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SERIES P: TELEPHONE TRANSMISSION QUALITY,
TELEPHONE INSTALLATIONS, LOCAL LINE
NETWORKS

Subscribers' lines and sets

**Transmission characteristics for telephone
band (300-3400 Hz) digital telephones**

ITU-T Recommendation P.310

ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

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ITU-T Recommendation P.310

Transmission characteristics for telephone band (300-3400 Hz) digital telephones

Summary

This Recommendation provides audio performance requirements and associated testing methods 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 ITU-T Recs G.711 (64 kbit/s, PCM) and G.726 (32 kbit/s, ADPCM).

Changes over the previous ITU-T Rec. P.310 (05/2000) are as follows:

- The weighted Terminal Coupling Loss (TCLw) requirement of digital handsets is increased from 40 dB to 45 dB to meet the G.131 requirement.
- A new test method is described as a suitable means to measure the increased TCLw limit.
- The Sidetone Masking Rating (STMR) requirement is modified. The STMR range is increased from 10 dB to 15 dB, to a range of 10 dB to 20 dB. This range now applies to all volume conditions.

These changes improve the speech quality of these digital handsets by reducing the occurrence of talker echo at the far-end, and improper sidetone levels at the near-end.

Source

ITU-T Recommendation P.310 was revised by ITU-T Study Group 12 (2001-2004) and approved under the WTSA Resolution 1 procedure on 16 March 2003.

FOREWORD

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ITU-T Recommendation P.310

Transmission characteristics for telephone band (300-3400 Hz) digital telephones

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 ITU-T Recs 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 ITU-T Rec. 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 References

The following ITU-T 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; 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. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

- [1] ITU-T Recommendation G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.
- [2] ITU-T Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*.
- [3] ITU-T Recommendation G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction*.
- [4] ITU-T Recommendation P.10 (1998), *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] ITU-T Recommendation G.712 (2001), *Transmission performance characteristics of pulse code modulation channels*.
- [7] ITU-T Recommendation G.223 (1988), *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*.
- [8] ITU-T Recommendation G.131 (1996), *Control of talker echo*.
- [9] ITU-T 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 Recommendation I.430 (1995), *Basic user-network interface – Layer 1 specification.*
- [12] ITU-T Recommendation P.64 (1999), *Determination of sensitivity/frequency characteristics of local telephone systems.*
- [13] ITU-T Recommendation P.79 (1999), *Calculation of loudness ratings for telephone sets.*
- [14] ITU-T Recommendation O.131 (1988), *Quantizing distortion measuring equipment using a pseudo-random noise test signal.*
- [15] ITU-T Recommendation O.41 (1994), *Psophometer for use on telephone-type circuits.*
- [16] ISO 1996-1:1982, *Acoustics – Description and measurement of environmental noise – Part 1: Basic quantities and procedures.*
- [17] ITU-T Recommendation P.57 (2002), *Artificial ears.*
- [18] ITU-T Recommendation P.51 (1996), *Artificial mouth.*
- [19] ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections.*
- [20] ITU-T Recommendation P.50 (1999), *Artificial voices.*
- [21] ITU-T Recommendation P.501 (2000), *Test signals for use in telephonometry.*
- [22] ITU-T Recommendation P.58 (1996), *Head and torso simulator for telephonometry.*

3 Definitions and abbreviations

3.1 Definitions

This Recommendation defines the following term:

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

3.2 Abbreviations

Relevant abbreviations in ITU-T Rec. P.10 [4] will apply.

This Recommendation uses the following abbreviations:

A/D	Analogue-to-Digital
D/A	Digital-to-Analogue
DTS	Digital Test Sequence
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
S_{je}	Receiving Sensitivity (Artificial Ear)
S_{JE}	Receiving Sensitivity (Real Ear)
SLR	Sending Loudness Rating
S_{mj}	Sending Sensitivity (Artificial Mouth)
S_{MJ}	Sending Sensitivity (Real Mouth)
STMR	SideTone Masking Rating
TCL	Terminal Coupling Loss
TCL _w	Weighted Terminal Coupling Loss

4 Sending Loudness Rating (SLR) and receiving Loudness Ratings (RLR)

In view of ITU-T Rec. G.111 [5], the following nominal values are recommended:

- SLR = 8 dB;
- RLR = 2 dB.

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 value of STMR shall be 10 dB to 20 dB when corrected to nominal values of SLR (8 dB) and RLR (2 dB).

NOTE 1 – For this correction, the formula $STMR = STMR_m - (SLR_m - 8 + RLR_m - 2)$ should be used. Index m indicates measured values.

Where a user-controlled receiving volume control is provided, the STMR shall be 10 dB to 20 dB for all volume control settings.

The value of LSTR shall be not less than 15 dB when corrected to nominal values of SLR (8 dB) and RLR (2 dB).

NOTE 2 – For this correction, the formula $LSTR = LSTR_m - (SLR_m - 8 + RLR_m - 2)$ should be used. Index m indicates measured values.

