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TELEPHONE TRANSMISSION QUALITY

OBJECTIVE ELECTRO-ACOUSTICAL MEASUREMENTS

METHODS FOR EVALUATING THE TRANSMISSION PERFORMANCE OF DIGITAL TELEPHONE SETS

ITU-T Recommendation P.66

(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 P.66 was revised by the ITU-T Study Group XII (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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.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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METHODS FOR EVALUATING THE TRANSMISSION PERFORMANCE OF DIGITAL TELEPHONE SETS

(Melbourne, 1988, amended at Helsinki, 1993)

1 Introduction

The CCITT recommends the following method to evaluate the voice transmission performance of a digital telephone set using encoding conforming to Recommendation G.711 (see also Recommendation P.31). A digital telephone set is one in which the A/D and D/A converters are built in and the connection to the network is via a digital bit-stream. This poses a fundamental problem in applying existing performance and measurement techniques, such as Recommendations P.64, P.34 and P.38, since these are generally given in terms of a transfer function of analogue input and output quantities, e.g. a frequency response. The principles involved in this Recommendation are applicable to handset, headset and hands-free operation; however, at present only procedures applicable to handset operation have been developed.

2 Approaches for testing digital telephones

There are two methods for evaluating the transmission performance of a digital telephone, the codec approach and the direct approach.

In the short term, use of the codec approach is advocated since many Administrations already have some experience with this methodology (see Recommendation O.133).

2.1 Codec approach

In this approach, shown in Figure 1, a codec is used to convert the companded digital input/output bit-stream of the telephone set to the equivalent analogue values, so that existing test procedures and equipment can be used. This codec should be a high-quality codec whose characteristics are as close as possible to ideal (see Clause 5).

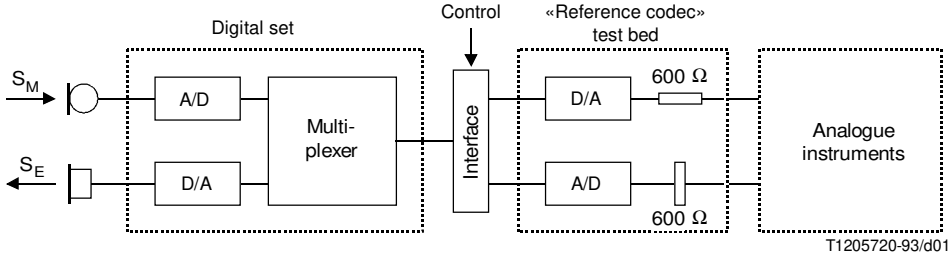

FIGURE 1/P.66

Digital telephone test arrangement (codec approach)

2.2 Direct digital processing approach

In this approach, shown in Figure 2, the companded digital input/output bit-stream of the telephone set is operated upon directly.

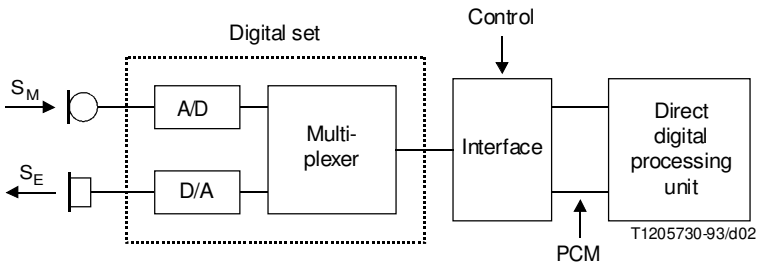

FIGURE 2/P.66

Digital telephone test arrangement
(direct digital processing approach)

3 Definition of 0 dB reference point

To preserve compatibility with existing codecs already in use in local digital switches, which are defined as a 0 dBr point, the codec (A- or μ-law) should be defined as follows:

– *D/A converter* – A digital test sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose r.m.s. value is 3.14 dB (A-law) or 3.17 dB (μ-law) below the maximum full-loaded capacity of the codec will generate 0 dBm across a 600 ohm load;

– *A/D converter* – A 0 dBm signal generated from a 600 ohm source will give the digital test sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose r.m.s. value is 3.14 dB (A-law) or 3.17 dB (μ-law) below the maximum full load capacity of the codec;

where DTS is defined as a periodic sequence of character signals as given in Tables 5/G.711 and 6/G.711.

