INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

P.310

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

(02/96)

TELEPHONE TRANSMISSION QUALITY

SUBSCRIBERS' LINES AND SETS

TRANSMISSION CHARACTERISTICS FOR TELEPHONE BAND (300-3400 Hz) DIGITAL TELEPHONES

ITU-T Recommendation P.310

(Previously "CCITT Recommendation")

FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). 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, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation P.310 was prepared by ITU-T Study Group 12 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 6th of February 1996.

NOTE

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

CONTENTS

SUMMARY

This Recommendation provides audio performance requirements and associated testing for telephone band (300-3400 Hz) digital telephones.

Requirements and test methods are specified for the major audio transmission parameters including Sending and Receiving Loudness Ratings, frequency response, noise, distortion, spurious signals, sidetone, echo path and delay.

This Recommendation is only applicable to digital telephones using encoding conforming to Recommendations G.711 (64 kbit/s, PCM) and G.726 (32 kbit/s, ADPCM). Lower coding rates, for example Recommendation G.726 (16 kbit/s, LD-CELP) are still under study.

TRANSMISSION CHARACTERISTICS FOR TELEPHONE BAND (300-3400 Hz) DIGITAL TELEPHONES

(Geneva, 1996)

1 Scope

This Recommendation deals with sending and receiving loudness ratings, sidetone masking rating, listener sidetone rating, sending and receiving sensitivity/frequency characteristics, noise and distortion characteristics, out-of-band signals, TCLw stability loss and delay of telephone band (300-3400 Hz) digital handset telephones using "Waveform" encoding according to Recommendation G.711 [1] (PCM at both 64 and 56 kbit/s) and G.726 [2] (ADPCM, 32 kbit/s).

The objective measurement methods for testing are covered in Annexes B and C.

The use of digital telephones using Recommendation G.728 [3] (LD-CELP, 16 kbit/s) and mobile/cordless telephones are under study.

Requirements applicable to low acoustic impedance transducers and digital telephone sets using non-linear techniques are under study.

The requirements listed in this Recommendation should also be used as the basis of requirements for other "Waveform" encoding schemes.

The values given in this Recommendation should be used for developing specifications which will include assigning tolerances, etc.

2 Normative References

The following Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision: all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

[1] CCITT Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies.*

[2] CCITT Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM).*

[3] CCITT Recommendation G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction.*

[4] ITU-T Recommendation P.10 (1993), *Vocabulary of terms on telephone transmission quality and telephone sets.*

[5] ITU-T Recommendation G.111 (1993), *Loudness Ratings (LRs) in an international connection.*

[6] CCITT Recommendation G.712 (1992), *Transmission performance characteristics of pulse code modulation.*

[7] CCITT Recommendation G.223 (1988), *Assumptions for the calculation of noise on hypothetical reference circuits for telephony.*

[8] CCITT Recommendation G.131 (1988), *Stability and echo.*

[9] CCITT Recommendation I.412 (1988), *ISDN user-network interfaces – Interface structures and access capabilities.*

[10] ITU-T Recommendation O.133 (1993), *Equipment for measuring the performance of PCM encoders and decoders.*

[11] ITU-T I.430-Series of Recommendations (1993), *Basic user-network interface – Layer 1 Specification.*

[12] ITU-T Recommendation P.64 (1993), *Determination of sensitivity/frequency characteristics of local telephone systems.*

[13] ITU-T Recommendation P.79 (1993), *Calculation of loudness ratings for telephone sets.*

[14] CCITT P-series Recommendations, Supplement 19 (1988), *Information on some loudness loss related ratings.*

[15] CCITT Recommendation O.131 (1988), *Quantizing distortion measuring equipment using a pseudo-random noise test signal.*

[16] ITU-T Recommendation O.41 (1994), *Psophometer for use on telephone-type circuits.*

[17] ISO 1996-1: 1982, *Acoustics – Description and measurement of environmental noise – Part 1: Basic quantities and procedures.*

[18] ITU-T Recommendation P.57 (1993), *Artificial ears.*

[19] ITU-T Recommendation P.51 (1993), *Artificial mouth.*

[20] ITU, *Handbook on Telephonometry* (1993).

[21] ISO 3: 1973, *Preferred numbers – Series of preferred numbers.*

[22] Delayed Contribution D.72, *Calculation of the signal-to-total noise ratio S/D (PCM G.711, 64 kbit/s, A-law.* FRG, Study Group 12, 4-15 September 1993.

[23] ITU-T *Recommendation G.122 (1993), Influence of national systems on stability and talker echo in international connections.*

3 Definitions and abbreviations

For the purposes of this Recommendation, the following definition applies.

3.1 acoustic Reference Level (ARL): Defined as the acoustic level at MRP which results in a –10 dBm0 output at the digital interface.

Relevant abbreviations in Recommendation P.10 [4] will apply.

A/D	Analogue-to-Digital
DTS	Digital Test Sequence
D/A	Digital-to-Analogue
ERP	Ear Reference Point
ETSI	European Telecommunications Standards Institute
ISDN	Integrated Services Digital Network
LRGP	Loudness Rating Guard-ring Position
LSTR	Listener Sidetone Rating
MRP	Mouth Reference Point
PABX	Private Automatic Branch Exchange
PCM	Pulse Code Modulation
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating

STMR Sidetone Masking Rating

S_{JE} Receiving Sensitivity (Real Ear)

S_{je} Receiving Sensitivity (Artificial Ear)

S_{MJ} Sending Sensitivity (Real Mouth)

S_{mj} Sending Sensitivity (Artificial Mouth)

TCL Terminal Coupling Loss

TCLw Weighted Terminal Coupling Loss

4 Sending loudness rating (SLR) and receiving loudness ratings (RLR)

In view of Recommendation G.111 [5], the following nominal values are recommended:

– SLR = 8 dB;

– RLR = 2 dB.

