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

Subscribers' lines and sets

Transmission characteristics for wideband (150-7000 Hz) digital hands-free telephony terminals

ITU-T Recommendation P.341

ITU-T P-SERIES RECOMMENDATIONS

TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

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ITU-T Recommendation P.341

Transmission characteristics for wideband (150-7000 Hz) digital hands-free telephony terminals

Summary

This revised Recommendation provides audio performance requirements for wideband audio (7 kHz) hands-free telephones. Associated test methods for verifying wideband audio performance are contained in Annex A.

Requirements and test methods are specified for the major audio transmission parameters affecting wideband audio, including levels, frequency response, noise, distortion, spurious signals and echo path. Wideband audio represents a considerable departure from traditional telephony, offering significantly improved quality.

The main revision encompassed by this version of the Recommendation is to adopt the wideband loudness rating algorithm as in Annex G/P.79.

Source

ITU-T Recommendation P.341 was approved on 6 June 2005 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. 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.

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

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, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation P.341

Transmission characteristics for wideband (150-7000 Hz) digital hands-free telephony terminals

1 Scope

This Recommendation provides audio performance requirements and test methods for hands-free telephones capable of transmitting an audio bandwidth extending beyond the conventional telephony bandwidth of 300 to 3400 Hz, to a bandwidth of approximately 150 to 7000 Hz. Such telephones are known as wideband audio telephones, and will make use of digital encoding schemes such as in ITU-T Rec. G.722 [1]. Wideband audio telephones are expected to be used in new services such as high quality audio conferencing, videoconferencing and multimedia applications.

The requirements listed in this Recommendation are primarily applicable to telephones using G.722 [1] encoding at 64 kbit/s, but should also be used as the basis of requirements for other wideband audio encoding schemes. This is still under study in ITU-T Study Group 12.

The measurement method for delay is still under study.

General information on HFTs, which includes switching characteristics, can be found in ITU-T Rec. P.340 [3] and information on acoustic echo controllers in ITU-T Rec. G.167 [16].

For loudspeaking telephones (see ITU-T Rec. P.10 [15]) which do not provide full hands-free operation, the relevant parts of this Recommendation may be used.

Conventional telephone band (300-3400 Hz) digital hands-free telephones using encoding according to ITU-T Recs G.711 [12] and G.726 [13] are covered by ITU-T Rec. P.342 [7].

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; 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.

[1] ITU-T Recommendation G.722 (1988), *7 kHz audio coding within 64 kbit/s*.

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

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

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

[5] ITU-T Recommendation P.57 (2002), *Artificial ears*.

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

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

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

[9] IEC 61672-2 (2003), *Electroacoustics – Sound level meters – Part 2: Pattern evaluation tests*.

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

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

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

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

[14] ITU-T Recommendation P.311 (2005), *Transmission characteristics for wideband (150-7000 Hz) digital handset telephones*.

[15] ITU-T Recommendation P.10 (1998), *Vocabulary of terms on telephone transmission quality and telephone sets*.

[16] ITU-T Recommendation G.167 (1993), *Acoustic echo controllers*.

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

3 Definitions and abbreviations

This Recommendation defines the following terms:

3.1 Acoustic Reference Level (ARL): The acoustic level at MRP which results in a –10 dBm0 output at the digital interface.

3.2 Hands-free Reference Point (HFRP): 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 measurement point No. 11 defined in ITU-T Rec. P.51 [4].

This Recommendation also uses the following abbreviations.

Relevant abbreviations in ITU-T Rec. P.10 [15] will also apply:

CSS Composite Source Signal

HFT Hands-free Terminal

MRP Mouth Reference Point

RLR Receiving Loudness Rating

SLR Sending Loudness Rating

TCL Terminal Coupling Loss

TCLw Weighted Terminal Coupling Loss

4 Sending characteristics

4.1 Loudness rating

Following the approach used for narrow-band hands-free telephones in ITU-T Rec. P.340 [3], the levels in the hands-free send direction are related to those in wideband handset mode (see ITU-T Rec. P.311 [14]) with an allowance of 5 dB for higher talking levels and the difference in speaking position. The value of SLR shall therefore be +9 dB, measured in terms of wideband loudness rating according to Annex G/P.79 [8].

