International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

P.310
(06/2009)

SERIES P: TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS

Voice terminal characteristics

Transmission characteristics for narrow-band digital handset and headset telephones

Recommendation ITU-T P.310

ITU-T

ITU-T P-SERIES RECOMMENDATIONS

TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS

Vocabulary and effects of transmission parameters on customer opinion of transmission quality	Series	P.10
Voice terminal characteristics	**Series**	**P.30** **P.300**
Reference systems	Series	P.40
Objective measuring apparatus	Series	P.50 P.500
Objective electro-acoustical measurements	Series	P.60
Measurements related to speech loudness	Series	P.70
Methods for objective and subjective assessment of speech quality	Series	P.80 P.800
Audiovisual quality in multimedia services	Series	P.900
Transmission performance and QoS aspects of IP end-points	Series	P.1000
Communications involving vehicles	Series	P.1100

For further details, please refer to the list of ITU-T Recommendations.

Recommendation ITU-T P.310

Transmission characteristics for narrow-band digital handset and headset telephones

Summary

Recommendation ITU-T P.310 provides audio performance requirements and associated testing methods for narrow-band (300-3400 Hz) digital handset and headset telephones.

Requirements and test methods are specified for the major audio transmission parameters including sending and receiving frequency response, loudness rating, noise, distortion, out-of-band signals, linearity, sidetone, echo and delay.

This Recommendation is only applicable to digital telephones using encoding conforming to Recommendations ITU-T G.711 (64 kbit/s, PCM) and ITU-T G.726 (32 kbit/s, ADPCM). IP terminals and wireless headsets are not covered in this recommendation.

Source

Recommendation ITU-T P.310 was approved on 22 June 2009 by ITU-T Study Group 12 (2009-2012) under Recommendation ITU-T A.8 procedures.

FOREWORD

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The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

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In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

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

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CONTENTS

Recommendation ITU-T P.310

Transmission characteristics for narrow-band digital handset and headset telephones

1 Scope

This Recommendation deals with sending and receiving frequency response, loudness rating, noise, distortion, out-of-band signals, linearity, sidetone, echo and delay of narrow-band (300-3400 Hz) digital handset and headset telephones using "Waveform" encoding according to [ITU-T G.711] (PCM at both 64 and 56 kbit/s) and [ITU-T G.726] (ADPCM, 32 kbit/s). IP terminals and wireless headsets are not covered in this Recommendation.

The use of digital telephones using [ITU-T G.728] (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.

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

[ITU-T G.131] Recommendation ITU-T G.131 (2003), *Talker echo and its control*.

[ITU-T G.711] Recommendation ITU-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[ITU-T G.712] Recommendation ITU-T G.712 (2001), *Transmission performance characteristics of pulse code modulation channels*.

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

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

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

[ITU-T I.430] Recommendation ITU-T I.430 (1995), *Basic user-network interface – Layer 1 specification*.

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

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

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

[ITU-T P.10] Recommendation ITU-T P.10/G.100 (2006), *Vocabulary for performance and quality of service*.

[ITU-T P.50] Recommendation ITU-T P.50 (1999), *Artificial voices*.

[ITU-T P.51] Recommendation ITU-T P.51 (1996), *Artificial mouth*.

[ITU-T P.57] Recommendation ITU-T P.57 (2009), *Artificial ears*.

[ITU-T P.58] Recommendation ITU-T P.58 (1996), *Head and torso simulator for telephonometry*.

[ITU-T P.64] Recommendation ITU-T P.64 (2007), *Determination of sensitivity/frequency characteristics of local telephone systems*.

[ITU-T P.79] Recommendation ITU-T P.79 (2007), *Calculation of loudness ratings for telephone sets*.

[ITU-T P.380] Recommendation ITU-T P.380 (2003), *Electro-acoustic measurements on headsets*.

[ITU-T P.501] Recommendation ITU-T P.501 (2007), *Test signals for use in telephonometry*.

[ITU-T P.502] Recommendation ITU-T P.502 (2000), *Objective test methods for speech communication systems using complex test signals*.

[ITU-T P.581] Recommendation ITU-T P.581 (2000), *Use of head and torso simulator (HATS) for hands-free terminal testing*

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

[ISO 1996-1] ISO 1996-1:2003, *Acoustics – Description, measurement and assessment of environmental noise – Part 1: Basic quantities and assessment procedures*.

3 Definitions and abbreviations

3.1 Definitions

This Recommendation defines the following terms:

3.1.1 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.

  Here DTS is defined as a periodic sequence of character signals as given in [ITU-T G.711].

– *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.

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

3.1.3 digital telephone set (DT): 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 bitstream.

3.1.4 HATS position: The HATS (head and torso simulator) position (see Annexes D and E of [ITU-T P.64]) is the correct handset position for measuring sensitivity and frequency response characteristics. The HATS position has been shown to be essentially identical to the LRGP (loudness rating guard-ring position), except for the mouth simulator direction, which has been corrected with a 19° downwards rotation to more closely match real talkers. For handsets with omnidirectional microphones, measurements on the two heads may differ slightly, typically less than 1 dB. For handsets with directional or noise-cancelling microphones, the differences will be larger, and the HATS position will give the more realistic results.

3.2 Abbreviations

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

This Recommendation uses the following abbreviations:

A/D	Analogue-to-Digital
ARL	Acoustic Reference Level
D/A	Digital-to-Analogue
DRP	earDrum Reference Point
DTS	Digital Test Sequence
ERP	Ear Reference Point
ERUP	Estimated Real Use Position
HATS	Head And Torso Simulator
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
RTP	Recommended Test Position
RWP	Recommended Wearing Position
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)
S_{si} (diff)	Sending Sensitivities for diffuse sound
S_{si} (direct)	Sending Sensitivities for direct sound

STMR SideTone Masking Rating

TCL Terminal Coupling Loss

TCLw Weighted Terminal Coupling Loss

Δ_{Sm} Difference between the sending sensitivities for diffuse and direct sound

4 Test configuration

4.1 Measurement 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.

ITU-T recommends both methods to evaluate the voice transmission performance of a digital handset or headset telephone using "Waveform" encoding conforming to [ITU-T G.711] (PCM at 64 kbit/s and 56 kbit/s) and [ITU-T G.726] (ADPCM, 32 kbit/s).

4.1.1 Direct digital processing approach

In this approach, shown in Figure 4-1, the companded digital input/output bitstream 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.