NOTES

1 The recommended values for SLR and RLR do not imply that echo control in the network can always be avoided.

2 The acoustic loss in the telephone set is an important factor in the echo path and will need careful consideration. A volume control in the telephone set will decrease the echo loss by the same amount as the gain is raised.

3 As a short-term objective, nominal values of SLR in the range 5 to 11 dB and nominal values of RLR in the range –1 to 5 dB. For digital telephones connected to a digital PABX (to which analogue telephones may also be connected), values at the lower end of the ranges above might be needed. The reason is to give customers the same receiving level as they are used to having with the analogue telephones. A receiving volume control might be considered.

5 Sidetone Masking Rating (STMR) and Listener Sidetone Rating (LSTR)

In view of the following considerations:

– the optimum STMR for conditions free from echo;

– the masking effect of sidetone on short delay talker echo;

– the difficulties of high ambient noise conditions;

– what subscribers are used to having with present analogue sets,

the following values are recommended:

– normalized values of STMR in the range 10 to 15 dB;

– normalized values of LSTR >15 dB,

(No maximum values for LSTR need to be imposed.)

NOTE – Normalized values of STMR and LSTR are those obtained by adjusting the measured values for a particular telephone set by the amount its SLR and RLR values deviate from the nominal value. For example, if the SLR is +3 dB relative to its nominal value and the RLR is –1 dB relative to its nominal value, 2 dB should be subtracted from the measured STMR and LSTR values to normalize them.

6 Sending and receiving sensitivity frequency characteristics for digital telephones

In view of the following considerations:

– the compatibility with analogue telephones in a mixed analogue digital network;

– the absence of line-length-dependent frequency distortion to be compensated for as with analogue telephones;

– the aim to achieve the best possible overall quality with the digital telephone, sending and receiving sensitivity/frequency characteristics as specified below are recommended:

– a substantially flat receiving frequency response S_{JE} between 300 Hz and 3400 Hz should be chosen;

– a nominal sending frequency response S_{MJ} within the area indicated in Figure 1;

– below 200 Hz, the send slope should fall by at least 6 dB/octave.

NOTES

1 S_{JE} and S_{MJ} (real ear and mouth) are normally estimated from measurements of S_{Je} and S_{mJ} (artificial ear and mouth) according to Annex B.

2 An expansion of the lower frequency range to 200 Hz will increase the naturalness of the speech.

3 The normal considerations for anti-aliasing filters must be applied to the frequency responses.

4 Marked peaks in the responses might cause stability problems and should therefore be avoided.

5 The preferred curves for S_{JE} and S_{MJ} defined in this way should be considered as a design objective. Individual microphone and receiver curves will, for several reasons, deviate more or less from the "ideal" curves. However, it is hardly possible to specify in a Recommendation concerning desirable frequency characteristics how much and in which way individual response curves may deviate from the target curve, without becoming unacceptable. For type approval of telephone sets, it is generally necessary to specify limits for the shape of sending and receiving frequency curves nationally, in the same way as tolerance limits for loudness ratings are usually specified. These limits are based on technical considerations as well as on cost of implementation, manufacturing tolerances and other economic factors.

6 For telephones using low acoustic impedance devices, Type 3 artificial ears [18] should be used. In this case the values in this Recommendation do not apply.

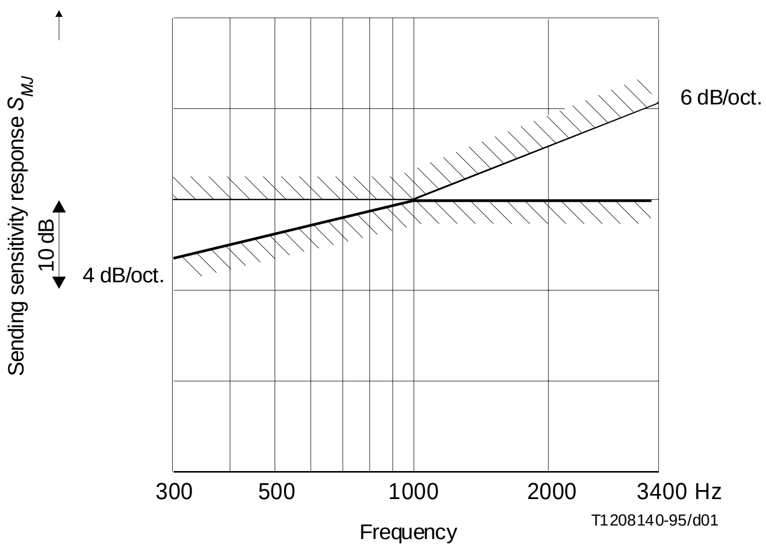

FIGURE 1/P.310

Nominal sending frequency response range

7 Noise characteristics at sending and receiving

In view of the following considerations:

– the compatibility with coder and decoder requirement according to Recommendation G.712 [6];

– certain additions of noise must be allowed for in the electrical and acoustical parts (see Annex C);

– the compatibility with existing analogue telephones, the following limits are recommended:

- sending noise level maximum –64 dBm0p;
- receiving noise level maximum –56 dBPa(A) if no user-controlled volume control is provided or when the volume control is set to nominal RLR value when driven by a PCM signal corresponding to the decoder output value No. 1 for A-law and 0 for μ-law.

NOTE – The noise levels are related to the long-term objective for SLR and RLR.

