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

Subscribers' lines and sets

Transmission characteristics for cordless and mobile digital terminals

ITU-T Recommendation P.313

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ITU-T Recommendation P.313

Transmission characteristics for cordless and mobile digital terminals

Summary

ITU-T Recommendation P.313 provides audio performance requirements for portable digital cordless and mobile handsets, headset and speakerphone sets. The requirements apply to narrow-band systems (3.1 kHz) regardless of the coding algorithm used in the terminal. Associated test methods are also given.

Requirements are specified for the major electro-acoustic performance parameters affecting audio quality, including sending and receiving levels, frequency responses, noise, sidetone, stability, echo path and delay. The requirements given in this Recommendation should ensure satisfactory service quality in a high percentage of installations under normal conditions.

Changes over the previous version of this Recommendation (2004) are as follows:

- Specifications for headset terminals have been added. These requirements are intended for mobile terminals which use headset.
- Specifications for hands-free terminals are recommended for handheld and desktop speakerphone set, and those for car mounted hands-free terminals will be removed to a separate Recommendation.
- The receiving masks for handset to use Type 3.2, Type 3.3 and Type 3.4 artificial ears are also recommended.

Testing methods for handsets have been updated to use test signals defined in ITU-T Recommendations P.50 and P.501.

Source

ITU-T Recommendation P.313 was approved on 16 March 2007 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

Keywords

Cordless, electro-acoustic measurements, hands-free, mobile, performance specification, speakerphone, terminals, wireless.

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ITU-T Recommendation P.313

Transmission characteristics for cordless and mobile digital terminals

1 General

1.1 Scope

This Recommendation deals with electro-acoustic performance parameters of portable digital cordless and mobile terminals. The requirements given below should ensure satisfactory voice service in a high percentage of installations under normal conditions, but other factors impacting the performance, such as the radio link, are not included.

This Recommendation includes specifications for handset, headset and speakerphone set of mobile terminals. These specifications may be applicable to digital cordless handset, headset and speakerphone sets. The requirements contained in this Recommendation apply only to narrow-band systems (3.1 kHz) regardless of the coding algorithm used.

Specifications for speakerphone sets are recommended for handheld terminals and desktops.

Specifications for car mounted hands-free terminals will be included in a separate Recommendation.

Requirements are given for handset, headset and speakerphone set in conjunction with a 0 dBr 4-wire reference (see Figure 1) base station having an appropriate air interface, and are specified irrespective of the particular technology and air interface.

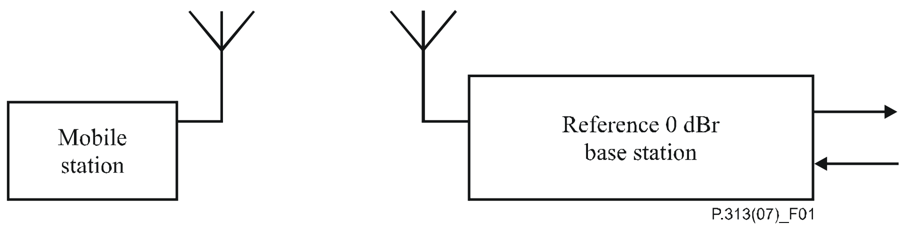

Figure 1 – Reference configuration

1.2 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.114] ITU-T Recommendation G.114 (2003), *One-way transmission time*.

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

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

[ITU-T G.168] ITU-T Recommendation G.168 (2004), *Digital network echo cancellers.*

[ITU-T G.174] ITU-T Recommendation G.174 (1994), *Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN.*

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

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

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

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

[ITU-T P.57] ITU-T Recommendation P.57 (2005), *Artificial ears*.

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

[ITU-T P.64] ITU-T Recommendation P.64 (1999), *Determination of sensitivity/frequency characteristics of local telephone systems.*

[ITU-T P.79] ITU-T Recommendation P.79 (1999), *Calculation of loudness ratings for telephone sets*.

[ITU-T P.310] ITU-T Recommendation P.310 (2003), *Transmission characteristics for telephone band (300-3400 Hz) digital telephones.*

[ITU-T P.330] ITU-T Recommendation P.330 (2003), *Speech processing devices for acoustic enhancement.*

[ITU-T P.340] ITU-T Recommendation P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals.*

[ITU-T P.342] ITU-T Recommendation P.342 (2000), *Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals.*

[ITU-T P.360] ITU-T Recommendation P.360 (2006), *Efficiency of devices for preventing the occurrence of excessive acoustic pressure by telephone receivers and assessment of daily noise exposure of telephone users.*

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

[ITU-T P.501] ITU-T Recommendation P.501 (2000), *Test signals for use in telephonometry*.

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

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

[ITU-T P.832] ITU-T Recommendation P.832 (2000), *Subjective performance evaluation of hands-free terminals.*

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

NOTE – The latest versions of the annexes to these Recommendations shall apply.

1.3 Definitions

Definitions from ITU-T Recommendation P.10/G.100.

1.3.1 handset: A device which includes telephone receiver and transmitter which is typically coupled to the ear by hand.

1.3.2 handset telephone: A telephone set equipped with a handset.

1.3.3 hands-free reference point (HFRP) (see ITU-T Recs P.340, P.341 and P.342): A point located on the axis of the artificial mouth, at 50 cm from the outer plane of the lip ring, where the level calibration is made, under free-field conditions. It corresponds to the measurement point 11, as defined in [ITU-T P.51].

1.3.4 hands-free terminal: A telephone set that does not require the use of hands during the communications session; examples are headset, speakerphone and group-audio terminal.

1.3.5 head and torso simulator (HATS) (see [ITU-T P.58]): Manikin extending downward from the top of the head to the waist, designed to simulate the sound pick-up characteristics and the acoustic diffraction produced by a median human adult and to reproduce the acoustic field generated by the human mouth.

1.3.6 headset: A device which includes telephone receiver and transmitter which is typically secured to the head or the ear of the wearer.

1.3.7 speakerphone set: A *telephone set* using a loudspeaker as a telephone receiver with or without an embedded microphone as a telephone transmitter; it may be used without the handset.

Definition from ITU-T Recommendation P.310

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

Definitions from ITU-T Recommendation P.380

1.3.9 recommended test position (RTP): Corresponds to the position in which the headset should be placed on HATS, e.g., as instructed by the manufacturer. In all cases, the RTP should resemble the RWP on humans.

1.3.10 recommended wearing position (RWP): Corresponds to the position in which a headset should be placed on humans according to the intended use (e.g., as instructed by the manufacturer in the user manual, etc.).

1.4 Abbreviations

This Recommendation uses the following abbreviations:

AEC	Acoustic Echo Controller
Ardt	Attenuation range in receiving direction during double talk
ARL	Acoustic Reference Level
Asdt	Attenuation range in sending direction during double talk
CL	Centre of lips of head and torso simulator
DRP	Eardrum Reference Point
ERP	Ear Reference Point
HATS	Head And Torso Simulator
HFRP	Hands-Free Reference Point
LR	Loudness Rating
LRGP	Loudness Rating Guard-ring Position
LSTR	Listener SideTone Rating

MRP	Mouth Reference Point
POI	Point of Interconnection
RLR	Receiving Loudness Rating
RTP	Recommended Test Position
RWP	Recommended Wearing Position
SLR	Sending Loudness Rating
SS	System Simulator
STMR	SideTone Masking Rating
TCLw	Terminal Coupling Loss weighted
TCLwdt	Weighted terminal coupling loss during double talk
TCLwst	Weighted terminal coupling loss during single talk
TELR	Talker Echo Loudness Rating
TELRdt	Talker Echo Loudness Rating during double talk
TRst-r	Build-up time, single talk, receive signal
TRst-s	Build-up time, single talk, send signal
UE	User Equipment
VAD	Voice Activity Detection

2 Test configuration

The general access to terminals is described in Figure 2. This can be made by using a HATS (head and torso simulator) or the LRGP position (loudness rating guard-ring position), with appropriate artificial ear and appropriate mountings for handset terminal in a realistic but reproducible way.

HATS is described in [ITU-T P.58], appropriate artificial ears are described in [ITU-T P.57] (Type 3.3 and Type 3.4 ears), a proper positioning of handset in realistic conditions can be found in [ITU-T P.64], the test setups for various types of speakerphone sets can be found in [ITU-T P.581].

LRGP is described in [ITU-T P.64], and the appropriate ears are described in [ITU-T P.57] (Type 1 and Type 3.2 ears), a proper positioning of handset in realistic conditions can be found in [ITU-T P.64].

NOTE – The detailed description of the artificial ears and their applicability can be found in [ITU-T P.57].

The preferred way of testing is the connection of a terminal to the system simulator (SS) with exact defined settings and access points. The test sequences are fed in either electrically using a reference codec or acoustically.