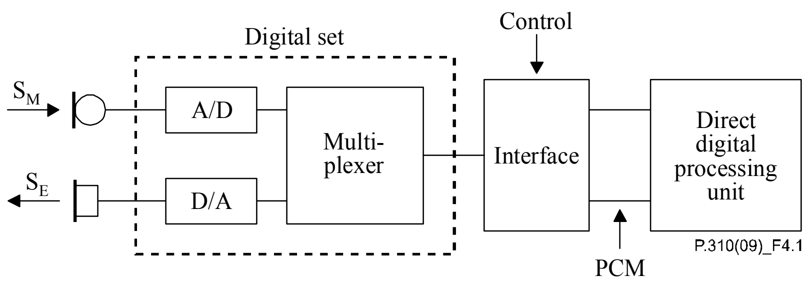

Figure 4-1 – Digital telephone test arrangement (direct digital processing approach)

4.1.2 Codec approach

In this approach, shown in Figure 4-2, a codec is used to convert the companded digital input/output bitstream of the telephone set to the equivalent analogue values, so that existing test procedures and equipment can be used. This codec should be a high-quality codec whose characteristics are as close as possible to ideal (see clause 4.2.2).

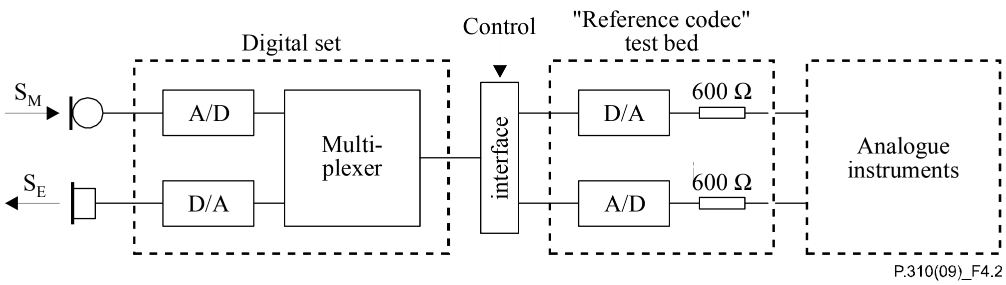

Figure 4-2 – Digital telephone test arrangement (codec approach)

NOTE – 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 I.412] for ISDN sets), to a form compatible with the test equipment. Interfaces can be applied for sending and receiving separately, taking into account telephone sets which are connected to various types of exchanges.

4.2 Test equipment

4.2.1 Electro-acoustic equipment

The preferred acoustical access to terminals is the most realistic simulation of the "average" subscriber. This is made by using HATS (head and torso simulator) with appropriate ear simulator and appropriate means to fix handset or headset in a realistic and reproducible way. HATS, including the artificial mouth and the artificial ear, shall comply with [ITU-T P.58]. The recommended ear simulators are type 3.3 and type 3.4 artificial ear, described in [ITU-T P.57].

The horizontal positioning of the HATS reference plane shall be guaranteed with ±2°. The exact calibration and equalization of HATS can be found in [ITU-T P.581].

4.2.2 Codecs

4.2.2.1 Ideal codec

The ideal codec consists of an independent encoder and decoder whose characteristics are hypothetical and comply with [ITU-T G.711]. The ideal encoder is a perfect analogue-to-digital converter preceded by an ideal low-pass filter (assumed to have no attenuation/frequency distortion and no envelope-delay distortion), and may be simulated by a digital processor. The ideal decoder is a perfect digital-to-analogue converter followed by an ideal low-pass filter (assumed to have no attenuation/frequency distortion and no envelope-delay distortion), and which may be simulated by a digital processor[1].

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

NOTE – For codecs conforming to [ITU-T G.726], a G.711/G.726 conversion will be applied.

4.2.2.2 Reference codec

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

For the reference codec, characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion, etc. should be better than the requirements specified in [ITU-T G.712], 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 4-3.

[1] This characteristic can be realized, for example, using over-sampling techniques and digital filters.

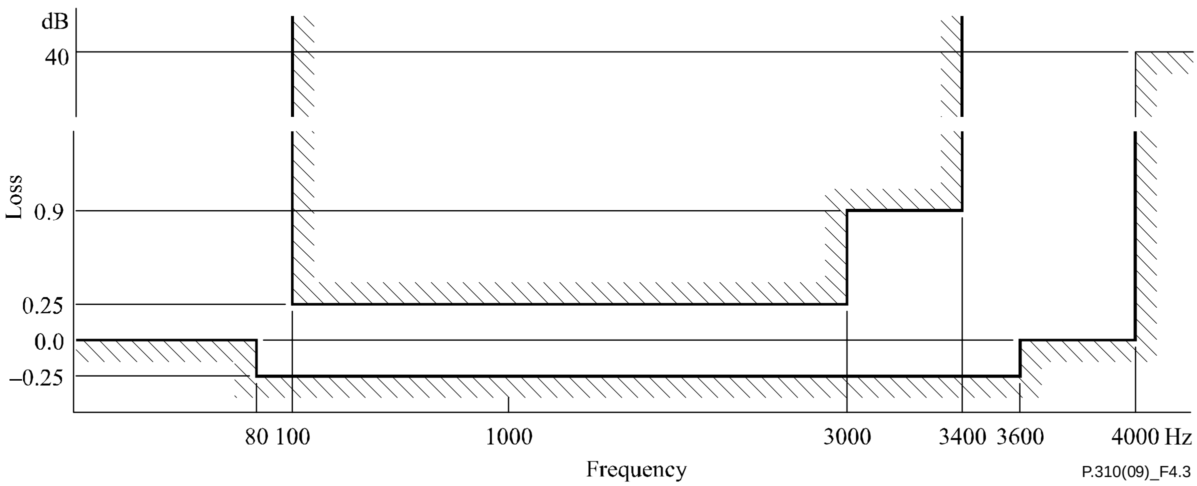

Figure 4-3 – Attenuation/frequency distortion of the sending or receiving sides of the reference codec

4.2.2.2.1 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 O.133].

4.2.2.2.2 Digital interface

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

4.3 Measurement set-up

All measurements are performed using a HATS. The type of ear simulator and application force shall be indicated in a test report. For handsets, if not stated otherwise, 8N application force shall be used to apply the handset against the artificial ear.

Proper positioning of handsets on HATS can be found in Annexes D and E of [ITU-T P.64].

Suggestions for positioning headsets on HATS are given in [ITU-T P.380]. The manufacturer should provide the recommended wearing position (RWP) in the user's guide, which describes a precise way the device should be placed on the user's head. From RWP, the recommended test position (RTP) information can be derived, as close as possible to RWP. Also the RTP description can be provided by the manufacturer which includes the information how the receiving part of the headset should be placed against or inside the ear simulator, and further describes the positioning and orientation of the microphone. If neither RTP nor RWP are provided by the manufacturer, the headset shall be tested in the estimated real use position (ERUP). The test lab shall define an ERUP that closely approximates real use. Natural headband pressure, or other positioning techniques normally used by a real user, shall be used. Because the headset position may affect test results, the test shall be repeated at least 5 times by completely repositioning the headset, following the rules described in [ITU-T P.380].

