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ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

P.342

(06/2009)

SERIES P: TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS

Voice terminal characteristics

Transmission characteristics for narrow-band digital loudspeaking and hands-free telephony terminals

Recommendation ITU-T P.342

ITU-T

ITU-T P-SERIES RECOMMENDATIONS

TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS

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Recommendation ITU-T P.342

Transmission characteristics for narrow-band digital loudspeaking and hands-free telephony terminals

Summary

Recommendation ITU-T P.342 provides audio performance requirements for digital hands-free and loudspeaking telephony terminals using, in the telephone band (300-3400 Hz), the waveform encoding according to Recommendations ITU-T G.711 (PCM at both 64 kbit/s and 56 kbit/s) and ITU-T G.726 (ADPCM 32 kbit/s).

This Recommendation does not deal with audio performance requirements for digital telephones using coding schemes other than waveform encoding and at bit rates lower than 32 kbit/s.

Source

Recommendation ITU-T P.342 was approved on 22 June 2009 by ITU-T Study Group 12 (2009-2012) under Recommendation ITU-T A.8 procedures.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

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

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

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

NOTE

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

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

CONTENTS

Recommendation ITU-T P.342

Transmission characteristics for narrow-band digital loudspeaking and hands-free telephony terminals

1 Scope

This Recommendation provides audio performance requirements for hands-free and loudspeaking telephones using, in the telephone narrow-band (300-3400 Hz), the waveform encoding according to [ITU-T G.711] (PCM at both 64 kbit/s and 56 kbit/s) and [ITU-T G.726] (ADPCM 32 kbit/s). Audio performance requirements for headset terminals are included in [ITU-T P.310]. IP terminals are not included in this Recommendation.

Audio performance requirements for digital telephones using coding schemes other than waveform encoding and at bit rates lower than 32 kbit/s are under study.

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 (1996), *Control of talker echo.*

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

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

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

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

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

[ITU-T P.57] Recommendation ITU-T P.57 (2005), *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.310] Recommendation ITU-T P.310 (2003), *Transmission characteristics for telephone-band (300-3400 Hz) digital telephones*.

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

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

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

[ISO 266] ISO 266:1997, *Acoustics – Preferred frequencies*.

[IEC 61672-1] IEC 61672-1:2002, *Electroacoustics – Sound level meters – Part 1: Specifications*.

3 Definitions and abbreviations

3.1 Definitions

This Recommendation defines the following terms:

3.1.1 double talk: An operation mode, where two users are speaking simultaneously.

3.1.2 ear-drum reference point (DRP): A point located at the end of the ear canal, corresponding to the ear-drum position.

3.1.3 ear reference point (ERP): A virtual point for geometric reference located at the entrance to the listener's ear, traditionally used for calculating telephonometric loudness ratings.

3.1.4 group-audio terminal: A speakerphone set primarily designed for use by several users, which will not be equipped with a handset.

3.1.5 hands-free reference point (HFRP): A point located on the axis of the artificial mouth, at 50 cm from the lip ring, where the level calibration is made in free field. It corresponds to the measurement point 11, as defined in [ITU-T P.51].

3.1.6 hands-free terminal (HFT): A telephone set that does not require the use of hands during the communications session. Examples are headset, speakerphone and group-audio terminal.

3.1.7 HATS hands-free reference point (HATS HFRP): Corresponds to a reference point "n" from [ITU-T P.58], where "n" is one of the points numbered from 11 to 17 and defined in Table 6A of [ITU-T P.58] (coordinates of far-field front point). The HATS HFRP depends on the location(s) of the microphones of the terminal under test: the appropriate axis lip-ring/HATS HFRP shall be as close as possible to the axis lip-ring/HFT microphone under test.

3.1.8 loudspeaking function: Function of a handset telephone using a loudspeaker associated with an amplifier as a telephone receiver. It shall be used with a handset to transmit send signals.

3.1.9 single talk: An operation mode, where only one user is speaking.

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

3.2 Abbreviations

This Recommendation uses the following abbreviations:

AEC Acoustic Echo Control

AGC Automatic Gain Control

CSS Composite Source Signal

DRP ear-Drum Reference Point

ERP Ear Reference Point

HATS Head And Torso Simulator

HFRP Hands-Free Reference Point

HFT Hands-Free Terminal

LRGP Loudness Rating Guard-ring Position

LST LoudSpeaking Terminal

MRP Mouth Reference Point

RLR Receiving Loudness Rating

SLR Sending Loudness Rating

TCL Terminal Coupling Loss

TCLw Weighted Terminal Coupling Loss

4 Test configuration

4.1 Electrical interface specifications

Clauses 4.1 to 4.2 of [ITU-T P.310] shall apply in this Recommendation.

4.2 Test arrangement

By choosing different test equipments, two methods can be used to make hands-free and loudspeaking measurement objectively: the conventional method (using a free-field microphone together with a discrete P.51 artificial mouth) and the HATS method.

The artificial mouth shall conform to [ITU-T P.51].

All measurement values produced by HATS are intended to be free-field equalized. If not specified, HATS's right ear is used for the receive measurement. More details such as the exact calibration and equalization procedures as well as the combination of the two ear signals for the purpose of measurements can be found in [ITU-T P.581].

All effects including test table and measurement equipment are considered a part of the measurement.

NOTE – It is recognized that the two test methods may give different results. While the HATS is intended to simulate the hands-free situation in a more realistic way, the conventional set-up is less sensitive to e.g., asymmetric constructions of devices under test. Differences in the tests results may result from the different orientation of the artificial mouth, from the diffraction effect of the HATS in sending and receiving and from the different physical position of the artificial head compared to the position of the free-field microphone. These effects and their impact on the measurement are still under study.