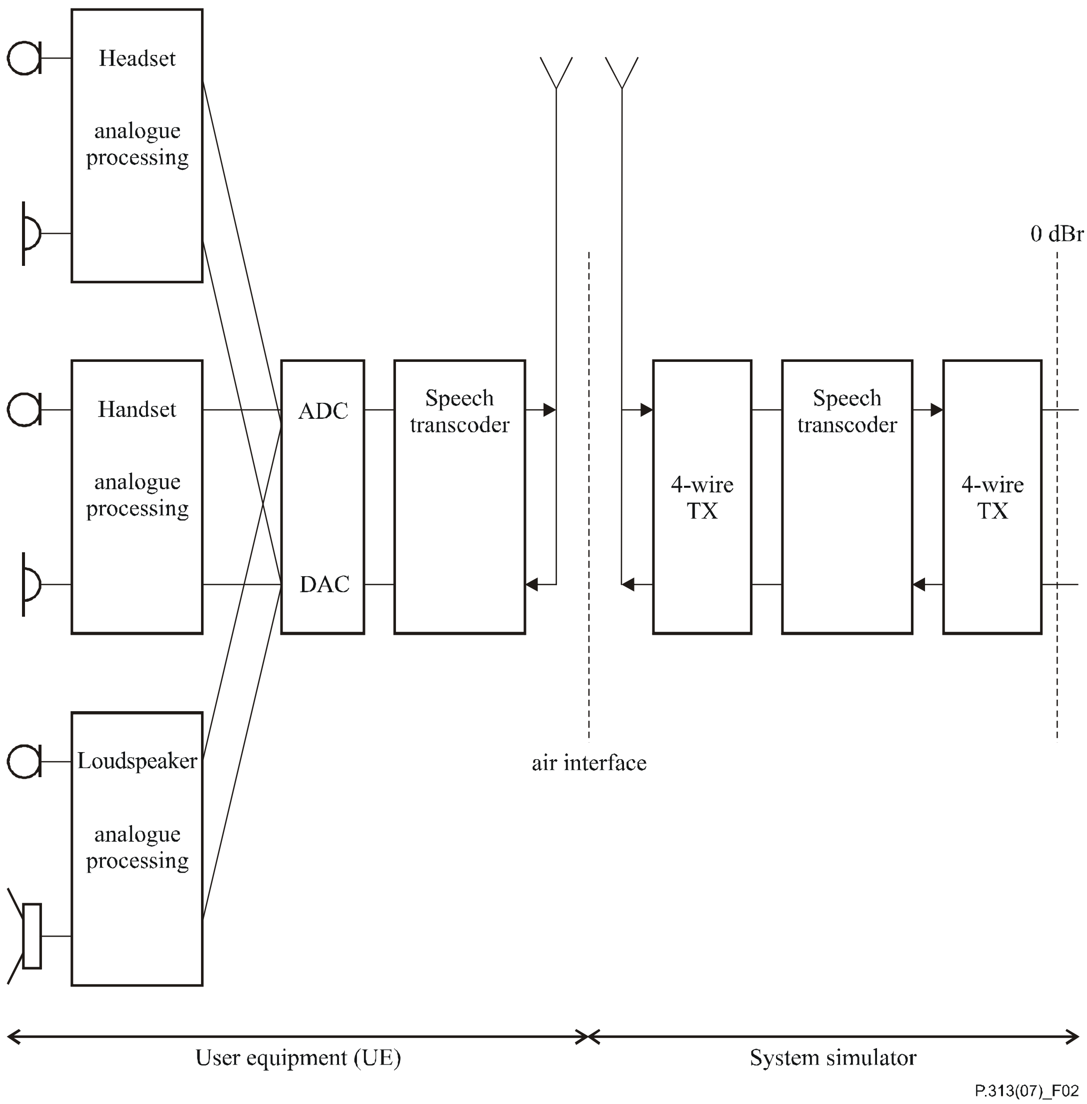

Figure 2 – Interfaces for specification and testing of terminal acoustic characteristics

2.1 Setup for handset terminal

HATS method: When using the HATS method, the handset is placed according to [ITU-T P.64]. The artificial mouth shall comply with [ITU-T P.58], the artificial ear shall comply with [ITU-T P.57], Type 3.3 or Type 3.4 ears should be used.

LRGP method: When using the LRGP method, the handset is placed in the LRGP position as described in [ITU-T P.64]. The artificial mouth shall comply with [ITU-T P.51]. The artificial ear shall comply with [ITU-T P.57], Type 1 or Type 3.2 ears should be used.

The type of artificial ear chosen will be used for all the tests.

2.2 Setup for headset

The test setup for headset terminals is defined in [ITU-T P.380].

NOTE 1 – As indicated in [ITU-T P.380], the measurement position for headset has to be specified and fully documented along with measurement.

NOTE 2 – Once the position of the headset has been specified, it should be used for all the tests.

2.3 Setup for speakerphone set

2.3.1 Desktop operated speakerphone set

HATS test equipment and setups for speakerphone set can be found in [ITU-T P.581].

Measurement setup using a free-field microphone and a discrete P.51 artificial mouth for desktop speakerphone set can be found in [ITU-T P.340].

The signal level for sending characteristics measurement shall be adjusted to –4.7 dBpa at MRP or –28.7 dBpa at HFRP or HATS HFRP.

The signal level for receiving characteristics measurement shall be adjusted to –16 dBm0.

2.3.2 Handheld speakerphone set

If HATS measurement equipment is used, it should be configured to the handheld speakerphone set according to Figure 3. The HATS should be positioned so that the HATS reference point is at a distance d_{HF} from the centre point of the visual display of the mobile station. The distance d_{HF} is specified by the manufacturer. A vertical angle θ_{HF} may be specified by the manufacturer.

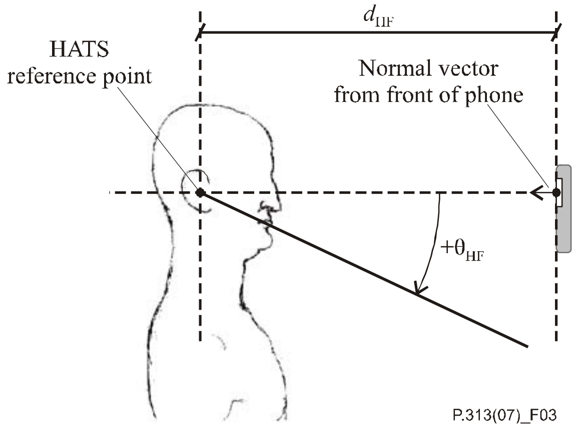

Figure 3 – Configuration of handheld loudspeaker UE relative to the HATS

If a free-field microphone together with a discrete P.51 mouth is used, they should be configured to the handheld speakerphone set as per Figure 4 for receiving measurements and Figure 5 for sending measurements. The measurement instrument should be located at a distance d_{HF} from the centre of the visual display of the mobile station. The distance d_{HF} is specified by the manufacturer, and $d_{HFR} = d_{HF}$, $d_{HFS} = d_{HF}$-d_{EM}, where d_{HFR} is the distance for receiving measurement, d_{HFS} is the distance for sending measurement, and d_{EM} is the distance from ERP to MRP.

The signal level for sending characteristics measurement shall be adjusted to –4.7 dBPa at MRP.

The signal level for receiving characteristics measurement shall be adjusted to –16 dBm0.

If the distance d_{HF} is not defined by the manufacturer, the default position as described below is used. The default position is 30 cm for d_{HFS}, which seems a typical value for a handheld phone.

With this value of d_{HFS}, we would obtain $d_{HF} = d_{HFR} = d_{HFS} + d_{EM} = 42$ cm.

$$d_{EM} = d_{EEP\text{-}CL} \times \cos(\theta_{HF})$$

with $d_{EEP\text{-}CL}$ = distance between EEP and centre lips = 13 cm nominal

$$\theta_{HF} = 24° \text{ nominal}$$

Resulting in $d_{EM} = 41.9$ cm.

The calibration for this default position is as follows:

– calibration and equalization at the MRP: –4.7 dBPa;

– calibration at 30 cm: –24.3 dBPa (attenuation is supposed to be 24 dB from MRP to HFRP located at 50 cm; so the attenuation at 30 cm is 4.4 dB less than for 50 cm).

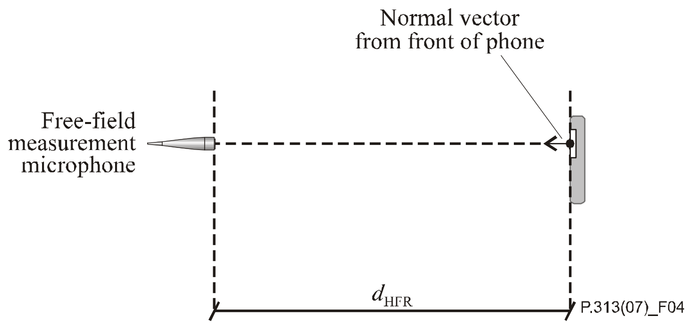

Figure 4 – Configuration of handheld speakerphone set, free-field microphone for receiving measurements

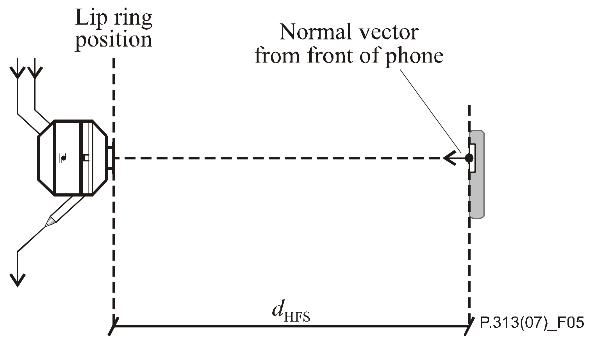

Figure 5 – Configuration of handheld speakerphone set, discrete P.51 artificial mouth for sending measurements

3 Test conditions

3.1 Test signal

In general the test signals used should be speech-like. The use of sine signals is not appropriate when speech processing and coding systems are implemented in the terminal, e.g., for speakerphone sets. Appropriate test signals can be found in [ITU-T P.50] and [ITU-T P.501]. The type of test signal used shall be stated in the test report.

When using speech-like signals, such as defined in [ITU-T P.50] and [ITU-T P.501], care should be taken not to overload the system under test due to the higher crest factor of the "speech-like signal".

3.2 Test environment

It is the responsibility of the test laboratory to ensure that the measurements done in the test room give identical results to those obtained in a free-field environment (e.g., an anechoic room).

3.2.1 Handset and headset terminal

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

For handset and headset measurements, the test room shall be practically free-field down to a lowest frequency of 275 Hz, the handset or the headset coupled with the HATS/LRGP 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 and noise contrast.

3.2.2 Speakerphone sets

When testing the speakerphone sets acoustic performance, care must be taken that, for example, noise levels are sufficiently low in order not to interfere with the measurements.

