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

Subscribers' lines and sets

Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals

ITU-T Recommendation P.342

(Formerly CCITT Recommendation)

ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

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ITU-T Recommendation P.342

Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals

Summary

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

This ITU-T 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

ITU-T Recommendation P.342 was revised by ITU-T Study Group 12 (1997-2000) and approved under the WTSC Resolution 1 procedure on 18 May 2000.

FOREWORD

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NOTE

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Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals

1 Scope

This ITU-T Recommendation provides audio performance requirements for loudspeaking and hands-free telephony terminals using, in the telephone band (300-3400 Hz), the waveform encoding according to ITU-T Recommendations G.711 [3] (PCM at both 64 kbit/s and 56 kbit/s) and G.726 [11] (ADPCM 32 kbit/s).

The test methods are described in Annex A.

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 Normative 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; all 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.

[1] ITU-T Recommendation P.310 (2000), *Transmission characteristics for telephone-band (300-3400 Hz) digital telephones*.

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

[3] CCITT Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies*.

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

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

[6] ITU-T Recommendation P.79 (1993), *Calculation of loudness ratings for telephone sets*.

[7] ISO 266:1977, *Acoustics – Preferred frequencies*.

[8] IEC 60651:1979, *Sound level metres*.

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

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

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

[12] ITU-T Recommendation G.131 (1996), *Control of talker echo*.

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

3 Definitions and abbreviations

3.1 Definitions

This ITU-T Recommendation defines the following terms:

3.1.1 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 Recommendation P.51 [5].

3.1.2 hands-free (telephone) (HFT) set: A telephone set using a loudspeaker associated with an amplifier as a telephone receiver and which may be used without a handset.

3.1.3 loudspeaking (telephone) (LST) set: A handset telephone using a loudspeaker associated with an amplifier as a telephone receiver.

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

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

3.2 Abbreviations

This ITU-T Recommendation uses the following abbreviations:

AEC Acoustic Echo Control

AGC Automatic Gain Control

CSS Composite Source Signal

LRGP Loudness Rating Guard-ring Position

MRP Mouth Reference Point

RLR Receiving Loudness Rating

SLR Sending Loudness Rating

TCL Terminal Coupling Loss

TCLw Weighted Terminal Coupling Loss

4 Sending characteristics

All sending characteristics are applicable only for hands-free telephones.

4.1 Sending loudness rating

The nominal value of SLR shall be +13 dB.

This value is derived from ITU-T Recommendation P.310 [1]. According to ITU-T Recommendation P.340 [4], the SLR of a hands-free telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

4.2 Sensitivity/frequency response

The sending sensitivity/frequency response shall be within the mask shown in Figure 1.

