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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



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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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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 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. 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
TCL _w	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.

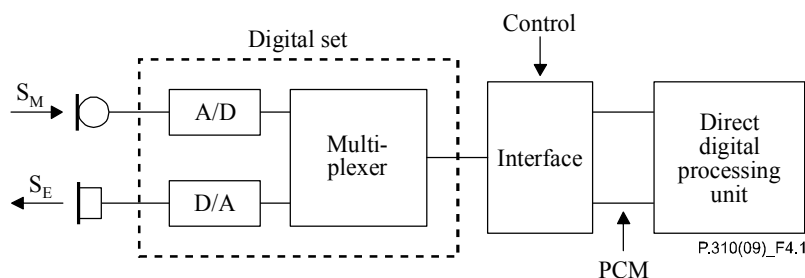


**Figure 4-1 – Digital telephone test arrangement
(direct digital processing approach)**

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).

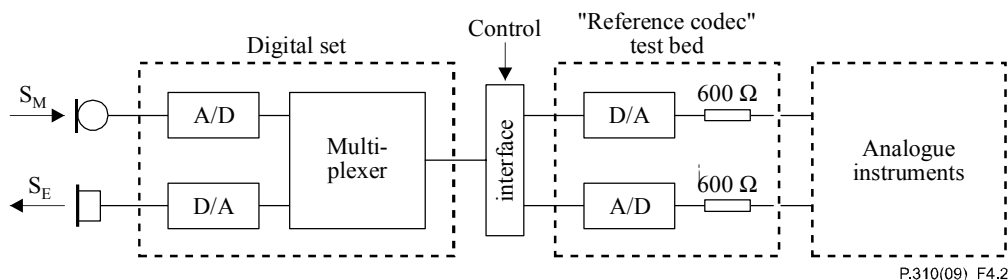


**Figure 4-2 – Digital telephone test arrangement
(codec approach)**

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 $\pm 2^\circ$. 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 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 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.

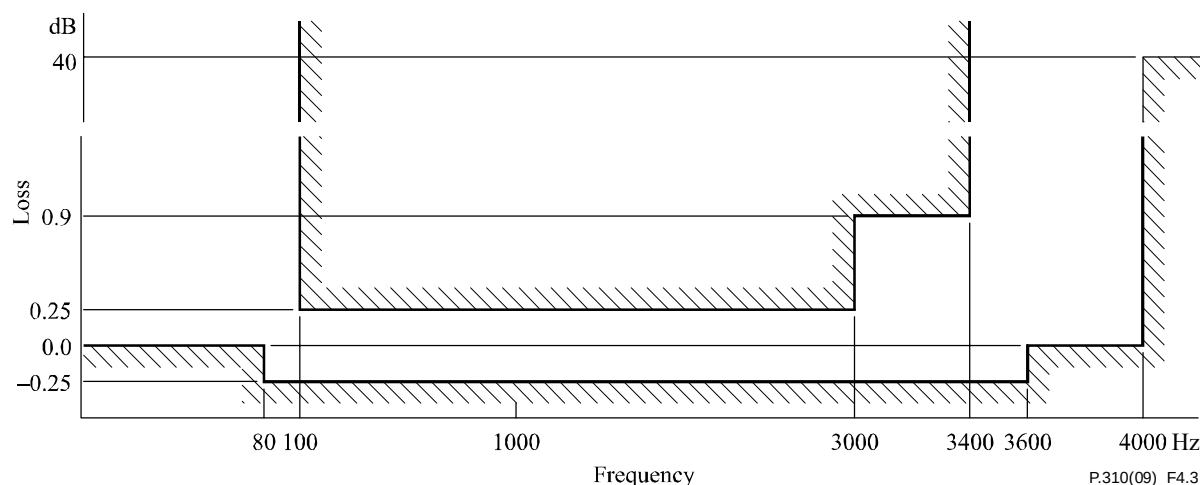


Figure 4-3 – Attenuation/frequency distortion of the sending or receiving sides of the reference codec

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.