The broadband noise level shall not exceed –70 dBPa(A) for all the tests, except for Background noise transmission.

The octave band noise level shall not exceed the values specified in Table 1.

Table 1 – Noise level

Centre frequency (Hz)	Octave band pressure level (dBPa)
63	–45
125	–60
250	–65
500	–65
1 k	–65
2 k	–65
4 k	–65
8 k	–65

4 Handset technical requirements

Unless otherwise specified, measurements defined in clause 4 shall be carried at the setting of the volume control where the RLR is as close as possible to nominal value.

4.1 Sending characteristics

4.1.1 Sending loudness rating (SLR)

In view of the following considerations:

– wireless terminals provide connectivity to the existing wireline-based international telecommunication networks;

– digital wireless access networks should provide the same signal levels as the digital access wireline networks,

the following loudness rating value is recommended:

– a nominal value of SLR = 8 dB.

NOTE – Several reports have indicated that sending speech levels in mobile networks are 5 dB higher on average than speech levels from fixed networks. The most likely cause is that mobile terminals are frequently used in noisy environments where users tend to talk louder. Manufacturers of mobile handsets should be aware that this may lead to saturation, clipping, and distortion of the signal and reduced speech quality. Changing SLR at this time is not recommended due to concerns of reduced speech quality when used at normal talker levels. However, the manufacturers of devices intended for use in noisy conditions can consider adopting a higher SLR.

The SLR shall be calculated, based on the measurements described in clause 4.1.2, according to [ITU-T P.79], using Equation 4-1,

$$LR = -\frac{10}{m} \cdot \log_{10} \sum_{i=N_1}^{N_2} 10^{0.1 \cdot m(S_i - W_i)} \tag{4-1}$$

with $m = 0.175$, over bands 4 to 17 and the send weighting factors from Table 1 of [ITU-T P.79].

4.1.2 Sending frequency responses

In view of the following considerations:

– the compatibility with wireline digital telephones and the mixed analogue/digital telephone network;

– the compatibility with most existing wireless systems;

– the aim to achieve the best possible overall quality with the cordless and mobile terminals,

sending nominal sensitivity/frequency response within the limits given in Table 2 is recommended.

Table 2 – 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	–

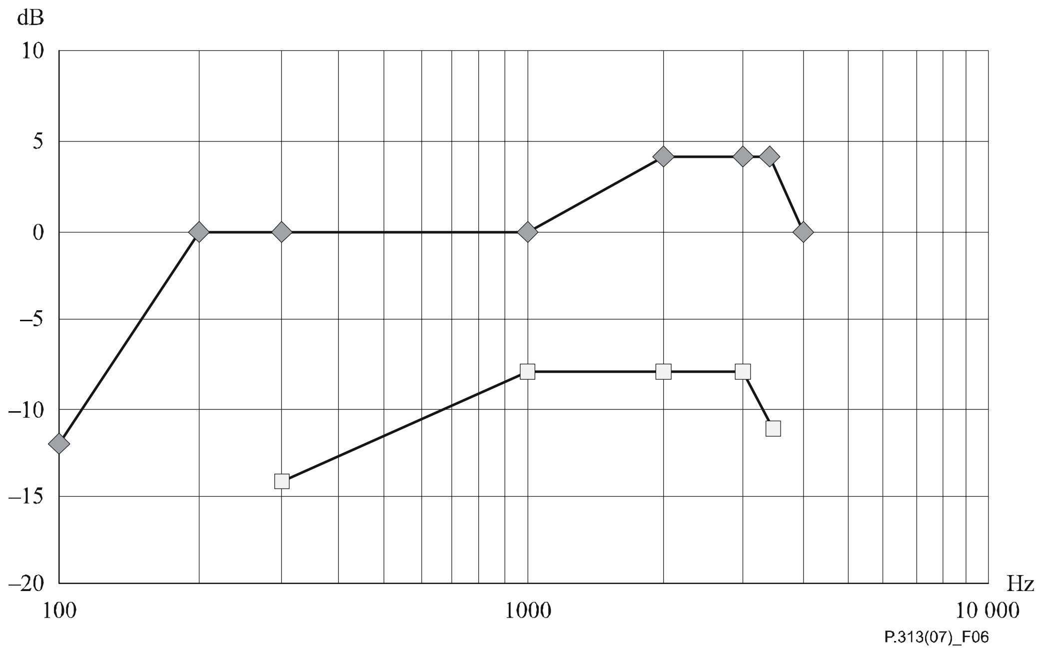

Figure 6 – Sending mask

4.1.2.1 Measurement method

The sending frequency response is measured according to [ITU-T P.64], using the measurement setup shown in Figure 7.

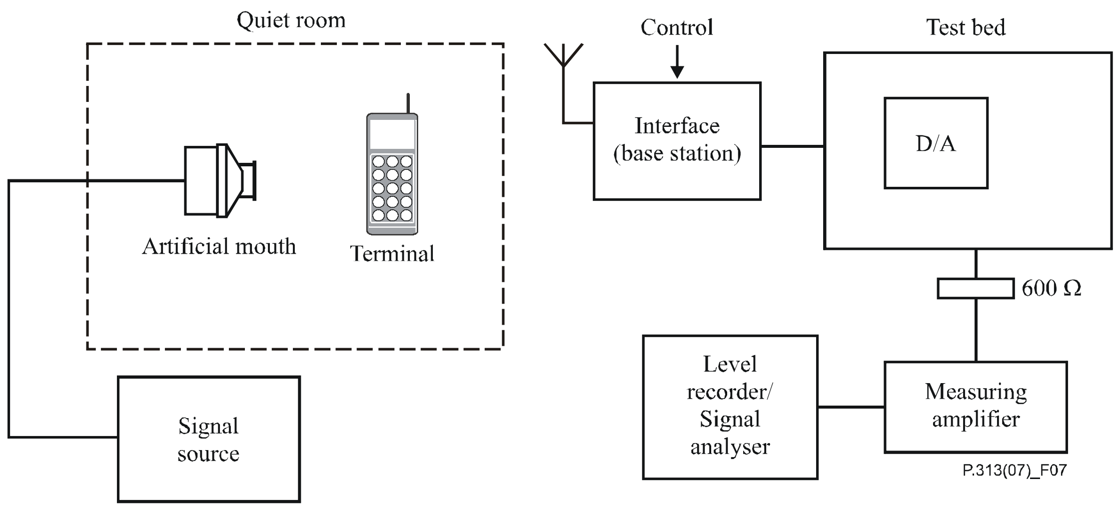

Figure 7 – Sending frequency response measurement method

NOTE – Figure 7 corresponds to LRGP position; a similar test configuration applies for HATS position.

An artificial speech signal (e.g., as specified in [ITU-T P.50] and [ITU-T P.501]) and a spectrum analyser should be used. Additional test methods may be found in [ITU-T P.502]. The test signal used shall be specified in the test report. The test signal level shall be –4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset is mounted according to [ITU-T P.64].

The sensitivity is expressed in terms of dBV/Pa.

4.1.3 Idle channel noise

The following limit is recommended:

– sending noise level maximum –64 dBm0p.

4.1.3.1 Measurement method

The handset is mounted according to [ITU-T P.64] in a quiet environment, as defined in clause 3.2, the sending noise level at the SS audio output is measured with apparatus including psophometric weighting described in [ITU-T O.41].

4.1.4 Distortion

The sending distortion includes non-linear distortion and quantizing distortion. The sending part shall meet the following distortion requirements:

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

Sending level dB relative to ARL	Sending ratio (dB)
–35	17.5
–30	22.5
–20	30.7
–10	33.3
0	33.7
+5	31.7

The sending distortion shall be measured between MRP and the SS audio output (output of the reference speech decoder of the SS). The ratio of signal-to-total distortion power measured with a psophometric noise weighting described in [ITU-T O.41] shall be above the limits given in Table 3, unless the sound pressure at MRP exceeds +10 dBPa.

4.1.4.1 Measurement method

The handset is mounted according to [ITU-T P.64] in a quiet environment as defined in clause 3.2.

The signal used is an activation signal followed by a sine-wave signal with a frequency in the range of 1004 Hz to 1025 Hz which is the actual test signal. The signal is applied at the MRP. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

The level of the actual test 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: –35, –30, –25, –20, –15, –10, –5, 0, 5 dB relative to ARL.

The total distortion power of the signal output of the SS shall be measured with the psophometric noise weighting described in [ITU-T O.41].

4.1.5 Linearity

NOTE – Linearity tests as described below assume linear and time invariant behaviour of the UE. If this can not be assured, 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].

If the system is intended to operate in a linear mode, the following characteristic is recommended:

– the linearity relative to the gain for acoustic reference level (ARL) should remain within the limits given in Table 4.

Table 4 – Linearity

Sending dB relative to ARL	Upper limit (dB)	Lower limit (dB)
+4	1	–2
–10	1	–2
–20	1	–5
–25	1	–8
–30	1	–12
< –30	6	–∞

The limits for intermediate levels can be found by drawing straight lines between the break points in the table on a linear (dB signal level) – linear (dB gain) scale.

4.1.5.1 Measurement method

The handset is mounted according to [ITU-T P.64] using a suitable artificial ear as recommended in [ITU-T P.57].

A sine-wave signal with a frequency in the range of 1004 Hz to 1025 Hz should be 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 should be applied at the following levels:

–30, –25, –20, –15, –10, –5, 0, 4 dB relative to ARL

The linearity relative to the gain for the ARL should be measured.

Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

NOTE 1 – In general, care must be taken in the case of time variant and/or non-linear terminals. In such cases, a sine wave may not be appropriate as the test signal; a speech-like test signal should be chosen as described in [ITU-T P.501] and [ITU-T P.50]. If the sine-wave 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. Additional test methods may be found in [ITU-T P.502]. The test signal used shall be specified in the test report.