Where a user-controlled receiving volume control is provided, the LSTR shall be not less than 15 dB at the setting where the RLR is equal to the nominal value.

6 Sending and receiving frequency responses

In view of the following consideration:

- the compatibility with analogue telephones and mobile telephones, the sensitivity masks for sending (as shown in Table 1 and Figure 1) and for receiving (as shown in Table 2 and Figure 2) are recommended.

Table 1/P.310 – Sending

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-12	
200	0	
300	0	-14
1000	0	-8
2000	4	-8
3000	4	-8
3400	4	-11
4000	0	

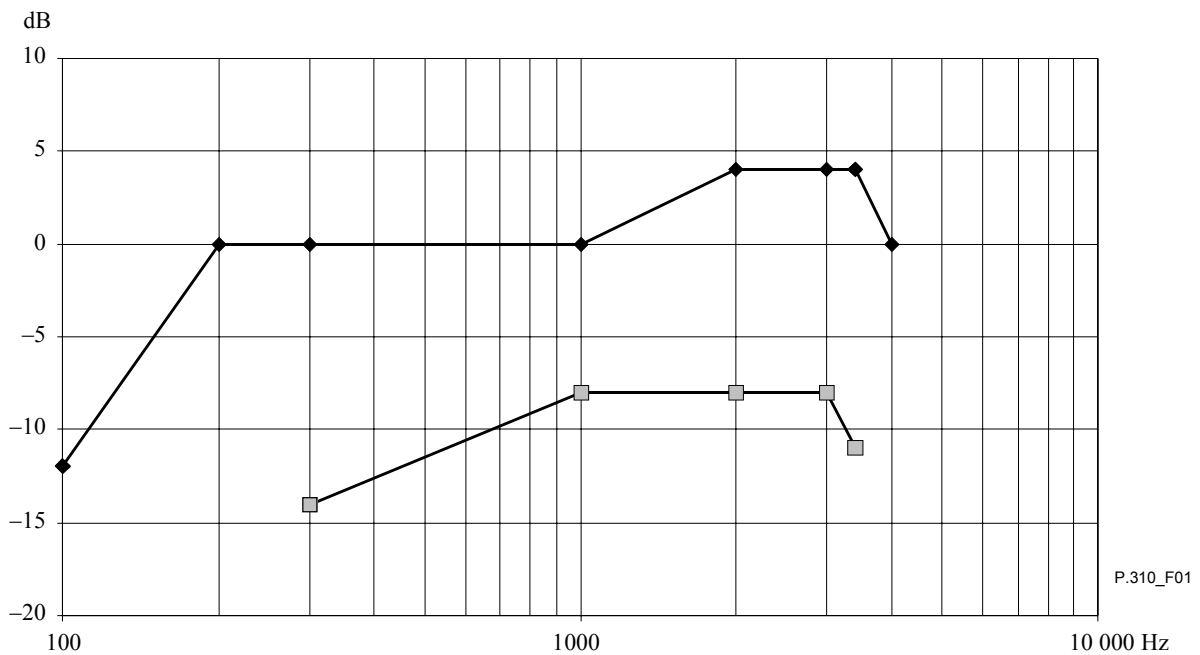


Figure 1/P.310 – Sending mask

Table 2/P.310 – Receiving

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-10	
200	2	
300	2	-9
2000	2	-7
3400	2	-12
4000	2	
8000	-18	