4 Definition of interfaces

The digital telephone test equipment will, in general, be connected to the telephone under test through an interface.

Such an interface should be able to provide all the signalling and supervisory sequences necessary for the telephone set to be working in all test modes. The interface must be capable of converting the digital output stream from the tested set (which may be in various formats, depending on the specific type of telephone set, e.g. conforming to Recommendation I.412 for ISDN sets), to a form compatible with the test equipment. Interfaces can be applied for sending and receiving separately, taking into account telephone sets which are connected to various types of exchanges.

5 Codec specification

5.1 Ideal codec

The ideal codec consists of an independent encoder and decoder whose characteristics are hypothetical and comply with Recommendation G.711. The ideal encoder is a perfect analogue-to-digital converter preceded by an ideal low-pass filter (assumed to have no attenuation/frequency distortion and no envelope-delay distortion), and may be simulated by a digital processor. The ideal decoder is a perfect digital-to-analogue converter followed by an ideal low-pass filter

(assumed to have no attenuation/frequency distortion and no envelope-delay distortion), and which may be simulated by a digital processor[1)] .

For the measurement of the sending side of a telephone set, the output digital signal is converted by the decoder to an analogue signal. The electrical characteristics of this output signal are measured using conventional analogue instruments. For the measurement of the receiving side of a telephone set, the analogue output from a signal source is converted to a digital signal by the ideal encoder and fed to the receiving input of the digital telephone set.

5.2 Reference codec

A practical implementation of an ideal codec may be called a reference codec (see 4/O.133).

For the reference codec, characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion, etc. should be better than the requirements specified in Recommendation G.714, so as not to mask the corresponding parameters of the set under test. A suitable reference codec may be realized by using:

1) at least 14 bit linear A/D and D/A converters of high quality, and transcoding the output signal to the A- or μ-law PCM format;

2) a filter response that meets the requirements of Figure 3.

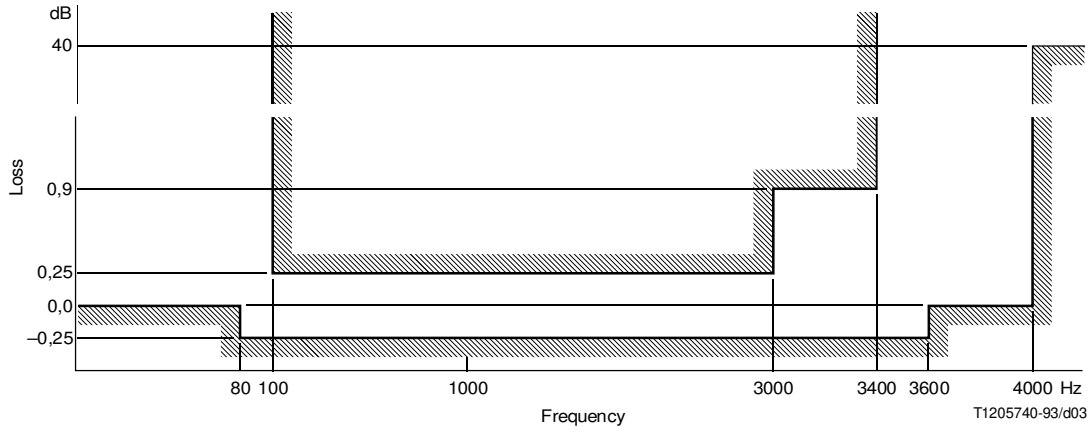

FIGURE 3/P.66

Attenuation/frequency distortion of the sending or receiving sides of the reference codec

5.2.1 Analogue interface

The output and input impedances return loss and longitudinal conversion losses of the analogue interface of the reference codec should be in accordance with 3.1.1/O.133.