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

4.1.6 Out-of-band signals

With any sine-wave 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 SS audio output 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 5.

Table 5 – Discrimination levels

Applied sine-wave frequency	Limit (minimum)
4.6 kHz	30 dB
8 kHz	40 dB

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

4.1.6.1 Measurement method

The handset is mounted according to [ITU-T P.64].

The signal used is 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 at the level of –4.7 dBPa which are the actual test signals. The test signal is applied at the MRP. Appropriate activation signals and signal combinations can be found in [ITU-T P.501].

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 of –4.7 dBPa, the level of any image frequencies at the SS audio output shall be measured.

4.2 Receiving characteristics

An application force of 6N should be used when measuring with Type 3.3 or 3.4 artificial ears. This force is considered to be a realistic average value.

4.2.1 Receiving loudness rating (RLR)

In view of the following considerations:

– wireless terminals provide connectivity to the existing wireline based international telecommunication networks;

– digital wireless terminals should be compatible with the digital access wireline networks,

the following loudness rating value is recommended:

– a nominal value of RLR = 2 dB.

The RLR shall be calculated, based on the measurements described in clause 4.2.3, according to [ITU-T P.79], using Equation 4-1, with m = 0.175, over bands 4 to 17 and the receive weighting factors from Table 1 of [ITU-T P.79]. The measured receiving sensitivity shall be corrected using the leakage correction from Table 2 of [ITU-T P.79].

NOTE – When Type 1 ear is used, the leakage correction shall be used to calculate RLR; when Type 3.2, 3.3 and 3.4 ears are used, no leakage correction shall be used.

4.2.2 Volume control

In view of the following considerations of mobile handsets:

– the mobile terminals are used often in noisy environment;

– the need to provide service for people with hearing impairments. For these applications, ITU-T Rec. P.370 applies.

Where a user-controlled receiving volume control is provided, the RLR shall meet the selected nominal value for at least one setting of the control. When the control is set to maximum, it is suggested that it should allow at least 12 dB volume increase relative to the nominal value of RLR = 2 dB, but the RLR shall not be less than (louder than) –13 dB at maximum position.

4.2.3 Receiving frequency responses

In view of the following considerations:

– the compatibility with wireline digital telephones and the mixed analogue/digital telephone network;

– the compatibility with most existing wireless systems;

– the aim to achieve the best possible overall quality with the cordless and mobile terminals.

For different types of artificial ears, the receiving sensitivity/frequency response within the limits given in Tables 6 and 7 is recommended.

Table 6 – Receiving-Type 1

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	–12	–
200	0	–
300	2	–7
500	(Note)	–5
1000	0	–5
3000	2	–5
3400	2	–10
4000	2	–
NOTE – The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) vs linear (dB) scale.		

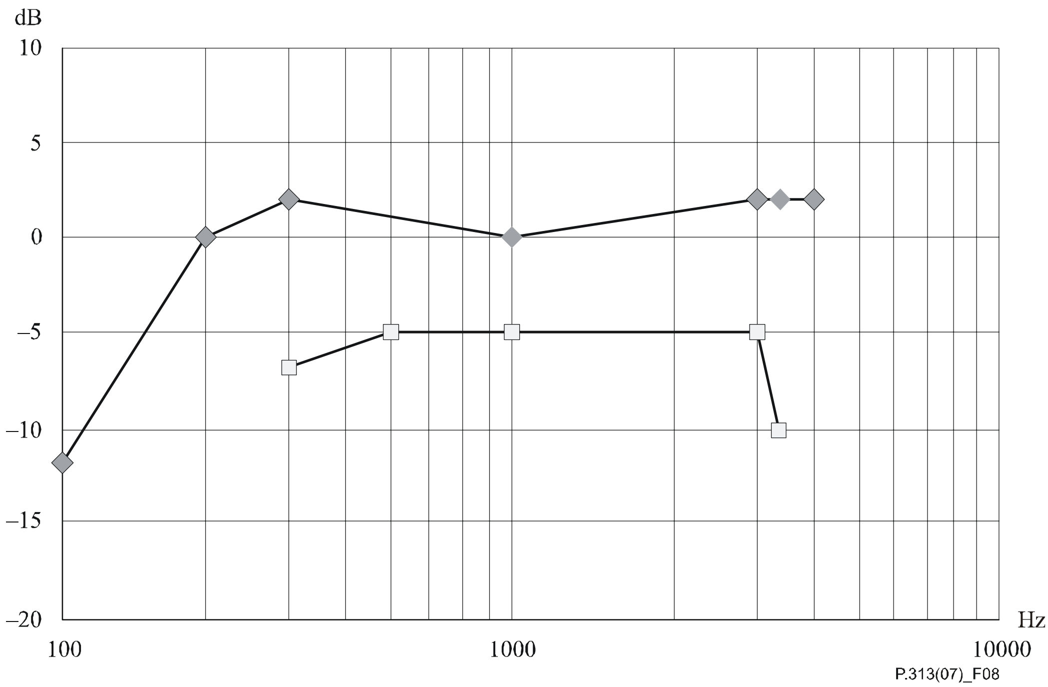

Figure 8 – Receiving mask-Type 1

Table 7 – Receiving-Type 3.2, 3.3, 3.4

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	–10	–∞
200	2	–∞
300	2	–9
1000	2	–7
3400	2	–12
4000	2	–∞
8000	–18	–∞

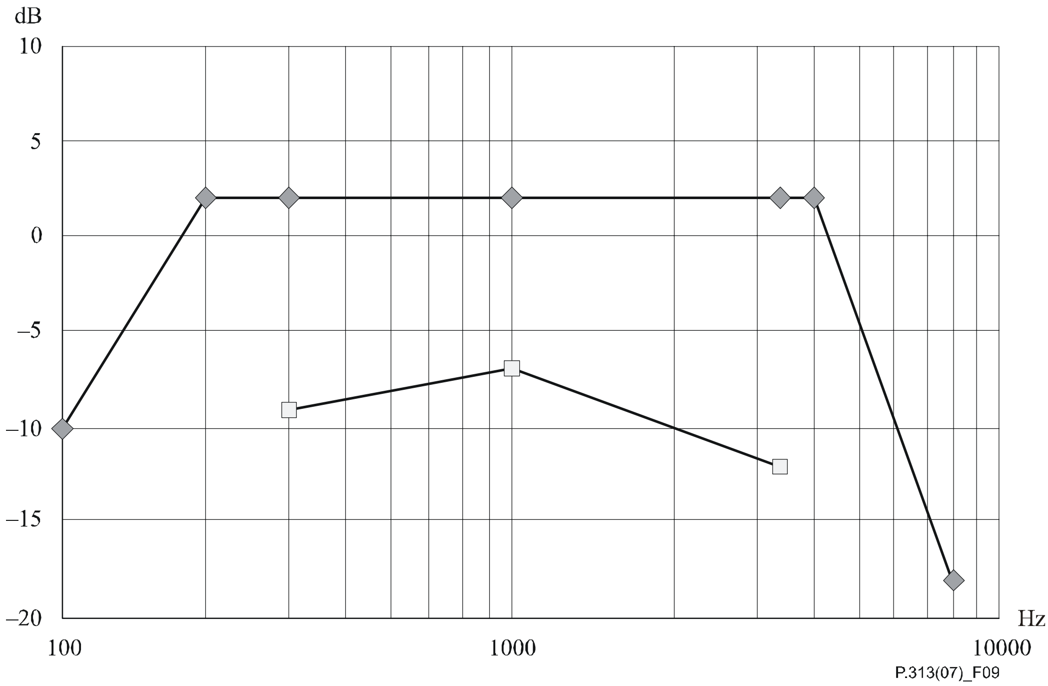

Figure 9 – Receiving mask-Type 3.2, 3.3, 3.4

NOTE – The following rules may be additionally applied:

In general, the frequency response should not introduce a strong rolloff at lower frequencies, regardless of which coupler is used for measurements. Signal levels at frequencies down to 300 Hz should not be attenuated more than 5 dB as compared to the level measured at 1 kHz. Too much emphasis at high frequencies should be also avoided. Compared to the level measured at 1 kHz, the emphasis introduced up to 3.4 kHz should not be more than 5 dB.

4.2.3.1 Measurement method

The handset should be mounted according to [ITU-T P.64], using a suitable artificial ear as recommended in [ITU-T P.57].

The receiving frequency response is measured, using the measurement setup shown in Figure 10. The test signal level shall be –16.0 dBm0. Terminals with adjustable receive levels shall be adjusted so that their RLR is as close as possible to the nominal value for this test.

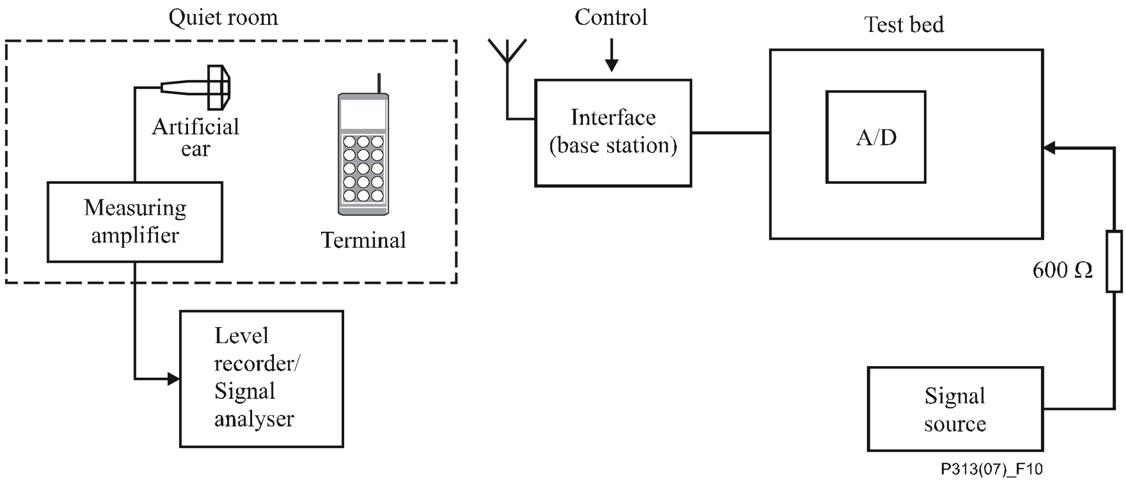

Figure 10 – Receive frequency response measurement method

NOTE – Figure 10 corresponds to LRGP position; a similar test configuration applies for HATS position.