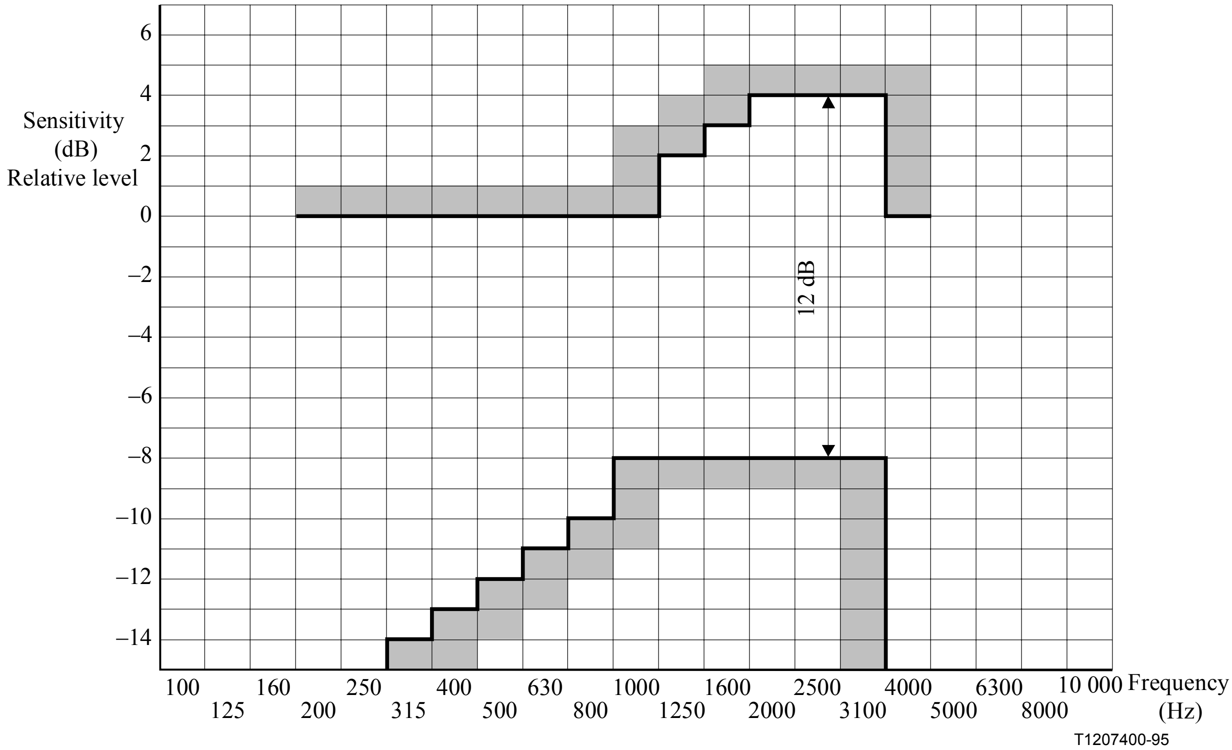


Figure 1/P.342 – Sending sensitivity/frequency mask for HFT

All sensitivity values are dB on an arbitrary scale.

Useful information on optimum frequency response can be found in ITU-T Recommendation P.340 [4].

4.3 Noise

The noise produced by the set in the sending path shall not exceed –64 dBm0p.

4.4 Harmonic distortion

The ratio of signal to harmonic distortion shall be above the mask defined in Table 1.

Table 1/P.342

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.	

4.5 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 2.

Table 2/P.342

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 Receiving characteristics

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.

5.1 Receiving loudness rating

The nominal value of RLR shall be +2 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 Recommendation P.310 [1]. According to ITU-T Recommendation P.340 [4], 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 Sensitivity/frequency response

The receiving sensitivity/frequency response shall be within the masks drawn in Figure 2.