4.2.1 Desktop hands-free terminal

4.2.1.1 Conventional method

If a free-field microphone together with a discrete P.51 artificial mouth is used, the desktop HFT is placed on a test table according to [ITU-T P.340]. As shown in Figure 1, the mouth axis and the microphone axis are coincident with the straight line drawn between point C and point B. For measurements in the sending direction, the artificial mouth is positioned at point C. For receiving direction, the artificial mouth shall be replaced by the measuring microphone. The centre of the microphone grid shall be positioned at point C, with its axis coincident with line CB.

For TCL and stability loss measurement, the set is positioned but not used.

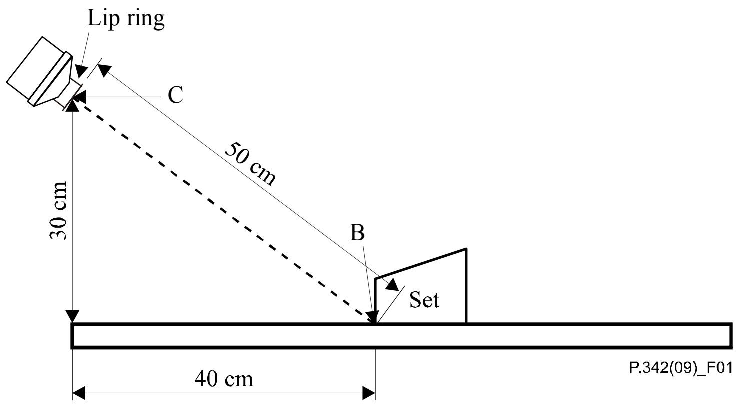


Figure 1 – Measurement configuration for the desktop hands-free terminal using a discrete P.51 artificial mouth and a free-field microphone, side view

4.2.1.2 HATS method

When a HATS is used, the centre of the lip-ring of HATS shall also be located at point C as defined in Figure 1, but the reference axis of the mouth ought to be horizontal, as illustrated in Figure 2.

For TCL and stability loss measurement, the set is positioned but not used.

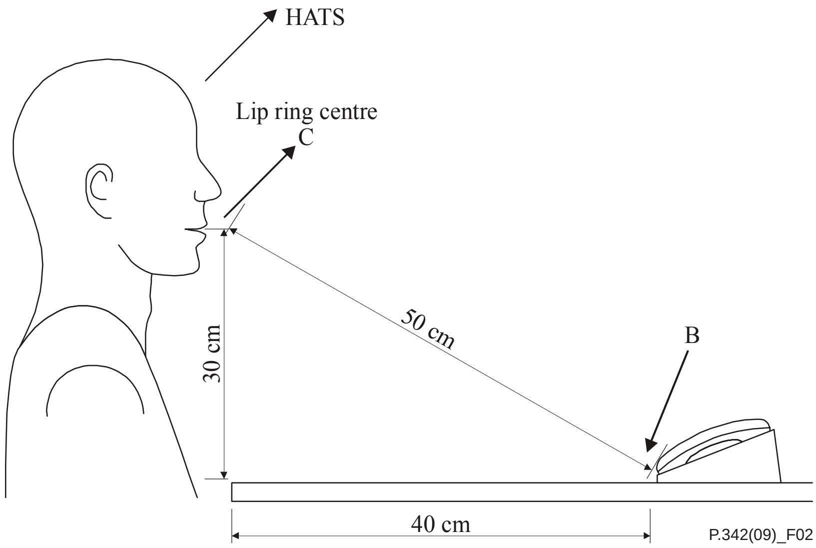


Figure 2 – Measurement configuration for the desktop hands-free terminal using HATS, side view

4.2.2 Double unit desktop speakerphone terminal

For desktop speakerphone terminals with detached microphone and speaker, the standard test position is shown in Figure 3. For desktop speakerphone terminals with more pieces, the test arrangement shall be modified to what it is stated in the instruction manual of the terminal under test (TUT).

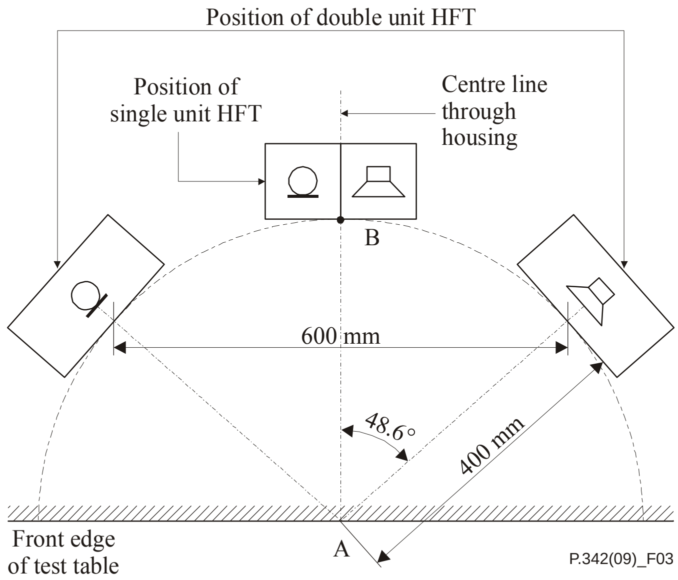


Figure 3 – Measurement configuration for the desktop speakerphone terminal with detached microphone and speaker (top view)

4.2.3 Other types of hands-free terminals

Group-audio terminals or speakerphones designed for non-desktop positioning, for example, videophony and multimedia terminals, should be tested with the appropriate position. This position shall be defined as the recommended test position (RTP). The RTP should be obtained from the manufacturer, and should be based on the product's intended use. If RTP has not been defined, then the HATS HFRP will be point n (far field) from Table 6A of [ITU-T P.58].

