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MAINTENANCE OF INTERNATIONAL SOUND-PROGRAMME AND TELEVISION TRANSMISSION CIRCUITS

LINE-UP AND SERVICE COMMISSIONING OF INTERNATIONAL VIDEOCONFERENCE SYSTEMS OPERATING AT TRANSMISSION BIT RATES OF 1544 AND 2048 kbit/s

ITU-T Recommendation N.86

(Previously "CCITT Recommendation")

FOREWORD

The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the International Telecommunication Union. The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation N.86 was revised by the ITU-T Study Group IV (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

In order not to delay publication of this Recommendation, no change has been made in the text to references containing the acronyms "CCITT, CCIR or IFRB" or their associated entities such as Plenary Assembly, Secretariat, etc. Future editions of this Recommendation will contain the proper terminology related to the new ITU structure.

2 In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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ABSTRACT

This Recommendation deals with the line-up and service commissioning of international videoconference systems routed over transmission paths operating at bit rates of 2048 and 1544 kbit/s.

Keywords

Audio tests, limits, line-up, primary rate videoconferencing systems, service commissioning, videoconference centre, videoconferencing, videoconference studio.

LINE-UP AND SERVICE COMMISSIONING OF INTERNATIONAL VIDEOCONFERENCE SYSTEMS OPERATING AT TRANSMISSION BIT RATES OF 1544 AND 2048 kbit/s

(Melbourne, 1988; revised at Helsinki, 1993)

1 General

This Recommendation deals with the line-up and service commissioning of international videoconference systems routed over transmission paths operating at transmission bit rates of 2048 and 1544 kbit/s. In this context an international videoconference system comprises the international videoconference connection and the videoconference rooms which are interconnected.

Figure 1 shows the constituent parts of an international videoconference connection. Recommendation H.110 [1] describes hypothetical reference connections for videoconferencing.

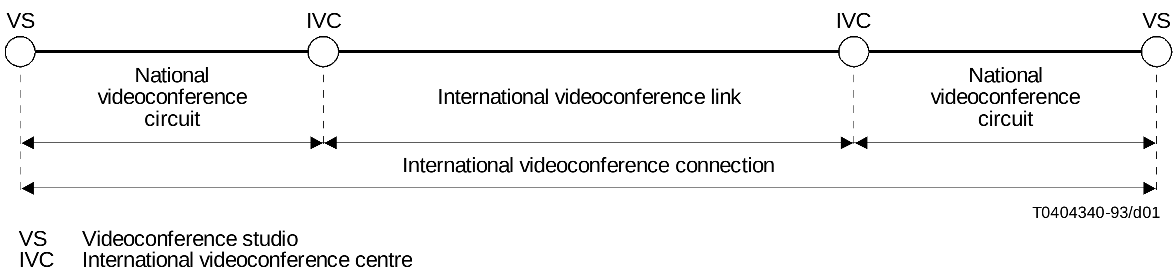

FIGURE 1/N.86

The constituent parts of an international videoconference connection

The video codecs are normally located within the videoconference studios but in some circumstances are located elsewhere so that the local tail serving the videoconference studio may be provided on wideband analogue (e.g. 5.5 MHz) or higher order digital transmission systems (e.g. 140 Mbit/s). Codecs are described in Recommendation H.120 [2].

The location of any 2048/1544 kbit/s remultiplexers which may be involved will be as agreed between the Administrations.

Clause 5 gives guidance on the setting up and testing arrangements for videoconference studios.

The international videoconference centre provides the interconnection point of the national videoconference circuit and the international link. This interconnection may be made manually or by automatic means.

Normally the international videoconference link will be common for all videoconference calls between the two Administrations concerned, whereas the national videoconference circuits will vary from call to call. Thus, in addition to the setting up and lining up of the constituent parts of the international videoconference connection, service commissioning tests are made between videoconference studios prior to the opening of an international videoconference service to ensure that a service can be satisfactorily sustained.

2 Setting up and lining up the constituent parts of the connection

2.1 National videoconference circuits

The national videoconference circuits should be set up and tested in accordance with the national procedures of the administrations concerned. This will include the line up of any sections which may not be provided as 2048 or 1544 kbit/s digital paths. The 2048 and 1544 kbit/s performance test limits to be met are given in Table 1 and it is recommended that two tests should be made, each of one hour's duration, on different days and at times that cover the peak traffic periods on the route concerned.

2.2 International link

The international videoconference link will only need to be set up and tested when establishing the first service between two Administrations. The procedures of Recommendation M.2110 [3] should apply. The test results should meet the performance limits given in Table 1.