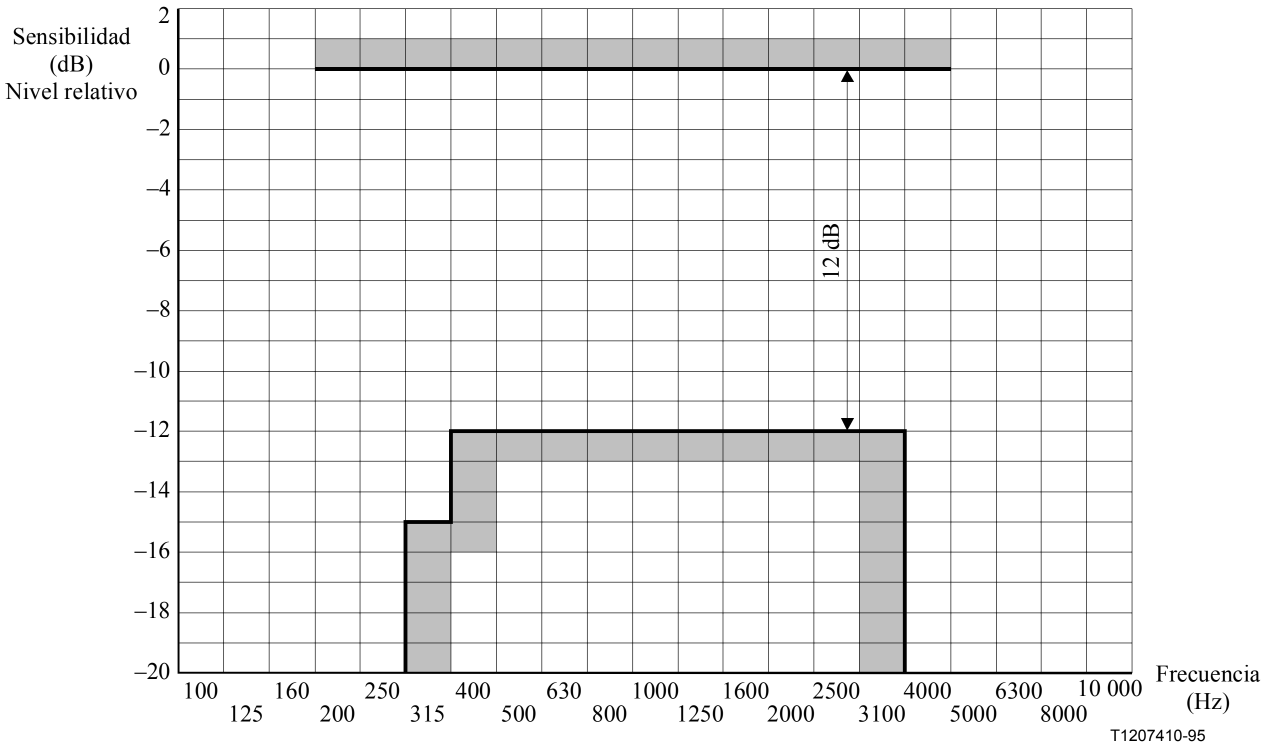


Figure 2/P.342 – Receiving sensitivity/frequency response

All sensitivities are dB on an arbitrary scale.

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

5.3 Noise

5.3.1 A-weighted

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

5.3.2 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.

5.4 Harmonic distortion

The ratio of signal to harmonic distortion shall be above the mask defined in Table 3.

Table 3/P.342

Frequency (Hz)	Signal to distortion ratio limit, receiving (dB)	
	Hands-free function	Loudspeaking function
315	26	20
400	26	26
500	30.5	30.5
1000	30.5	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 signal frequencies are limited at 1 kHz. Limits above 1 kHz are for further study.

5.5 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 4.

Table 4/P.342

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.	

6 Echo path loss characteristics

6.1 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 to 16) dB + (2 to 0) dB = 41 dB.

However, in order to meet G.131 [12] 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 then 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 5 with the receive volume control in its maximum setting.

Table 5/P.342

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 6 may apply.

Table 6/P.342

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

6.2 Stability loss

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.

7 Delay

The total delay shall be less than 8 ms (5 ms for the telephone set to allow digital signal processing and 3 ms for the air path) for digital telephones using G.711 [3] coding, and 8.75 ms for digital telephones using G.726 [11] coding.

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

NOTE – An extra delay could result from the AEC processing in the processing unit. For end-to-end digital communications, the delay should be no more than 16 ms in each direction of speech transmission (see Appendix II in [4]).

ANNEX A

Test methods

A.1 Electrical interface specifications

Subclauses B.2 to B.5/P.310 [1] shall apply in this ITU-T Recommendation.

A.2 Test conditions

A.2.1 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 A.1, inside a sphere centred at point B in Figure A.1, with one metre radius, in absence of the test table.

Table A.1/P.342

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 which are numbered (1) to (7) in Figure A.1, 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 the axis (7). Measurement points along each axis, taken from the front plan 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 A.2.