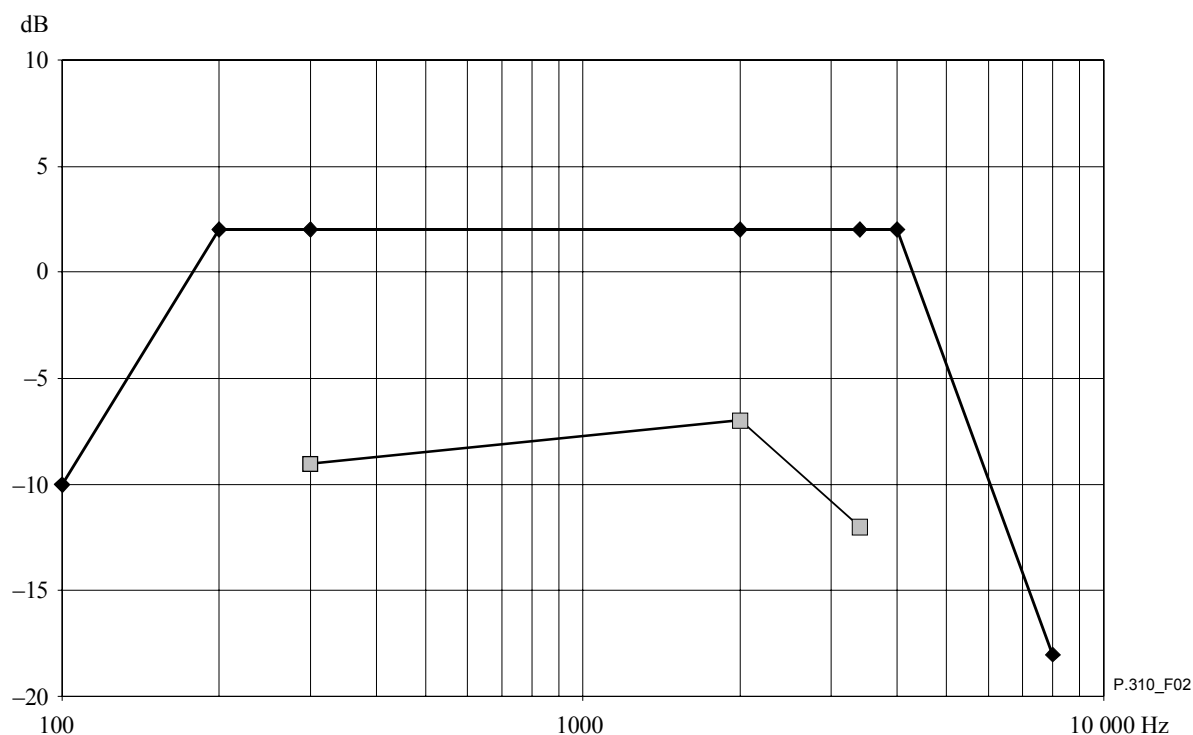


Figure 2/P.310 – Receiving mask

7 Noise characteristics at sending and receiving

In view of the following considerations:

- the compatibility with coder and decoder requirement according to ITU-T Rec. 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 dBm_{0p};
- 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 ITU-T Rec. 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 ITU-T Rec. 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 3 and 4 for ITU-T Recs 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.

**Table 3/P.310 – A-law limits for signal-to-total distortion ratio
(ITU-T Rec. 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 4/P.310 – A-law limits for signal-to-total distortion ratio
(ITU-T Rec. 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

8.1.2 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 3 and 4 for ITU-T Recs 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 ITU-T Rec. G.223 [7]) shall be above the limits given in Tables 5, 6 and 7 for ITU-T Recs 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 5/P.310 – Limits for signal-to-total distortion ratio
(ITU-T Rec. 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 6/P.310 – Limits for signal-to-total distortion ratio
(ITU-T Rec. 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 7/P.310 – Limits for signal-to-total distortion ratio
(ITU-T Rec. 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

8.2.2 Receiving

The ratio of signal-to-total distortion power measured in the artificial ear with the proper noise weighting (see ITU-T Rec. G.223 [7]) shall be above the limits given in Tables 5, 6 and 7 for ITU-T Recs 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.

9 Out-of-band signals

In view of the following considerations:

- the compatibility with coder and decoder requirements according to ITU-T Rec. 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 8.

Table 8/P.310 – Discrimination levels – Sending

Applied sinewave frequency	Limit (minimum) (Note)
4.6 kHz	30 dB
8.0 kHz	40 dB
NOTE – 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 [17] shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in Table 9.

Table 9/P.310 – Discrimination levels – Receiving

Image signal frequency	Equivalent input signal level (Note)
4.6 kHz	-35 dBm0
8.0 kHz	-50 dBm0
NOTE –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;
- that the far end talker echo should be controlled under all volume control settings, and for the range of transducers sensitivities as long as the handset is properly used;
- the far end terminal may be connected via a mobile or a IP network which introduces long talker echo path delay;
- what is practically obtainable in real use where the customer himself chooses the way to hold his handset,

the following limit is recommended:

In order to meet the G.131 [8] talker echo objective requirements, the weighted Terminal Coupling Loss (TCLw) should be greater than 45 dB when measured under free field conditions and with SLR normalized to SLR = +8 dB and RLR = +2 dB. For example, if the measured TCLw is 48 dB, the measured SLR is +9 dB and the measured RLR is +3 dB, then the normalized value of $TCLw = 48 \text{ dB} + (8 - 9) \text{ dB} + (2 - 3) \text{ dB} = 46 \text{ dB}$.

For handsets fitted with a volume control, the TCLw shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control. This TCLw value shall be normalized by considering the SLR and RLR values measured with the volume control at its nominal position. Permanently increasing SLR as the RLR is decreased when increasing the receive volume level is not recommended.