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. 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 ERP. Information about correction factors are available in [ITU-T P.57].

4.2.4 Idle channel noise

The maximum (acoustic) noise level at the handset terminal when no signal is transmitted to the input of the SS shall be as follows:

– If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured at the ear reference point (ERP) contributed by the receiving equipment alone shall not exceed –57 dBPa(A).

– Where a volume control is provided, the measured noise shall also not exceed –54 dBPa(A) at the maximum setting of the volume control.

4.2.4.1 Measurement method

With the handset mounted according to [ITU-T P.64], using a suitable artificial ear as recommended in [ITU-T P.57], placed in a quiet environment, as defined in clause 3.2, the receiving noise level at ERP is measured with apparatus including psophometric weighting described in [ITU-T O.41].

4.2.5 Distortion

The receiving distortion includes non-linear distortion and quantizing distortion.

The receiving part between the SS audio input (input of the reference speech encoder of the SS) and ERP shall meet the requirements at the nominal setting of the volume control.

The ratio of signal-to-total distortion power measured with the psophometric noise weighting described in [ITU-T O.41] shall be above the limits given in Table 8. For sound pressures exceeding +10 dBPa at the ERP, there is no distortion requirement.

Table 8 – Limits for signal-to-total distortion ratio

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
–45	17.5
–40	22.5
–30	30.5
–20	33.0
–10	33.5

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.

4.2.5.1 Measurement method

The handset should be mounted according to [ITU-T P.64], using a suitable artificial ear as recommended in [ITU-T P.57].

The signal used is an activation signal followed by a sine-wave signal with a frequency in the range of 1004 Hz to 1025 Hz which is the actual test signal. The signal is applied at input of the SS at the following levels: –45, –40, –35, –30, –25, –20, –15, –10 dBm0. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

The total distortion power shall be measured at the ERP with the psophometric noise weighting.

4.2.6 Linearity

If the system is intended to operate in a linear mode, the following is recommended:

– the linearity relative to the gain at an input level of –10 dBm0 should be within the limits given in Table 9.

Table 9 – Variation of gain with input level, receiving

Receiving level at the digital interface	Upper limit (dB)	Lower limit (dB)
–6 dBm0	1	–2
–50 dBm0	1	–2
–50 dBm0	1	–∞

The limits for intermediate levels can be found by drawing straight lines between the break points in the table on a linear (dB signal level) – linear (dB gain) scale.

4.2.6.1 Measurement method

The handset should be mounted according to [ITU-T P.64] using a suitable artificial ear as recommended in [ITU-T P.57].

A digitally simulated sine-wave signal with a frequency in the range of 1004 Hz to 1025 Hz should be applied at the digital interface at the following levels: –50, –45, –40, –35, –30, –25, –20, –15, –10, –6 dBm0.

NOTE 1 – In general, care must be taken in the case of time variant and/or non-linear terminals. In such cases, a sine wave may not be the appropriate test signal; a more speech-like test signal should be chosen as described in [ITU-T P.501] and [ITU-T P.50]. The test signal used shall be specified in the test report.

The linearity relative to the gain at an input level of –10 dBm0 should be measured using the artificial ear.

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

4.2.7 Spurious out-of-band signals

The level of out-of-band signals at the ERP shall meet the following requirements:

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3.4 kHz and at a level of –5 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4.6 to 8 kHz measured selectively at the ERP shall be at least lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in Table 10.

Table 10 – Discrimination levels

Image signal frequency	Equivalent input signal level
4.6 kHz	–35 dBm0
8 kHz	–45 dBm0

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

4.2.7.1 Measurement method

The handset should be mounted according to [ITU-T P.64], using a suitable artificial ear as recommended in [ITU-T P.57].

The signal used is an activation signal followed by a sine-wave signal at the frequencies 500, 1000, 2000, and 3150 Hz which is the actual test signal. The signal is applied at the input of the SS at the level of –5 dBm. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501]. For test signals at frequencies 500, 1000, 2000 and 3150 Hz, the level of spurious out-of-band image signals at frequencies of up to 8 kHz shall be measured selectively at the ERP.

4.3 Sidetone characteristics

4.3.1 Sidetone masking rating (STMR)

In view of the following considerations:

– the optimum STMR for conditions free from echo;

– the difficulties of high ambient noise conditions,

the following is recommended:

– the value of the STMR shall be in the range of 10 dB to 20 dB.

Where a user-controlled receiving volume adjustment is provided, the STMR shall meet the requirement given above at the setting where the RLR is equal to the nominal value (2 dB).

4.3.1.1 Measurement methods

A test signal level of –4.7 dBPa shall be applied at the MRP. For each frequency given in Table 3 of [ITU-T P.79], bands 1 to 20, the sound pressure (at the ERP) shall be measured.

The handset should be mounted according to [ITU-T P.64], using a suitable artificial ear as recommended in [ITU-T P.57].

The test setup shown in Figure 11 is used to measure the sidetone frequency response. The sidetone path loss L_{meST} and the STMR shall be calculated according to [ITU-T P.79], using Equation 4-1 ($m = 0.225$) and the weighting factors in Table 3 of [ITU-T P.79].

Techniques other than the swept sine-wave can be used. 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. Additional test methods may be found in [ITU-T P.502]. The test signal used shall be specified in the test report.

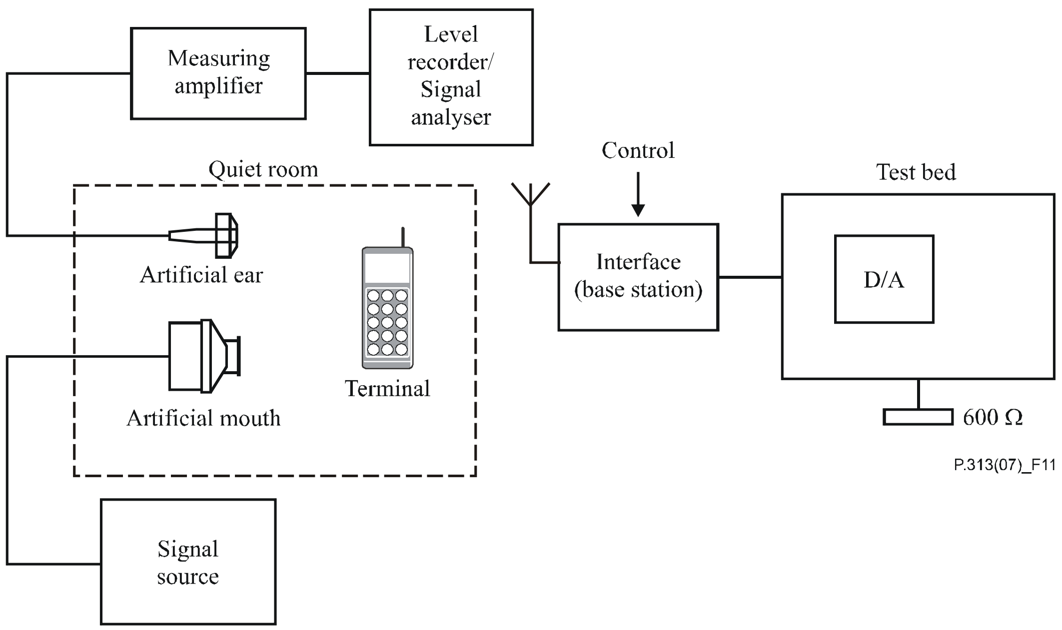

Figure 11 – Sidetone measurement method

NOTE – Figure 11 corresponds to LRGP position; a similar test configuration applies for HATS position.

4.3.2 D-factor

In view of the following considerations:

– mobile sets are being used often in noisy environments;

– the difficulties of high ambient noise conditions,

the following is recommended:

– the value of the weighted average D ("D-factor") of the difference of the send sensitivities between direct and diffuse sound 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 – Terminals designed for quiet environments (e.g., some indoor applications) may have lower D-factor limits, but the D value should not be less than –3 dB.

4.3.2.1 Measurement method

The handset should be mounted according to [ITU-T P.64].

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

The sound pressure level of the diffuse sound field shall be adjusted in the range of –54 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-factor is the difference of the sending sensitivities between direct and diffuse sound.

The diffuse sound sending sensitivity shall be used for the calculation as S_{si}(diff).

The direct sound sending sensitivity S_{si}(direct) shall be measured according to clauses 4.1.1 and 4.1.2.

The D-factor is computed with S_{si}(diff) and S_{si}(direct) from Formulae E-3 and E-2 of [ITU-T P.79] and the coefficients K_i in Table E.1 of [ITU-T P.79].

4.4 Noise contrast and comfort noise

In some circumstances, such as application of voice-operated devices, the continuous background noise present, regardless of whether the users are talking or not, may be interrupted. This switching on and off is annoying to the users and may in fact degrade speech intelligibility. To reduce this effect, noise contrast should be minimized by increasing signal-to-noise ratio.

Comfort noise may be injected during silent periods to reduce the impairments created by the noise contrast. This may create undesirable performance degradation by itself if not done properly, due to the level or spectrum contents differences between the injected and the transmitted noise. Efforts should be made to match the characteristics of the injected comfort noise to the transmitted noise to reduce any perceptible contrast between them.

4.4.1 Measurement method

For further study.