Table A.2/P.342

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

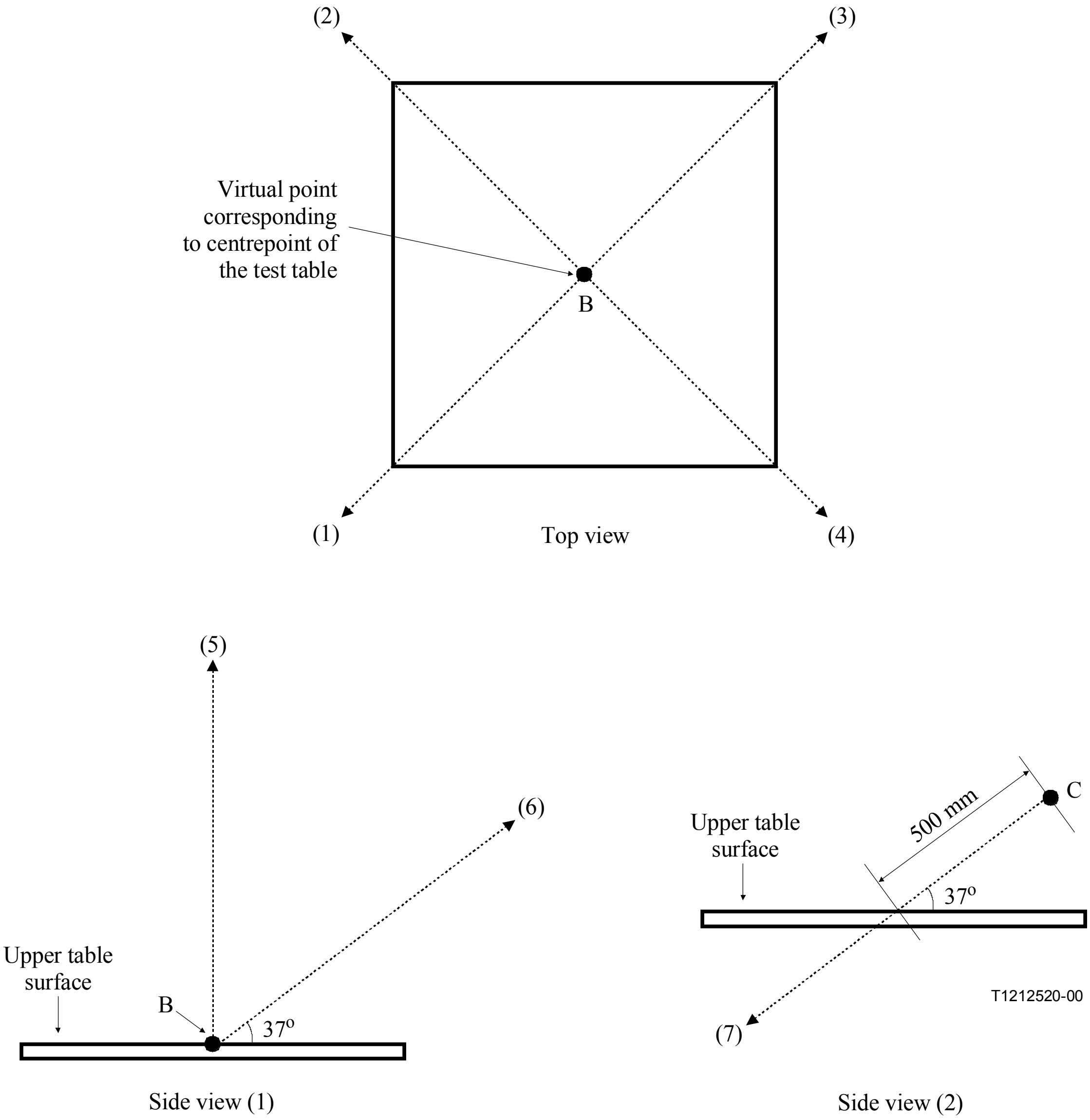
NOTE (informative) – A room including the test arrangement fulfilling the following requirements probably meets the satisfactory conditions.

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 ~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) ≤ 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 A.1/P.342 – Verification of the free-field conditions

A.2.2 Test arrangement

A.2.2.1 Hands-free terminal

The HFT is placed on a test table according to ITU-T Recommendation P.340 [4].

The artificial mouth axis and the microphone axis are coincident with the straight line drawn between point C and point B (see Figure A.2).