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 coding, decoding and filtering according to ITU-T Recs G.712 [6] and G.726 [2], as appropriate;
- 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 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

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 [18]. 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 – Variation 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	$-\infty$
-40	1	$-\infty$
<-40	2	$-\infty$

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****B.1 Introduction**

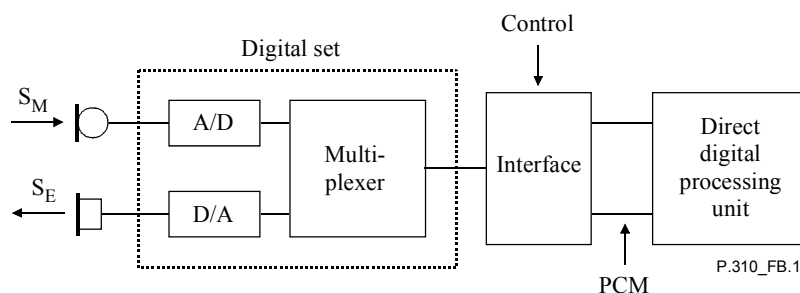
The ITU-T recommends the following method to evaluate the voice transmission performance of a digital telephone set using "Waveform" encoding conforming to ITU-T Recs G.711 [1] (PCM at 64 kbit/s and 56 kbit/s) and G.726 [2] (ADPCM, 32 kbit/s). A digital telephone set is one in which the A/D and D/A converters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

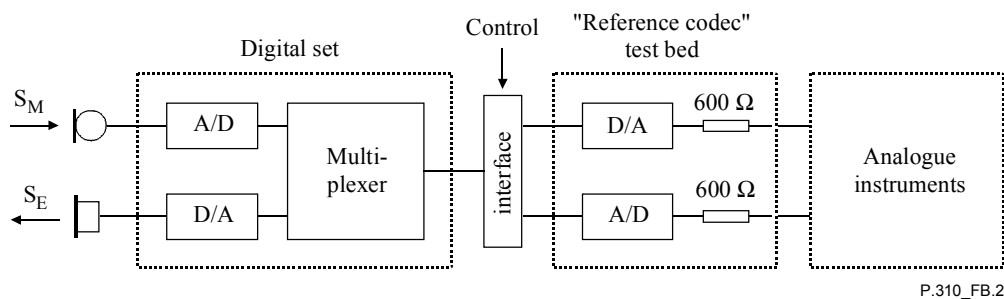
In this approach, shown in Figure B.1, the companded digital input/output bit-stream of the telephone set is operated upon directly. The advantage is that most of the test signals, if sampled at 8 kHz, can be generated and analysed without the need for resampling and A/D or D/A conversion.



**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 dB_r 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 ITU-T Rec. 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 ITU-T Rec. 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 ITU-T Rec. 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 processor¹.

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 ITU-T Rec. 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 ITU-T Rec. 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 ITU-T Rec. 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.

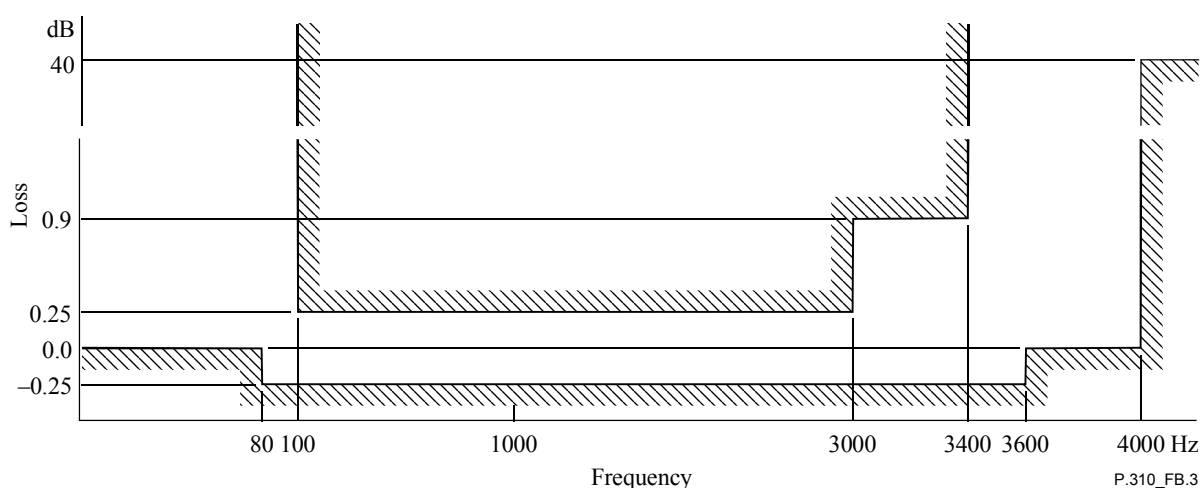


Figure B.3/P.310 – Attenuation/frequency distortion of the sending or receiving sides of the reference codec

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

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 ITU-T Rec. O.133 [10].

B.5.2.2 Digital interface

The fundamental requirements for the reference codec digital interface are given in the appropriate ITU-T Recommendations (e.g., ITU-T Rec. I.430 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 ITU-T Rec. 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 $\pm 2\%$ 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 ITU-T Rec. P.64 [12] using the measurement setup shown in Figure B.4.