TABLE 1/N.86

Performance test limits[a)]

	Nominal bit rate[b)] (kbit/s)	Bit error ratio (BER)	Max. errors in 1 hour	Severely errored events[c)] in 1 hour	Error-free seconds (EFS) (%)
National videoconference circuit	2048	1×10^{-6}	6 912	0	92
	1544	1×10^{-6}	5 530	0	92
International videoconference link	2048	1×10^{-6}	6 912	2	92
	1544	1×10^{-6}	5 530	2	92
International videoconference connection	2048	3×10^{-6}	20 736	2	92
	1544	3×10^{-6}	16 589	2	92

a) The limits are provisional and subject to further study.

b) Structured formatting required with a consequent reduction in actual test bit rate as follows:

At 2048 kbit/s, test bit rate = 1920 kbit/s (time slots 1-15 and 17-31 only);

At 1544 kbit/s, test bit rate = 1536 kbit/s (8 kbit/s used for frame alignment).

When working through a 2048/1544 kbit/s remultiplex or the test signal at 2048 kbit/s side should be restricted as specified in Recommendation H.130 for four times 384 kbit/s working, i.e. TS 1-15 and 17-25. The limits will be as specified for 1544 kbit/s systems.

c) Severely errored events are defined by the particular data tester used, e.g. 20 000 errors in 100 000 bits. A continuous period of up to 10 seconds, during which severely errored transmission persists, will be considered as a single severely errored event (for further study).

NOTES

1 In addition to the above limits the BER shall be no worse than 1×10^{-5} over any 5-minute period during the tests (5760 errors at 2048 kbit/s and 4608 errors at 1544 kbit/s). If this test fails, then corrective action shall be taken on the offending section.

2 For loop-tests, the above limits should be doubled (92% EFS becoming 84% EFS).

3 Performance check, codec-to-codec

The constituent parts of the connection having been satisfactorily lined up and connected together at the international videoconference centres, three tests (each of one hour's duration) may (if required) be made between the codecs. The tests should be made on different days and at times to cover the peak traffic periods for the route. The digital test equipment should be connected at the digital line side of the codecs, as close to the codecs as possible. Each test should meet the data performance limits given in Table 1.

Where loopback arrangements exist, loopback measurements may be made in order to obtain reference measurements for subsequent maintenance. Care must be taken to avoid the concurrent operation of more than one loopback device where several devices are used on the same circuit.

4 Digital test equipment

The digital test equipment required for the above tests shall be capable of transmitting and receiving a test pattern within a signal structured in accordance with Recommendation G.732 [4] for 2048 kbit/s interfaces or Recommendation G.733 [5] for 1544 kbit/s interfaces. The nature of the test pattern is undefined but should be the subject of further study.

If compatible digital test equipment is not available at both ends of the link, circuit or connection under test, then one set of test equipment should be used to transmit and receive, with the distant end being looped-back.

5 Videoconference studios

All videoconference studios that will be used for international videoconference calls should comply with agreed design standards.

To ensure proper interworking between studios, the audio part should comply with the values given in 5.1 and 5.2. Test equipment for audio testing is listed in Annex A.

5.1 Local audio test

i) Connect the white noise generator, filter, amplifier and loudspeaker in series. With the SPLM placed as in diagram a) of Figure 2, adjust the white noise level to measure 90 dBSPL on the SPLM.

ii) The loudspeaker should now be positioned relative to the ORP (see Note) as in diagram b) of Figure 2 and the audio levels at the input to the codec should have an average value of –9 dBm with respect to all seating positions. Any adjustment to achieve this should be made immediately prior to the codec [see diagram b) of Figure 2].

NOTE – The ORP is the point located 1200 mm from the floor level and 150 mm to the rear of the working edge of the conference desk and on the centre line of each seating position.

iii) The white noise generator, filter and amplifier should be used to simulate the nominal received level (–9 dBm) from a codec into the studio loudspeaker. Measure the SPL at the ORP which should be in the range of 67 to 75 dBA, depending on the acoustic conditions [see diagram c) of Figure 2].

The audio level at the input to the codec due to loudspeaker/microphone acoustic coupling should be measured at less than –40 dBm.

5.2 Audio tests (studio-to-studio) (Figure 3)

In the following tests, each end will be required to send and measure audio levels at the codec interface. For ease of testing each end has been designated either A or B and this should be decided prior to any testing.

The received level at "A" will be referred to as studio Level 1 and the received level at "B" will be referred to as studio Level 2.