Use of the codec test approach means that test procedures for digital telephone sets in general follow those for analogue sets (see [ITU-T P.64]).

The reference codec should meet the requirements of clause 4.2.2.2. An important difference, however, concerns the test circuits themselves (see Figures 5-2 and 5-4 to 5-7).

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

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

4.4 Test environment

The test laboratory shall ensure that all measurements performed in the test room give identical results to those obtained in a free-field environment (e.g., an anechoic room).

The environmental conditions for testing handset and headset terminals are specified as follows:

– The test room shall be practically free-field down to a lowest frequency of 275 Hz, the handset or headset coupled with the HATS shall lie totally within this free-field volume. This is met if deviations from the ideal free-field conditions are less than ±1 dB.

– The ambient noise level shall be less than –64 dBPa(A) for all the tests, except for tests requesting the generation of ambient noise, e.g., D-Factor measurement.

– The test environment shall be free of mechanical disturbances.

4.5 Test signal and level

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 P.50] and [ITU-T P.501] should be applied. Test house and manufacturer should ensure that the appropriate type of test signal is chosen.

Unless stated otherwise, the test signal shall be set to –4.7 dBPa at MRP for the sending direction and –16 dBm0 for the receiving direction. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free-field conditions at MRP.

For a correct activation of the system, an artificial voice according to [ITU-T P.50] or a speech like test signal as described in [ITU-T P.501] shall be used for activation. The level of this activation signal shall be –4.7 dBPa at MRP.

The use of test signals and levels should be stated in the test report.

NOTE 1 – The use of sine signals may not be appropriate when speech processing and coding systems are implemented in the terminal. For example, in distortion measurement, if a sine wave is not usable, an alternative test signal could be a band-limited noise signal centred on the test frequencies.

NOTE 2 – It should be ensured that the test signal is treated by speech processing algorithms as a speech-like signal, and not a noise-like signal.

NOTE 3 – 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.

4.6 Accuracy of test equipment

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

Table 1 – Minimum measurement accuracy

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	±5%
Frequency	±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	±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 ±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.	

5 Handset technical requirements

5.1 Sending characteristics

5.1.1 Sending frequency response

In view of the following consideration:

– Unlike other standards, this Recommendation does not use ERP as the reference point for receiving, but uses the diffuse-field instead. With the concept of diffuse-field based receive measurements, a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse-field based receiving frequency response; the sensitivity mask of Table 2 is recommended.

Table 2 – Sending frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	−18	
300	5	−5
3400	5	−5
4000	5	
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (Hz) scale.		

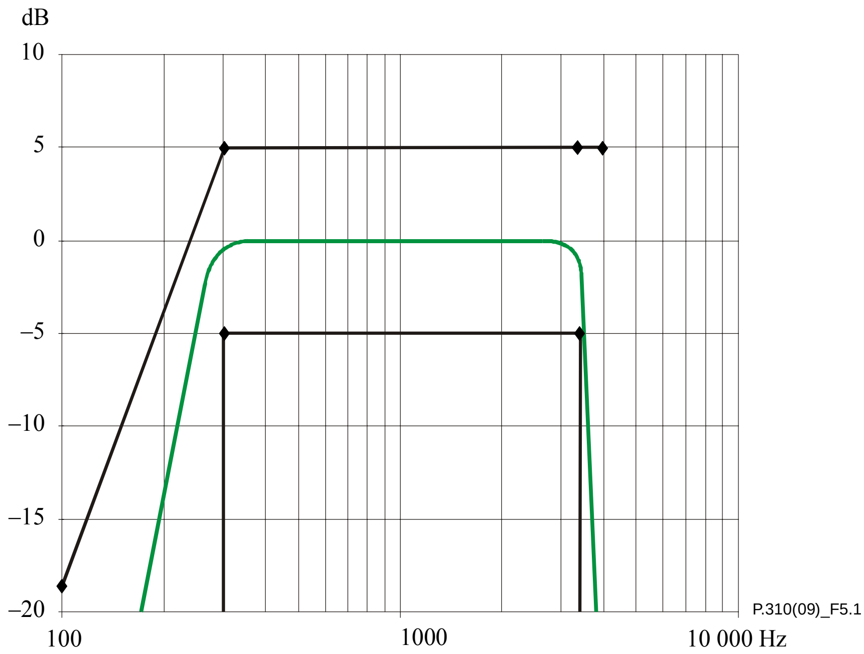

Figure 5-1 – Sending mask

5.1.1.1 Measurement method

The sending frequency characteristic is measured according to [ITU-T P.64] using the measurement set-up shown in Figure 5-1.

A sinewave or an artificial speech (e.g., as specified in [ITU-T P.50] and [ITU-T P.501]) generator and a spectrum analyser should be used. Additional test methods may be found in [ITU-T P.502]. The test signal level shall be –4.7 dBPa at MRP. The test signal level is averaged over the complete test signal sequence. The sensitivity is expressed in terms of dBV/Pa.

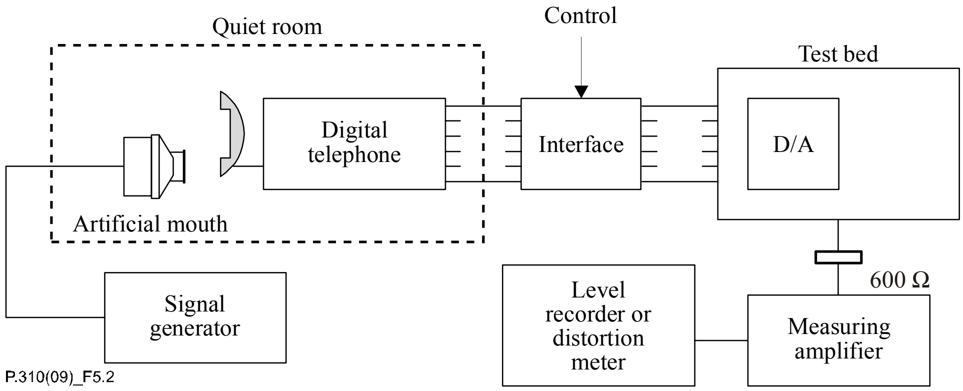

Figure 5-2 – Measurements of sending frequency characteristic, HATS position

5.1.2 Sending loudness rating (SLR)

– The nominal value of SLR shall be ±8 dB, with a tolerance of ±3 dB.

5.1.2.1 Measurement method

This should be calculated from the sensitivity/frequency characteristic determined in clause 5.1.1.1 by means of [ITU-T P.79].

NOTE – Other methods for calculating loudness ratings used by some Administrations for their own internal planning purposes can be found in [b-Teleph].

5.1.3 Sending noise

In view of the following considerations:

– compatibility with coder and decoder requirements according to [ITU-T G.712];

– certain additions of noise must be allowed for in the electrical and acoustical parts (see clause 4.6);

– compatibility with existing analogue telephones,

the following limits are recommended.

