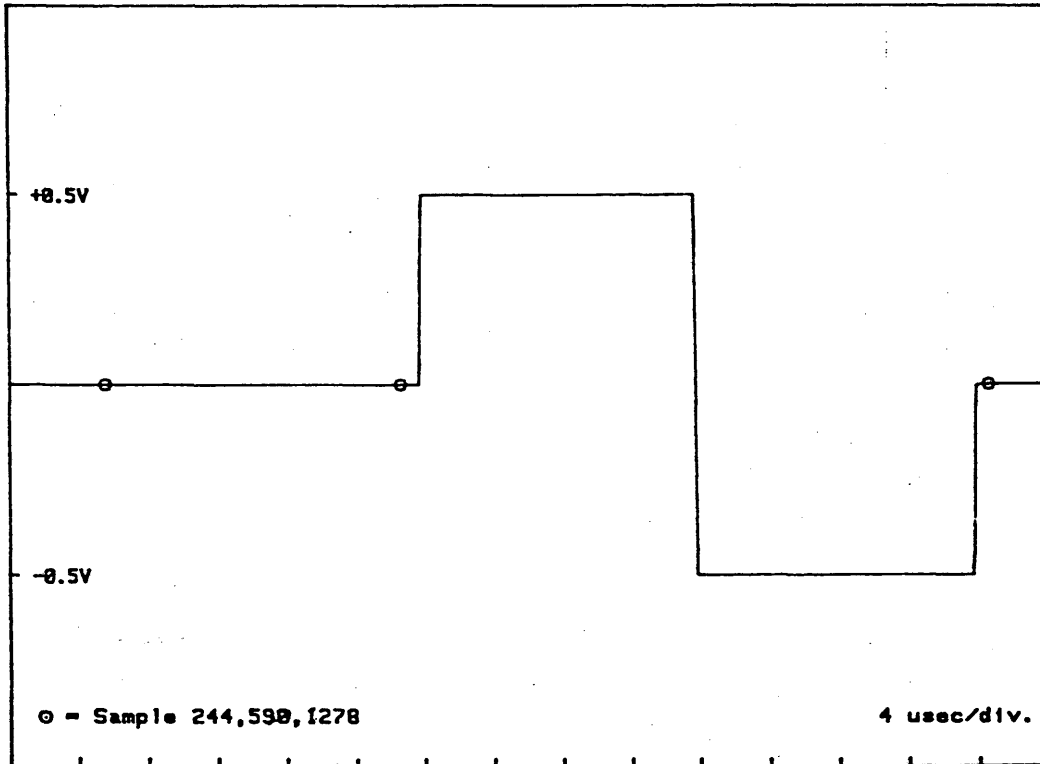


FIGURE 3b

Test signal 1b, odd frame

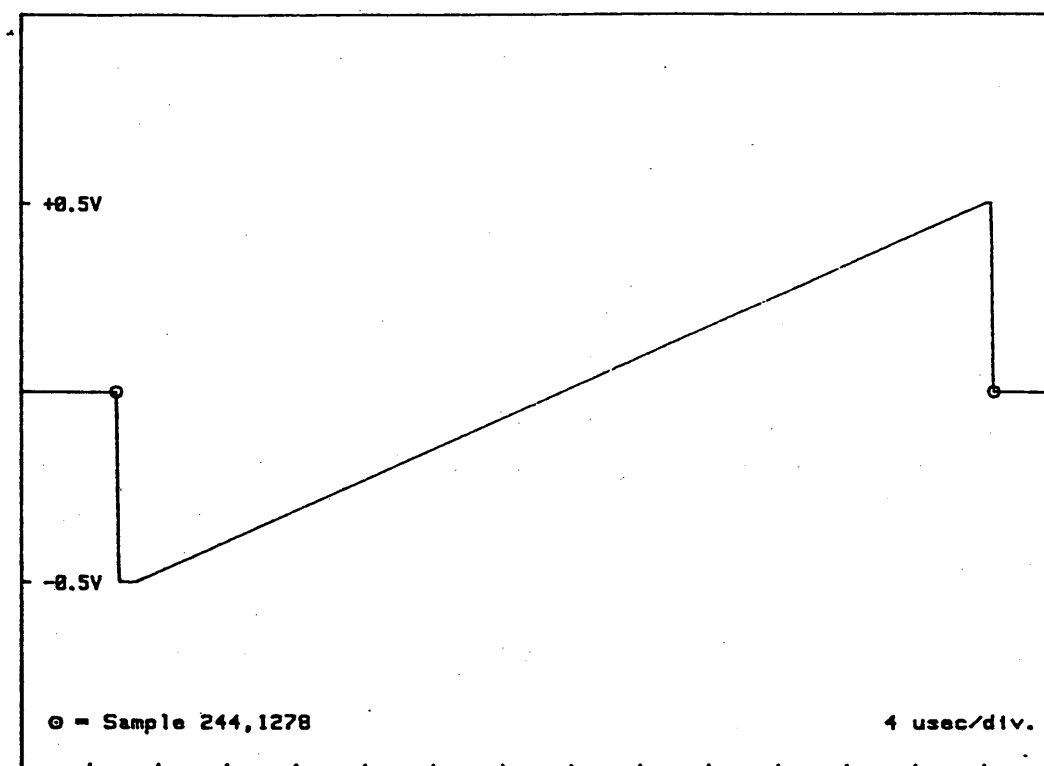


FIGURE 4a

Test signal 2a, even frame

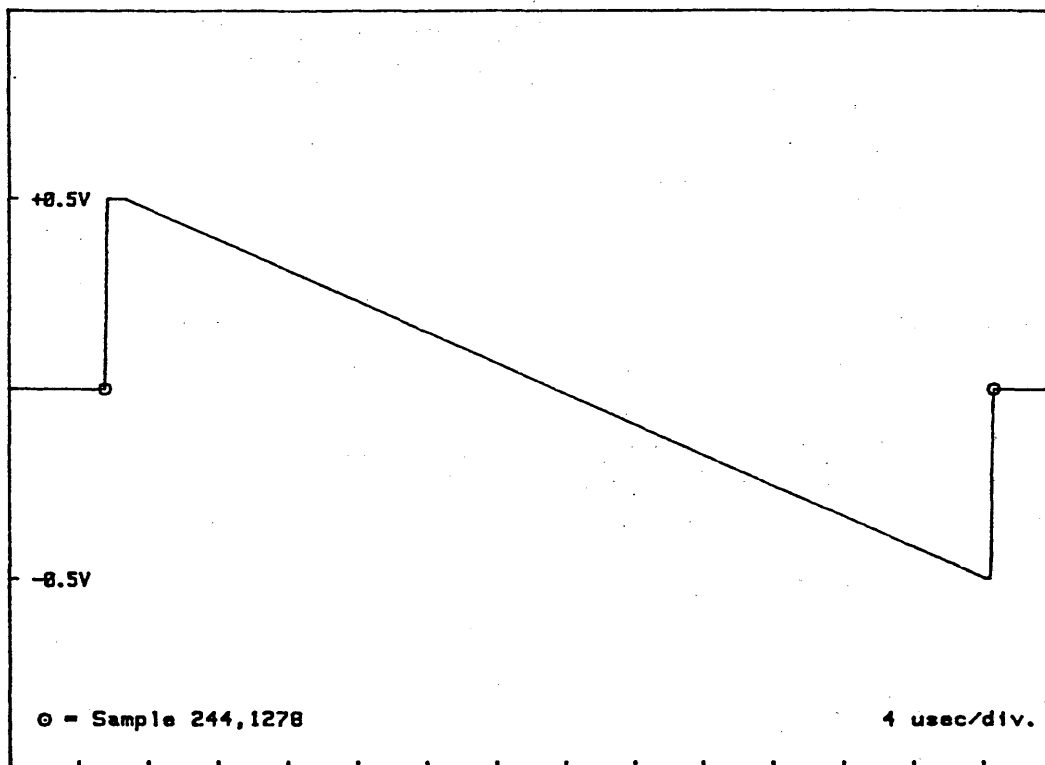


FIGURE 4b

Test signal 2b, odd frame

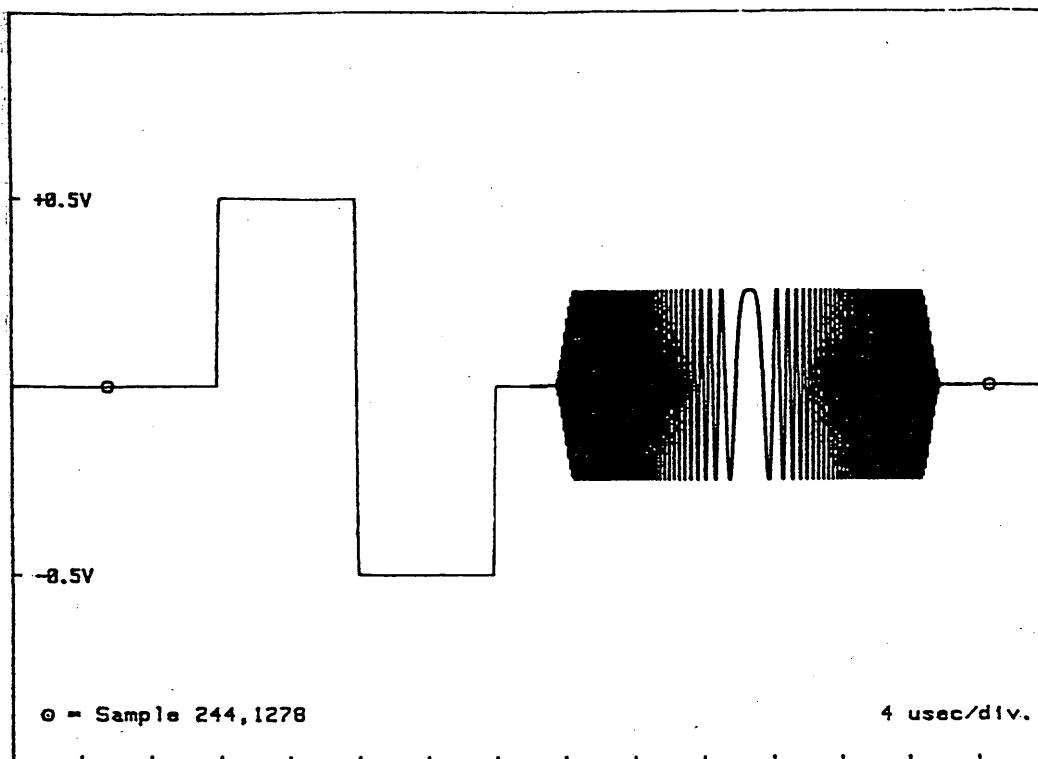


FIGURE 5a

Test signal 3a, even frame, positive polarity
of complex wobblelation signal

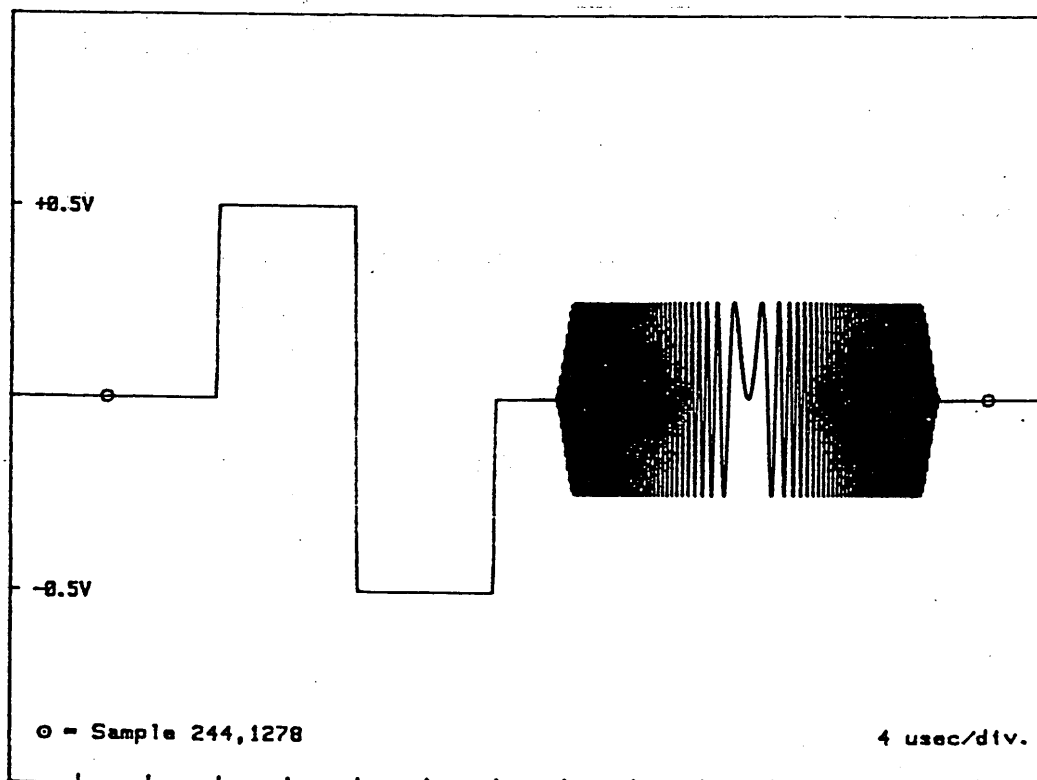