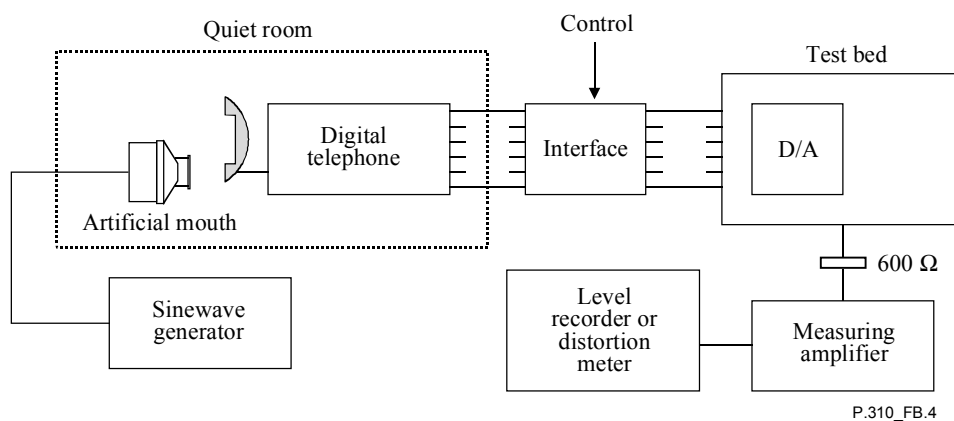


Figure B.4/P.310 – Measurements of sending frequency characteristic

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 ITU-T Rec. P.79 [13].

NOTE – Other methods for calculating loudness ratings used by some Administrations for their own internal planning purposes can be found in the *Handbook on Telephony* (ITU-T, 1992).

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 precalibration 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 ITU-T Rec. O.131 [14]. 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 ITU-T Rec. O.131 [14]).

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 ITU-T Rec. O.41 [15].

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 -64 dBPa (A)), the noise level at the digital output is measured with apparatus including psophometric weighting according to ITU-T Rec. O.41 [15].

NOTE – The ambient noise criterion will be met if the ambient noise does not exceed NR20 [16].

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 [17].

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 ITU-T Rec. P.64 [12] using the measurement setup shown in Figure B.5.

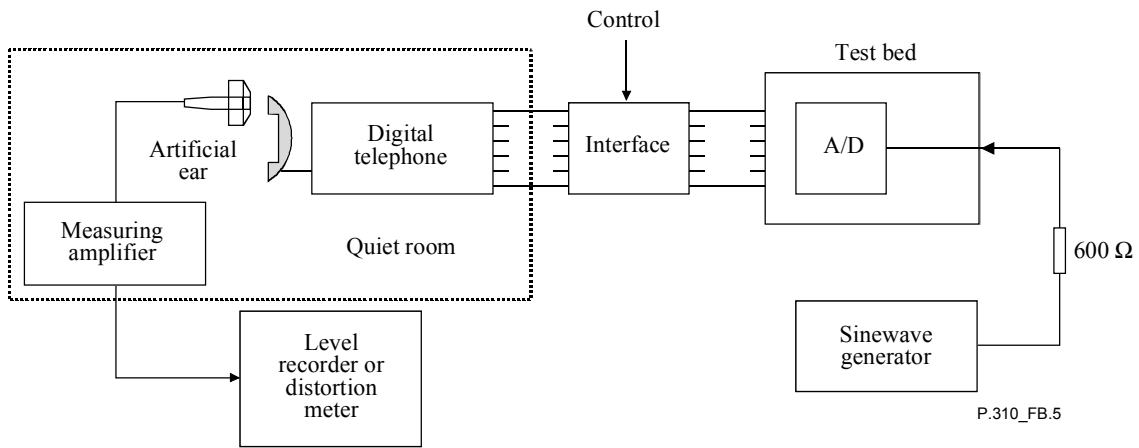


Figure B.5/P.310 – Measurement of receiving frequency characteristic

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 ITU-T Rec. P.79 [13].

NOTE – Other methods for calculating loudness rating used by some Administrations for their own internal planning purposes can be found in the *Handbook on Telephonometry* (ITU-T, 1992).

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 [17]. A digitally simulated band-limited noise signal corresponding to ITU-T Rec. O.131 [14] 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 [17] (see ITU-T Rec. O.131 [14]).

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 [18].

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 [17] 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 [18].

B.6.2.4 Noise

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear [17]. 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 -64 dBPa (A).

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 [17]. 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 [18] and the artificial ear [17] located relative to each other in accordance with ITU-T Rec. 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 ITU-T Rec. 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.