Headset terminal positioning is described in [ITU-T P.310].

4.2.4 Loudspeaking function

4.2.4.1 Conventional method

When a free-field microphone together with a discrete P.51 artificial mouth is used, measurements in the sending direction shall be made with the handset placed at LRGP as described in [ITU-T P.64].

Receiving measurements of loudspeaking function are made with the same test position employed in desktop speakerphone terminal measurement, except that the handset is taken off the cradle and placed out of the way during measurement.

TCL measurements of LST are the same except for the positioning of the handset. Figure 4 shows a recommended test position for making echo path loss measurements of LST. The handset earphone "centre" shall be placed at point C with the microphone vertical below the earphone. The meaning of "centre" is the centre of the surface of the handset earphone which is placed normally against the ear. This surface is set at 90 degrees relative to the loudspeaker.

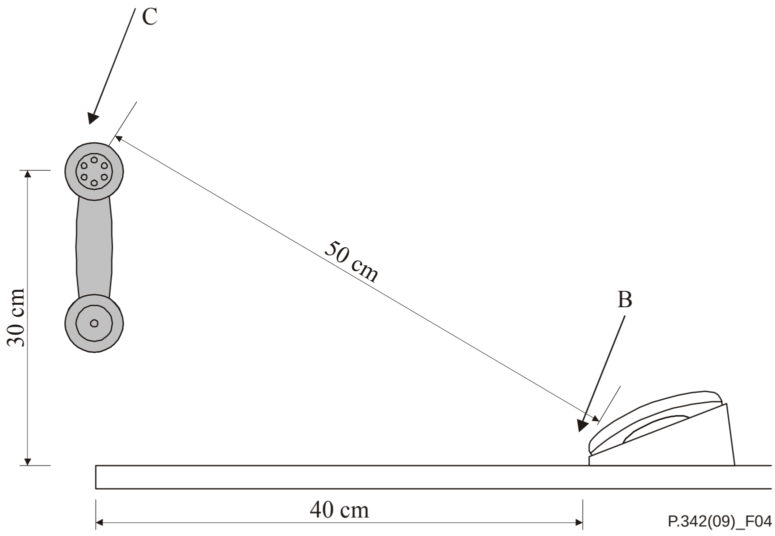


Figure 4 – Standard test position for the LST (side view)

4.2.4.2 HATS method

When a HATS is used, the set shall be positioned as shown in Figure 2. Measurement is performed with one ear while the handset is placed on the other ear. The ear used for measurement will be specified in the test report.

For stability loss measurement, the handset is placed at 50 cm beside the terminal, with the transducers facing the table.

4.3 Test conditions

4.3.1 Test room

1) For the repeatability of the tests, the environment for most of the measurements shall be free-field (anechoic) down to the lowest frequency of the 1/3-octave band centred at 200 Hz.

   Satisfactory free-field conditions exist where errors, due to the departure from ideal conditions, do not exceed the values defined in Table 1, inside a sphere centred at point B in Figure 5, with one metre radius, in the absence of the test table.

Table 1 – Free-field conditions

1/3-octave band centre frequency (Hz)	Allowable departure (dB)
<630	±1.5
800 to 5000	±1
>6300	±1.5

   The test signal level for verification of the free field is –20 dBPa.

   Verification of the free field is made along the seven axes numbered (1) to (7) in Figure 5, with the sound source placed at positions equivalent to B or C, as appropriate. When placed at point B, the artificial mouth reference axis shall be perpendicular to the test table surface. When placed at point C, the artificial mouth reference axis shall be coincident with axis (7). Measurement points along each axis, taken from the front plane of the artificial mouth lip-ring, are at the distances of 315 mm, 400 mm, 500 mm, 630 mm, 800 mm and 1000 mm.

2) The broadband noise level shall not exceed −70 dBPa(A). The octave band noise level shall not exceed the values specified in Table 2.