5.2.2 Digital interface

The fundamental requirements for the reference codec digital interface are given in the appropriate Recommendations (e.g. I.430-Series Recommendations for ISDN telephone sets).

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1) This characteristic can be realized, for example, using oversampling techniques and digital filters.

6 Measurement of digital telephone transmission characteristics

Use of the codec test approach means that test procedures for digital telephone sets in general follow those for analogue sets (see Recommendation P.64). The reference codec should meet the requirements of Clause 5. An important difference, however, concerns the test circuits themselves, see Figures 4 to 10.

The set is connected to the interface and is placed in the active call state.

NOTE – When measuring digital telephone sets, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance on the frequencies of ± 2% which may be used to avoid this problem, except for 4 kHz where only the –2% tolerance may be used.

6.1 Sending

6.1.1 Sending frequency characteristic

The sending frequency characteristic is measured according to Recommendation P.64 using the measurement set-up shown in Figures 4 or 5, depending on the excitation signal used.

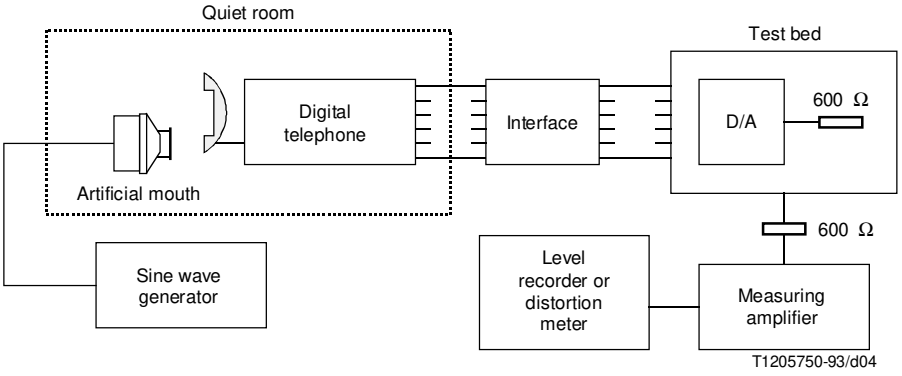

FIGURE 4/P.66

Measurements of sending frequency characteristic – Swept sine wave technique

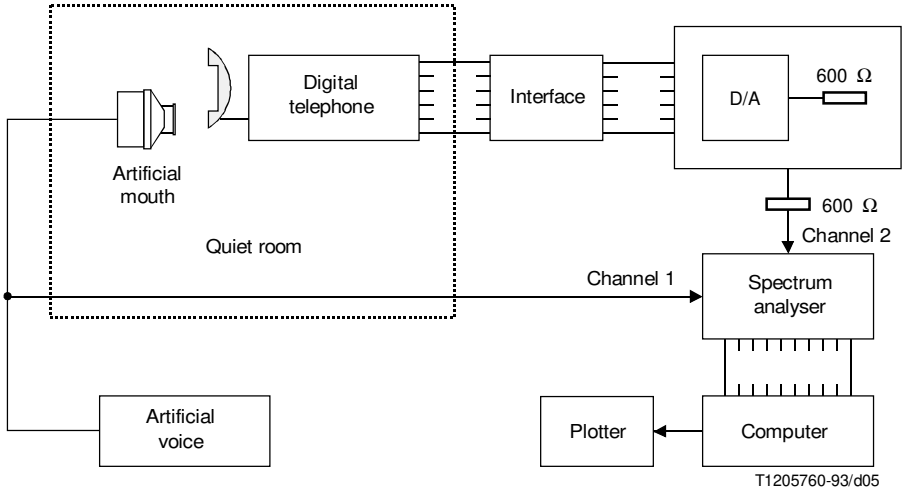

FIGURE 5/P.66

Measurement of sending frequency characteristic – Wideband technique

6.1.2 Send loudness rating

This should be calculated from the sensitivity/frequency characteristic determined in 6.1.1 by means of Recommendation P.79.