4.5 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 user himself chooses the way to hold his handset,

the following limit is recommended:

In order to meet the G.131 talker echo objective requirements, the weighted terminal coupling loss (TCLw) should be greater than 46 dB when measured under free-field conditions.

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. 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.

4.5.1 Measurement method

TCL_w is measured in free air in such a way that the inherent mechanical coupling of the handset is not affected.

Noise and reflections in the test space must not influence the measurement. The test should be performed in the environment as defined in clause 3.2.

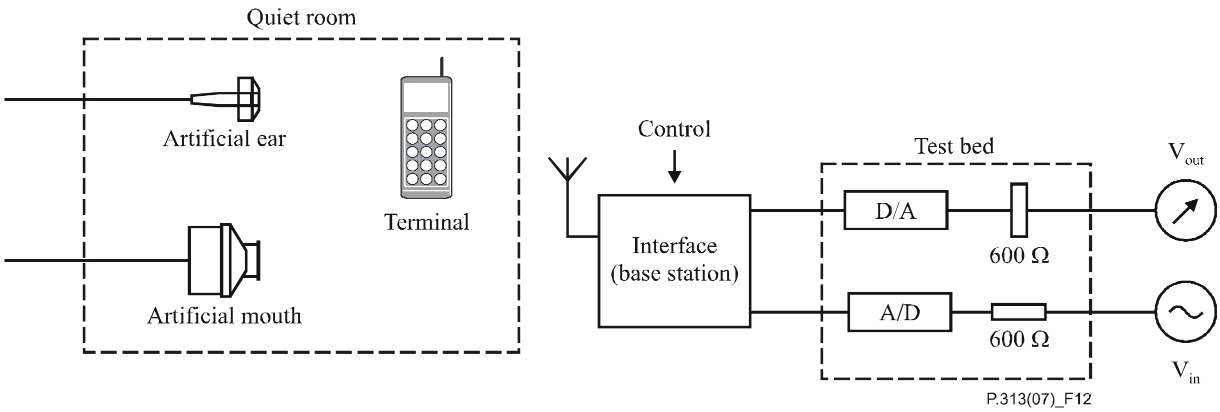

Figure 12 – Terminal coupling loss measurement method

NOTE 1 – Figure 12 corresponds to LRGP position; a similar test configuration applies for HATS position.

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 12.

The test signal used is described in Annex A of [ITU-T P.501].

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

Terminals with adjustable receive levels shall be tested at the nominal setting. For the nominal setting, adjust the level so that the RLR is as close as possible to the nominal RLR value.

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.

4.6 Stability loss

The minimum stability loss at any volume control setting should be at least 6 dB.

4.6.1 Measurement method

The stability measurement is made at an input signal of –10.0 dBm0, at one-twelfth octave intervals for frequencies from 200 Hz to 4 kHz. With the handset and transmission circuit fully active, measure the attenuation from the digital input to the digital output using the following method.

Place the handset in the reference corner, as shown in Figure 13, with the earcap and mouthpiece facing a hard, smooth surface. The handset shall be placed along the diagonal from the apex of the reference corner to the outside corner, with the earcap end of the handset 250 mm from the apex. The telephone set shall be fully active.

The reference corner consists of three, smooth, hard surfaces of perpendicular planes extending 0.5 m from the apex of the corner.

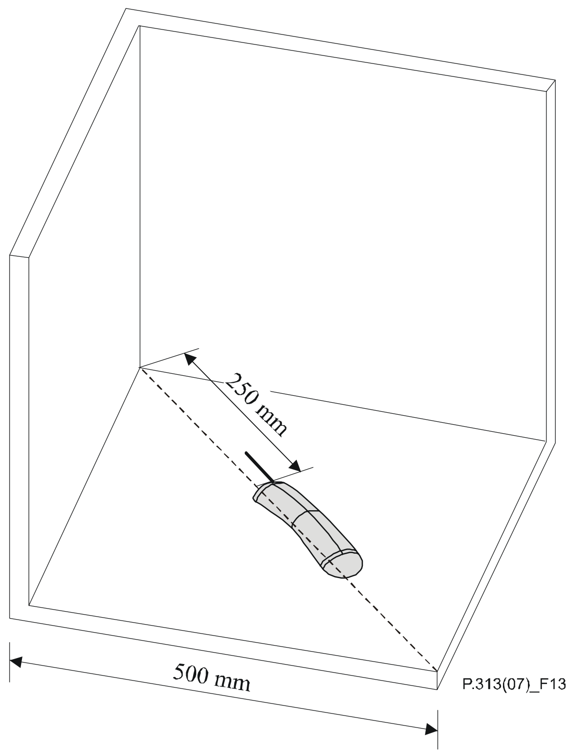

Figure 13 – Reference corner

4.7 Delay

In view of the following considerations:

– that delay has impact on echo performance and the dynamics of voice conversation;

– the amount of delay introduced by wireless systems depends on specific technology and may be inherent to the adopted coding technique,

the following is recommended:

– delay added by the terminal equipment should be minimized in accordance with the guidelines provided in [ITU-T G.114] even with the use of echo control;

– 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, should be less than 20 ms ([ITU-T G.174]);

NOTE – It is recognized that some existing systems will not meet the above limit.

– terminal manufacturers must ensure that appropriate echo control measures are in place according to the guidelines provided in [ITU-T G.131]. This may include, for example, meeting limits specified in clause 4.5.

4.7.1 Measurement method

The delay shall be measured according to [ITU-T P.310].

4.8 Speech clipping

In view of the following considerations:

– that wireless systems may employ a variety of speech interpolation techniques as well as being susceptible to bursts of errors in the radio channel;

– excessive loss of speech signal may affect quality of a connection;

– subjective impact of clipping depends upon duration of clipping, percentage of speech clipped, frequency of clipping and overall speech activity, the following is recommended for speech clipping, i.e., loss of speech:

– no speech loss occurrences longer than 64 ms should be present;

– speech loss periods shorter than 64 ms should be kept below 0.2 per cent of active speech.

NOTE – Per cent of clipped speech is 100 times the product of the frequency of speech clipping times clipping duration, divided by the speech activity factor.

4.8.1 Measurement methods

For further study.

4.9 Acoustic safety of telephone user

In order to ensure safety and to minimize annoyance to the user, the terminal shall meet limits specified in [ITU-T P.360].

4.9.1 Measurement method

The maximum steady state acoustic pressure is measured by applying maximum positive digital code to the receive input defined for the handset under test. The test procedure is in accordance with clause 4.2.3.1 for the receive characteristics except that the acoustic pressure in the artificial ear is measured with a sound level meter.

The measurement shall refer to [ITU-T P.360].

Terminals with adjustable receive levels shall be tested at the maximum setting.

5 Headset technical requirements

Unless otherwise specified, measurements defined in clause 5 will be carried at the setting of the volume control where the RLR is as close as possible to nominal value.

5.1 Headset sending characteristics

5.1.1 Sending loudness rating

The nominal values of SLR shall be: SLR = 8 dB.

5.1.1.1 Measurement method

The SLR should be measured using the methods given in [ITU-T P.380] and computed using the methods given in [ITU-T P.79].

The SLR shall be calculated based on the measurements described in clause 5.1.2.

5.1.2 Sending frequency response

The sending sensitivity/frequency characteristics shall be as follows:

The sending sensitivity frequency response, measured from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) with microphone at RTP position, shall be within a mask, which can be drawn between the points given in Table 11. The mask is drawn with straight lines between the breaking points in Table 11 on a logarithmic (frequency) – linear (dB sensitivity) scale.

Table 11 – Sending sensitivity/frequency mask-headset

Frequency (Hz)	Upper limit	Lower limit
100	–12	–
200	0	–
300	0	–12
1000	0	–6
2000	4	–6
3000	4	–6
3400	4	–9
4000	0	–

All sensitivity values are expressed in dB on an arbitrary scale.

5.1.2.1 Measurement method

The headset should be mounted according to [ITU-T P.380] using a HATS.

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. Additional test methods may be found in [ITU-T P.502]. The test signal used shall be specified in the test report. The test signal level shall be –4.7 dBPa, measured at MRP. The test signal level is averaged over the complete test signal sequence.

The sensitivity is expressed in terms of dBV/Pa.

5.1.3 Idle channel noise

The maximum noise level produced by the apparatus at the output of the SS under silent conditions in the sending direction shall not exceed –64 dBm0p.

5.1.3.1 Measurement method

The headset should be mounted according to [ITU-T P.380] using a HATS in a quiet environment, as defined in clause 3.2, the sending noise level at the SS audio output is measured with apparatus including psophometric weighting described in [ITU-T O.41].

5.1.4 Distortion

The sending part shall meet the following distortion requirements:

The sending distortion shall be measured between MRP and the SS audio output (output of the reference speech decoder of the SS). The ratio of signal-to-total distortion power measured with the a psophometric noise weighting described in [ITU-T O.41] shall be above the limits given in Table 12, unless the sound pressure at MRP exceeds +10 dBPa.

Table 12 – Limits for signal-to-total distortion ratio

Sending level dB relative to ARL	Sending ratio (dB)
–35	17.5
–30	22.5
–20	30.7
–10	33.3
0	33.7
+5	31.7

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.

5.1.4.1 Measurement method

The headset should be mounted according to [ITU-T P.380] using a HATS.

The signal used is an activation signal followed by sine-wave signal with a frequency in the range of 1004 Hz to 1025 Hz which is the actual test signal. The signal is applied at the MRP. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

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: –35, –30, –25, –20, –15, –10, –5, 0, 5 dB relative to ARL.

The total distortion power of the signal output of the SS shall be measured with the psophometric noise weighting described in [ITU-T O.41].

5.2 Headset receiving characteristics

5.2.1 Receiving loudness rating

The nominal values of RLR shall be: RLR = 2 dB.