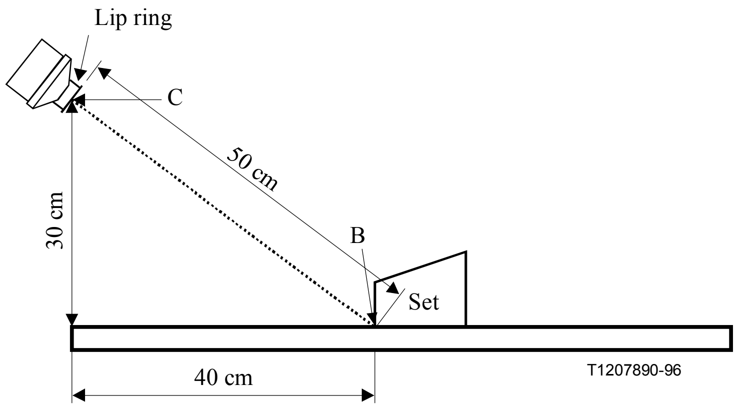


Figure A.2/P.342 – Measurement configuration

For stability control, the different pieces of the HFT (if the HFT is built in two or more pieces) shall be placed as close as possible to each other, but without modifying the normal use of the HFT.

NOTE – If HFT is implemented in two or more pieces, care should be taken to ensure that the test arrangement does not modify the normal use of the HFT. The case of special terminals (multifunctions, etc.) including the hands-free function is for further study.

A.2.2.2 Loudspeaking function

The set is placed on a test table according to ITU-T Recommendation P.340 [4].

For TCL measurements, 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.

For stability measurements, the handset shall be placed as defined in ITU-T Recommendation P.310 [1].

The set shall be placed symmetrically to the axis of the handset. The front side of the terminal is directed towards the corner formed by the three surfaces with its front edge at a distance of 1 m from this corner.

For the test of all characteristics except TCL and stability loss, the handset shall be placed in the LRGP on the test head. The centre of the artificial mouth lip-ring shall be placed at point D, as shown in Figure A.3.