NOTE – Other methods for calculating loudness ratings used by some Administrations for their own internal planning purposes can be found in Supplement No. 19 to Series P Recommendations.

6.1.3 Distortion

Method 1 – Noise

The input at MRP is a band-limited noise signal corresponding to Recommendation O.131. ARL is defined as the acoustic level, at MRP, that produces –10 dBm0 at the terminal input. The test signal is then applied relative to ARL at –45, –40, –35, –30, –24, –20, –17, –10, –5, 0, 4, 7 dB. The input sound pressure level is limited at +5 dBPa for this measurement.

The ratio of the signal to total distortion power of the digital signal output is measured (see Recommendation O.131).

Method 2 – Sinewave

A sinewave signal with a frequency in the range of 1004 Hz to 1025 Hz is applied at MRP. ARL is defined as the acoustic level, at MRP, that produces –10 dBm0 at the terminal output. The test signal is then applied relative to ARL at –35, –30, –25, –20, –15, –10, –5, 0, 7, 10 dB. (A-Law) and –37.5, –32.5, –22.5, +7.5 dB (μ-Law). The input sound pressure level is limited at +10 dBPa for this measurement.

The ratio of the signal to total distortion power of the digital signal output is measured with a psophometric noise weighting according to Recommendation P.53.

6.1.4 Noise

With the handset mounted at LRGP and the earpiece sealed to the knife-edge of the artificial ear in a quiet environment (ambient noise less than 30 dBA), the noise level at the digital output is measured with apparatus including psophometric weighting according to Recommendation P.53.

NOTE – The ambient noise criterion will be met if the ambient noise does not exceed NR20 [ISO Recommendation 1996, 1971].

6.1.5 Discrimination against out-of-band input signal

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

For an input frequency of 1 kHz at a level of –4.7 dBPa at the MRP, a reference level is measured at the digital interface.

For input signals at frequencies of 4.65 kHz, 5 kHz, 6 kHz, 6.5 kHz, 7 kHz and 7.5 kHz at the level specified in 2.3/P.31 the level of any image frequencies at the digital interface is measured.

6.2 Receiving

6.2.1 Receiving frequency characteristic

The receiving frequency characteristic is measured according to Recommendation P.64 using the measurement set-up shown in Figures 6 or 7, depending on the excitation signal used.

6.2.2 Receiving loudness rating

This should be calculated from the sensitivity/frequency characteristic determined in 6.2.1 by means of Recommendation P.79.

NOTE – Other methods for calculating loudness rating used by some Administrations for their own internal planning purposes can be found in Supplement No. 19 to Series P Recommendations.