FIGURE 5b

Test signal 3a, odd frame, positive polarity
of complex wobblelation signal

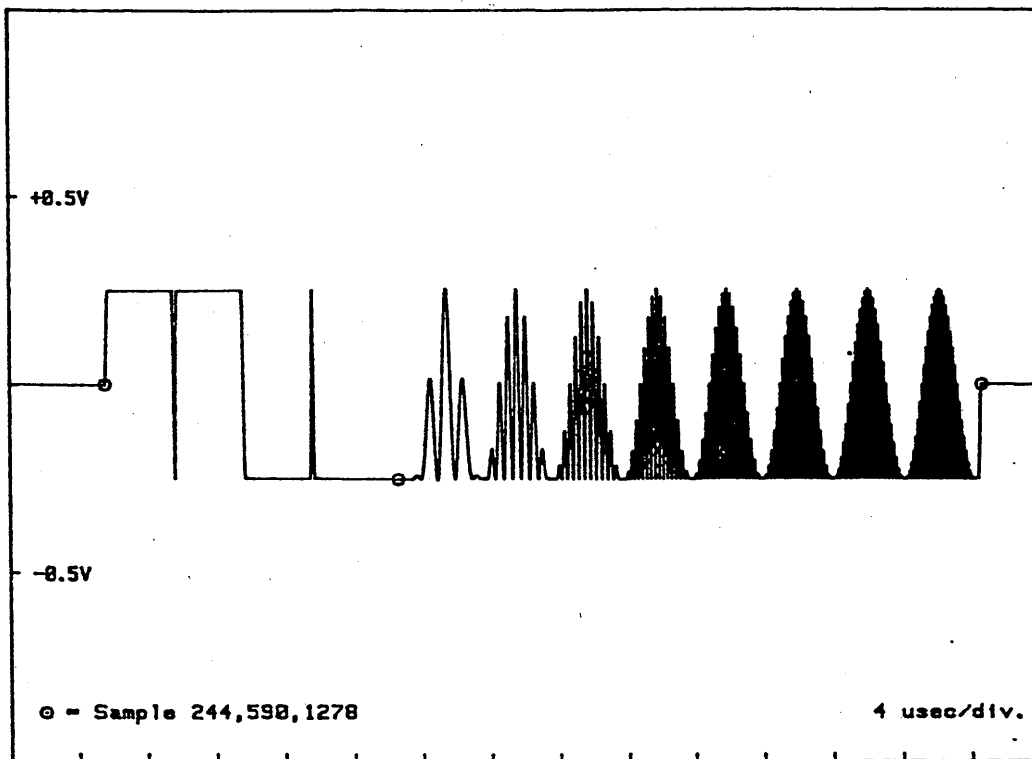


FIGURE 6

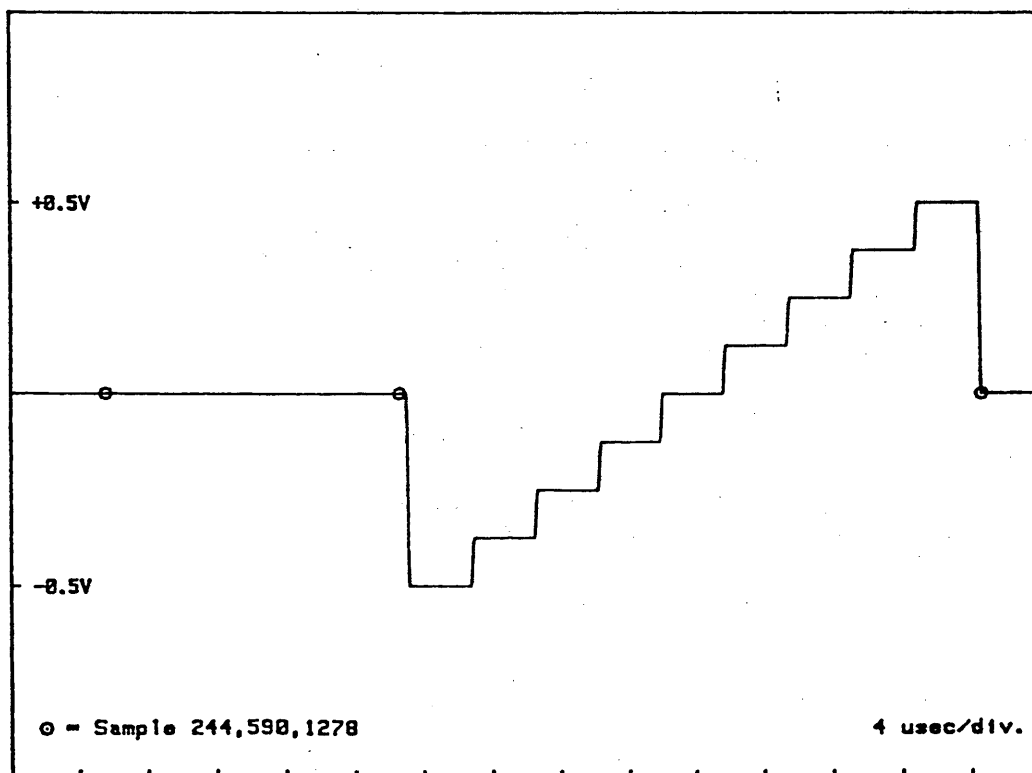
Test signal 4

FIGURE 7

Test signal 5

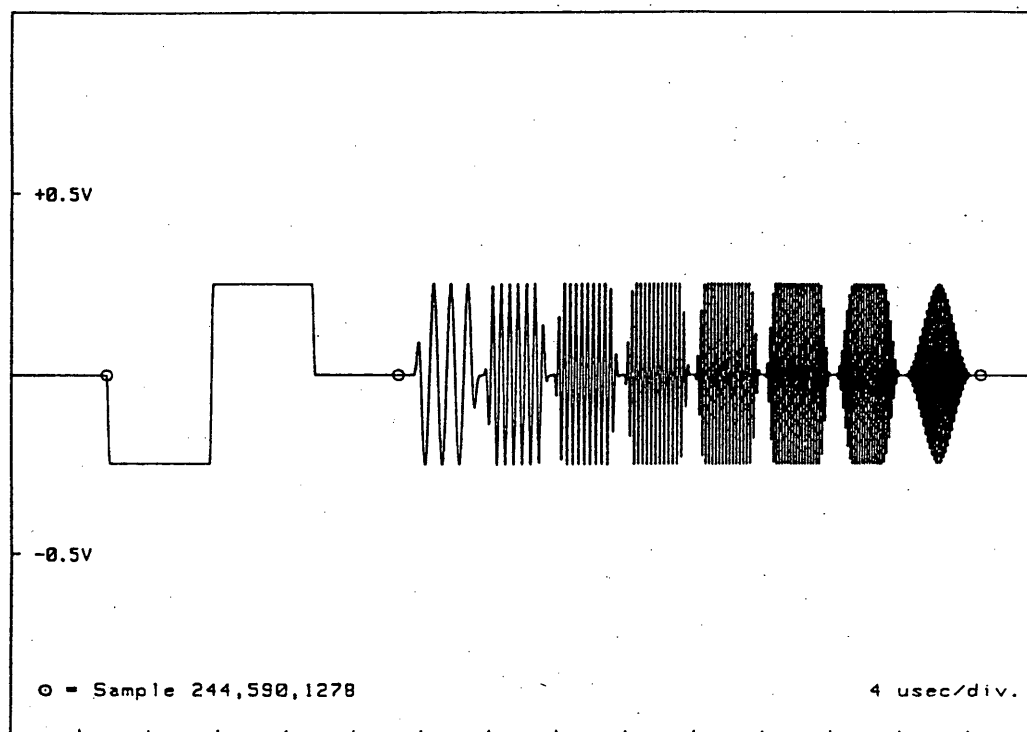


FIGURE 8

Test signal 6

TABLE IIa

Definition of signal 1, even frames

k = 255 to 612: level 0 mV
 k = 612 to 616: transition from 0 mV to -500 mV
 k = 616 to 773: level -500 mV
 k = 773 to 779: pulse (base -500 mV; peak +500 mV)
 k = 779 to 936: level -500 mV
 k = 936 to 940: transition from -500 mV to +500 mV
 k = 940 to 1097: level +500 mV
 k = 1097 to 1103: pulse (base +500 mV, peak -500 mV)
 k = 1103 to 1260: level +500 mV
 k = 1260 to 1264: transition from +500 mV to 0 mV
 k = 1264 to 1292: level 0 mV