8 Distortion characteristics at sending and receiving

In view of the following considerations:

- compatibility with coder and decoder requirement according to Recommendation G.712 [6];
- certain additions of distortion must be allowed for in the electrical and acoustical parts (see Annex C);
- compatibility with existing analogue telephones,

the following limits are recommended:

Two different sets of values are recommended relating to two different measuring methods (see Recommendation G.712 [6]). Either is acceptable.

NOTE – ETSI have found it desirable to use both the noise method (Method 1) and the sinewave method (Method 2) for the following reasons:

- The “Sinewave” method (nominally 1 kHz) is effective for the measurement of the coding distortion and overload distortion.
- The “Noise” method being more speech-like and of lower frequency content is more likely to indicate imperfections, including inter-modulation distortion, in the transducers as well as the coding.

8.1 Method 1 (Noise method)

The “Noise” method is used routinely for A-law codecs.

8.1.1 Sending

The ratio of signal-to-total distortion (harmonic and quantizing) power of the digitally encoded signal output by the terminal equipment shall be above the limits given in Tables 1 and 2 for Recommendations G.711 [1] (64 kbit/s) and G.726 [2] (32 kbit/s) respectively, unless the sound pressure at the MRP exceeds +5 dBPa.

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) – linear (dB ratio) scale.

8.1.2 Receiving

The ratio of signal-to-total distortion (harmonic and quantizing) power of the signal in the artificial ear [18] shall be above the limits given in Tables 1 and 2 for Recommendations G.711 [1] (64 kbit/s) and G.726 [2] (32 kbit/s) respectively, unless the signal in the artificial ear exceeds +5 dBPa or is less than –50 dBPa.

8.2 Method 2 (Sinewave method)

8.2.1 Sending

The ratio of signal-to-total distortion power measured with the proper noise weighting (see Recommendation G.223 [7]) shall be above the limits given in Tables 3, 4 and 5 for Recommendations G.711 [1] (64 kbit/s), G.711 (56 kbit/s) and G.726 [2] (32 kbit/s) respectively, unless the sound pressure at MRP exceeds +10 dBPa.

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) – linear (dB ratio) scale.

TABLE 1/P.310

A-law limits for signal-to-total distortion ratio
(Recommendation G.711, 64 kbit/s) for method 1

Sending level dB relative to ARL	Receiving level at the digital interface (dBm0)	Sending ratio (dB)	Receiving ratio (dB)
–45	–55	5.0	5.0
–30	–40	20.0	20.0
–24	–34	25.5	25.0
–17	–27	30.2	30.6
–10	–20	32.4	33.0
0	–10	33.0	33.7
+4	–6	33.0	33.8
+7	–3	23.5	24.0

TABLE 2/P.310

A-law limits for signal-to-total distortion ration
(Recommendation G.726, 32 kbit/s) for method 1

Sending level dB relative to ARL	Receiving level at the digital interface (dBm0)	Sending ratio (dB)	Receiving ratio (dB)
–45	–55	5.0	5.0
–30	–40	20.0	20.0
–24	–34	25.3	24.8
–17	–27	29.7	30.1
–10	–20	31.6	32.3
0	–10	32.1	32.9
+4	–6	32.1	32.9
+7	–3	22.9	23.4

8.2.2 Receiving

The ratio of signal-to-total distortion power measured in the artificial ear with the proper noise weighting (see Recommendation G.223 [7]) shall be above the limits given in Tables 3, 4 and 5 for Recommendations G.711 [1] (64 kbit/s), G.711 (56 kbit/s) and G.726 [2] (32 kbit/s) respectively, unless the signal in the artificial ear exceeds +10 dBPa or is less than –50 dBPa.

TABLE 3/P.310

Limits for signal-to-total distortion ratio (Recommendation G.711, 64 kbit/s) for method 2

Sending level (dB relative to ARL)	Receiving level at the digital interface (dBm0)	Sending ratio (dB)	Receiving ratio (dB)
–35	–45	17.5	17.5
–30	–40	22.5	22.5
–20	–30	30.7	30.5
–10	–20	33.3	33.0
0	–10	33.7	33.5
+7	–3	31.7	31.2
+10	0	25.5	25.5

TABLE 4/P.310

Limits for signal-to-total distortion ratio (Recommendation G.711, 56 kbit/s) for method 2

Sending level (dB relative to ARL)	Receiving level at the digital interface (dBm0)	Sending ratio (dB)	Receiving ratio (dB)
–35	–45	15.3	15.3
–30	–40	20.3	20.3
–20	–30	27.5	27.4
–10	–20	28.5	28.4
0	–10	28.6	28.6
+7	–3	27.9	27.7
+10	0	24.2	24.2

TABLE 5/P.310

Limits for signal-to-total distortion ratio (Recommendation G.726, 32 kbit/s) for method 2

Sending level (dB relative to ARL)	Receiving level at the digital interface (dBm0)	Sending ratio (dB)	Receiving ratio (dB)
–35	–45	17,3	17,3
–30	–40	22,3	22,3
–20	–30	29,3	29,2
–10	–20	31,1	30,9
0	–10	31,3	31,2
+7	–3	30,0	29,7
+10	0	25,0	25,0

9 Out-of-band signals

In view of the following considerations:

- the compatibility with coder and decoder requirements according to Recommendation G.712 [6];

- compatibility with existing practice in the mixed analogue – digital network in use today,

the following limits are recommended.

9.1 Sending

With any sinewave signal above 4.6 kHz and up to 8 kHz applied at the MRP at a level of –4.7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (–4.7 dBPa at MRP) by at least the amount (in dB) specified in Table 6.