– sending noise level: maximum −64 dBm0p;

– no peaks in any 1/3-octave band, with the level of 10 dB higher than the average noise spectrum in the frequency domain, shall occur.

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

5.1.3.1 Measurement method

The sending noise shall be measured in a quiet environment as defined in clause 4.4.

After a correct activation, the noise level at the digital output is measured in the frequency range from 100 Hz to 4000 Hz with apparatus including psophometric weighting described in [ITU-T O.41].

NOTE – The ambient noise criterion will be met if the ambient noise does not exceed NR20 [ISO 1996-1].

5.1.4 Sending distortion

In view of the following considerations:

– compatibility with coder and decoder requirements according to [ITU-T G.712];

– certain additions of distortion must be allowed for in the electrical and acoustical parts (see clause 4.6);

– compatibility with existing analogue telephones,

two different sets of values are recommended relating to two different measuring methods (the "Sinewave" method and the "Noise" method; see [ITU-T G.712]). Either is acceptable.

NOTE – ETSI have found it desirable to use both the noise method and the sinewave method for the following reasons:

• The "Sinewave" method 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.

5.1.4.1 Sending distortion using "Noise" method

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

The ratio of signal-to-total distortion (harmonic and quantizing) power of the digitally encoded signal output by the terminal equipment measured with the proper noise weighting (see [ITU-T O.41]) shall be above the limits given in Tables 3 and 4 for [ITU-T G.711] (64 kbit/s) and [ITU-T G.726] (32 kbit/s), respectively.

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 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 64 kbit/s) for "Noise" method

Sending level dBPa at MRP	Sending ratio (dB)
–45	5.0
–30	20.0
–24	25.5
–17	30.2
–10	32.4
0	33.0
+4	33.0

Table 4 – Limits for signal-to-total distortion ratio ([ITU-T G.726], 32 kbit/s) for "Noise" method

Sending level dBPa at MRP	Sending ratio (dB)
–45	5.0
–30	20.0
–24	25.3
–17	29.7
–10	31.6
0	32.1
+4	32.1

5.1.4.1.1 Measurement method

The input at MRP is a band-limited noise signal corresponding to [ITU-T O.131]. The test signal is then applied at –45, –40, –35, –30, –24, –20, –17, –10, –5, 0, 4 dBPa.

The ratio of the signal-to-total distortion power of the digital signal output with a psophometric noise weighting according to [ITU-T O.41] is measured (see [ITU-T O.131]).

5.1.4.1.2 Sending distortion using "Sinewave" method

The ratio of signal-to-total distortion power measured with the proper noise weighting (see [ITU-T O.41]) shall be above the limits given in Tables 5, 6 and 7 for [ITU-T G.711] (64 kbit/s), [ITU-T G.711] (56 kbit/s) and [ITU-T G.726] (32 kbit/s) respectively. The input sound pressure level is limited at +10 dBPa for this measurement.

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 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 64 kbit/s) for "Sinewave" method

Frequency (Hz)	Sending level (dBPa at MRP)	Sending ratio (dB)
315	–4.7	26
510	–4.7	30.5
1020	–35	17.5
	–30	22.5
	–20	30.7
	–10	33.3
	0	33.7
	+7	31.7
	+10	25.5

Table 6 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 56 kbit/s) for "Sinewave" method

Sending level (dBPa at MRP)	Sending ratio (dB)
–35	15.3
–30	20.3
–20	27.5
–10	28.5
0	28.6
+7	27.9
+10	24.2

Table 7 – Limits for signal-to-total distortion ratio ([ITU-T G.726], 32 kbit/s) for "Sinewave" method

Sending level (dBPa at MRP)	Sending ratio (dB)
–35	17.3
–30	22.3
–20	29.3
–10	31.1
0	31.3
+7	30.0
+10	25.0

5.1.4.2.1 Measurement method

After a correct activation of the system, a sinewave signal at frequencies of 315, 510 and 1020 Hz is applied at MRP respectively. The signal level shall be calibrated to –4.7 dBPa at MRP for all frequencies, except for the sinewave signal with a frequency of 1020 Hz that shall be applied at MRP at the following levels: –35, –30, –25, –20, –15, –10, –5, 0, 7, 10 dBPa. 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 O.41].

NOTE – In cases where the sound pressure exceeds +6 dBPa, the linearity of the artificial mouth should be checked as it exceeds the ITU-T 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.

5.1.5 Sending out-of-band signals

In view of the following considerations:

– compatibility with coder and decoder requirements according to [ITU-T G.712];

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

the following limits are recommended:

With any sinewave signal above 4.6 kHz and up to 8 kHz applied at 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 1020 Hz (–4.7 dBPa at MRP) by at least the amount (in dB) specified in Table 8.

Table 8 – Discrimination levels – Sending out-of-band signals

Applied sinewave frequency (kHz)	Limit (minimum) (dB) (Note)
4.6	30
8.0	40
NOTE – The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.	

5.1.5.1 Measurement method

The reference signal used is an activation signal followed by a sine wave at 1020 Hz. The reference signal is applied at MRP (–4.7 dBPa at MRP). The level is measured at the digital interface.

The actual test signals used are an activation signal followed by a sine wave with the frequencies of 4.65 kHz, 5 kHz, 6 kHz, 6.5 kHz, 7 kHz and 7.5 kHz (–4.7 dBPa at MRP) respectively. The level of any image frequencies at the digital interface is measured.

5.1.6 Sending linearity

Considering that some non-linear techniques may be used in digital telephones, for example:

– automatic gain control;

– compressor/expander techniques,

these devices may be deliberately designed non-linear over the input level range specified, and may have dynamic characteristics (e.g., attack and hangover time).

For digital telephones that are intended to have linear input-versus-output characteristics, the linearity relative to the gain for ARL should remain within the limits given in Table 9. For intermediate levels, the same limits for gain variation apply.

Table 9 – Sending linearity

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	–∞

NOTE 1 – In cases where the sound pressure at MRP exceeds +6 dBPa, the linearity of the artificial mouth should be checked, as it exceeds the ITU-T 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.

NOTE 2 – If a digital telephone has specifically designed non-linear characteristics, alternative test methods based on more complex test signals may be needed for these tests. The principles are described in [ITU-T P.501] and [ITU-T P.502].

5.1.6.1 Measurement method

After a correct activation of the system, a sinewave signal with a frequency in the range 1004 Hz to 1025 Hz is applied at MRP. The level of this signal is adjusted until the output of the terminal is –10 dBm0. The level of the signal at 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 1 – If the sinewave techniques cannot be used (which is likely if the terminal has a noise reduction device), another appropriate technique should be applied. For example, an artificial speech generator (e.g., as specified in [ITU-T P.50] and [ITU-T P.501]) and a spectrum analyser can be used. The test signal used shall be specified in the test report.