TABLE IIb

Definition of signal 1, odd frames

k = 255 to 612: level 0 mV
 k = 612 to 616: transition from 0 to +500 mV
 k = 616 to 936: level +500 mV
 k = 936 to 940: transition from +500 mV to -500 mV
 k = 940 to 1260: level -500mV
 k = 1260 to 1264: transition from -500 mV to 0 mV
 k = 1264 to 1292: level 0 mV

TABLE III

Definition of signals 2a and 2b

Signal 2a: even frames

k = 225 to 244: level 0 mV
 k = 244 to 248: transition from 0 mV to -500 mV
 k = 248 to 268: level -500 mV
 k = 268 to 1268: -500 mV to +500 mV ramp
 k = 1268 to 1274: level +500 mV
 k = 1274 to 1278: transition from +500 mV to 0 mV
 k = 1278 to 1292: level 0 mV

Signal 2b (odd frames) has timing as above, levels inverted.

TABLE IV

Definition of signals 3a and 3b

k = 255 to 370: level 0 mV
 k = 370 to 374: transition from 0 mV to +500 mV
 k = 374 to 532: level +500 mV
 k = 532 to 536: transition from +500 mV to -500 mV
 k = 536 to 694: level -500 mV
 k = 694 to 698: transition from -500 mV to 0 mV
 k = 698 to 739: level 0 mV
 k = 739 to 1251: complex wobulation of amplitude +250 mV
 k = 739 to 1251: transmitted in a 4-frame sequence: real part positive, imaginary part positive, real part negative, imaginary part negative
 k = 1251 to 1292: level 0 mV

TABLE V

Definition of signal 4

k - 225 to 244: level 0 mV
 k - 244 to 248: transition from 0 mV to +250 mV
 k - 248 to 324: level +250 mV
 k - 324 to 330: pulse (base +250 mV; peak -250 mV)
 k - 330 to 406: level +250 mV
 k - 406 to 410: transition from +250 mV to -250 mV
 k - 410 to 486: level -250 mV
 k - 486 to 492: pulse (base -250 mV; peak +250 mV)
 k - 492 to 607: level -250 mV
 k - 607 to 688: pulse modulated at 1 MHz
 k - 688 to 690: level -250 mV
 k - 690 to 771: pulse modulated at 2 MHz
 k - 771 to 773: level -250 mV
 k - 773 to 854: pulse modulated at 3 MHz
 k - 854 to 856: level -250 mV
 k - 856 to 937: pulse modulated at 4 MHz
 k - 937 to 939: level -250 mV
 k - 939 to 1020: pulse modulated at 5 MHz
 k - 1020 to 1022: level -250 mV
 k - 1022 to 1103: pulse modulated at 6 MHz
 k - 1103 to 1105: level -250 mV
 k - 1105 to 1186: pulse modulated at 7 MHz
 k - 1186 to 1188: level -250 mV
 k - 1188 to 1269: pulse modulated at 8 MHz
 k - 1269 to 1274: level -250 mV
 k - 1274 to 1278: transition from -250 mV to 0 mV
 k - 1278 to 1292: level 0 mV

TABLE VI

Definition of signal 5

k - 225 to 598: level 0 mV
 k - 598 to 602: transition from 0 mV to -500 mV
 k - 602 to 674: level -500 mV
 k - 674 to 678: transition from -500 mV to -375 mV
 k - 678 to 749: level -375 mV
 k - 749 to 753: transition from -375 mV to -250 mV
 k - 753 to 824: level -250 mV
 k - 824 to 828: transition from -250 mV to -125 mV
 k - 828 to 899: level -125 mV
 k - 899 to 903: transition from -125 mV to 0 mV
 k - 903 to 974: level 0 mV
 k - 974 to 978: transition from 0 mV to +125 mV
 k - 978 to 1049: level +125 mV
 k - 1049 to 1053: transition from +125 mV to +250 mV
 k - 1053 to 1124: level +250 mV
 k - 1124 to 1128: transition from +250 mV to +375 mV
 k - 1128 to 1199: level +375 mV
 k - 1199 to 1203: transition from +375 mV to +500 mV
 k - 1203 to 1274: level +500 mV
 k - 1274 to 1278: transition from +500 mV to 0 mV
 k - 1278 to 1292: level 0 mV

TABLE VII

Definition of signal 6

k - 225 to 244: level 0 mV
k - 244 to 248: transitions from 0 mV to -250 mV
k - 248 to 365: level -250 mV
k - 365 to 369: transitions from -250 mV to +250 mV
k - 369 to 486: level +250 mV
k - 486 to 490: transitions from +250 mV to 0 mV
k - 490 to 607: level 0 mV
k - 607 to 688: burst 1 MHz
k - 688 to 690: level 0 mV
k - 690 to 771: burst 2 MHz
k - 771 to 773: level 0 mV
k - 773 to 854: burst 3 MHz
k - 854 to 856: level 0 mV
k - 856 to 937: burst 4 MHz
k - 937 to 939: level 0 mV
k - 939 to 1020: burst 5 MHz
k - 1020 to 1022: level 0 mV
k - 1022 to 1103: burst 6 MHz
k - 1103 to 1105: level 0 mV
k - 1105 to 1186: burst 7 MHz
k - 1186 to 1188: level 0 mV
k - 1188 to 1269: burst 8 MHz
k - 1269 to 1292: level 0 mV

BIBLIOGRAPHY

CCIR Documents

[1986-90]: a. CMTT/190 (France); b. CMTT/208 (Federal Republic of Germany).

ANNEX II

MEASUREMENT TECHNIQUES FOR TRANSMISSION CIRCUITS
CARRYING MAC/PACKET SIGNALS1. General remarks

Most of the measurement techniques described below use the test signals in Annex I of this report. These methods are based largely on digital signal processing techniques.

2. Measurements related to the vision signal2.1 Low-frequency noise

Low-frequency noise (below line frequency) is measured on a 50% white picture signal. High-frequency noise is reduced by averaging the signal value within each line. The spectral noise density can be estimated by Fourier transform of the 625 signal values derived from one frame. The obtained analysis pitch is 25 Hz, allowing separation between the different sources of noise in the frequency band 25 Hz to 7.8 kHz.

2.2 High-frequency noise

High-frequency noise measurement can be carried out using test signals No. 2a and 2b (ramp signals).

One method consists of estimating the noise level by averaging a large number of observations of the same complete test line and subtracting this average waveform from each observation. The result is the true noise content. From this the true spectral noise density can be calculated.

The other method for weighted and unweighted random noise measurement uses the 200 kHz highpass filter as specified in Recommendation 569 to remove the ramp signal. The filter also reduces the influence of static non-linearities to the noise measurement [Henschke, 1988, 1989]. This technique was introduced in the Federal Republic of Germany in 1986 because measurements on a blank line in mixed analogue and digital transmission systems were unsatisfactory.