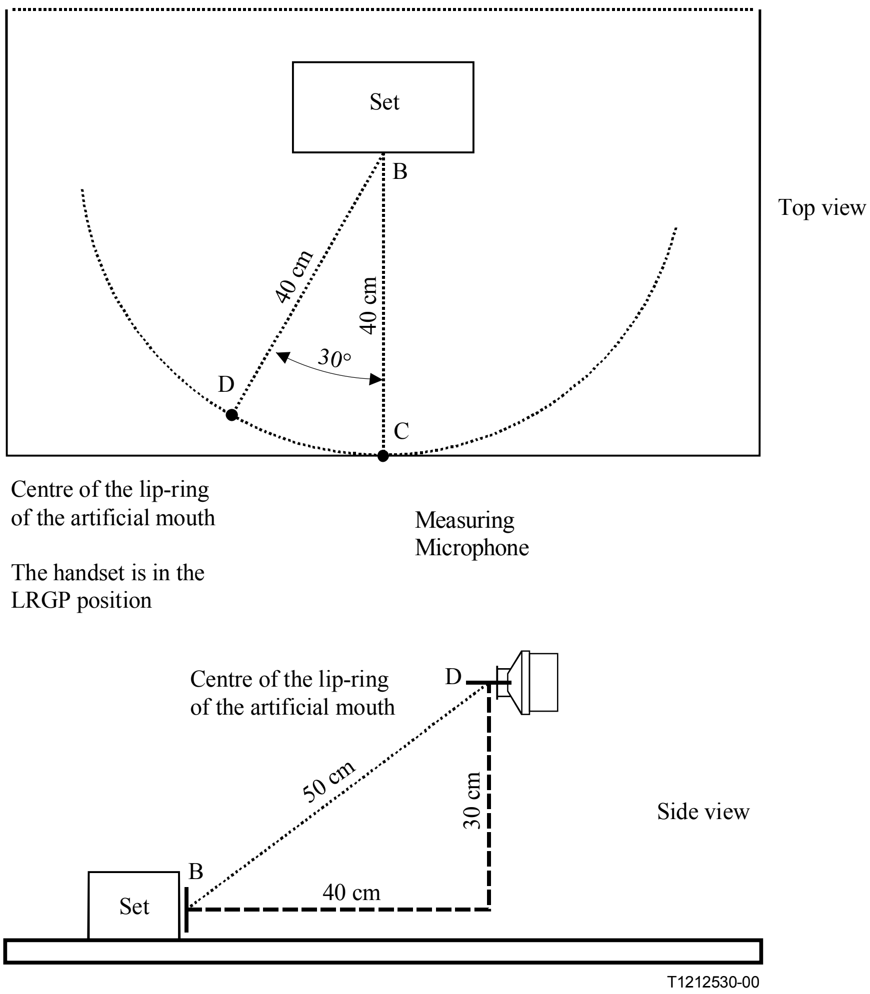


Figure A.3/P.342 – Measurement position, LST

A.2.3 Electroacoustic equipment

The artificial mouth shall conform to ITU-T Recommendation P.51 [5].

Sound level measurement equipment shall conform to IEC 60651 [8].

A.2.4 Test signals

The test signal levels specified in this annex 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 A.2.4.1 and A.2.4.2, at a rate of 250 ms (± 5 ms) ON and 150 ms (± 5 ms) OFF.

– A complex signal as defined in ITU-T Recommendation P.501 [13] (e.g. CSS).

For HFT incorporating adaptative AGC, AEC or other non-linear functions, the results may differ with 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 annex.

A.2.4.1 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 the 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.

A.2.4.2 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.

A.2.5 Test signal levels

A.2.5.1 Sending

Unless specified otherwise, the test signal level shall be –4.7 dBPa at the MRP. The characteristics of the artificial mouth shall be according to ITU-T Recommendation P.51 [5].

The output signal from the artificial mouth is calibrated under free field conditions at the MRP, such that the spectrum corresponds to A.2.4 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 the MRP is then recorded and the level is adjusted to −28.7 dBPa at the HFRP.

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

A.2.5.2 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.

A.2.6 Accuracy of calibrations

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

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:

Quantity	Accuracy
Sound pressure level at the 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.	

A.3 Transmission requirements testing

Unless stated otherwise, the tests are performed with the volume control (if provided) at its maximum.

A.3.1 Sensitivity/frequency response

The set is placed according to A.2.2. The test signal is specified in A.2.4 and the test level is adjusted according to A.2.5.

Measurements are made of the 1/3-octave band levels within 200-4000 Hz defined as band No. 4-17 in ITU-T Recommendation P.79 [6].

A.3.1.1 Sending

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:

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

where:

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

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

A.3.1.2 Receiving

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:

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.

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

A.3.2 Loudness ratings

A.3.2.1 Sending Loudness Rating (SLR)

The SLR shall be calculated according to ITU-T Recommendation P.79 [6] using the sending sensitivity values measured in the band No. 4-17 (see A.3.1.1).

A.3.2.2 Receiving Loudness Rating (RLR)

The RLR(cal) shall be calculated according to ITU-T Recommendation P.79 [6] using the receiving sensitivity values measured in the band No. 4-17 (see A.3.1.2).

The RLR shall then be computed as RLR(cal) –14 dB (according to ITU-T Recommendation P.340 [4]), and without the L_E factor.

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 (A.3.1.2) has to be measured at least at this setting. Test signal level is specified in A.2.5.2.

A.3.3 Terminal coupling loss

The set is placed according to A.2.2.

The test signal is specified in A.2.4.

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 Recommendation G.122 [2], 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.

A.3.4 Stability loss

The test signal is specified in A.2.4.

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.

A.3.5 Harmonic distortion

The set is placed according to A.2.2.