Table 2 – Octave band 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

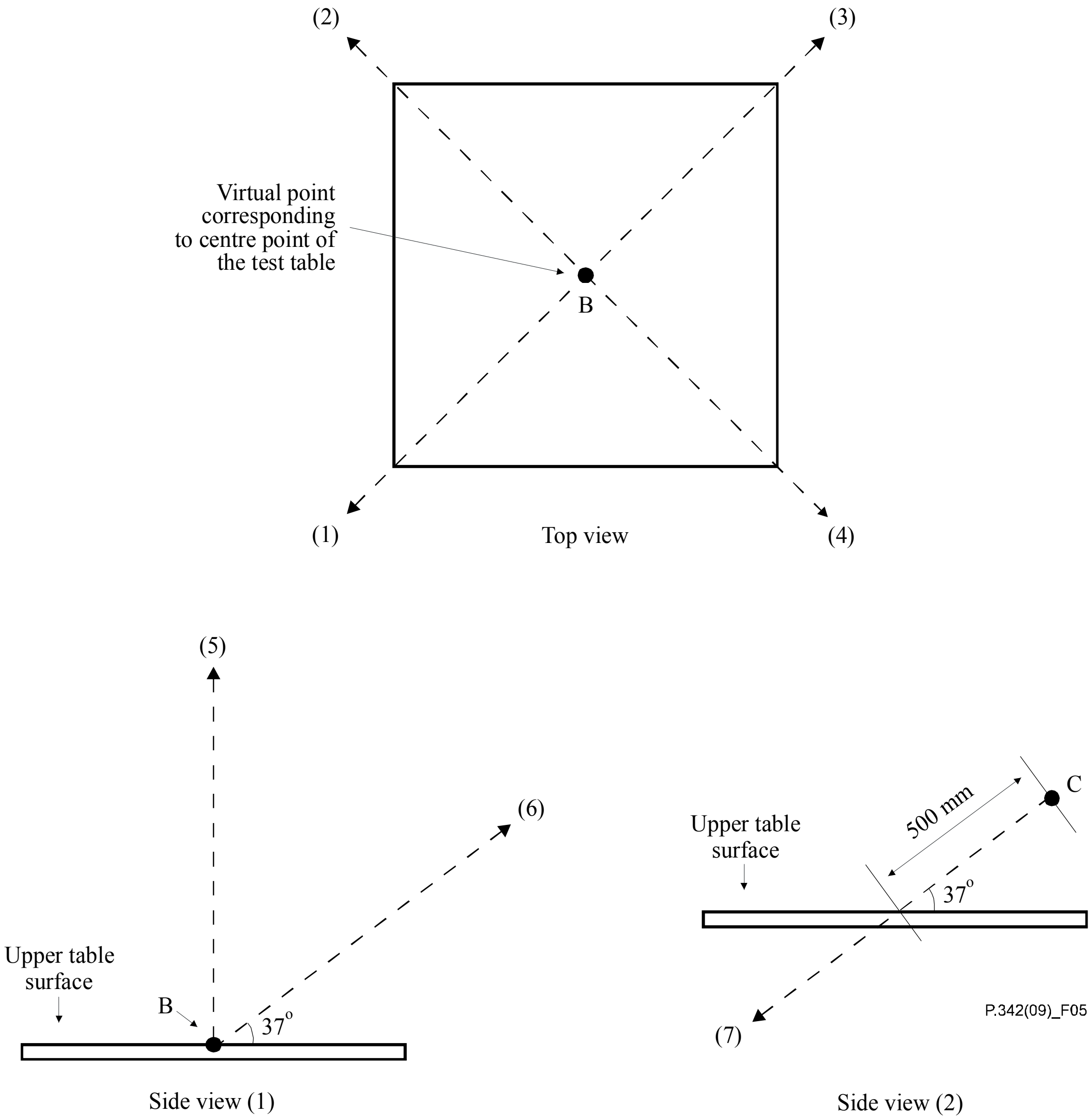
NOTE (informative) – A room including the test arrangement fulfilling the following requirements probably meets the satisfactory conditions.

1) Dimensions of the room: height ≥ 2.2 m; volume V ≥ 30 m^3.

2) The table should be placed horizontally in the centre of the test room and there should be an inclination of 0°~30° between the table and the ceiling.

3) The reverberation time T, measured at points B and C, should satisfy the following inequality:

$T(s) \leq 0.0033\ V\ (m^3)$; which is based on a calculation with the radius of 50 cm.



NOTE 1 – Axes (1) to (7) are used in the determination of free-field conditions for a 1 m radius sphere.
NOTE 2 – Axes (1) to (4) are in the horizontal plane occupied by the test table surface.
NOTE 3 – Axis (5) is perpendicular to the horizontal plane occupied by the test table surface.
NOTE 4 – Measurements of free-field sound pressure are made in the absence of the test table.

Figure 5 – Verification of the free-field conditions

Sound level measurement equipment shall conform to [IEC 61672-1].

4.4 Test signals

The test signal levels specified in this clause are referred to the active part of the signal.

In order to ensure that the test is representative of the normal operation, the test signal has two functions:

– terminal activation; and

– providing the measurement stimulus without adversely affecting the activation.

It shall be checked that both functions are correctly achieved.

Appropriate types of test signal are:

– Switched ON/OFF signals, as defined in clauses 4.4.1 and 4.4.2, at a rate of 250 ms (±5 ms) ON and 150 ms (±5 ms) OFF.
– A complex signal as defined in [ITU-T P.501] (e.g., CSS).

For HFT incorporating adaptative AGC, AEC or other non-linear functions, the results may differ for the two signals.

A complex signal shall be used for equipment incorporating AEC functions and may be used when the switched signals do not activate properly the terminal for all tests described in this Recommendation.

4.4.1 Broadband signal

One possible broadband signal shall be a Gaussian pink noise, with a crest factor of 11 dB ± 1 dB.

The bandwidth of the broadband signal shall correspond to the 14 1/3-octave bands from 200 Hz to 4 kHz.

The 1/3-octave spectrum of electrically generated pink noise shall be equalized within ±1 dB, while the acoustically generated shall be equalized at MRP within ±3 dB.

The slope outside the bandwidth shall be at least 8 dB/1/3-octave.

Broadband signals are used for testing sensitivity/frequency response, loudness ratings, TCL, TCLw and stability.

4.4.2 Sinusoidal and narrow-band signals

– Sinusoidal signals are used for testing harmonic distortion and delay.
– Narrow-band noise signals (100 Hz bandwidth) are used for testing out-of-band signals.

4.5 Test signal levels

4.5.1 Sending

Unless specified otherwise, the test signal level shall be −4.7 dBPa at MRP. The characteristics of the artificial mouth shall be according to [ITU-T P.51].