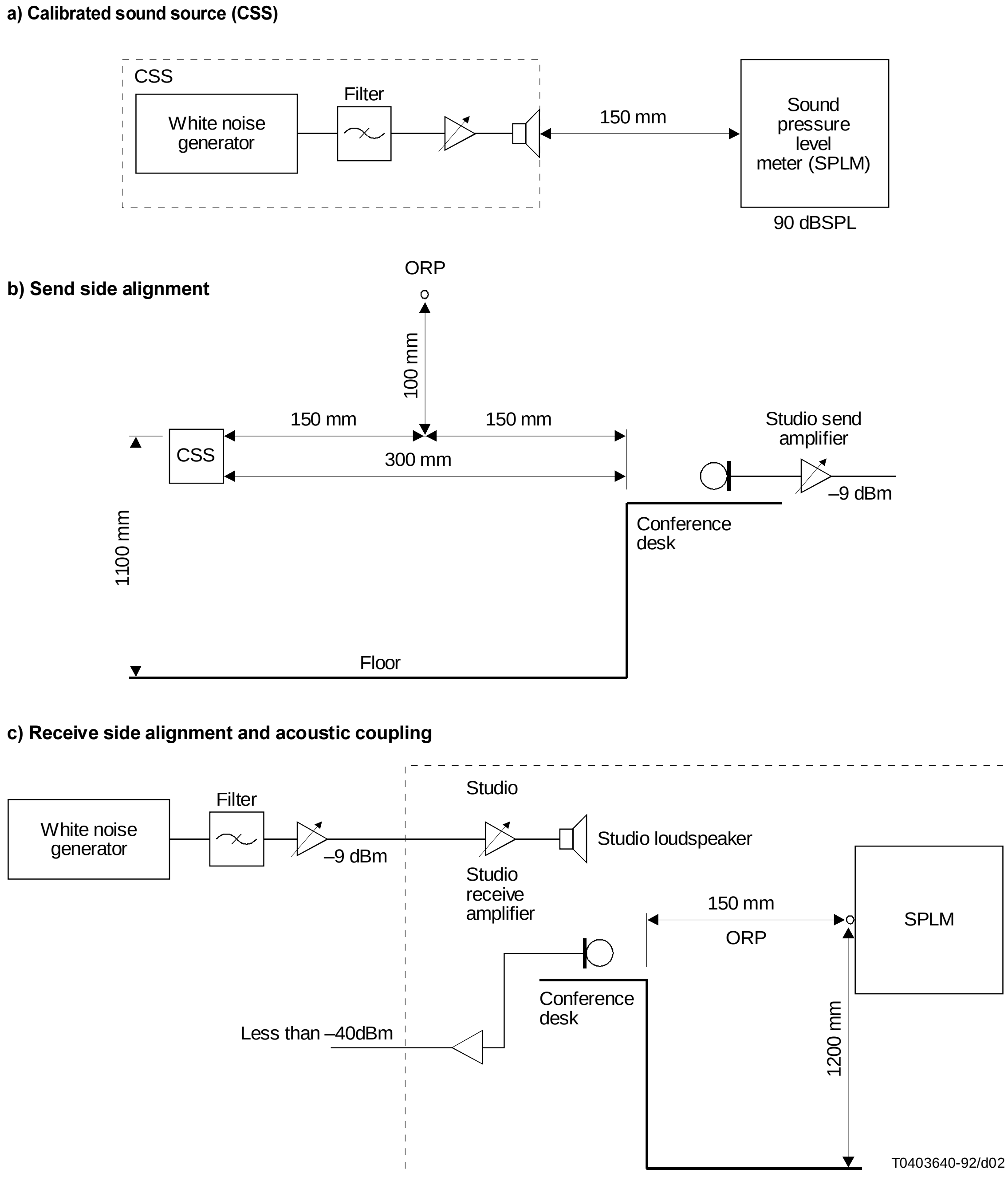

FIGURE 2/N.86

Audio alignment

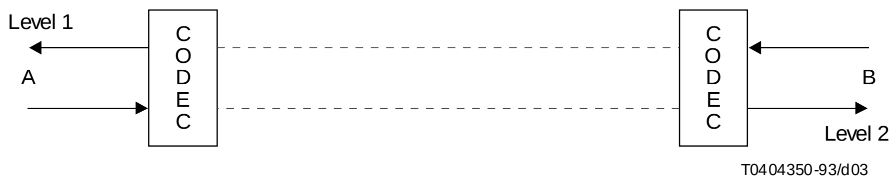

FIGURE 3/N.86

Designations for levels and ends for audio tests

The white noise source referred to below is positioned as in diagram b) of Figure 2 and adjusted as described in 5.1, ii).

– *Electrical noise*

Conditions:	A and B mute room microphones. A and B enable echo canceller.
Measurements:	Level 1 = electrical noise from Studio B. Level 2 = electrical noise from Studio A.
Specification:	Level 2 < –53 dBm (flat) at audio output of codec. Level 1 < –53 dBm (flat) at audio output of codec.