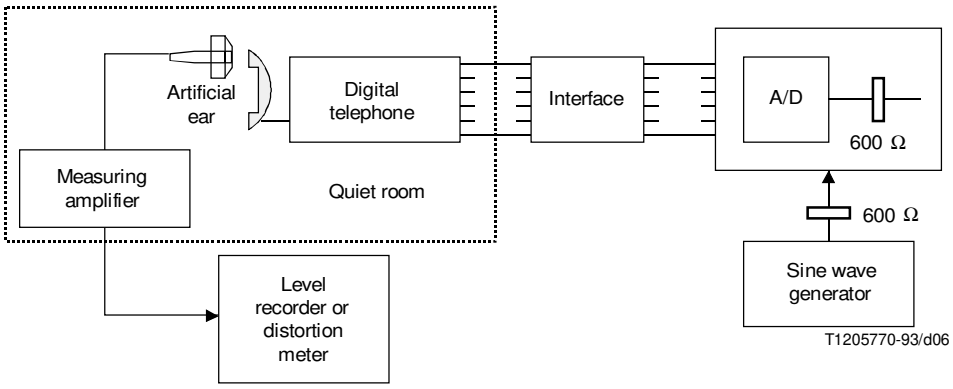

FIGURE 6/P.66

Measurement of receiving frequency characteristic – Swept sine wave technique

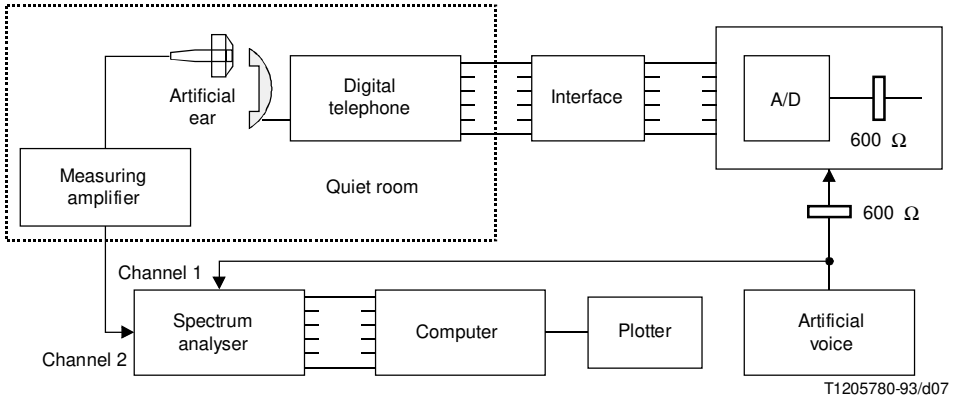

FIGURE 7/P.66

Measurement of receiving frequency characteristic – Wideband technique

6.2.3 Distortion

Method 1 – Noise

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A digitally simulated band-limited noise signal corresponding to Recommendation O.131 is applied to the digital interface at the following levels: –55, –50, –45, –40 –34, –30, –27, –20, –15, –10, –6, –3 dBm0.

The ratio of the signal to total distortion power is measured in the artificial ear (see Recommendation O.131).

Method 2 – Sinewave

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A digitally simulated sinewave signal in the range 1004 Hz to 1025 Hz is applied to the digital interface at the following levels: –45, –40, –35, –30, –25, –20, –15, –10, –3 , 0 dBm0 (A-Law) and –47.5, –42.5, –32.5, –2.5 dBm0 (μ-Law).

The ratio of the signal-to-total distortion power is measured in the artificial ear with A-weighting applied.

6.2.4 Noise

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A signal corresponding to either decoder output value number 1 (A-Law), or decoder output value 0 (μ-Law), is applied at the digital interface. The A-weighted noise level is measured in the artificial ear.

The ambient noise for this measurement shall not exceed 30 dBA.

6.2.5 Spurious out-of-band signals

The handset is mounted at LRGP and the earpiece is sealed to the knife- edge of the artificial ear. For input signals at the frequencies 500, 1000, 2000, 3150 Hz applied at the level specified in Recommendation P.31, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively in the artificial ear.

6.3 Sidetone

Provision should be made for driving the microphone of the telephone set under test as described in 6.1 and measuring the receiver output as described in 6.2. The recommended method of measuring sidetone is with the microphone and receiver mounted in the same handset, and using a test fixture which includes the artificial mouth and the artificial ear located relative to each other in accordance with Recommendation P.64.

NOTE – Care should be taken to avoid mechanical coupling between the artificial mouth and the artificial ear.

6.3.1 Sidetone frequency characteristic

6.3.1.1 Talker sidetone frequency characteristic

The talker sidetone frequency characteristic is measured according to Recommendation P.64 using the measurement set-up of Figure 8 or 9 depending on the excitation signal used. The reference codec is not used in this measurement but may remain in the test circuit, with no external coupling path.