TABLE 6/P.310

Discrimination levels – Sending

Applied sinewave frequency	Limit (minimum)[a)]
4.6 kHz	30 dB
8.0 kHz	40 dB

a) The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.

9.2 Receiving

With a digitally-simulated sinewave signal in the frequency range of 300 Hz to 3400 Hz and at a level of 0 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4.6 kHz to 8 kHz measured selectively in the artificial ear [18] shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in Table 7.

TABLE 7/P.310

Discrimination levels – Receiving

Image signal frequency	Equivalent input signal level[a)]
4.6 kHz	–35 dBm0
8.0 kHz	–50 dBm0

a) The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.

10 Weighted Terminal Coupling Loss (TCLw)

In view of the following considerations:

- the aim to achieve as high an acoustic coupling loss as possible to minimize degradation caused by echo;
- what is practically obtainable in real use where the customer himself chooses the way to hold his handset,

the following limit is provisionally recommended:

The weighted Terminal Coupling Loss (TCLw) should be greater than 40 dB when measured under free field conditions and with SLR + RLR normalized to OLR = +10 dB.

However, in order to meet the G.131 [8] talker echo objective requirements, a weighted terminal coupling loss greater than 45 dB is desirable and should be striven for.

NOTE – For practical reasons it may be necessary, for handsets fitted with a volume control, for the TCLw to be not less than 35 dB for the higher gain settings above the nominal setting of the volume control.

11 Stability loss

In view of the following considerations:

- the aim to achieve a good stability;
- what is practically obtainable with normal type of handsets and transducers,

the following limit is recommended:

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be preferably at least 10 dB but not less than 6 dB at all frequencies in the range of 200 Hz to 4 kHz with SLR + RLR normalized to OLR = +10 dB.

NOTE – Those handsets that are fitted with a volume control should maintain stability throughout the volume control range.

12 Delay

In view of the audio group delay introduced by:

- the group delay introduced by the coding, decoding and filtering according to Recommendations G.712 [6] and G.726 [2], as appropriate;
- the group delay introduced by the airpaths and transducers involved,

the following is recommended:

The sum of the group delays from the mouth reference point to the digital interface and from the digital interface to the ear reference point shall not exceed 2.0 ms for digital telephones using Recommendation G.711 encoding and 2.75 ms for G.726 encoding.

13 Input versus output (amplitude) characteristics

Non-linear techniques may be used, e.g. automatic volume control or compressor/expander techniques. These devices may deliberately be non-linear over the input level range specified and may have dynamic characteristics (e.g. attack and hang over time).

At present there are no ITU-T recommended characteristics or verification test methods for such devices in digital telephones (under study). Unless a digital telephone has specifically designed non-linear characteristics, it is desirable to meet the variation of gain characteristics given in Annex A.

Annex A

Variation of gain with input level

(This annex forms an integral part of this Recommendation)

A.1 Sending direction

For digital telephones that are intended to have linear input versus output characteristics the gain variation relative to the gain for ARL should remain within the limits given in Table A.1. For intermediate levels, the same limits for gain variation apply.

NOTE – In cases where the sound pressure exceeds +6 dBPa the linearity of the artificial mouth should be checked, as it exceeds the P.51 limits [19]. For good performance, in this case, it is recommended to use a suitable individual pre-calibration of the artificial mouth for compensation of the deviation of the measured data by taking into account the calibration results.

TABLE A.1/P.310

Variaton of gain with input level – sending

Sending dB relative to ARL	Upper limit (dB)	Lower limit (dB)
13	0.5	–0.5
0	0.5	–0.5
–30	0.5	–0.5
–30	1	–∞
–40	1	–∞
<–40	2	–∞

A.2 Receiving direction

For digital telephones that are intended to have linear input versus output characteristics the gain variation relative to the gain at an input level of –10 dBm0, should be within the limits given in Table A.2. For intermediate levels, the same limits for gain variation apply.

TABLE A.2/P.310

Variation of gain with input level – receiving

Receiving level at the digital interface (dBm0)	Upper limit (dB)	Lower limit (dB)
+3	0.5	–0.5
–10	0.5	–0.5
–40	0.5	–0.5
–40	1	–1
–50	1	–1
<–50	2	–2

Annex B

Objective measurement methods for testing

(This annex forms an integral part of this Recommendation)

B.1 Introduction

The ITU-T recommends the following method to evaluate the voice transmission performance of a digital telephone set using "Waveform" encoding conforming to Recommendations G.711 [1] (PCM at 64 kbit/s and 56 kbit/s) and G.726 [2] (ADPCM, 32 kbit/s). 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.

B.2 Approaches for testing digital telephones

In general there are two approaches for evaluating the transmission performance of a digital telephone, the direct approach and the codec approach. The direct approach is in principle the most accurate although the use of the codec approach may sometimes be advantageous.

B.2.1 Direct digital processing approach

In this approach, shown in Figure B.1, the companded digital input/output bit-stream of the telephone set is operated upon directly. The advantage is that most of the test signals, if sampled at 8 kHz, can be generated and analysed without the need for resampling and A/D or D/A conversion.

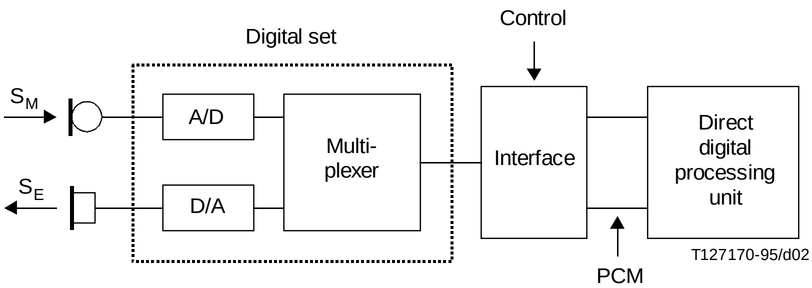

FIGURE B.1/P.310

Digital telephone test arrangement
(direct digital processing approach)

B.2.2 Codec approach

In this approach, shown in Figure B.2, 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 B.5).