The signal to harmonic distortion ratio is measured with sinusoidal test signals at 315, 400, 500, 630, 800 and 1000 Hz, switched ON/OFF at zero crossings at a rate defined in A.2.4.

The harmonics are measured selectively up to 3.15 kHz.

A.3.5.1 Sending

The sinusoidal signal level, calibrated at Hands-Free Reference Point (HFRP), shall be −20 dBPa.

A.3.5.2 Receiving

The sinusoidal signal level is calibrated to −20 dBm0.

A.3.6 Out-of-band signals

The set is placed according to A.2.2 test arrangement.

A.3.6.1 Sending

For a correct activation of the HFT, an activation signal with characteristics as the test signals specified in A.2.4 shall be applied at a level according to A.2.5.1.

The out-of-band test signal applied shall be a narrowband 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 A.2.5.1.

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.

A.3.6.2 Receiving

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.

A.3.7 Noise

To ensure that the set is correctly stated for the sending direction and the receiving direction respectively, the test signal specified in A.2.4 shall be applied with a level as specified in A.2.5 for activation.

The set is placed according to A.2.2.

A.3.7.1 Sending

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 Recommendation G.223 [9], and according to ITU-T Recommendation O.41 [10] regarding dynamic requirements.

The idle mode noise shall be measured 500 ms after interrupting the activation signal.

A.3.7.2 Receiving

The noise shall be measured 500 ms after interrupting the activation signal.

A weighting is specified in IEC 60651 [8].

A.3.8 Delay

The following method is defined in ITU-T Recommendation P.310 [1].

The test arrangement is shown in Figure A.4.

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

Measurements shall be made with pairs of sine wave signals.

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

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

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

The measurements are executed in the following steps:

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

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

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

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

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

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

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

where:

D_e is the audio group delay of the test equipment.

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

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

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

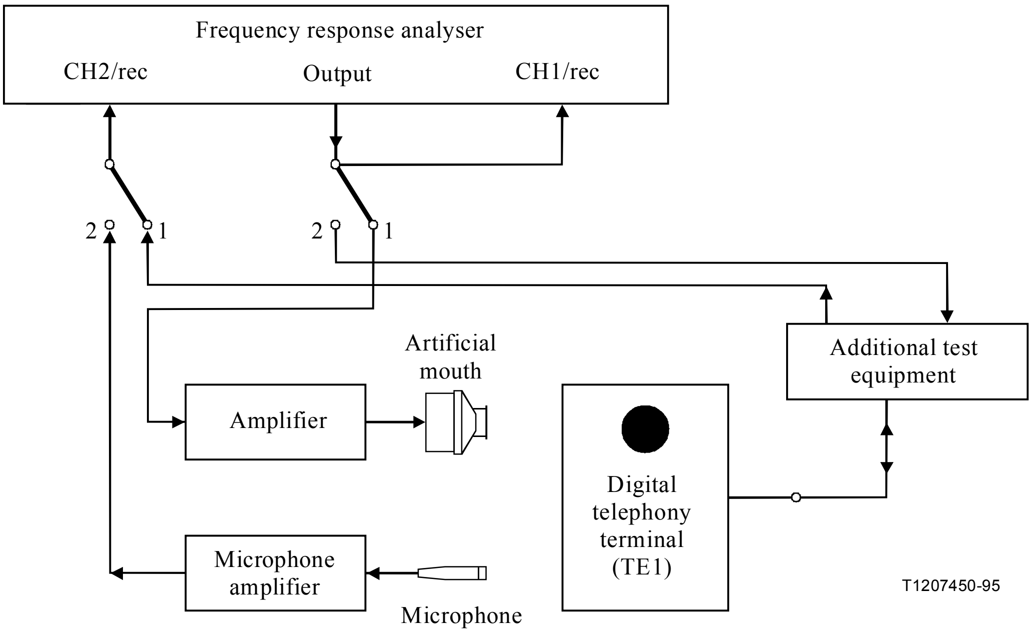


Figure A.4/P.342 – Configuration for delay measurements

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