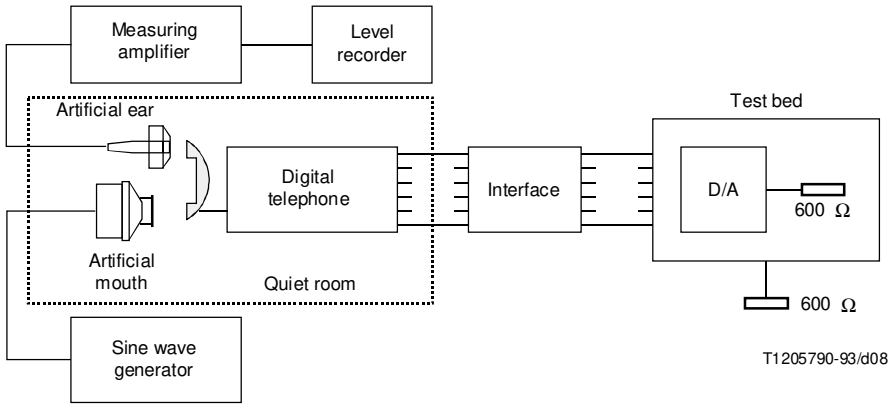

FIGURE 8/P.66

Measurement of talker sidetone frequency characteristic – Swept sine wave technique

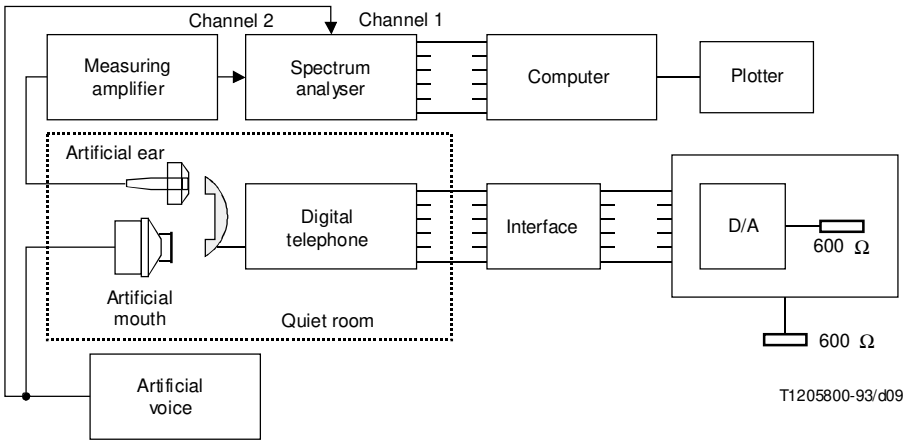

FIGURE 9/P.66

Measurement of talker sidetone frequency characteristic – Wideband technique

6.3.1.2 Listener sidetone frequency characteristic

The listener sidetone frequency characteristic is measured according to Recommendation P.64 using the measurement set-up of Figure 10. The reference codec is not used in this measurement but may remain in the test circuit, with no external coupling path.

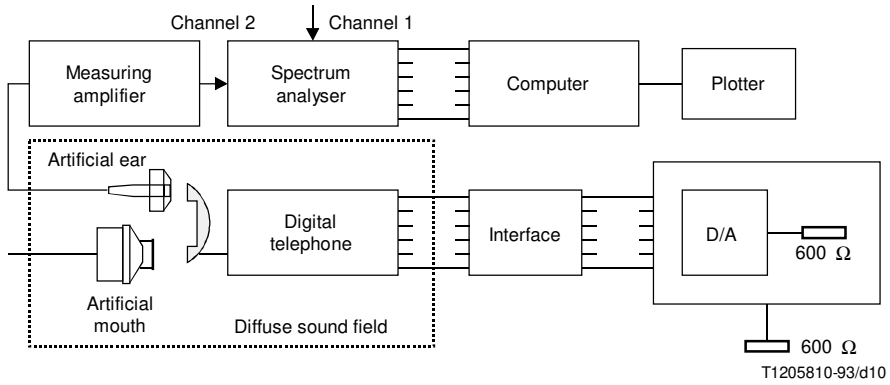

FIGURE 10/P.66

Measurement of listener sidetone frequency characteristic

6.3.2 Sidetone masking rating

This should be calculated from the sensitivity/frequency characteristic determined in 6.3.1.2 by means of Recommendation P.79.