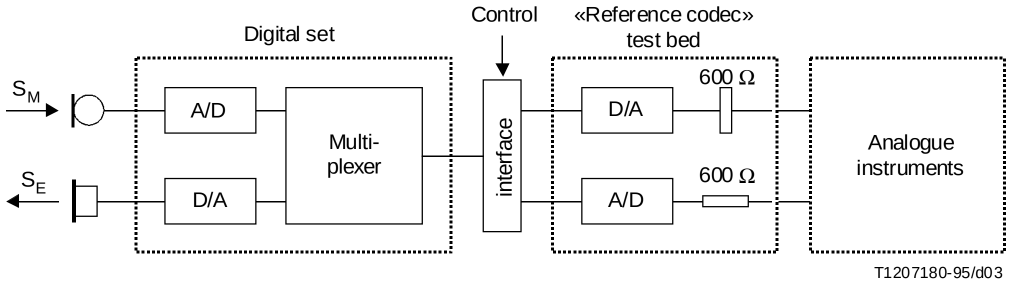

FIGURE B.2/P.310

Digital telephone test arrangement (codec approach)

B.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.

  Where DTS is defined as a periodic sequence of character signals as given in Recommendation G.711 [1].

- *A/D converter* – A 0 dBm signal generated from a 600 ohm source will give the digital test sequence 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.

B.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 [9] 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.

B.5 Codec specification

B.5.1 Ideal codec

The ideal codec consists of an independent encoder and decoder whose characteristics are hypothetical and comply with Recommendation G.711 [1]. 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)].

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

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.

NOTE – For codecs conforming to Recommendation G.726, a G.711/G.726 conversion will be applied.

B.5.2 Reference codec

A practical implementation of an ideal codec may be called a reference codec (see Recommendation O.133 [10]).

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.712 [6], 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 B.3.

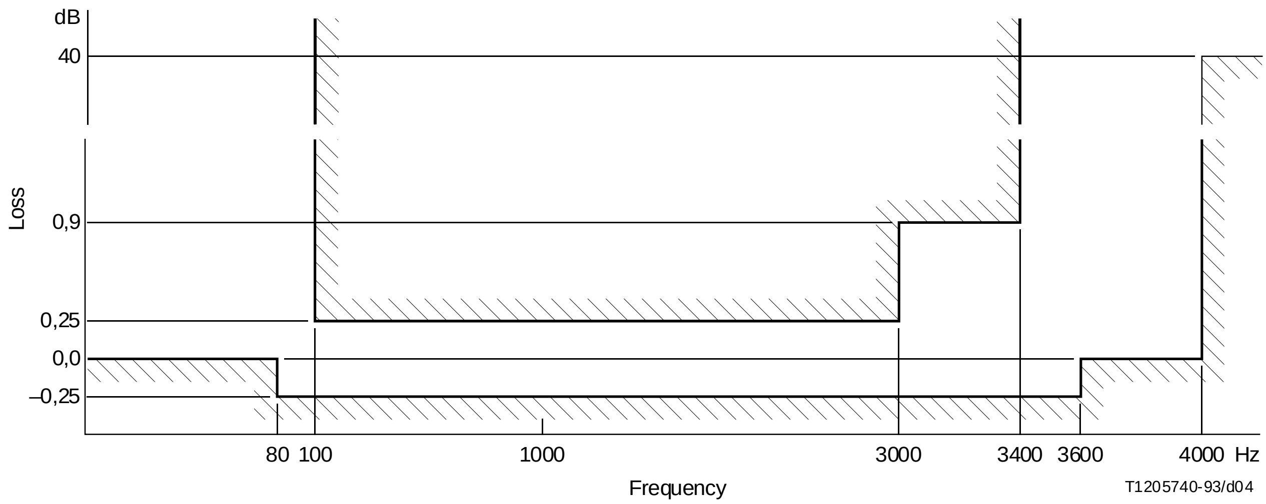

FIGURE B.3/P.310

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

B.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 Recommendation O.133 [10].

B.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 [11]).

B.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 [12]). The reference codec should meet the requirements of B.5. An important difference, however, concerns the test circuits themselves, see Figures B.4 to B.7.

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.

Unless stated otherwise the test signal level shall be –4.7 dBPa for the sending direction and –15.8 dBm0 for the receiving direction.

Handsets fitted with a volume control on receiving shall be set as close as possible to the nominal and any residential difference from the nominal value will be corrected by the normalization process.

B.6.1 Sending

B.6.1.1 Sending frequency characteristic

The sending frequency characteristic is measured according to Recommendation P.64 [12] using the measurement setup shown in Figure B.4.