NOTE 2 – Selective measurements may be used to avoid the effects of ambient noise.

5.2 Receiving characteristics

5.2.1 Receiving frequency response

In view of the following consideration:

– Unlike other standards, this Recommendation does not use ERP as the reference point for receiving, but uses the diffuse-field instead. With the concept of diffuse-field based receive measurements, a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse-field based receiving frequency response; the sensitivity mask of Table 10 is recommended.

Table 10 – Receiving frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	
300	4	–4
3400	4	–4
4000	4	
NOTE – The limit curves shall be determined by straight lines joining successive coordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. It is a floating or 'best fit' mask.		

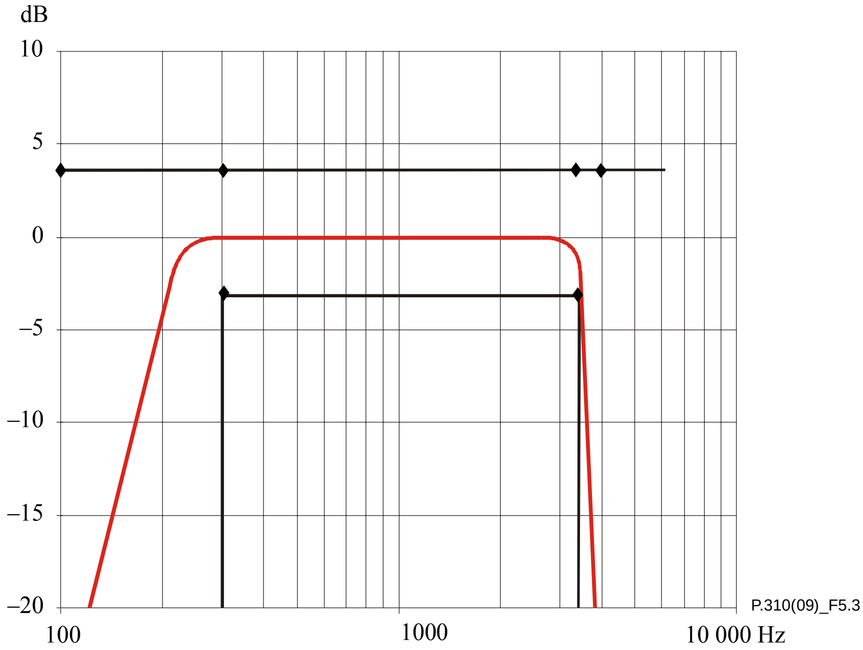

Figure 5-3 – Receiving mask with HATS, diffuse-field equalized

5.2.1.1 Measurement method

The receiving frequency characteristic is measured according to [ITU-T P.64] using the measurement set-up shown in Figure 5-4.

A sinewave or an artificial speech generator (e.g., as specified in [ITU-T P.50] and [ITU-T P.501]), a spectrum analyser and a diffuse-field equalized HATS are used. Additional test methods may be found in [ITU-T P.502]. The test signal level shall be –16.0 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence. The sensitivity is expressed in terms of dBPa/V, referred to the HATS Reference Point (HRP). Information about HATS diffuse-field frequency response is available in [ITU-T P.58].

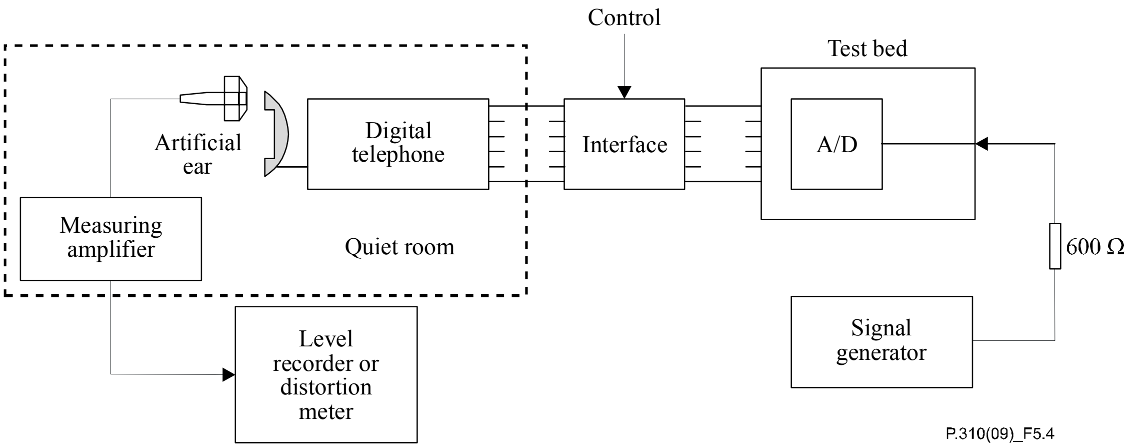

Figure 5-4 – Measurement of receiving frequency characteristic, HATS position

5.2.2 Receiving loudness rating (RLR)

– The nominal value of RLR shall be +2 dB, with a tolerance of ±3 dB

5.2.2.1 Measurement method

This should be calculated from the sensitivity/frequency characteristic determined in clause 5.2.1.1 by means of [ITU-T P.79]. The DRP-ERP correction as defined in [ITU-T P.57] is applied.

NOTE – Other methods for calculating loudness rating used by some Administrations for their own internal planning purposes can be found in [b-Teleph].

5.2.3 Receiving noise

When driven by a signal corresponding to a decoder value quiet code, the maximum acoustic noise level at ERP shall be as follows:

– if no user-controlled volume control is provided or when the volume control is set to nominal RLR value, the measured receiving noise at ERP shall not be greater than −57 dBPa(A);

– if a volume control is provided, the measured receiving noise at ERP shall not be greater than –54 dBPa(A) at the maximum setting of the volume control;

– no peaks in any 1/3-octave band with the level of 10 dB higher than the average noise spectrum in the frequency domain shall occur.

5.2.3.1 Measurement method

The receiving noise shall be measured in a quiet environment as defined in clause 4.4.

The receiving noise level at ERP is measured with A-weighting. The DRP-ERP correction as defined in [ITU-T P.57] is applied.

5.2.4 Receiving distortion

5.2.4.1 Receiving distortion using "Noise" method

The ratio of signal-to-total distortion (harmonic and quantizing) power of the signal measured at ERP with A-weighting shall be above the limits given in Tables 11 and 12 for [ITU-T G.711] (64 kbit/s) and [ITU-T G.726] (32 kbit/s), respectively, unless the signal in the artificial ear exceeds +5 dBPa or is less than −50 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 11 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 64 kbit/s) for "Noise" method

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
–55	5.0
–40	20.0
–34	25.0
–27	30.6
–20	33.0
–10	33.7
–6	33.8
–3	24.0

Table 12 – Limits for signal-to-total distortion ratio ([ITU-T G.726], 32 kbit/s) for "Noise" method

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
–55	5.0
–40	20.0
–34	24.8
–27	30.1
–20	32.3
–10	32.9
–6	32.9
–3	23.4

5.2.4.1.1 Measurement method

A digitally simulated band-limited noise signal corresponding to [ITU-T O.131] is applied to the digital interface at the following levels: –55, –50, –45, –40, –34, –30, –27, –20, –15, –10, –6, –3 dBm0.