5.2.1.1 Measurement method

The RLR shall be calculated, based on the measurements described in clause 5.2.2, according to [ITU-T P.79].

For binaural earphones, the receiving sensitivity equals the power sum of that measured with the left ear and right ear individually.

5.2.2 Receiving frequency response

The receiving sensitivity/frequency characteristics shall be as follows:

The receiving sensitivity frequency response, measured from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the ERP, shall be within a mask, which can be drawn with straight lines between the breaking points in Table 13 on a logarithmic (frequency) – linear (dB sensitivity) scale.

For binaural earphones, the left earphone and right earphone should meet the requirement specified in Table 13.

Table 13 – Receiving sensitivity/frequency mask-headset

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	–10	–∞
200	2	–∞
300	2	–9
1000	2	–7
3400	2	–12
4000	2	–∞
8000	–18	–∞

NOTE 1 – All sensitivity values are expressed in dB on an arbitrary scale.

NOTE 2 – The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.

5.2.2.1 Measurement method

The headset should be mounted at the test position using a HATS as specified in [ITU-T P.380].

The receive frequency response is measured according to [ITU-T P.64] using the measurement setup shown in Figure 10.

The test signal level shall be –16.0 dBm0. Terminals with adjustable receive levels shall be adjusted so that their RLR is as close as possible to the nominal value for this test.

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. The test signal level shall be –16 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 ERP. Information about correction factors is available in [ITU-T P.57].

5.2.3 Idle channel noise

The maximum (acoustic) noise level at the headset UE when no signal is applied to the input of the SS shall be as follows:

– If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured at the ear reference point (ERP) contributed by the receiving part alone shall not exceed –57 dBPa(A).

For binaural earphones, each receiver shall not exceed –60 dBPa(A).

– Where a volume control is provided, the measured noise shall also not exceed –54 dBPa(A) at the maximum setting of the volume control.

For binaural earphones, each receiver shall not exceed –57 dBPa(A).

5.2.3.1 Measurement method

The headset should be mounted at test position using a HATS as specified in [ITU-T P.380] in a quiet environment, as defined in clause 3.2, the receiving noise level at ERP is measured with apparatus including psophometric weighting described in [ITU-T O.41].

5.2.4 Distortion

The receiving part between the SS audio input (input of the reference speech encoder of the SS) and ERP shall meet the requirements in this clause at the nominal setting of the volume control:

The ratio of signal-to-total distortion power measured with the proper noise weighting described in [ITU-T O.41] shall be above the limits given in Table 14 when the sound pressure at ERP is up to +10 dBPa. For sound pressures exceeding +10 dBPa at the ERP, there is no distortion requirement.

Table 14 – Limits for signal-to-total distortion ratio

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
–45	17.5
–40	22.5
–30	30.5
–20	33.0
–10	33.5

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.

5.2.4.1 Measurement method

The headset should be mounted at test position according to [ITU-T P.380] using a HATS.

The signal used is an activation signal followed by a sine-wave signal with a frequency in the range of 1004 Hz to 1025 Hz which is the actual test signal. The signal is applied at the input of the SS at the following levels: –45, –40, –35, –30, –25, –20, –15, –10 dBm0. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

The total distortion power shall be measured at the ERP with the psophometric noise weighting.

5.3 Sidetone characteristics

The talker sidetone masking rating (STMR) for headset shall be within the range from 10 dB to 20 dB.

5.3.1 Measurement method

The headset should be mounted at test position according to [ITU-T P.380] using a HATS.

The test setup shown in Figure 11 is used to measure the sidetone frequency response. The sidetone path loss L_{meST} and the STMR shall be calculated according to [ITU-T P.79], using Equation 4-1 ($m = 0.225$) and the weighting factors in Table 3 of [ITU-T P.79].

5.3.2 D-factor

In view of the following considerations:

– mobile sets are being used often in noisy environments;

– the difficulties of high ambient noise conditions,

the following is recommended:

– the value of the weighted average D ("D-factor") of the difference of the send sensitivities between direct and diffuse sound 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 headset performance in noisy conditions is the D-factor.

NOTE 2 – Headsets designed for quiet environments (e.g., some indoor applications) may have lower D-factor limits, but the D value should not be less than –3 dB.

5.3.2.1 Measurement method

The headset is mounted according to [ITU-T P.64].

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

The sound pressure level of diffuse sound field shall be adjusted in the range of −54 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-factor is the difference of the sending sensitivities between direct and diffuse sound.

The diffuse sound sending sensitivity shall be used for the calculation as S_{si}(diff).

The direct sound sending sensitivity S_{si}(direct) shall be measured according to clauses 5.1.1 and 5.1.2.

The D-factor is computed with S_{si}(diff) and S_{si}(direct) from Formulae E-3 and E-2 of [ITU-T P.79] and the coefficients K_i in Table E.1 of [ITU-T P.79].

5.4 Weighted terminal coupling loss (TCLw)

The TCLw for a headset UE shall be 46 dB.

Headset with adjustable receive levels shall be tested at the nominal setting. For the nominal setting, adjust the level so that the RLR is as close as possible to the nominal RLR value.

5.4.1 Measurement method

The headset should be mounted at test position according to [ITU-T P.380] using a HATS.

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.

The test signal used is described in Annex A of [ITU-T P.501].

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

5.5 Acoustic safety of telephone user

In order to ensure safety and to minimize annoyance to the user, the terminal shall meet limits specified in [ITU-T P.360].

5.5.1 Measurement method

The maximum steady state acoustic pressure is measured by applying maximum positive digital code to the receive input defined for the handset under test. The test procedure is in accordance with clause 5.2.2.1 for the receive characteristics except that the acoustic pressure in the artificial ear is measured with a sound level meter.

The measurement shall refer to [ITU-T P.360].

Terminals with adjustable receive levels shall be tested at the maximum setting.

6 Speakerphone sets technical requirements

The requirements and test methods in this Recommendation are only for desktop operated and handheld loudspeaker mobile terminal. The transmission characteristics and test methods specified in [ITU-T P.340] and [ITU-T P.342] can be used for the two kinds of speakerphone set.

Speakerphone sets use speech processing devices for acoustic enhancement (SPDA) to control acoustic echo, reduce background noise transmission, etc. These parameters are defined in [ITU-T P.330].

Appropriate test signals are described in [ITU-T P.50] and [ITU-T P.501]. Test methods appropriate for parameters defined in this Recommendation may be found in [ITU-T P.502]. Proper use of HATS testing may be found in [ITU-T P.581].

Unless otherwise specified, measurements defined in clause 6 will be carried at the setting of the volume control where the RLR is as close as possible to nominal value.

6.1 Loudspeaker sending characteristics

6.1.1 Sending loudness rating (SLR)

According to [ITU-T P.340], the SLR of a loudspeaker telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

Therefore, the nominal value of SLR for handheld and desktop speakerphone set shall be +13 dB.

6.1.1.1 Measurement method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The SLR shall be calculated according to [ITU-T P.79].

6.1.2 Sending frequency response

In view of the following considerations:

– ideally, the frequency response should be flat for optimal voice quality;

– many manufacturers use microphones and analogue filters to reduce noise,

the sending tolerance mask of the handheld and desktop speakerphone set for the nominal sensitivity/frequency response mask shown in Table 15 is recommended. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) – linear (dB sensitivity) scale.

Table 15 – Speakerphone sending frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
200	0	–
250	0	–
315	0	–14
400	0	–13
500	0	–12
630	0	–11
800	0	–10
1000	0	–8
1300	2	–8
1600	3	–8
2000	4	–8
2500	4	–8

Table 15 – Speakerphone sending frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
3100	4	–8
4000	0	–
NOTE – As stated in [ITU-T P.340], the interval between 200-300 Hz makes a significant contribution to the naturalness of the transmitted speech. However, the noise energy in a noise environment in that frequency range might be significantly higher than the speech energy. Therefore, the manufacturers should consider this trade-off between naturalness of the transmitted speech and background noise transmitted when determining the optimal frequency response in the sending direction.		

6.1.2.1 Measurement methods

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

An artificial voice according to [ITU-T P.50] or a speech-like test signal as described in [ITU-T P.501] can be used for the test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free-field conditions at the MRP. The test signal level shall be –4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to the required value and the spectrum is not altered.

The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as a reference to determine the sending sensitivity S_{mJ}.

The test procedure principle, as described in clause 4.1.2.1, shall be used.

6.1.3 Idle channel noise

The following limit is recommended:

– send noise level maximum –64 dBm0p.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

6.1.3.1 Measurement methods

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The environment shall comply with the conditions described in clause 3.2.

The psophometric noise level at the output of the SS is measured. The psophometric filter is described in [ITU-T O.41].

6.1.4 Non-linear distortion

The ratio of signal to non-linear distortion shall be above the mask defined in Table 16.

Table 16 – Signal to distortion ratio limit, sending

Frequency (Hz)	Signal to distortion ratio limit, sending (dB)
315	26
400	30.5
1000	30.5
NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale.	

6.1.4.1 Measurement methods

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315, 400, 500, 630, 800 and 1000 Hz which is the actual test signal. The signal is applied at the MRP. The test signal level is –4.7 dBPa. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

The signal to harmonic distortion ratio is measured selectively up to 3.15 kHz.

6.1.5 Out-of-band signals in sending direction

With any 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 shall be below the level obtained for the reference signal by at least the amount (in dB) specified in Table 17.

Table 17 – Out-of-band signal limit, sending

Frequency (kHz)	Signal limit (dB)
4.6	30
8	40
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (kHz) scale.	

6.1.5.1 Measurement methods

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The signal used is 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 at the level of –4.7 dBPa which are the actual test signals. The test signal is applied at the MRP. Appropriate activation signals and signal combinations can be found in [ITU-T P.501].

6.2 Speakerphone set receiving characteristics

6.2.1 Receiving loudness rating (RLR)

For desktop speakerphone set, the nominal value of RLR shall be 2 dB.