2.3 Dynamic non-linearity

When comparing the shape of the two polarities of the Blackman pulse and the Hamming slopes in test signal No. 1 information on high-frequency non-linearity can be obtained.

2.4 Static non-linearity

Static non-linearity is measured using test signals No. 2a and 2b. The two polarities are used to separate linear from non-linear effects.

One method is to sample the ramp signals and to remove the noise by averaging. After removal of linear distortion the processed waveform is approximated by a polynomial of degree k . Analysis of the coefficients of the

polynomial provides information on the overall non-linear characteristic. It should be noted that any difference between the above mentioned processed waveform and the polynomial indicates the presence of quantization distortion.

The second method also uses the ramp signals but is analogous to the luminance non-linearity measurement on the 5-riser staircase described in Recommendation 569 and provides comparable results. The ramp level is measured at timing instants which are equally separated from one-another by $9\ \mu\text{s}$, and centered with respect to the active ramp period of $50.5\ \mu\text{s}$. The amplitude value at each of the 6 timing instants is calculated as the arithmetic mean of the ramp signal from $0.5\ \mu\text{s}$ before to $0.5\ \mu\text{s}$ after the timing instant. This reduces the influence of quantization errors and superimposed high-frequency noise. The six samples are processed to a single figure as described in Recommendation 569, § 2.9.

2.5 Amplitude and phase/group delay frequency response

The complex wobblelation signal (test signal No. 3) is processed through a Fourier transform into amplitude and phase frequency response. The latter can be processed further into group delay. Prior to applying the sampled complex wobblelation signal to the FFT, non-linearities can be minimized using the two opposite polarities and noise can be reduced by averaging.

3. Specific data signal measurements

3.1 Bit error ratio measurement

The bit error ratio can be measured during programme transmission on synchronization words, the Golay code on packet headers, and on dummy packets. In the case of duobinary coding, the violation rate of this code gives an excellent estimate of the bit error ratio.

3.2 Decoding margin measurement

Bit error ratio measurements give no indication on the actual safety margin for possible additional distortions.

One method of deriving such information is to measure the error rate using variable deviations from the threshold levels until a predefined error rate is reached. A second method adds Gaussian noise to the received signal until the predefined error rate is reached. Alard [1987] describes solutions which use digital processing.

3.3 Analysis of eye diagrams

The analysis of the eye diagrams of a two-level or a duobinary data signal - either made by oscilloscope or microprocessor means - provides eye-height and eye-width indications and additional information about the location of the optimum sampling phase and threshold levels with respect to their nominal values.

REFERENCES

Alard, M. [1987] Automatic measurement methods for the MAC/packet family. Third International Conference on Television Measurements, Montreux 18-20 June 1987, IERE Pub. No. 74, 168-179.

Henschke, W. [October 1988] *Proposal for Luminance Signal-to-Noise Measurement when processing through Codecs*. IEEE - Doc. G-2.1.5./205 (25.Oct.1988).

Henschke, W. [February 1989] *SNR Measurement through Codecs. Comparison of the Difference Method and the 200 kHz High-pass Method with regard to Non-linearities*. IEEE - Doc. G-2.1.5/217 (17.Feb.1989).

BIBLIOGRAPHY

CCIR Documents

[1986-90]: a. CMTT/190 (France); b. CMTT/208 (Federal Republic of Germany).

ANNEX III

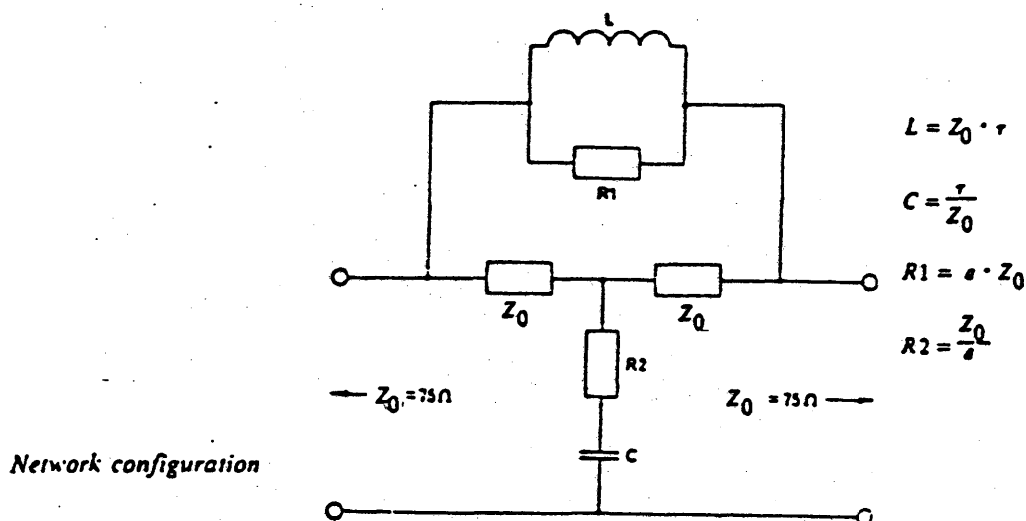


FIGURE 9a

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: $A_{\infty} = 20 \log (1 + a)$

where:

$$\tau = \frac{245}{1.5} \text{ ns}; \quad a = 4.5 \quad (A_{\infty} = 14.8 \text{ dB})$$

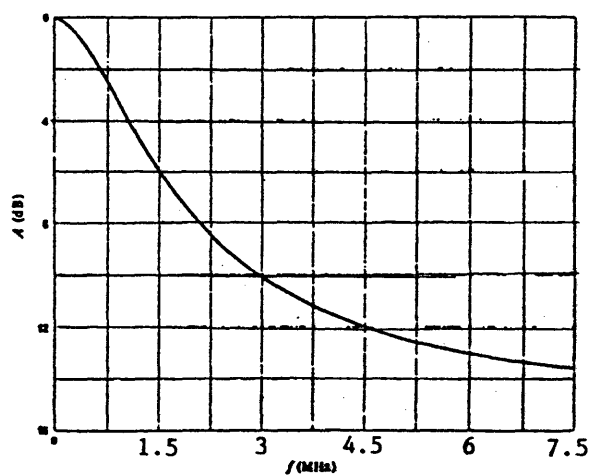


FIGURE 9b

Unified MAC weighting filter for random noise in MAC circuits with compression ratio of 1.5:1

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DECISIONS

DECISION 18-6*

**DIGITAL SYSTEMS FOR THE TRANSMISSION OF
SOUND-PROGRAMME AND TELEVISION SIGNALS**

(1974-1976-1978-1981-1983-1989)

The CMTT,

CONSIDERING

- (a) that CCITT Study Group XVIII is studying the problems associated with the establishment of an integrated services digital network (ISDN) for the transmission of all kinds of signals, including those for sound-programme and television;
- (b) that CCITT Study Group XV is studying digital multiplex and line systems, including equipment for the digital transmission of sound-programme and television signals, and coding for interactive audio-visual services;
- (c) that CCITT Study Group I is studying the end-to-end service quality of all interactive audio-visual services supported on the ISDN;
- (d) that there is an urgent need to ensure that the standards adopted for the integrated services digital network shall be compatible with the requirements necessary for the transmission of sound-programme and television signals;
- (e) that CCIR Study Groups 10 and 11 are studying the use of digital techniques in programme production and emission;
- (f) 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;
- (g) that the CMTT is studying the performance requirements applicable to _____ digital transmission circuits for sound-programmes and television, either for contribution or distribution circuits;
- (h) that the CMTT has to consider the transition period during which technological changes will take place;
- (j) that there is a need to co-ordinate the studies undertaken by Study Groups 10 and 11 and the CMTT, together with those in progress in CCITT Study Groups XV and XVIII,

DECIDES

1. that a Joint Interim Working Party CMTT-10-11/2 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 maximum compatibility between the Recommendations of the CCIR and the CCITT related to the digital transmission of sound-programme and television signals;

* This Decision should be brought to the attention of CCIR Study Groups 10 and 11 and CCITT Study Groups I, IV, XV and XVIII.