The total distortion power shall be measured at ERP with A-weighting. The ratio of the signal-to-total distortion power is calculated (see [ITU-T O.131]). The DRP-ERP correction defined in [ITU-T P.57] is applied.

5.2.4.2 Receiving distortion using the "Sinewave" method

The ratio of signal-to-total distortion power measured at ERP with A-weighting shall be above the limits given in Tables 13, 14 and 15 for [ITU-T G.711] (64 kbit/s), [ITU-T G.711] (56 kbit/s) and [ITU-T G.726] (32 kbit/s) respectively, unless the signal in the artificial ear exceeds +10 dBPa or is less than –50 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 13 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 64 kbit/s) for "Sinewave" method

Frequency (Hz)	Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
315	−16	20
510	−16	30.5
1020	−45	17.5
	−40	22.5
	−30	30.5
	−20	33.0
	−10	33.5
	−3	31.2
	0	25.5

Table 14 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 56 kbit/s) for "Sinewave" method

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
−45	15.3
−40	20.3
−30	27.4
−20	28.4
−10	28.6
−3	27.7
0	24.2

Table 15 – Limits for signal-to-total distortion ratio ([ITU-T G.726], 32 kbit/s) for "Sinewave" method

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
−45	17.3
−40	22.3
−30	29.2
−20	30.9
−10	31.2
−3	29.7
0	25.0

5.2.4.2.1 Measurement method

After a correct activation of the system, a digitally simulated sinewave signal at frequencies of 315, 510 and 1020 Hz is applied to the digital interface respectively. The signal level is –16 dBm0, except for the sinewave signal with a frequency of 1020 Hz that shall be applied to the digital interface at the following levels: –45, –40, –35, –30, –25, –20, –15, –10, –3, 0 dBm0.

The total distortion power shall be measured at ERP with A-weighting. The DRP-ERP correction as defined in [ITU-T P.57] is applied. The ratio of the signal-to-total distortion power is calculated.

5.2.5 Receiving out-of-band signals

In view of the following considerations:

– compatibility with coder and decoder requirements according to [ITU-T G.712];

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

the following limits are recommended:

Any spurious out-of-band image signal above 4.6 kHz and up to 8 kHz measured selectively shall be lower than the in-band acoustic level measured with a reference signal (1000 Hz, sinewave). The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in Table 16:

Table 16 – Discrimination levels – Receiving out-of-band signals

Image signal frequency	Minimum level difference
4.6 kHz	35 dB
8.0 kHz	45 dB
NOTE – The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.	

5.2.5.1 Measurement method

The actual test signals used are a digitally activation signal followed by a digitally simulated sine wave with the frequencies of 500 Hz, 1000 Hz, 2000 Hz and 3150 Hz (–16 dBm0, at the digital reference point or the equivalent analogue point) respectively. The level of spurious out-of-band image signals at frequencies up to 8 kHz are measured selectively at the ear simulator.

The measured spurious out-of-band signals shall be compared to the measured in-band 1000 Hz signal.

NOTE – Depending on the type of codec, the test signal used may need to be adapted.

5.2.6 Receiving linearity

Considering that some non-linear techniques may be used in digital telephones, for example:

– automatic gain control;

– compressor/expander techniques,

these devices may be deliberately designed non-linear over the input level range specified and may have dynamic characteristics (e.g., attack and hangover time). Unless a digital telephone has specifically designed non-linear characteristics, the following limits are recommended.

NOTE – If a digital telephone has specifically designed non-linear characteristics, alternative test methods based on more complex test signals may be needed for these tests. The principles are described in [ITU-T P.501] and [ITU-T P.502].

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 17. For intermediate levels, the same limits for gain variation apply.

Table 17 – Receiving linearity

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

5.2.6.1 Measurement method

After a correct activation of the system, 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. The measured sound pressure level should be transferred from DRP to ERP according to [ITU-T P.57].

NOTE 1 – If the sinewave techniques cannot be used (which is likely if the terminal has a noise reduction device), another appropriate technique should be applied. For example, an artificial speech generator (e.g., as specified in [ITU-T P.50] and [ITU-T P.501]) and a spectrum analyser can be used.

NOTE 2 – Selective measurements may be used to avoid the effects of ambient noise.

5.3 Sidetone characteristics

5.3.1 Sidetone masking rating (STMR)

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

NOTE – For this correction, the formula STMR = STMRm – (SLRm – 8 + RLRm – 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. It is preferable to have a constant STMR independent of the volume control setting.

5.3.1.1 Measurement method

The test signal to be used for the measurements shall be the artificial voice according to [ITU-T P.50]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free-field conditions at MRP. The test signal level shall be –4.7 dBPa at MRP. The test signal level is averaged over the complete test signal sequence.

The talker sidetone sensitivity/frequency characteristic is measured according to [ITU-T P.64] using the measurement set-up of Figure 5-5. The reference codec is not used in this measurement but may remain in the test circuit, with no external coupling path.

Sidetone masking rating (STMR) should be calculated from the sensitivity/frequency characteristic determined from above by means of [ITU-T P.79]. The measured sound pressure should be transferred from DRP to ERP.

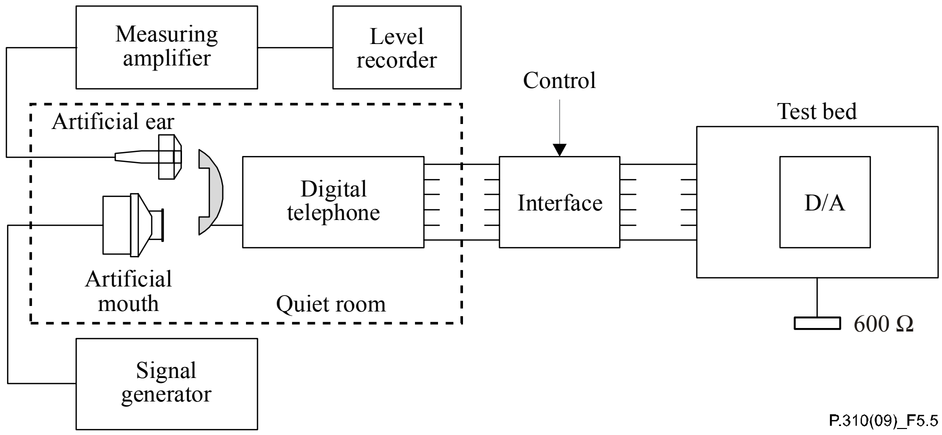

Figure 5-5 – Measurement of talker sidetone sensitivity/frequency characteristic, HATS position

5.3.2 D-factor

In view of the following considerations:

– the difficulties of high ambient noise conditions;

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

the value of D-factor shall not be less than 0 dB. As the long-term objective, the value of +3 dB is recommended.