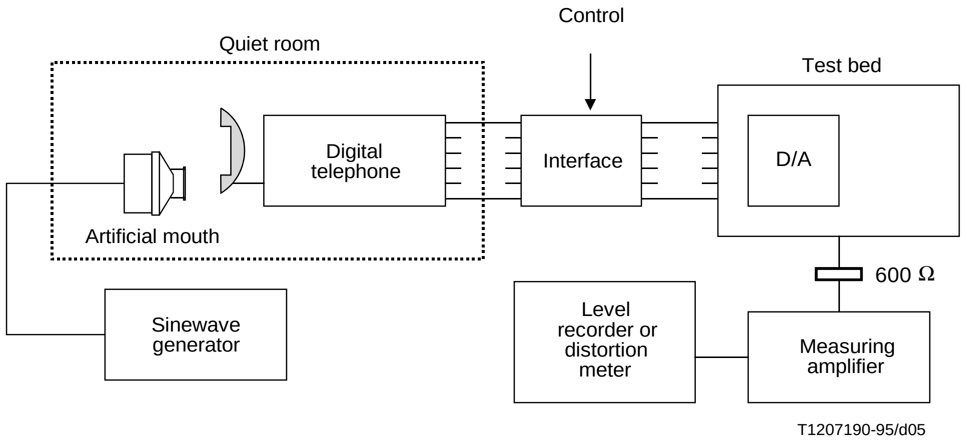

FIGURE B.4/P.310

Measurements of sending frequency characteristic

B.6.1.2 Sending loudness rating

This should be calculated from the sensitivity/frequency characteristic determined in B.6.1.1 by means of Recommendation P.79 [13].

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 P-Series Recommendations [14] and the *Handbook on Telephonometry* [20].

B.6.1.3 Distortion

NOTE – In cases where the sound pressure exceeds +6 dBPa the linearity of the artificial mouth should be checked as it exceeds the P.51 limits. For good performance, in this case, it is recommended to use a suitable individual pre-calibration of the artificial mouth for compensation of the deviation of the measured data by taking into account the calibration results.

Method 1 – Noise

The input at MRP is a band-limited noise signal corresponding to Recommendation O.131 [15]. 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 [15]).

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. 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 O.41 [16].

B.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 O.41 [16].

NOTE – The ambient noise criterion will be met if the ambient noise does not exceed NR20 [17].

B.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 [18].

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 9.1, the level of any image frequencies at the digital interface is measured.

B.6.2 Receiving

B.6.2.1 Receiving frequency characteristic

The receiving frequency characteristic is measured according to Recommendation P.64 [12] using the measurement setup shown in Figure B.5.

B.6.2.2 Receiving loudness rating

This should be calculated from the sensitivity/frequency characteristic determined in B.6.2.1 by means of Recommendation P.79 [13].

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 P-Series Recommendations [14] and the *Handbook on Telephonometry* [20].

B.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 [18]. A digitally simulated band-limited noise signal corresponding to Recommendation O.131 [15] 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 [18] (see Recommendation O.131 [15]).

NOTE 1 – In cases where the sound pressure exceeds +6 dBPa the linearity of the artificial mouth should be checked, as it exceeds the P.51 limits [19].

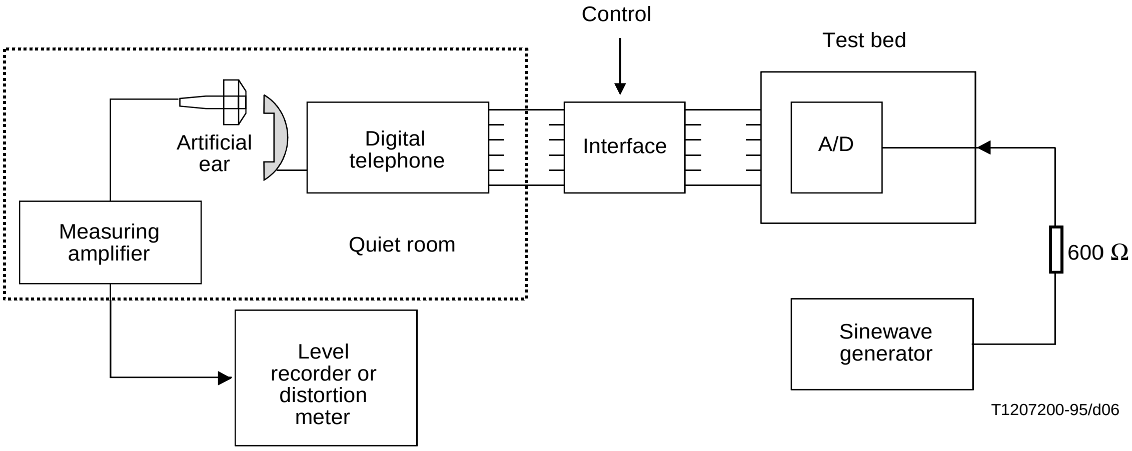

FIGURE B.5/P.310

Measurement of receiving frequency characteristic

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.

The ratio of the signal-to-total distortion power is measured in the artificial ear [18] with A-weighting applied.

NOTE 2 – In cases where the sound pressure exceeds +6 dBPa the linearity of the artificial mouth should be checked, as it exceeds the P.51 limits [19].

B.6.2.4 Noise

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear [18]. 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.

B.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 [18]. For input signals at the frequencies 500, 1000, 2000 and 3150 Hz applied at the level specified in 9.2, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively in the artificial ear.

B.6.3 Sidetone

Provision should be made for driving the microphone of the telephone set under test as described in B.6.1 and measuring the receiver output as described in B.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 [19] and the artificial ear [18] located relative to each other in accordance with Recommendation P.64 [12].

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

B.6.3.1 Sidetone frequency characteristic

B.6.3.1.1 Talker sidetone frequency characteristic

The talker sidetone frequency characteristic is measured according to Recommendation P.64 [12] using the measurement setup of Figure B.6. The reference codec is not used in this measurement but may remain in the test circuit, with no external coupling path.