- 1.3 to cooperate with CCITT Study Group I to ensure appropriate compatibility between interactive audio-visual services and the broadcasting services;
2. that the Chairman of JIWP CMTT-10-11/2 shall be a representative of the Administration of the United Kingdom;
3. that JIWP CMTT-10-11/2 shall be composed of the Chairmen and Vice-Chairmen of CCIR Study Groups 10, 11 and the CMTT 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 JIWP CMTT-10-11/2 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 Joint Interim Working Party before transmission to CCITT Study Groups I, XV and 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 Joint Interim Working Party must take account of meetings of the Study Groups concerned.

— JIWP CMTT-10-11/2 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

JIWP CMTT-10-11/2 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 Joint Interim Working Party CMTT-10-11/2

1. Specification of all ISDN network aspects (CCITT Study Group XVIII); specification of transmission equipment and systems (CCITT Study Group XV); and specifications relating to telecommunication services (CCITT Study Group I).
- 2.* Coding methods, including redundancy reduction and standards to be adopted for foreseen application of:
 - 2.1 Sound-broadcasting and television (see Note 1)
 - 2.1.1 studio installations including recording { sound (SG 10)
television (SG 11);
 - 2.1.2 terrestrial radio-relay links
 - 2.1.3 cable transmission links { (CMTT);
 - 2.1.4 satellite transmission links
 - 2.1.5 broadcasting from terrestrial transmitters and from satellites { sound (SG 10)
television (SG 11).
 - 2.1.6 B-ISDN (distribution) (CMTT)
 - 2.2 Interactive audio-visual services (Study Group XV)
- 3.* Compatibility of the relevant coding methods (see § 2) for:
 - 3.1 studio installations, including recording;**
 - 3.2 terrestrial radio-relay links;
 - 3.3 cable transmission links;
 - 3.4 satellite transmission links;
 - 3.5 broadcasting from terrestrial transmitters and from satellites between themselves and with the other services envisaged under § 1
 - 3.6 interactive audio-visual services
- 4.* Study of ways of protecting against digital transmission impairments (e.g. errors, jitter, slip) for:

	<i>Studio</i>	<i>Transmission link</i>
4.1 sound programmes	(SG 10)	(CMTT).
4.2 television	(SG 11)	
- 4.3 Interactive audio-visual services (Study Group XV)

* Close cooperation between Joint Interim Working Party CMTT-10-11/2 and Study Groups XV and XVIII is essential for these studies.

** Aspects of compatibility relating to the exchange of recordings are matters for Study Groups 10 and 11.

5. Study of transmission methods over various transmission media _____ for:
 - 5.1 studio installation, including recording | sound (SG 10)
television (SG 11);
 - 5.2 terrestrial radio-relay links (SG 9);
 - 5.3 optical and metallic 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).
- 5.6 multiplex equipment and digital line systems on optical fibre cables in local networks, including ISDN access (Study Group XV).
6. Equipment for digital coding and multiplexing 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 | (CMTT).
 - 7.2 television
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 services 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).

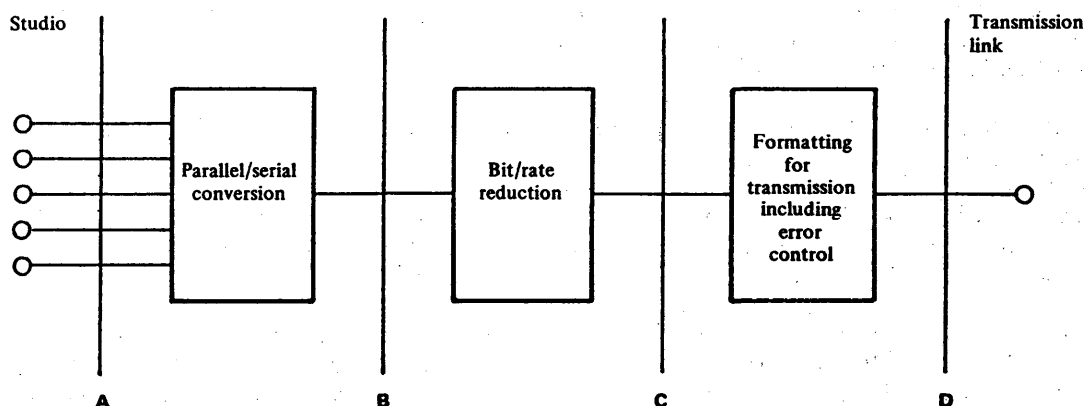
13. Interfaces

- 13.1 within studios and within broadcasting transmitters (Study Groups 10 and 11).
- 13.2 within transmission links other than in the B-ISDN (CMTT).
- 13.3 between studios and transmission links and between transmission links and broadcasting transmitters (CMTT, see Note 2).
- 13.4 within the B-ISDN including the B-ISDN user-network interface (Study Group XVIII).

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. Prime responsibility for interface D is in CCITT Study Group XVIII.

DECISION 67-2

DIGITAL TRANSMISSION OF COMPONENT-CODED TELEVISION
AND HIGH DEFINITION TELEVISION SIGNALS

(1986-1987-1989)

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 CMTT has adopted two ——— new Recommendations 721 ——— and 723 ——— for the transmission of component coded television signals at 32, 34 and 44 Mbit/s and at 140 Mbit/s;
- (d) that CMTT has adopted Study Programme 22A/CMTT for the digital transmission of HDTV signals;
- (e) 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;
- (f) that bit-rate reduction equipments will be important for the transmission of television and HDTV 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 and HDTV signals;
- (g) that in Recommendation G.702, CCITT Study Group XVIII has recommended the hierarchical bit rates for digital networks;
- (h) that in Recommendation G.802, CCITT Study Group XVIII has recommended a specific digital hierarchy for interworking between networks using different hierarchies;
- (j) that for the ISDN, CCITT Study Group XVIII plans to specify broadband services and related broadband channels, channel structures and access capabilities;
- (k) 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 continued. 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 and HDTV in component or multiplexed analogue component form for transmission, with particular emphasis on the following factors:

- the methods for use on contribution circuits and on primary and secondary 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 or HDTV 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, International Organizations and Recognized Private Operating Agencies have indicated as of November 1987 their participation in the work of Interim Working Party CMTT/2.