NOTE 1 – The key parameter for the handset performance in noisy conditions is the D-factor.

NOTE 2 – When a telephone set is connected into the telecom network, there is a firm relation between STMR and LSTR: D = LSTR − STMR, here LSTR is the listener sidetone loudness rating.

5.3.2.1 Measurement method

The measurement method is defined according to Annex E of [ITU-T P.79].

For linear microphones and circuitry, D-factor is computed directly from measurements of the difference Δ_{Sm} between the sending sensitivities for diffuse and direct sound, S_{si} (diff) and S_{si} (direct), respectively.

$$\Delta_{Sm} = S_{si}\text{ (diff)} - S_{si}\text{ (direct)}$$

Both S_{si} (diff) and S_{si} (direct) can be measured according to clause 5.1.1, using the measurement set-up of Figure 5-6 except that diffuse sound is used as the input signal when measuring S_{si} (diff). The sound pressure level of the diffuse sound field shall be adjusted in the range of –54 dBPa(A) to –29 dBPa(A) at MRP in the absence of artificial mouth. The actual level and type of noise should always be stated in quoting test results.

D is computed as a weighted average of $\Delta_{Sm,}$

$$D = -\sum_{i=1}^{N} K_i . \Delta_{Sm}$$

The coefficients K_i are given in Table E.1 of [ITU-T P.79].

NOTE – For sets with non-linear microphones and/or non-linear circuitry, *D* depends on the room noise level.

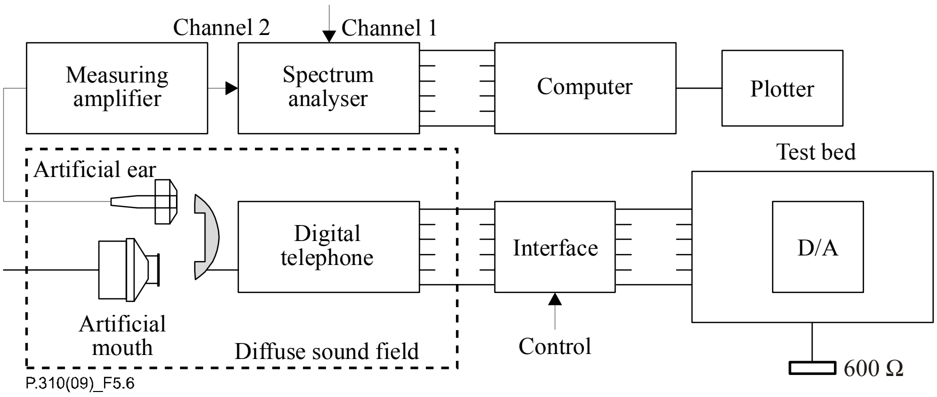

Figure 5-6 – Measurement of listener sidetone sensitivity/frequency characteristic, HATS position

5.4 Echo path loss characteristics

5.4.1 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 an 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 ITU-T G.131 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 dB + (8 – 9) dB + (2 – 3) dB = 46 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. Permanently increasing SLR as the RLR is decreased when increasing the receive volume level is not recommended.

NOTE – In consideration of the increasing delays introduced by modern networks, a higher TCLw value than that specified here may be necessary for proper operation with these networks.

5.4.1.1 Measurement method

Terminal coupling loss (TCL) is measured 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. The test should be performed in the environment as defined in clause 4.4.

The test is performed with the handset mounted at HATS position according to [ITU-T P.64].

The test signal may be a composite source signal (CSS) as defined in [ITU-T P.501] with a length of 4096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The low crest factor is achieved by random alternation of the phase between –180° and 180°. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second. The test signal level is –3 dBm0. The calibration shall be determined during the ON portions of the signal. The test signal shall be band-limited to 100 Hz ~ 4000 Hz. A logarithmically distributed multi-sinewave is equally well applicable with G.711 codecs.

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 invoking the test signal for at least 2 seconds before the actual measurement occurs.

The attenuation from digital input to digital output is measured at one-twelfth octave frequencies as given by the R.40-series of preferred numbers in [ISO 3] for frequencies from 300 to 3350 Hz, using the measurement arrangement shown in Figure 5-7.

The TCLw is calculated according to clause B.4 of [ITU-T G.122] (trapezoidal rule).

NOTE 1 – The echo impairment perceived by the person at the opposite end of the connection from a telephone set is a function of the magnitude of the talker echo signal as well as the talker echo path delay. The echo signal becomes more disturbing as the talker echo path delay increases. Thus, a telephone set with adequate TCLw performance on low delay connections may provide satisfactory performance while the same may not be true for connections that have a long delay.

NOTE 2 – There might be problems measuring 46 dB TCL in the case where sophisticated coding with limited dynamic range is used. In such cases, typically speech or speech-like test signals need to be used that themselves have crest factors in the range of 15 dB, which reduces the measurement dynamic by the same amount. In such cases, the signal measured in the sending direction should be evaluated more carefully in order to find whether an echo signal is present, or whether this signal is completely masked by the noise signal introduced by the codec. If the signal measured in the sending direction is completely masked by the noise, the requirement can be considered to be fulfilled. If this is not the case, more sophisticated measurement procedures such as time averaging (in order to improve the signal-to-noise ratio) need to be applied in order to achieve reliable measurement results.

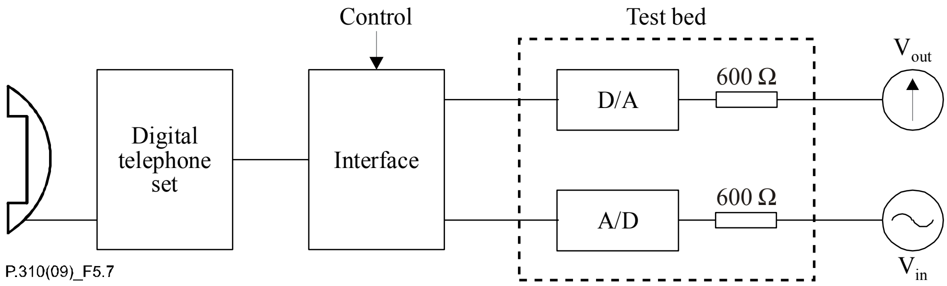

Figure 5-7 – Measurement of terminal coupling loss, HATS position

5.4.2 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.