6.3.3 Listener sidetone rating

This should be calculated from the sensitivity\frequency characteristic determined in 6.3.1.2 by means of Recommendation P.79.

6.4 Terminal coupling loss

Terminal Coupling Loss (TCL) is measured in free-air in such a way that the inherent mechanical coupling of the handset is not affected.

When performing tests, the test space acoustics must not have a dominating influence. It is recommended for objective measurements that the test space be practically free-field (anechoic) down to a lowest frequency of 275 Hz, and be such that the test handset lies totally within the free-field volume. This is met by having a reverberation distance r ≥ 50 cm.

NOTE – A method of verifying the reverberation distance will be found in the second edition of the *Handbook of Telephonometry*.

The test is performed with the handset suspended in a noose around the earcap with the handset cord hanging freely from below the handset (see Figure 11).

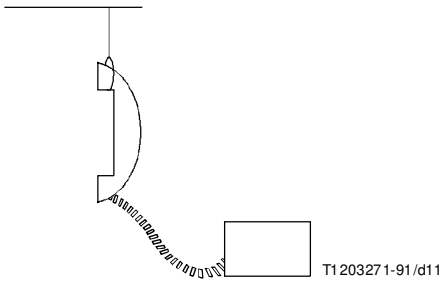

FIGURE 11/P.66

The attenuation from digital input to digital output is measured at one twelfth octave frequencies as given by the R.40-Series of preferred numbers in ISO 3 for frequencies from 300 to 3400 Hz, using the measurement arrangements shown in Figure 12. The ambient noise level shall be less than 30 dBA.

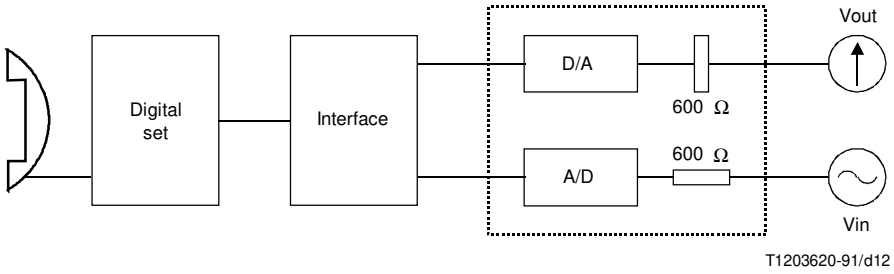

FIGURE 12/P.66

6.5 Delay

The handset is mounted at LRGP. The earpiece is sealed to the knife-edge of the artificial ear. The signal is looped at the digital interface so that the sending signal is fed directly back to the receiving path.

NOTES

1 This method of direct measurement, where the signal is looped at the digital interface, can be used where the sidetone is quiet.

2 If the sidetone is found to be a problem, then the measurement may be made with the direction of transmission separated by looping back to a second terminal.

The acoustic input level shall be ARL as defined in 6.1.3.

For each of the nominal frequencies F_0 given in Table 1, the delay at each value of F_0 is derived from the measurements at the corresponding values for F_1 and F_2.

TABLE 1/P.66

Frequencies for delay measurements

F_0 (Hz)	F_1 (Hz)	F_2 (Hz)
500	475	525
630	605	655
800	775	825
1000	975	1025
1250	1225	1275
1600	1575	1625
2000	1975	2025
2500	2475	2525

The measurement configuration is shown in Figure 13.

For each value of F_0, the delay is evaluated as follows:

1) output the frequency F_1 from the spectrum analyser;

2) measure the phase shift in degrees between ch1 and ch2 (P_1);

3) output the frequency F_2 from the spectrum analyser;

4) measure the phase shift in degrees between ch1 and ch2 (P_2);

5) compute the delay in milliseconds from the formula:

$$D = \frac{-1000 \times (P_2 - P_1)}{360 \times (F_2 - F_1)}$$

NOTE – When using this formula care must be taken that both P_1, P_2 and (P_1 - P_2) are in the range of 0 to 360 degrees; any negative value must first be adjusted by adding 360. The delay of the electro-acoustic equipment shall not be deducted from the calculated delay.