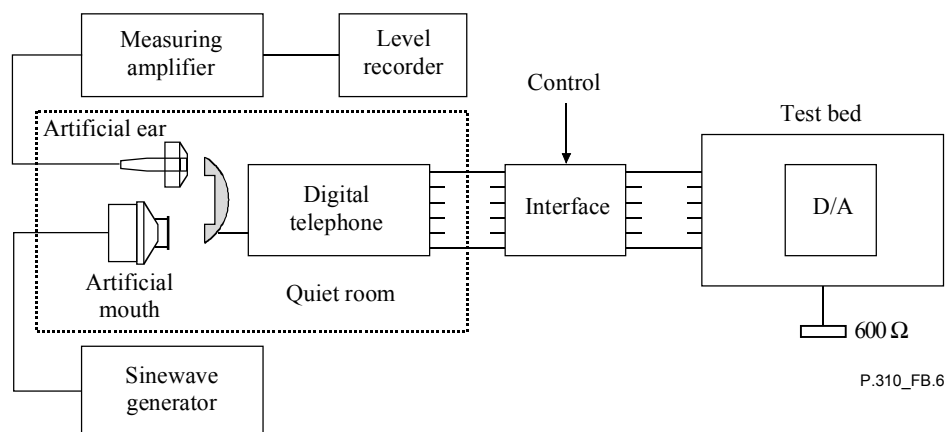


Figure B.6/P.310 – Measurement of talker sidetone frequency characteristic

B.6.3.1.2 Listener sidetone frequency characteristic

The listener sidetone frequency characteristic is measured according to ITU-T Rec. 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.

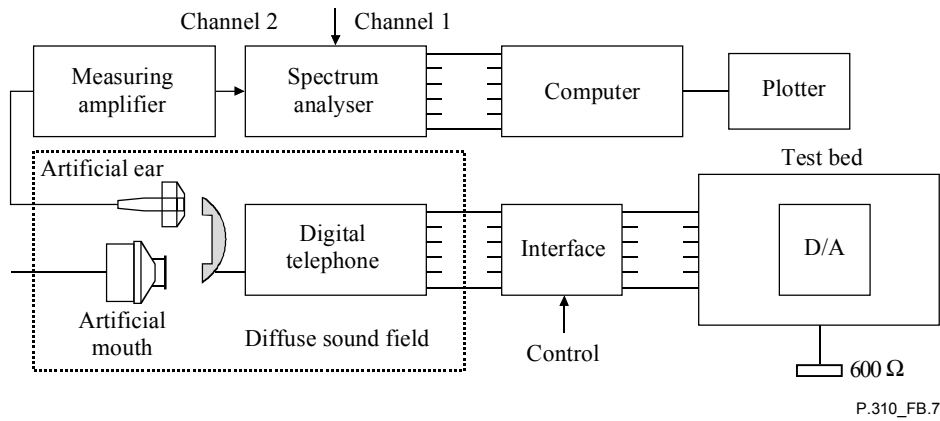


Figure B.7/P.310 – Measurement of listener sidetone frequency characteristic

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 ITU-T Rec. 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 ITU-T Rec. P.79 [13]. In case LSTR measurement is not applicable due to noise problems, LSTR can be estimated by measuring the D-factor.

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 $r \geq 50$ cm.

NOTE – A method of verifying the reverberation distance will be found in the *Handbook on Telephony* (ITU-T, 1992).

The test is performed with the handset suspended in a noose around the earcup with the handset cord hanging freely from below the handset (see Figure B.8).

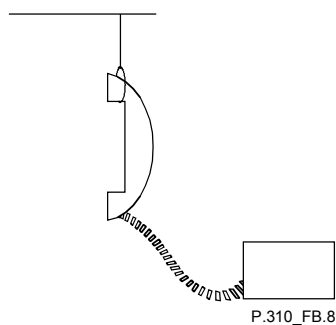


Figure B.8/P.310 – Handset test position

A PN-sequence test signal with a low crest factor or a logarithmically distributed multi sinewave is equally well applicable for TCL measurement with high dynamic range.

For non-linear and/or time variant systems, it must be ensured that the equipment under test is under "steady state conditions". Depending on the task e.g., echo cancellers should be fully converged. This can be achieved by using training sequences e.g., using artificial voice, CS-signals (as described in ITU-T Rec. P.168) or other speech like test signal before inserting the actual test signal.

The PN-sequence test signal is a PN-sequence according to ITU-T Rec. P.501 with a length of 4096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 1 second. The test signal level is -3 dBm0.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[[A + \mu_{AM} \cos(2\pi t \times f_{AM})] \times \cos(2\pi t \times f_{0i}) \right]$$

with:

$$\begin{aligned} A &= 0,5 \\ f_{AM} &= 4 \text{ Hz}, \mu_{AM} = 0.5 \\ f_{0i} &= 250 \text{ Hz} \times 2^{(i/3)}, \quad i = 1..12 \end{aligned}$$

The test signal level should be adjusted to -3 dBm0.

Use the measurement arrangements shown in Figure B.9. The ambient noise level shall be less than -64 dBPa (A).

The TCL_w is calculated according to B.4/G.122 [19] (trapezoidal rule).