5.2.1 "A" end audio tests

a) *Level check*

Conditions:	B sends white noise source at –9 dBm level. B enables studio microphone. A mutes studio microphone. A and B enable echo cancellers.
Measurements:	i) A measures Level 1 at audio output of codec. ii) A measures sound pressure level at ORP.
Specification:	i) Level 1 at audio output of codec = –9 dBm. ii) Level at ORP = 67 to 75 dBA.

b) *Echo check*

Conditions:	A sends white noise at –9 dBm level. B enables studio microphones. A and B enable echo cancellers.
Measurements:	A measures Level 1 (echo).
Specification:	Level 1 < –40 dBm at audio output of codec.

c) *Echo return loss measurement*

Conditions:	A sends white noise at –9 dBm level. B enables studio microphones. A enables echo canceller. B disables echo canceller.
Measurements:	A measures Level 1.
Specification:	Level 1 < –15 dBm at audio output of codec.

d) *Crosstalk check*

Conditions:	A sends white noise at –9 dBm level. B mutes studio microphones. A and B enable echo canceller.
Measurements:	A measures Level 1 (crosstalk).
Specification:	Level 1 < –50 dBm.

5.2.2 "B" end audio test

a) *Level check*

Conditions:	A sends white noise source at –9 dBm level. A enables source microphone. B mutes studio microphone. B and A enable echo cancellers.
Measurements:	i) B measures Level 1 at audio output of codec. ii) B measures sound pressure level at ORP.
Specification:	i) Level 2 at audio output of codec = –9 dBm. ii) Level at ORP = 67 to 75 dBA.

b) *Echo check*

Conditions: B sends white noise at –9 dBm.
A enables studio microphones.
B and A enable echo cancellers.

Measurements: B measures Level 1 (echo).

Specification: Level 2 < –40 dBm at audio output of codec.

c) *Echo return loss measurement*

Conditions: B sends white noise at –9 dBm level.
A enables studio microphones.
B enables echo canceller.
A disables echo canceller.

Measurements: B measures Level 2.

Specification: Level 2 < –15 dBm at audio output of codec.

d) *Crosstalk check*

Conditions: B sends white noise at –9 dBm level.
A mutes studio microphones.
B and A enable echo canceller.

Measurements: B measures Level 2 (crosstalk).

Specification: Level 2 < –50 dBm.

6 Service commissioning tests

6.1 General

The international videoconference connection having been satisfactorily tested, functional video and audio service commissioning tests should be undertaken between the videoconference studios.

6.2 Test videoconference studios

The videoconference studio chosen by an Administration for commissioning tests should be typical of all the other studios used for the service. This studio should then serve as a reference studio for any future tests between videoconference studios with other Administrations.

The reference studio for each administration should be identified to all other administrations. The parameters of this studio should also be shared with all other Administrations.

6.3 Commissioning test

The purpose of the tests is to demonstrate that the international videoconference system performs adequately when the constituent parts are connected together. The tests include a subjective assessment of the main functions of each videoconference studio and selected objective tests. The tests are not intended to be exhaustive, but should serve as sample checks in compliance with the standards and as a confident indicator to both Administrations before the opening of an international videoconference service.

7 Abbreviations

For the purposes of this Recommendation, the following abbreviations used:

BER Bit error ratio

CSS Calibrated sound source

EFS Error-free seconds

ORP Optical reference point

SPL	Sound pressure level
SPLM	Sound pressure level meter
TS	Time slots
SES	Severely errored seconds

Annex A

Test equipment for videoconference studio audio tests

(This annex forms an integral part of this Recommendation)

To conduct the test it will be necessary to have the following test equipment:

a) a white noise generator set to a bandwidth of 50 Hz-10 kHz;

b) a filter which produces a flat (within 3 dB) white or pink noise response within the band 250-3000 Hz rolling off at 48 dB/octave outside these limits;

c) an audio amplifier and associated loudspeaker with the following properties:

   i) the loudspeaker must be able to deliver a sound pressure level of at least 100 dBSPL at 150 mm from the loudspeaker in the axis of the loudspeaker;

   ii) the acoustic properties must be similar to the average human mouth (as regards the law of decreasing acoustic pressure in the axis of emission and the law of directivity);

   iii) the loudspeaker must be single and small (diameter less than 15 cm) conforming to DIN 45 500;

d) a sound pressure level meter (with A-weighting and linear scale);

e) a level measuring set.

References

[1] CCITT Recommendation H.110, *Hypothetical reference connections for videoconferencing using primary digital group transmission*.

[2] CCITT Recommendation H.120, *Codecs for videoconferencing using primary digital group transmission*.

[3] CCITT Recommendation M.2110, *Bringing international digital blocks, paths and sections into service.*

[4] CCITT Recommendation G.732, *Characteristics of primary PCM multiplex equipment operating at 2048 kbit/s*.

[5] CCITT Recommendation G.733, *Characteristics of primary PCM multiplex equipment operating at 1544 kbit/s*.

[6] CCITT Recommendation H.130, *Frame structures for use in the international interconnection of digital codecs for videoconferencing or visual telephony*.