The output signal from the artificial mouth is calibrated under free-field conditions at MRP, such that the spectrum corresponds to clause 4.4 and the total level in the frequency range corresponding to the 1/3-octave bands from 200 Hz to 4000 Hz is −4.7 dBPa.

The spectrum at MRP is then recorded and the level is adjusted to −28.7 dBPa at the HFRP (HATS HFRP for HATS method).

The spectrum at MRP and the actual level at MRP (measured in 1/3-octaves) are used as references for calculating SLR and response characteristics.

4.5.2 Receiving

Unless specified otherwise, the applied test signal level at the digital input shall be −30 dBm0, as far as the user-controlled receiving volume control is set at its maximum.

For measurements with the volume control at its minimum position, a test signal level of −15 dBm0 shall be used.

4.6 Accuracy of calibrations

Unless specified otherwise, the accuracy of measurements made by test equipment shall be better than:

Table 3 – Measurements accuracy

Item	Accuracy
Electrical signal power	±0.2 dB for levels ≥–50 dBm
Electrical signal power	±0.4 dB for levels ≤–50 dBm
Sound pressure	±0.7 dB
Time	±5%
Frequency	±0.2%

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Table 4 – Signals accuracy

Quantity	Accuracy
Sound pressure level at MRP	±1 dB
Electrical excitation level	±0.4 dB
Frequency generation	±2% (Note)
The measurement results shall be corrected for the measured deviations from the nominal level.	
NOTE – At 4 kHz a tolerance of −2% may be used.	

5 Technical requirements

5.1 Sending characteristics

All sending characteristics are applicable only for hands-free telephones. Sending characteristics for the loudspeaking terminal can be found in [ITU-T P.310].

5.1.1 Sending sensitivity/frequency response

The sending sensitivity/frequency response shall be within the mask shown in Figure 6.

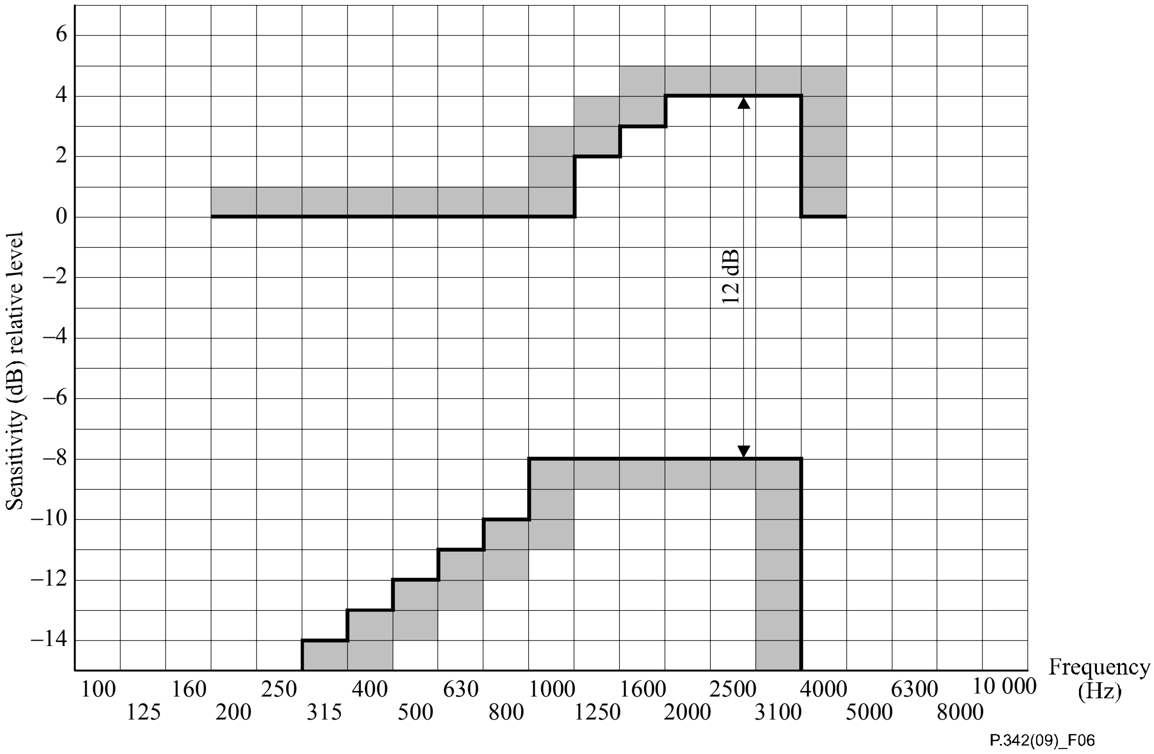


Figure 6 – Sending sensitivity/frequency mask for HFT

All sensitivity values are dB on an arbitrary scale.

Useful information on optimum frequency response can be found in [ITU-T P.340].

5.1.1.1 Measurement method

The set is placed according to clause 4.2. The test signal is specified in clause 4.4 and the test level is adjusted according to clause 4.5.

Measurements are made of the 1/3-octave band levels within 200-4000 Hz defined as band No. 4-17 in [ITU-T P.79].

The sensitivity for each 1/3-octave band is expressed in dBV/Pa (i.e., dB relative to 1 V/Pa) and is defined as:

$$S_{mJ} = 20 \log V_s - 20 \log P_{MRP} + \text{Corr} - 24$$

where:

S_{mJ} is the sending sensitivity from mouth to junction point.

V_s is the measured output voltage (in volts) at the digital interface.