Administrations:

Germany (Federal Republic of)
 Australia
 Canada
 Denmark
 United States of America
 Finland
 France
 India
 Italy
 Japan
 Norway
 Netherlands
 German Democratic Republic
 United Kingdom
 USSR

International Organizations and Recognized Private Operating Agencies:

INTELSAT
 NANBA
 EBU

CTNE
 CIT-ALCATEL

Chairman of Interim Working Party CMTT/2:

A representative of the Administration of the
Federal Republic of Germany - Dr. Stenger

Vice-Chairmen of Interim Working Party CMTT/2:

A representative of the Administration of Japan - Dr. Murakami
A representative of the Administration of Canada - Mr. K. Davies

ANNEX II

Future activities of Interim Working Party CMTT/2

The terms of reference of Interim Working Party CMTT/2 are 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 and for HDTV in component or multiplexed analogue component form for transmission.

Interim Working Party 11/7 is responsible for defining the subjective quality requirements for such codecs, and for defining the subjective test conditions, supplying still and moving test pictures and carrying out the subjective assessment of the quality of the coded pictures, and the subjective assessment of the effects of transmission impairments, such as errors, which will be specified by Interim Working Parties CMTT/2 and CMTT/3.

1. Contribution/distribution codecs for television

In order to resolve the still open parameters in _____
Recommendations 721 and 723, the following actions are required:

1.1 Contribution codecs at rates near 140 Mbit/s

A multiplexing scheme should be commonly agreed and tests with a complete codec should be carried out possibly before the Plenary Assembly 1990. A special meeting of Interim Working Party 11/7 will be invited in order to evaluate the codec presented for formal tests.

1.2 Contribution/distribution codecs at 32, 34 and 44 Mbit/s rates

It is intended to solve the still open parameters on VLC and video multiplex before the Plenary Assembly as stated in Report 1235, Chapter 7.

2. It is assumed, that a "compromise" codec as specified in Recommendation 723 could be implemented in hardware and made available for tests at the end of 1990 or at the beginning of 1991. The codec should then be presented to IWP 11/7 for final acceptance tests taking into account the requirements as stated in Report 1211.

3. Secondary distribution and HDTV codecs

Reports 1092 and 1239 are brought to the attention of administrations; further contributions are requested.

DECISION 68-2*

TELEVISION AND SOUND-PROGRAMME SIGNALS
IN THE BROADBAND ISDN

(1986-1987-1989)

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 the CMTT requires information on the required capacity for broadband channels transmitting digital television and sound-programme signals in the ISDN;
- (d) that the CCITT 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 the CCITT;
 - 1.3 to maintain liaison with CCITT Study Group XVIII and any other CCITT bodies 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 the CCITT, subject to these results being subsequently approved by the CMTT at its next meeting;
5. that IWP CMTT/3 shall provide appropriate and timely input directly to the relevant Study Groups of CCITT so that this input may be incorporated in the CCITT documentation;
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.

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



8. that a representative of CCIR Study Group 10 and a representative of CCIR Study Group 11 should be members;

9. that Special Rapporteurs or other representatives from the CCITT should be invited to participate as may be required for discussions on particular subjects.

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

ANNEX II

FUTURE ACTIVITIES OF IWP CMTT/3

From the nature of its terms of reference, the timing of the activities of IWP CMTT/3 is affected by the frequency and dates of the meetings of CCITT Study Groups - particularly Study Group XVIII.

Contributions to, and meetings of, IWP CMTT/3 need therefore to be scheduled so as to allow for the most effective interworking and communication with CCITT Study Groups XV and XVIII.

DECISION 76-1

SATELLITE NEWS GATHERING (SNG)

(Question 13/CMTT, Study Programmes 13H/CMTT and 22D/CMTT)

(1987-1989)

CCIR Study Groups 4, 10, 11 and CMTT,

CONSIDERING

- (a) that satellite transmissions using transportable or portable earth stations is an invaluable and at times the only viable solution to the timely transmission of television news from remote locations;
- (b) that throughout the world, where news events take place, standardized and uniform technical and operating procedures should be established to ensure prompt activation of SNG service;
- (c) that there was a unanimous Recommendation of the Fifth World Conference of Broadcasting Unions in Prague, 1986, on the use of international auxiliary broadcast frequencies and transportable earth stations;
- (d) that the special characteristics needed for transportable or portable transmitting earth stations may necessitate acceptance of performance objectives differing from those specified in Recommendation 567 for general purpose satellite television connections;
- (e) that there will be a requirement to provide, on the same uplink service, auxiliary circuits for programme and technical coordination, and for the management of the transportable earth station interface;
- (f) that SNG services from any specific location will be classified as occasional and/or temporary;
- (g) that the very nature of SNG requires that earth stations be activated in an expedient manner, the philosophy of which is not compatible with the long advance notice periods normally established for fixed permanent services;
- (h) that it is expected in the future that HDTV transmissions could originate from portable uplink facilities;
- (j) that the ITU Constitution states in its Preamble: "...fully recognizing the sovereign right of each State to regulate its telecommunication..."

DECIDE

1. that Joint Interim Working Party JIWP CMTT-4-10-11/1 be continued, to prepare, within the Terms of Reference of Study Groups CMTT, 4, 10 and 11, an overall strategy for SNG transmissions, proposals intended to solve the technical, operating and organizational aspects associated with the use of transportable or portable transmitting earth stations for SNG;
2. that the Terms of Reference of the JIWP should be as follows:
 - 2.1 to define the technical quality specifications for the programme video signal acceptable for SNG;
 - 2.2 to define the number and the technical quality specifications for the programme audio signals acceptable for SNG;
 - 2.3 to define the number and the technical quality specifications of auxiliary circuits required for SNG operations;
 - 2.4 to study uniform technical parameters for SNG that may be applicable on a geographically wide scale;
 - 2.5 to study uniform operating procedures for SNG that may also be applicable on a geographically wide scale;
 - 2.6 to study the overall transmission and performance objectives for HDTV transmission by portable satellite earth stations for SNG;
 - 2.7 to determine the technical characteristics of the specific equipment required to meet the objectives in 2.6;
 - 2.8 to identify the operational requirements related to HDTV transmission by portable satellite earth stations for SNG;
 - 2.9 to prepare draft Recommendations on the study items above;
 - 2.10 to investigate and prepare a Report on means to simplify the procedures required to obtain, as expeditiously as possible, temporary authorization to operate SNG facilities;
3. that, in accordance with §2.3.3 of CCIR Resolution 24, Study Groups CMTT, 4, 10 and 11 are jointly entrusted with the responsibility of the work of the JIWP, and in accordance with §2.3.9 of Resolution 24, CMTT is designated to coordinate the work of the JIWP. The JIWP shall submit its reports and results of its work to joint meetings of these Study Groups, when possible;
4. that the work of the JIWP should be completed in the course of the current Study Period; a report and the results of its work shall be submitted to the Final Study Group Meetings in 1993;
5. that the JIWP should, as far as possible, work by correspondence; however it may meet when this is considered necessary by its Chairman, by the Chairmen of Study Groups CMTT, 4, 10 and 11, and by the Director, CCIR;