For handheld speakerphone set, the nominal value of RLR shall be 6 dB.

Due to the varying levels of background noise, it is recommended that a user-specific volume control is provided. The RLR value shall be met for at least one setting of the volume control.

This value is derived from [ITU-T P.310]. According to [ITU-T P.340], the volume control range should span the value of the receiving loudness rating which is equal to that of the corresponding handset telephone, as well as an RLR value about 10 dB lower.

As a short-term objective, the minimum RLR shall be –18 dB. This corresponds to a 20 dB gain over nominal levels.

6.2.1.1 Measurement methods

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The RLR shall be calculated according to [ITU-T P.79].

6.2.2 Receiving frequency response

In view of the following considerations:

– ideally, the frequency response should be flat for optimal voice quality,

the receiving tolerance mask for handheld speakerphone set shown in Table 18 and for desktop speakerphone set shown in Table 19 are recommended. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) – linear (dB sensitivity) scale.

Table 18 – Receiving frequency response-handheld

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
200	0	–
250	0	–
315	0	–
400	0	–
500	0	–
630	0	–
800	0	–10
1000	0	–12
1300	0	–12
1600	0	–12
2000	0	–12
2500	0	–12
3100	0	–12
4000	0	–

Table 19 – Receiving frequency response-desktop

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
200	0	–
250	0	–
315	0	–15
400	0	–12
500	0	–12
630	0	–12
800	0	–12
1000	0	–12
1300	0	–12
1600	0	–12
2000	0	–12
2500	0	–12
3100	0	–12
4000	0	–

6.2.2.1 Measurement method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

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 the test. The type of test signal used shall be stated in the test report. The test signal level shall be –16 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 test procedure described in clause 4.2.3.1 shall be used.

6.2.3 Idle channel noise

The following limit is recommended at the nominal RLR setting:

– receive noise level maximum –53 dBPa(A).

Spectral peaks shall be less than 10 dB above the average noise spectrum. A narrow-band FFT analysis with a frequency resolution of 32 Hz or less is recommended.

The noise produced by the speakerphone terminal providing volume control shall be less than –45 dBPa(A) with the volume control set to its maximum level (minimum RLR setting).

6.2.3.1 Measurement method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The environment shall comply with the conditions described in clause 3.2.

Under quiet conditions, as defined in clause 3.2, the receiving noise level at ERP is measured with apparatus including psophometric weighting described in [ITU-T O.41].

6.2.4 Non-linear distortion

The ratio of signal to harmonic distortion shall be above the mask defined in Table 20.

Table 20 – Signal to distortion ratio limit, receiving

Frequency (Hz)	Signal to distortion ratio limit, receiving (dB)
315	26
400	26
500	30.5
1000	30.5
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (Hz) scale.	

The requirement only applies for 1 kHz frequency for handheld loudspeaker.

6.2.4.1 Measurement method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The measurement methodology only applies for 1 kHz frequency for handheld loudspeaker.

The signal used is an activation signal followed by a series of sine-wave signal with a frequency at 315, 400, 500, 630, 800 and 1000 Hz, which is the actual test signal. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501]. The signal level shall be calibrated to –16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 3.15 kHz.

6.2.5 Linearity in receiving direction

The audio components of the speakerphone terminal should operate in a linear fashion. This is not only for optimal voice quality, but also for optimal performance of the AEC within the speakerphone terminal. Specific requirements are for further study.

6.2.6 Spurious out-of-band signals

The test signal is a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3.4 kHz and at a level of –5 dBm0 applied at the digital interface.

The level of spurious out-of-band image signals in the frequency range of 4.6 to 8 kHz measured selectively at the artificial ear shall be at least lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in Table 21.

Table 21 – Discrimination levels

Image signal frequency (kHz)	Equivalent input signal level (dBm0)
4.6	–35
8	–45
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (kHz) scale.	

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

6.2.6.1 Measurement Method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The signal used is an activation signal followed by a sine-wave signal at the frequencies 500, 1000, 2000 and 3150 Hz which is the actual test signal. The signal is applied at the input of the SS at the level of –5 dBm. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501]. For the test signals at the frequencies 500, 1000, 2000 and 3150 Hz, the level of spurious out-of-band image signals at frequencies of up to 8 kHz shall be measured selectively at the artificial ear.

6.3 Weighted terminal coupling loss

In order to meet the G.131 talker echo objective requirements, the weighted terminal coupling loss during single talk (TCLwst) should be greater than 46 dB when measured under free-field conditions.

For terminals fitted with a volume control, the TCLwst shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.

The limits for weighted terminal coupling loss during double talk (TCLwdt) are defined in clause 6.5.

6.3.1 Measurement method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

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.

The test signal used is described in Annex A of [ITU-T P.501].

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

6.4 Switching characteristics

Many mobile speakerphone terminals use a voice activity detector (VAD) and a loss controller to control acoustic echo. The VAD distinguishes silent periods (no active speech signal), single-talk periods (near-end speech periods or far-end speech periods) and double-talk periods (near-end and far-end speech signals active at the same time). According to the state determined by the VAD, the loss controller reduces the acoustic echo level by inserting variable losses on the received and/or transmitted audio signals. (The VAD and loss controller may also be used to control electrical echo from the network if the mobile terminal operates on an analogue wireless network without echo control.) An AEC and/or NLP may also be used to control acoustic echo. However, most devices with an AEC and/or NLP also require a loss controller to control residual acoustic echo. Although no VAD is perfect and instantaneous, errors by the VAD may lead to speech clipping and acoustic echo.

The switching parameters and switching functions are described in detail in [ITU-T P.330] and [ITU-T P.340].

The minimum activation level in the sending direction shall be ≤ –20 dBPa. The build-up time for activation in the sending direction shall be:

$$TRst\text{-}s \leq 50 \text{ ms}$$

The minimum activation level in the receiving direction shall be ≤ –35.7 dBm0. The build-up time for activation in the receiving direction shall be:

$$TRst\text{-}r \leq 50 \text{ ms}$$

Requirements for the attenuation and build-up time for receive-to-send states (and send-to-receive states) are for further study.

6.4.1 Measurement method

The setup for desktop speakerphone set is described in [ITU-T P.581].

The setup for handheld speakerphone set is described in Figure 3.

The build-up time TRst-s and TRst-r shall be measured according to [ITU-T P.340].

The echo loss measurement during double talk should be carried out according to [ITU-T P.502].

6.5 Double talk performance

The speech quality during double talk is mainly determined by two parameters: talker echo loudness rating (impairment caused by echo during double talk, and related to TCLwdt) and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions, the TELR should be high and the attenuation inserted should be as low as possible. Terminals that do not allow double talk should provide a good echo attenuation realized by high attenuation.

The most important parameters to determine the quality during double talk are as follows:

– Attenuation range in sending direction during double talk, Asdt.
– Attenuation range in receiving direction during double talk, Ardt.
– Echo attenuation during double talk.

The requirements for Asdt and Ardt at nominal RLR for each category of speakerphone terminals are provided in Table 4 of [ITU-T P.340]. Higher values for Asdt and Ardt are permitted at lower RLR (due to increasing the received volume level) without affecting the behaviour characteristics. However, the TELRdt must maintain the requirements in [ITU-T P.340] for all received volume settings.

Subjective evaluation methods for double talk quality, as well as single talk quality, are described in [ITU-T P.832].

NOTE – Acoustic echo control is more easily and effectively done in terminals. However, network equipment, as described in [ITU-T G.168], may also include acoustic echo control processing. This leads to tandeming issues of acoustic echo control functionalities. As specified in [ITU-T G.168], the added network component shall prevent any degradation of the overall perceived quality. In practice, however, double-talk capability may be reduced due to network equipment.

6.5.1 Measurement method

The setup for desktop speakerphone set is described in [ITU-T P.581].

The setup for handheld speakerphone set is described in Figure 3.

The attenuation range in sending and receiving direction during double talk, Asdt and Ardt, shall be measured according to [ITU-T P.340].

6.6 Background noise transmission and comfort noise injection

Mobile speakerphone terminals are typically used in noisy environments.

Most speakerphone terminals have some type of noise reduction capability. The main purpose of a noise reduction (NR) system in a device is to reduce the annoying and fatiguing effects of the transmitted background noise. The techniques used to reduce background noise may be classified as analogue only, digital only, and combined analogue and digital techniques. These techniques are described in [ITU-T P.330].

Some parameters describing NR and some requirements are also given in [ITU-T P.330]. There is usually a trade-off between the level of digital noise reduction and the voice quality. Too much noise reduction may reduce the transmitted voice level and distort the speech signal.

Without comfort noise, the acoustic echo control system (loss controller, VAD, NLP, and AEC) will reduce and/or distort the background noise transmitted during receive and double talk states. This can be annoying to the far-end talker. Some speakerphone terminals inject comfort noise in order to maintain a constant level of background noise transmitted. Some parameters for comfort noise are described in [ITU-T P.330].

The level of comfort noise should be within a range of +2 and –5 dB from the original (transmitted) background noise.

6.6.1 Measurement methods

For further study.

6.7 Delay

Additional processing delay is usually required in a speakerphone terminal in order to control acoustic echo and noise. Other acoustic enhancement features, such as a receive-end equalizer, may further increase delay.

It is recommended that the delay in the sending direction due to the terminal be ≤ 30 ms. This is measured from the MRP to the POI (electrical reference point), while subtracting the system delay of the network simulator.

It is recommended that the delay in the receiving direction due to the terminal be ≤ 30 ms. This is measured from the POI to the DRP (eardrum reference point) while subtracting the system delay of the network simulator.

6.7.1 Measurement method

The setup for desktop speakerphone set is described in clause 2.3.1.

The setup for handheld speakerphone set is described in clause 2.3.2.

The delay of speakerphone set shall be measured according to Annex A of [ITU-T P.342].

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