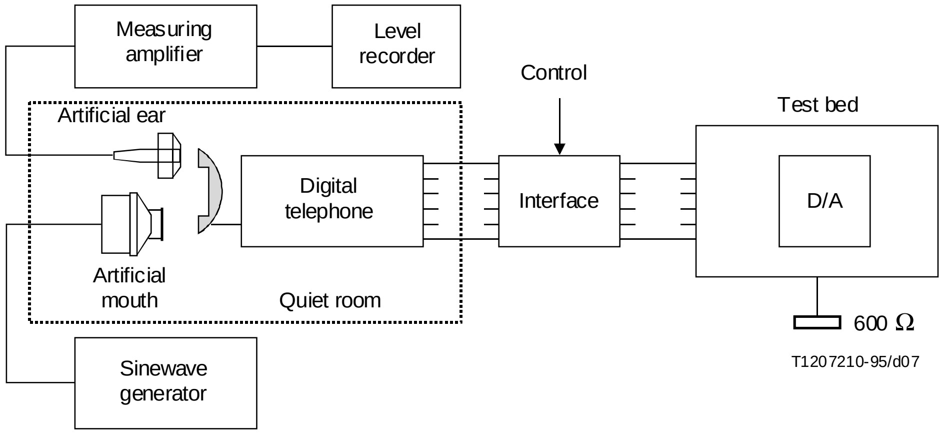

FIGURE B.6/P.310

Measurement of talker sidetone frequency characteristic

B.6.3.1.2 Listener sidetone frequency characteristic

The listener sidetone frequency characteristic is measured according to Recommendation P.64 [12] using the measurement set-up of Figure B.7. The diffuse sound field shall be band-limited (50 Hz to 10 kHz) pink noise within ± 3 dB with a level of –24 dBPa(A) ± 1 dB. The reference codec is not used in this measurement but may remain in the test circuit, with no external coupling path.

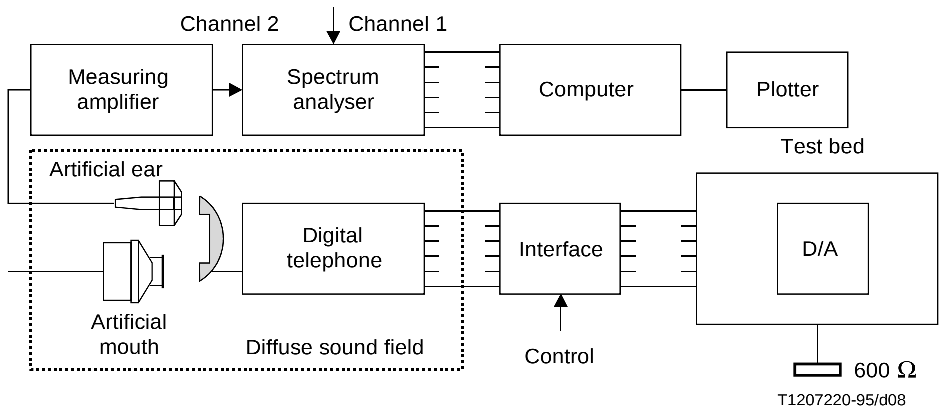

FIGURE B.7/P.310

Measurement of listener sidetone frequency characteristic

B.6.3.2 Sidetone masking rating

This should be calculated from the sensitivity/frequency characteristic determined in B.6.3.1.2 by means of Recommendation P.79 [13].

B.6.3.3 Listener sidetone rating

This should be calculated from the sensitivity/frequency characteristic determined in B.6.3.1.2 by means of Recommendation P.79 [13].

B.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 on Telephonometry* [20].

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 B.8).

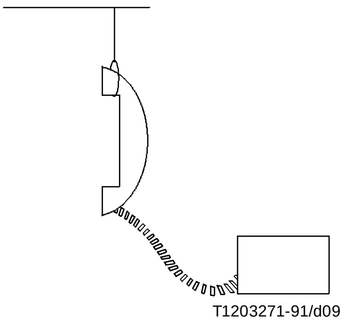

FIGURE B.8/P.310

Handset test position

The attenuation from digital input to digital output is measured using a pure tone with a level of 0 dBm0 at one twelfth octave frequencies as given by the R.40-series of preferred numbers in ISO 3 [21] for frequencies from 300 to 3350 Hz, using the measurement arrangements shown in Figure B.9. The ambient noise level shall be less than 30 dBA.

The TCLw is calculated according to B.4/G.122 [23] (traperoidal rule).

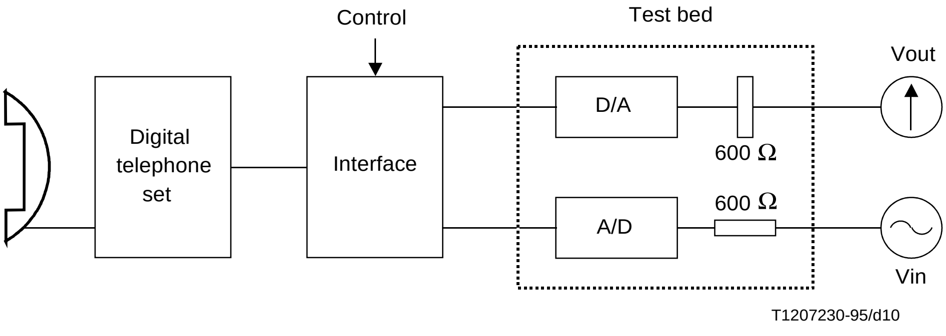

FIGURE B.9/P.310

Measurement of terminal coupling loss

B.6.5 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 B.10.

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 B.10.