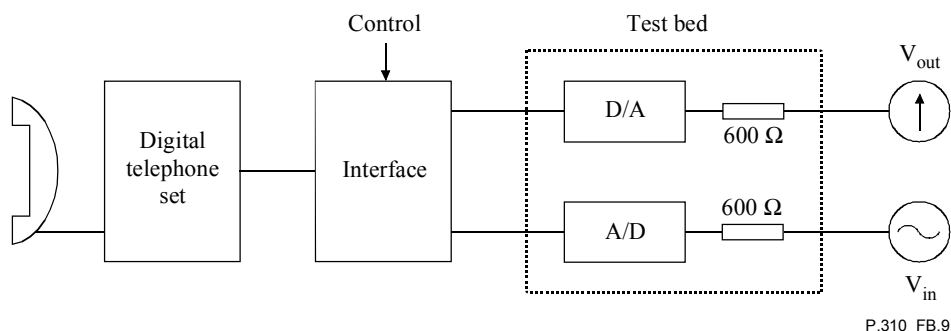


Figure B.9/P.310 – Measurement of terminal coupling loss

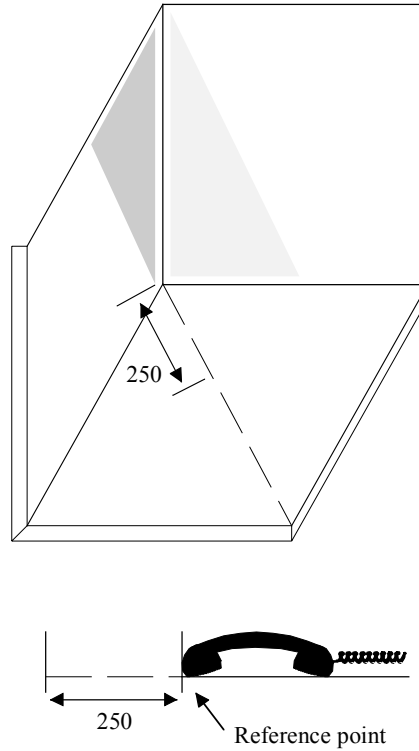
B.6.5 Stability

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.



P.310_FB.10

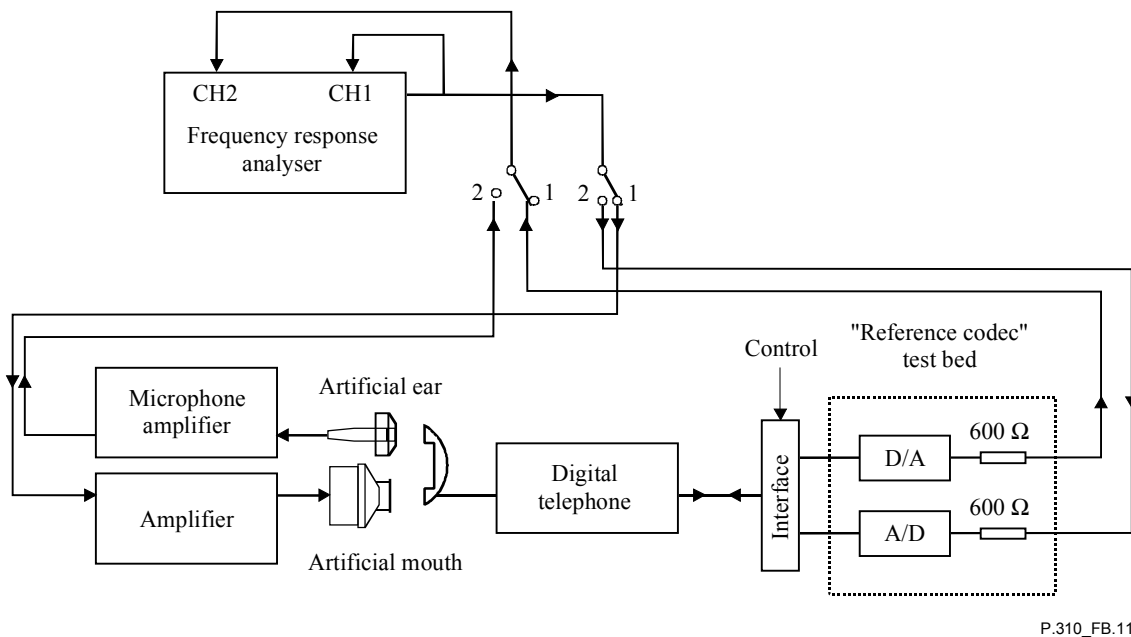
Figure B.10/P.310 – Reference corner

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 test arrangement is shown in Figure B.11.



P.310_FB.11

Figure B.11/P.310 – Measurement of delay

The audio group delay (D) in the sending and receiving direction shall be measured separately from MRP to digital interface (D_s) and from digital interface to measurement microphone (D_r).

Measurements shall be made with pairs of sinewave signals.

The nominal frequencies are 500 Hz, 630 Hz, 800 Hz, 1 kHz, 1.25 kHz, 1.6 kHz, 2 kHz and 2.5 kHz.

The audio group delay is derived from the measurement of the phase shift between the sending signal on channel 1 (CH1) of the measurement equipment and the receiving signal on channel 2 (CH2) of this equipment. For each of the frequencies f_0 the phase shift is measured at the frequencies f_1 and f_2 . f_1 and f_2 yield as follows: $f_1 = f_0 - 50$ Hz and $f_2 = f_0 + 50$ Hz.