P_{MRP} is the applied sound pressure (in Pa) at MRP.

Corr is 20 log (P_{MRP}/P_{HFRP}) of the used artificial mouth.

NOTE – The value of Corr is the value given in the calibration chart of the artificial mouth (24.0 dB is the ideal value).

5.1.2 Sending loudness rating

The nominal value of SLR shall be +13 dB, with a tolerance of ±3 dB.

This value is derived from [ITU-T P.310]. According to [ITU-T P.340], the SLR of a hands-free telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

5.1.2.1 Measurement method

The SLR shall be calculated according to [ITU-T P.79] using the sending sensitivity values measured in the band No. 4-17 (see clause 5.1.1.1).

5.1.3 Sending noise

The noise produced by the set in the sending path from 100 Hz to 4 kHz shall not exceed –64 dBm0p. No peaks in any 1/3-octave band with the level of 10 dB higher than the average noise spectrum in the frequency domain shall occur.

5.1.3.1 Measurement method

To ensure that the set is correctly stated for the sending direction, the test signal specified in clause 4.4 shall be applied with a level as specified in clause 4.5 for activation.

The set is placed according to clause 4.2.

The noise level shall be measured in a quiet environment (ambient noise less than −64 dBPa(A)) at the digital output with a measurement equipment including psophometric weighting according to [ITU-T G.223], and according to [ITU-T O.41] regarding dynamic requirements.

The idle mode noise shall be measured 500 ms after interrupting the activation signal, averaging over a minimum period of 5 s.

5.1.4 Sending distortion

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

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

Table 5 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 64 kbit/s)

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

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

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

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

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

5.1.4.1 Measurement method

The set is placed according to clause 4.2.

After a correct activation of the system, a sine-wave signal at frequencies of 315, 510 and 1020 Hz is applied at MRP respectively. The signal level shall be calibrated to –4.7 dBPa at MRP for all frequencies, except for the sine-wave signal with a frequency of 1020 Hz that shall be applied at MRP at the following levels: −20, −15, −10, −5, 0, 7, 10 dBPa. The input sound pressure level is limited at +10 dBPa for this measurement.

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

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

5.1.5 Sending out-of-band signals

With any signal above 4.6 kHz and up to 8 kHz, 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 8.

Table 8 – Sending limit for out-of-band signal

Frequency (kHz)	Out-of-band signal limit, sending (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.	

5.1.5.1 Measurement method

The set is placed according to clause 4.2.

For a correct activation of the HFT, an activation signal with the characteristics of the test signals specified in clause 4.4 shall be applied at a level according to clause 4.5.1. Level of this activation signal shall be –4.7 dBPa at MRP. The activation signal acts as the reference signal.

The out-of-band test signal applied shall be a narrow-band signal (100 Hz bandwidth) centred on 4.65 kHz, 5 kHz, 6 kHz, 6.5 kHz, 7 kHz and 7.5 kHz, respectively, and with a level according to clause 4.5.1. The levels of these signals shall be –4.7 dBPa at MRP.

The complete test signal is constituted by t_1 ms of activation signal, t_2 ms of out-of-band test signal and another time t_1 ms of activation signal, where t_1 may be 250 ms and t_2 should be <150 ms.

The output level of the activation signal (during t_1) and the output level of any image frequency (during t_2) shall be measured at the digital interface. The time t_2 depends on the integration time of the analyser.

The observation of the output signal during sending of the activation signal permits to control if the set is correctly activated.

5.2 Receiving characteristics

All receiving characteristics are applicable for both hands-free and loudspeaking terminals.

Where a user-controlled receiving volume control is provided, the recommended values apply when the volume control is set at its maximum, unless stated otherwise.

5.2.1 Receiving sensitivity/frequency response

The receiving sensitivity/frequency response shall be within the masks drawn in Figure 7.