5.4.2.1 Measurement method

The signal used in stability measurement is the same with TCLw, with a different input signal level of 0 dBm0.

The measurement is made 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. Stability is the least value of the attenuation in the band 200 to 4000 Hz.

Method 1

a) The handset shall be positioned on one inside surface at the corner formed by the intersection of three perpendicular, smooth, hard surfaces. Each surface shall extend 0.5 m from the vertex 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 5-8.

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

   i) the mouthpiece and ear-cap shall face towards the surface;

   ii) the handset shall be placed centrally on the diagonal line with the ear-cap 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 5-8.

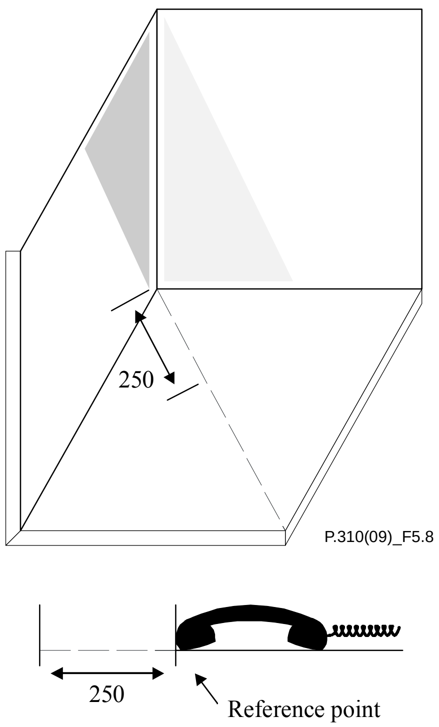

Figure 5-8 – Reference corner and reference position

Method 2

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

5.5 Delay

Delay is a complex end-to-end issue. Certain aspects of delay can be optimized in digital terminals, such as the internal hardware/firmware delay. Other aspects are also important sources of delay, but they are a function of the selected codec, so they cannot be optimized in digital terminals.

Considering the delay, referred to in this Recommendation,

– the coding, decoding and filtering according to [ITU-T G.712] and [ITU-T G.726], as appropriate;

– the air paths and transducers involved,

the following is recommended:

The sum of the delays from MRP to the digital interface (sending direction), and from the digital interface to the artificial ear (receiving direction), shall not exceed 25 ms for digital telephones using G.711 encoding, and 30 ms for G.726 encoding.

5.5.1 Measurement method

5.5.1.1 Sending delay measurement

The test signal to be used for the measurements shall be a composite source signal (CSS) as described in [ITU-T P.501]. The handset terminal is set-up as described in clause 4.3. The delay is calculated using the cross-correlation function between the signal at the electrical test point and the signal at MRP. The measurement is corrected by the delay introduced by the test equipment. The delay is expressed in ms, determined from the maximum of the cross-correlation function.

5.5.1.2 Receiving delay measurement

The test signal to be used for the measurements shall be a composite source signal (CSS) as described in [ITU-T P.501]. The handset terminal is set-up as described in clause 4.3. The delay is calculated using the cross-correlation function between the signal at the electrical test point and the signal arriving at the artificial ear. The measurement is corrected by the delay introduced by the test equipment. The delay is expressed in ms, determined from the maximum of the cross-correlation function.

6 Headset technical requirements

Most headset technical requirements are the same as those of the handset terminal. To condense contents and avoid redundance, this clause only covers those which are particular to headsets.

The headset should be mounted at the test position using a HATS as specified in [ITU-T P.380]. For binaural headsets, the following applies:

– For headsets that do not have left and right ear wearing indication, they can be tested on a HATS that is equipped with one functional ear simulator. The primary receiver (where the microphone system is located) should be positioned on the functional ear simulator for all tests. The secondary receiver shall be moved into position for evaluating frequency response, RLR and distortion measurements.

– For headsets that are designed to be worn in only one manner or fashion, a HATS with two functional ear simulators shall be used. The two receivers shall be tested separately. The secondary receiver need only be tested for frequency response, RLR and distortion.

6.1 Sending characteristics

Refer to clause 4.3 for headset measurement set-up. Refer to clause 5 for sending requirements and measurements.

6.2 Receiving characteristics

6.2.1 Receiving loudness rating (RLR)

– the nominal value of RLR shall be +2 dB, with a tolerance of ±3 dB;

– for binaural earphones, the nominal value of RLR shall be 8 dB for each earphone, with a tolerance of ±3 dB.

6.2.1.1 Measurement method

This should be calculated from the sensitivity/frequency characteristic determined in clause 5.1.1.1 by means of [ITU-T P.79]. The DRP-ERP correction, as defined in [ITU-T P.57], is applied.

NOTE – Other methods for calculating loudness rating used by some Administrations for their own internal planning purposes can be found in [b-Teleph].

6.2.2 Receiving noise

When driven by a signal corresponding to a decoder value quiet code, the maximum acoustic noise level at ERP shall be as follows:

– if no user-controlled volume control is provided or when the volume control is set to nominal RLR value, the measured receiving noise at ERP shall not be greater than –57 dBPa(A);

  For binaural headsets, each earphone shall not exceed –60 dBPa(A).

– if a volume control is provided, the measured receiving noise at ERP shall not be greater than –54 dBPa(A) at the maximum setting of the volume control;

  For binaural headsets, each earphone shall not exceed –57 dBPa(A).

– no peaks in any 1/3-octave band with the level of 10 dB higher than the average noise spectrum in the frequency domain shall occur.

6.2.2.1 Measurement method

The receiving noise shall be measured in a quiet environment as defined in clause 4.4.

The receiving noise level at ERP is measured with A-weighting. The DRP-ERP correction as defined in [ITU-T P.57] is applied.

6.3 Echo path loss characteristics

6.3.1 Stability loss

With the headset positioned on the defined surface as specified in clause 5.4.2.1, 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.

6.3.1.1 Measurement method

The signal used in stability measurement is the same with TCLw, with a different input signal level of 0 dBm0.

The measurement is made at one-twelfth octave intervals for frequencies from 200 Hz to 4000 Hz. With the headset and the transmission circuit fully active, the attenuation from digital input to digital output is measured under one of the following conditions. Stability is the least value of the attenuation in the band 200 to 4000 Hz.

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

– the microphone and the receiver shall face towards the surface;
– the headset receiver shall be placed centrally at the reference point as shown in Figure 5-8;
– the headset microphone is positioned as close as possible to the receiver.

Annex A

Distortion allowances

(This annex forms an integral part of this Recommendation)

In producing the distortion characteristics at sending and receiving 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 dBm0p.

The total contribution of these factors is calculated using power summation.

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.

Bibliography

[b-Teleph] ITU-T Handbook on Telephonometry (1992), *Measurement Method: Telephonometry*.

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Printed in Switzerland
Geneva, 2010