NOTE 1 – If the phase shift of f_2 and f_1 is greater than 180 degrees, then the frequency step should be reduced (e.g., 10 Hz).

The measurements are executed in the following steps:

- 1) output the sinewave test signal with the frequency f_1 on CH1;
- 2) measure the phase shift in degrees between CH1 and CH2 (p_1);
- 3) output the sinewave test signal with the frequency f_2 on CH1;
- 4) measure the phase shift in degrees between CH1 and CH2 (p_2);
- 5) compute the audio group delay in milliseconds from the following formula:

$$D(f_0) = \frac{-1000 \times (p_2 - p_1)}{360 \times (f_2 - f_1)}$$

Negative values on p_1 and p_2 achieved at steps 2 and 4 correspond to a lagging of CH2 relative to CH1. Care shall be taken that no errors occur when the phase shift p passes 0° , or a multiple of 360° .

Finally, the average D of all values $D(f_0)$ for the different frequencies f_0 is calculated.

The audio group delay introduced by the artificial mouth shall be measured by mounting a microphone at the MRP. The audio group delay of all additional test equipment between the interface provided for the connection to a digital network and the digital input (CH2), respectively output (CH1), of the test equipment shall be determined. The values of these audio group delays are needed for the correction of the measurement results.

The audio group delay of the item under test is deducted from the formula:

$$D = D_s + D_r = D_{sm} + D_{rm} - D_e$$

where:

D_e is the audio group delay of the test equipment;

D_{sm} is the measurement audio group delay in the sending direction;

D_{rm} is the measurement audio group delay in the receiving direction.

NOTE 2 – A new methodology for testing delay is under study.

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 [17].

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 [17].

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

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. Information on the calculation process and assumptions used can be found in [I.2].

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

NOTE 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.

NOTE 3 – Room noise at ≤ -64 dBPa (A) has no significant effect.

Annex D

Test equipment requirements

D.1 Electro-acoustic equipment

The artificial ear used should conform to ITU-T Rec. P.57 [17]. The artificial mouth should either conform to ITU-T Rec. P.51 [18] or, in case where HATS is used for the measurements, it should conform to ITU-T Rec. P.58 [22].

It is recognized that for most handset designs, the applicable artificial ear is type 1 (P.57). However, when type 1 artificial ear is not applicable, it is advisable to check the terminal performance with other types of artificial ears specified in ITU-T Rec. P.57 [17] such as types 3.2, 3.3 or 3.4.

When using a type 1 or 3.2 artificial ear, the handset is mounted in the LRGP position, as described in ITU-T Rec. P.64 [12].

When type 3.3 or 3.4 artificial ear is used, the handset is mounted on the HATS as described in Annex D/P.64 or Annex E/P.64.

The sound pressure measurements shall be referred to the Ear Reference Point (ERP) by the correction characteristic specified in ITU-T Rec. P.57.

When type 3.2, type 3.3 or type 3.4 artificial ear is used, no leakage correction shall be made in the calculations of RLR and STMR (i.e., $LL_e = 0$).

D.2 Test signals

In general, the test signals as described in this Recommendation should be applied. The use of the proposed test signals requires a linear and time invariant operation of the equipment under test. This cannot be ensured in all cases. For devices where the transmission properties are level and signal-dependent, alternative test signals should be chosen. In this case a more speech-like test signal such as described in ITU-T Recs P.50 [20] and P.501 [21] should be applied. The use of the alternative

test signals should be stated in the test report. Test house and manufacturer should ensure that the appropriate type of test signal is chosen.

D.3 Accuracy of test equipment

Unless specified otherwise, the accuracy of the measurements made by the test equipment shall be better than given in Table D.1.

Table D.1/P.310

Quantity	Accuracy
Electrical Signal Power	± 0.2 dB for levels ≥ -50 dBm ± 0.4 dB for levels < -50 dBm
Sound pressure	± 0.7 dB
Time	$\pm 5\%$
Frequency	$\pm 0.2\%$
Sound pressure level at MRP	± 3 dB for 100 Hz to 200 Hz ± 1 dB for 200 Hz to 4 Hz ± 3 dB for 4 kHz to 8 kHz
Electrical excitation levels	± 0.4 dB (see Note 1)
Frequency generation	$\pm 0.2\%$ (see Note 2)
NOTE 1 – Across the whole frequency range.	
NOTE 2 – When measuring sampled systems, it is advisable to avoid measuring at sub-multiples frequency. There is a tolerance of $\pm 2\%$ on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2% tolerance may be used.	

Appendix I

Bibliography

- [I.1] Handbook on Telephony, ITU-T, 1992.
- [I.2] Study Group 12, Delayed Contribution D.72 (1993): *Calculation of the signal-to-total noise ratio S/D (PCM G.711, 64 kbit/s, A-law)*; Federal Republic of Germany.

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