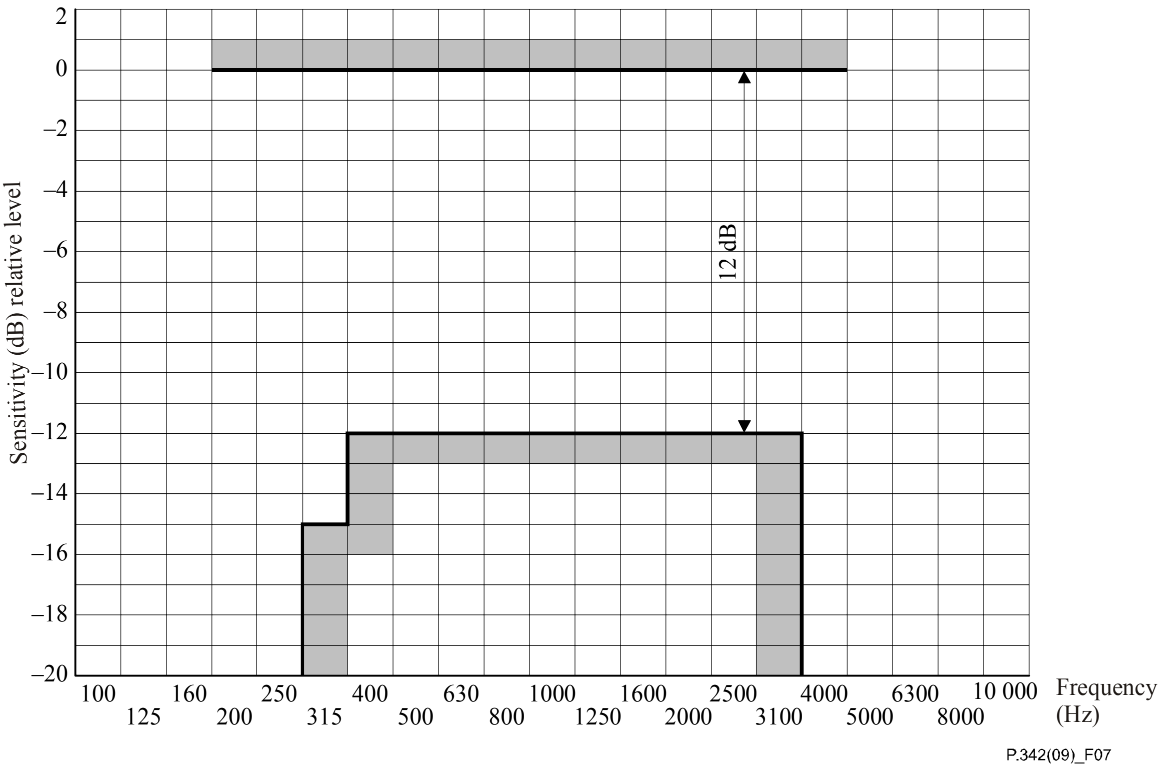


Figure 7 – Receiving sensitivity/frequency response

All sensitivities are dB on an arbitrary scale.

The optimum frequency response is a flat curve between 300 and 3400 Hz.

5.2.1.1 Measurement method

The set is placed according to clause 4.2. The test signal is specified in clause 4.4 and the test level is adjusted according to clause 4.5.

When a free-field microphone is used as the receiving test equipment, the sensitivity for each 1/3-octave band is expressed in dBPa/V (i.e., dB relative to 1 Pa/V), and is defined as:

$$S_{Je} = 20 \log P_{MP} - 20 \log V_r$$

where:

S_{Je} is the receiving sensitivity from junction point to the measuring microphone.

P_{MP} is the measured sound pressure (in Pa) at the microphone position.

V_r is the voltage (in volts) applied at the digital interface.

When a HATS is used instead of a free-field microphone, the sensitivity S_{Je} shall be substituted by S_{Jeff}, which is defined as:

$$S_{Jeff} = 20 \log P_{eff} - 20 \log V_r$$

where:

S_{Jeff} is the receiving sensitivity from junction to HATS ear with free-field correction.

P_{eff} is the measured sound pressure converted from drum reference point (DRP) to free-field.

In case of devices not provided with manual volume control, the measurement is repeated for excitation levels of –30 dBm0 and –15 dBm0.

NOTE – RLR is checked at –30 dBm0 (nominal input level) and –15 dBm0 excitation level to ensure linearity in this range of levels. This is required since some measurements require –15 dBm0 excitation level to ensure a proper measurement (e.g., TCLw). However, these measurements are referred to nominal SLR and RLR values where such linearity is assumed. If a difference between measurements at –30 dBm0 and –15 dBm0 excitation level is detected, the results need to be corrected accordingly.

Example:

TCLw (measured) = 30 dB

RLR (–30 dBm0) = 2 dB; RLR (–15 dBm0) = 4 dB; → Difference on RLR = 2 dB

⇒ TCLw (measured) + Difference on RLR = 30 dB + 2 dB = 32 dB = TCLw

5.2.2 Receiving loudness rating

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

The RLR value shall be met for at least one setting of the volume control (when manually operated).

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.

5.2.2.1 Measurement method

The RLR(cal) shall be calculated according to [ITU-T P.79] using the receiving sensitivity values measured in the band No. 4-17 (see clause 5.2.1.1).

The RLR shall then be computed as RLR(cal) minus 14 dB (according to [ITU-T P.340]), and without the L_E factor. To compute RLR using the combination of left and right ear signals from HATS, the correction factor has to be 8 dB, instead of 14 dB.

Where a user-controlled receiving volume control is provided, it is necessary to verify that the nominal RLR value is met for at least one setting of this control. This implicates that the RLR has to be determined not only at the maximum setting but at least also at the minimum setting of the volume control. Accordingly, the receiving sensitivity (clause 5.2.1) has to be measured at least at this setting. Test signal level is specified in clause 4.5.2.

5.2.3 Receiving noise

5.2.3.1 A-weighted

The noise level shall not exceed –49 dBPa(A).

5.2.3.2 1/3-octave band spectrum

The level in any 1/3-octave band, between 100 Hz and 10 kHz shall not exceed a value of –59 dBPa.

5.2.3.2.1 Measurement method

To ensure that the set is correctly stated for the receiving direction, the test signal specified in clause 4.4 shall be applied with a level as specified in clause 4.5 for activation.

The set is placed according to clause 4.2.

The noise shall be measured 500 ms after interrupting the activation signal, averaging over a minimum period of 5 s.

A weighting is specified in [IEC 61672-1].

5.2.4 Receiving distortion

The ratio of signal-to-total distortion shall be above the mask defined in Tables 9, 10 and 11 for [ITU-T G.711] (64 kbit/s), [ITU-T G.711] (56 kbit/s) and [ITU-T G.726] (32 kbit/s) respectively, unless the signal in the artificial ear exceeds +10 dBPa or is less than –50 dBPa.