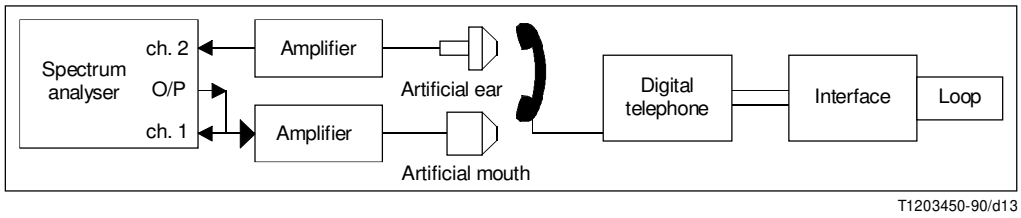

FIGURE 13/P.66

6.6 Stability

The measurement is made at an input signal level of 0 dBm0, at one-twelfth octave intervals for frequencies from 200 Hz to 4000 Hz. With the handset and the transmission circuit fully active, the attenuation from digital input to digital output is measured under one of the following conditions.

Method 1

a) The handset shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0.5 m from the apex of the corner. One surface shall be marked with a diagonal line extending from the corner and a reference position 250 mm from the corner formed by the three surfaces, as shown in Figure 14.

b) The handset with the transmission circuit fully active, shall be positioned on the defined surface as follows:

   i) the mouthpiece and earcap shall face towards the surface;

   ii) the handset shall be placed centrally on the diagonal line with the earcap nearest to the apex of the corner;

   iii) the extremity of the handset shall coincide with the normal to the reference point, as shown in Figure 14.

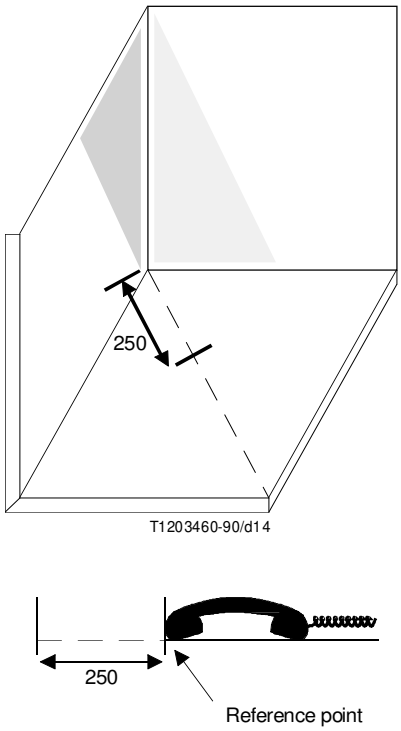

FIGURE 14/P.66

Reference corner

Method 2

The handset, with the transmission circuit fully active, is placed with the earcap and mouthpiece facing a hard, smooth surface free of any other object for 0.5 m.

Annex A

Measurement of variation of gain with input level

(This annex forms an integral part of this Recommendation)

A.1 Sending

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A sinewave signal with a frequency in the range 1004 Hz to 1025 Hz is applied at the MRP. The level of this signal is adjusted until the output of the terminal is –10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

–45, –40, –35, –30, –25, –20, –15, –10, –5, 0, 5, 10, 13 dB relative to ARL.

The variation of gain relative to the gain for ARL is measured.

NOTE – Selective measurements may be used to avoid the effects of ambient noise.

A.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A digitally simulated sinewave signal with a frequency in the range 1004 Hz to 1025 Hz shall be applied at the digital interface at the following levels:

–55, –50, –45, –40, –35, –30, –25, –20, –15, –10, –5, 0, 3 dBm0.

The variation of gain relative to the gain at an input level of –10 dBm0 shall be measured in the artificial ear.

NOTE – Selective measurement may be used to avoid the effects of ambient noise.