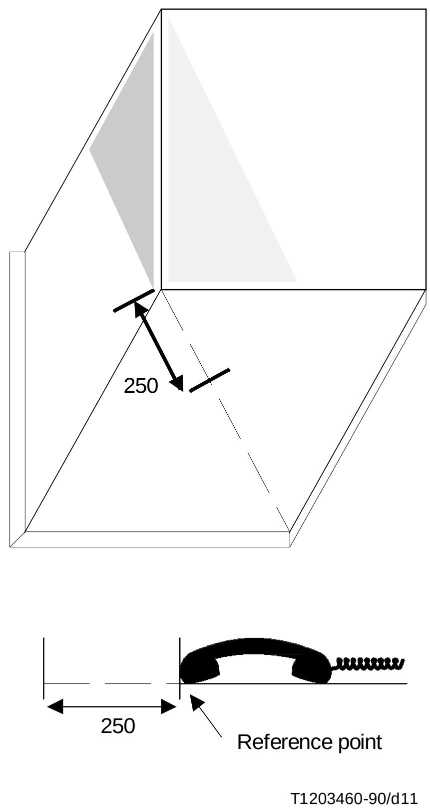

FIGURE B.10/P.310

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.

B.6.6 Delay

The handset is mounted at LRGP. The earpiece is sealed to the knife-edge of the artificial ear. The delay in the sending and receiving directions shall be measured separately from MRP to the digital interface (D_s) and from the digital interface to ERP (D_r).

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

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

The measurement configuration is shown in Figure B.11.

TABLE B.1/P.310

Frequencies for audio group 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

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

1) output the frequency F_1 from the frequency response analyser;
2) measure the phase shift in degrees between ch1 and ch2 (P_1);
3) output the frequency F_2 from the frequency response analyser;
4) measure the phase shift in degrees between ch1 and ch2 (P_2);
5) compute the audio group delay in milliseconds from the formula:

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

NOTE 1 – 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 audio group delay of the electro-acoustic equipment shall be deducted from the calculated delay. The group delay of all additional test equipment shall be determined.

The delay is calculated from the formula:

$$D = D_s + D_r - D_E$$

where D_E is the delay of the test equipment.

NOTE 2 – The method of direct measurement, where the signal is looped at the digital interface, can be used where the sidetone is quiet.

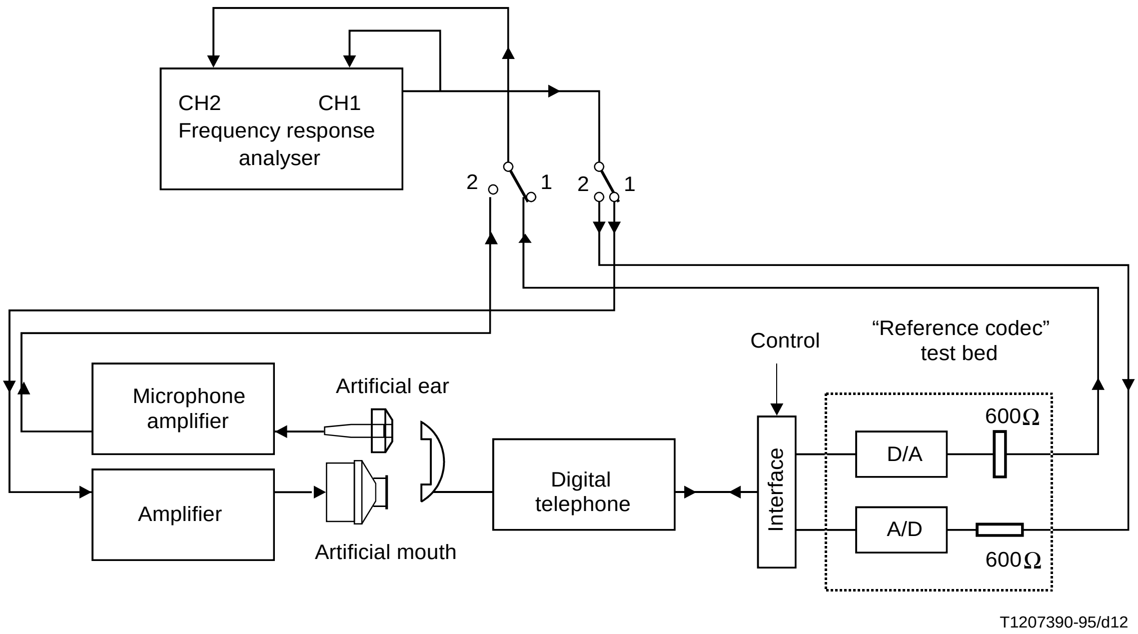

FIGURE B.11/P.310

Measurement of delay

B.6.7 Input versus output (amplitude) characteristics

B.6.7.1 Designed non-linearity

Under study.

B.6.7.2 Linear

B.6.7.2.1 Sending

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear [18].

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.

B.6.7.2.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear [18].

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.

Annex C

Distortion allowances

(This annex forms an integral part of this Recommendation)

In producing the distortion characteristics at sending and receiving (see clause 8) allowances for non-linear products, the following ways have been accounted for:

- The transducers (microphone and earphone) have a 1% distortion allowance for most input levels. The exceptions are the highest and lowest input levels which have been allowed 5% and the second lowest to have 2%.
- Noise levels for sending and receiving are equivalent to –64 dBmp.

The total contribution of these factors is calculated using power summation and information on the calculation process and assumptions used can be found in [22].

NOTES

1 This is of particular interest for developing specifications for other types of "Waveform" codecs not covered in this Recommendation.

2 It may be prudent to allow 0.2 – 0.4 dB to the final calculation to account for other sources of non-linearity, e.g. artificial mouth, amplifiers.

3 Room noise at ≤ 30 dBA has no significant effect.