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

Table 9 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 64 kbit/s)

Frequency (Hz)	Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
315	–16	26
510	–16	30.5
1020	–30	30.5
	–20	33.0
	–10	33.5
	–3	31.2
	0	25.5

Table 10 – Limits for signal-to-total distortion ratio ([ITU-T G.711], 56 kbit/s)

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
–30	27.4
–20	28.4
–10	28.6
–3	27.7
0	24.2

Table 11 – Limits for signal-to-total distortion ratio ([ITU-T G.726], 32 kbit/s)

Receiving level at the digital interface (dBm0)	Receiving ratio (dB)
–30	29.2
–20	30.9
–10	31.2
–3	29.7
0	25.0

5.2.4.1 Measurement method

The set is placed according to clause 4.2.

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

The total distortion power shall be measured at ERP with A-weighting. The DRP-ERP correction as defined in [ITU-T P.57] is applied. The ratio of the signal-to-total distortion power is calculated. The duration of the sine-wave shall be less than 1 s. Appropriate signals for activation and signal combinations can be found in [ITU-T P.501].

5.2.5 Receiving out-of-band signals

Any spurious out-of-band image signals in the frequency range from 4.6 to 8 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in Table 12.

Table 12 – Receiving limit for out-of-band signal

Frequency (kHz)	Out-of-band signal limit, receiving (dB)
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.	

5.2.5.1 Measurement method

The set is placed according to clause 4.2.

For a correct activation of the HFT, an activation signal with characteristics as the test signals specified in clause 4.4 shall be applied at a level according to clause 4.5.2.

For input narrow-band signals centred on 500 Hz, 1000 Hz, 2000 Hz and 3150 Hz, applied at the level of –30 dBm0, the level of any out-of-band signals at frequencies up to 8 kHz shall be measured selectively.

5.3 Echo path loss characteristics

5.3.1 Terminal coupling loss

The weighted terminal coupling loss (TCLw) should be greater than 40 dB when measured under field conditions and with SLR normalized to SLR = +13 dB and RLR = +2 dB. For example, if the measured TCLw is 42 dB, the measured SLR is +16 dB and the measured RLR is 0 dB, then the normalized value of TCLw = 42 dB + (13 – 16) dB + (2 – 0) dB = 41 dB.

However, in order to meet ITU-T G.131 talker echo objective requirements, a TCLw greater than 45 dB is desirable and should be striven for.

NOTE – The perceived echo impairment, by the person at the opposite end of the connection from a telephone set that has a TCLw of less than 45 dB, is a function of the magnitude of the talker echo signal as well as the talker echo path delay. A telephone set that has a TCLw of less than 45 dB will provide an echo signal that becomes more disturbing as the talker echo path delay increases. Thus, a telephone set that has a TCLw of less that 45 dB may provide satisfactory performance on low delay connections while the same may not be true for connections that have a long delay.

It is assumed that this requirement is met if TCL and TCLw, respectively, meet the values of Table 13 with the receive volume control in its maximum setting.

Table 13 – Terminal coupling loss for maximum volume setting

TCL (1/3-octave band)	TCLw
>25 dB	>35 dB
NOTE – These values assume no other echo control in the connection.	

If information is available in the terminal about the one-way transmission time of the connection, and if the terminal operates in double talk, then the limits defined in Table 14 may apply.

Table 14 – Possible terminal coupling loss vhen one-way transmission time is known

	One-way transmission time	TCLw
Single talk	≤10 ms	≥25 dB
Double talk	≤10 ms	≥19 dB (Note)
NOTE – To achieve MOS ≥ 4. Further information is found in [ITU-T P.340].		

5.3.1.1 Measurement method

The set is placed according to clause 4.2.

The test signal is specified in clause 4.4.

The test signal level shall be –15 dBm0.

TCL shall be measured as attenuation from the digital input to the digital output, at the 14 1/3-octave bands between 200 Hz and 4 kHz.

The TCLw (before normalization) shall be calculated from [ITU-T G.122], with the following formula:

$$TCLw = -10\log_{10}\left(\frac{1}{14}\sum_{i=1}^{14} A_i\right)$$

where A_i is the output/input power ratio at the i-th 1/3-octave band.

5.3.2 Stability loss

Stability loss is defined as the minimum loss from the digital input (receive) to the digital output (send), at any test frequency. The attenuation from the digital input to the digital output shall be, at any time, at least 6 dB, for all frequencies in the range of 200 Hz to 4 kHz.

5.3.2.1 Measurement method

The set is placed according to clause 4.2.

The test signal is specified in clause 4.4.

The test signal level shall be –15 dBm0.

Stability loss shall be measured as attenuation from the digital input to the digital output by a selective analyser with a bandwidth of 80 Hz ± 10 Hz, between 200 Hz and 4 kHz.

5.4 Delay

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

Measurements shall be performed on the two paths separately. The total delay is the summation of these two values.

5.4.1 Measurement method

5.4.1.1 Sending delay measurement

The test signal to be used for the measurements shall be a composite source signal (CSS) as described in [ITU-T P.501]. The hands-free terminal is set up as described in clause 4.2. The delay is calculated using the cross-correlation function between the signal at the electrical test point and the signal at MRP. The measurement is corrected by the delay introduced by the test equipment. The delay is expressed in milliseconds, determined from the maximum of the cross-correlation function.

5.4.1.2 Receiving delay measurement

The test signal to be used for the measurements shall be a composite source signal (CSS) as described in [ITU-T P.501]. The handset terminal is set-up as described in clause 4.2. The delay is calculated using the cross-correlation function between the signal at the electrical test point and the signal acquired by the free-field microphone or artificial ear. The measurement is corrected by the delay introduced by the test equipment. The delay is expressed in milliseconds, determined from the maximum of the cross-correlation function.

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Printed in Switzerland
Geneva, 2010