6. that the JIWP will be chaired by:

Mr. Joseph A. COLSON (NANBA)
Chairman, NANBA Technical Committee
CTV Television Network Limited
42 Charles Street East
Toronto, CANADA M4Y 1T5

Telephone : +1 416 928 6170
Telex : 06-22080
Telefax : +1 416 928 0907

and the Vice-chairmen will be:

Mr. D. PHAM THAT (EBU)
Télédiffusion de France (TDF)
Direction de l'Equipement
21-27 rue Barbes
B. P. 518
F-92542 MONTRouGE CEDEX
France

Telephone: +33 1 46 57 1115
Telex : tedif 250738f
Telefax : +33 1 46 54 3341

Mr. Gabor HEGYI (OIRT)
Administration centrale des
Postes et Telecommunications de Hongrie
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Hungary

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Telex : 221193 gentel h
Telefax : +36 156 0855

and

Mr. Kuniharu ASANO (ABU)
Senior Engineer
Engineering Planning Bureau
Japan Broadcasting Corporation (NHK)
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TOKYO 150
Japan

Telephone : +81 3 465 1234
Telex : 22377 Radio NHK
Telefax : +81 3 485 0952

DECISION 77-1*

TRANSMISSION OF DIGITAL SOUND PROGRAMME SIGNALS OF DIGITAL STUDIO
QUALITY ON CIRCUITS USING THE H₁ CHANNEL

(1987-1989)

The CMTT,

CONSIDERING

- (a) that Decision 18-6 of the CMTT states that Study Group 10 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 sound programme signals;
- (b) that the general principles for the digital transmission of sound programme signals, originated in conformance with Recommendation 646, are contained in Recommendation 659;
- (c) that it is desirable to avoid sampling frequency conversions when transporting signals of digital studio quality over digital circuits;
- (d) that Recommendation 647 specifies auxiliary information that should accompany each digital audio signal;
- (e) that bit rate reduction techniques can be effective for digital sound signals to provide the additional capacity required in the H₁₁ channel for error protection and auxiliary data;
- (f) that in Recommendation G.702, CCITT Study Group XVIII has recommended the hierarchical bit rates for digital networks;

* This Decision should be brought to the attention of CCIR Study Group 10.

- (g) that in Recommendation G.802, CCITT Study Group XVIII has recommended a digital hierarchy for interworking between networks using different hierarchies;
- (h) that for the ISDN, CCITT Study Group XVIII plans to specify broadband services and related broadband channels, channel structures and access capabilities;
- (j) that Decision 68 of the CMTT has established IWP CMTT/3 to maintain liaison with the CCITT;
- (k) that the Recommendation of a suitable transmission system should be achieved during the study period 1986-90,

DECIDES

1. that IWP CMTT/4 should be continued. 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 and multiplexing digital studio sound signals, originated in conformance with Recommendation 646 for transmission over H₁ channels, maintaining the sampling frequency of 48 kHz with particular emphasis on the following factors:
 - 1.1 the need to facilitate the international exchange of digital sound-programme signals between countries using H₁₁ and H₁₂ channels as described in CCITT Recommendation I.412;
 - 1.2 the required performance objectives in terms of audio signal quality, delay, jitter, transmission errors and transmission bit and/or frame slips;
 - 1.3 the method of transmitting the auxiliary signals specified in Recommendation 647;
 - 1.4 the possible use of encryption;
2. that appropriate parts of the work should be carried out in close collaboration with Interim Working Party CMTT/3 which should serve as liaison with CCITT Study Group XVIII to ensure consistency between the studies;
3. that close liaison with CCIR Study Group 10 should be maintained;
4. that IWP CMTT/4 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 could be advantageous for IWP CMTT/3 and IWP CMTT/4 to meet at the same place and time;
5. that the result of work of IWP CMTT/4 should be presented to the CMTT at the interim meeting of the 1990-1994 study period;
6. that the Chairman, the Vice-Chairman, and the composition of IWP CMTT/4 shall be as shown in Annex I.

ANNEX I

The following Administrations and International Organizations have indicated that they wished to participate in the work of Interim Working Party CMTT/4:

Administrations:

Germany (Federal Republic of)
Australia
Denmark
United States of America
France
Italy
Japan
Norway
Netherlands
German Democratic Republic
United Kingdom
Switzerland
USSR

International Organizations:

EBU
OIRT

Chairman of Interim Working Party CMTT/4:

A. WEISSER
Télédiffusion de France
10, Rue d'Oradour-sur-Glane
F-75732 PARIS Cedex 15
France

Vice-Chairman of Interim Working Party CMTT/4:

(Representative of USA, to be nominated)

DECISION 98*

LOW BIT-RATE DIGITAL AUDIO CODING SYSTEMS

(1989)

The CCIR,

CONSIDERING

(a) that recent progress in digital audio coding techniques allows for a very considerable degree of bit-rate reduction to be achieved while maintaining high quality;

(b) that information gathered by the CCIR on various aspects of digital audio coding systems is contained in Reports 799, 953, 1068, 1070, 1095, 1236 and 1199;

(c) that applications exist for low bit-rate audio coding systems for digital audio broadcasting, television sound (especially HDTV and EDTV) and for storage of signals;

(d) that the studies concerning digital techniques in sound are in progress both in Study Groups 10 and the CMTT, being matters related to studio installations and broadcasting dealt with in Study Group 10 and matters related to transmission dealt with by the CMTT,

DECIDES

1. that Joint Interim Working Party 10-CMTT/1 be established within the general terms of reference of Study Groups 10 and CMTT:

1.1 to define the sound quality and other requirements for each application, including:

- studio installation including recording,
- transmission,
- broadcasting from terrestrial transmitters and from satellites;

1.2 to define bit-rate reduction techniques that would satisfy the quality level and other requirements for each of the applications listed in 1.1 while making efficient use of storage or transmission media;

* This Decision should be brought to the attention of the IEC and ISO.

- 1.3 to combine in a single CCIR Report the results of studies on digital audio coding systems which are now dispersed in a number of Reports;
- 1.4 to present a draft Recommendation by the 1991 Interim Meeting;
2. that JIWP 10-CMTT/1 should liaise, where appropriate, with JIWP 10-11/6 (former IWP 11/4), IWP 10/12 (multi-channel sound systems) and JIWP 10-11/3;
3. that JIWP 10-CMTT/1 should, as far as possible, work by correspondence, however, it may meet upon the proposal of its Chairman following consultation with the Chairmen of Study Groups 10 and the CMTT and the Director of the CCIR;
4. that the Chairman and the composition of the JIWP 10-CMTT/1 shall be as shown in Annex I.

ANNEX I

The following Administrations and International Organizations have indicated that they wish to participate in the work of JIWP 10-CMTT/1:

Administrations:

Germany (Federal Republic of)
Australia
Canada
Denmark
United States of America
France
Italy
Japan
German Democratic Republic
United Kingdom
Sweden
Switzerland
USSR

International organizations:

EBU
OIRT

Chairman of JIWP 10-CMTT/1:

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10 Rue D'Oradour-sur-Glane
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Vice-Chairmen of JIWP 10-CMTT/1:

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