NOTE – The overload point for wideband audio is currently defined as +9 dBm0 [1]. Should this overload point be changed in future revisions of [1], then the Loudness Rating requirements of this Recommendation should also be revised accordingly. The same concept applies in case this Recommendation is used for specifying the electroacoustic requirements of digital telephone sets using wideband audio coders with a different overload point.

4.2 Sensitivity/frequency characteristics

The sending sensitivity/frequency characteristics shall fall between the upper and lower limits given in Table 1, and shown in Figure 1. All sensitivities are in dB on an arbitrary scale.

Table 1/P.341 – Sending sensitivity/frequency characteristics

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	–∞
125	4	–7
200	4	–4
1000	4	–4
5000	(Note)	–4
6300	9	–7
8000	9	–∞
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.		

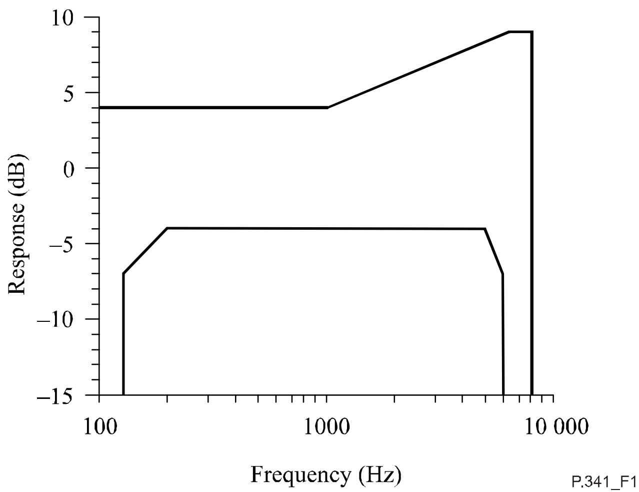


Figure 1/P.341 – Hands-free sending characteristic

4.3 Noise

With the microphone acoustically muted (equivalent to an ambient noise level of <30 dBA), the noise in the sending direction at the digital interface shall not exceed –68 dBm0 (A-weighted).

4.4 Distortion

The distortion in the sending direction shall be measured in terms of the total distortion (harmonic and quantizing) arising from the application of 200 Hz, 1 kHz and 6 kHz tones applied separately. The limits shall be as shown in Table 2.

Table 2/P.341 – Distortion in sending direction

Sending level (dB re ARL)	Signal-to-distortion ratio limit (dB)		
	200 Hz	**1 kHz**	**6 kHz**
+18 to –20	29.0	35.0	29.0
–30	25.0	26.5	25.0
–46	11.0	12.5	11.0

NOTE 1 – These limits only apply up to the maximum sound pressure level which can be produced by the artificial mouth (+10 dBPa).

NOTE 2 – The limits for signal-to-total distortion ration for intermediate sending levels lie on straight lines drawn between the given values on a linear (dB sending level) – linear (dB ratio) scale.

4.5 Discrimination against out-of-band input signals

The level of any in-band image frequencies resulting from application of input signals at 8 kHz and above shall be attenuated by at least 25 dB compared to the output level of a 1-kHz input signal.

5 Receiving characteristics

5.1 Loudness rating

Following the approach used for narrow-band hands-free telephones in ITU-T Rec. P.340 [3], the levels in the hands-free receiving direction are, in principle, the same as those in wideband handset mode. The nominal value of RLR shall then be +2 dB.

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

For HFTs equipped only with automatic (receiving) gain control, the RLR measured with an input signal of −15 dBm0 shall be higher by 10 to 15 dB than the RLR measured with an input signal of –30 dBm0. The nominal RLR shall be included in the measured range. The RLR measured with an input signal of –30 dBm0 shall be –7 dB.

NOTE – The overload point for wideband audio is currently defined as +9 dBm0 [1]. Should this overload point be changed in future revisions of [1], then the Loudness Rating requirements of this Recommendation should also be revised accordingly. The same concept applies in case this Recommendation is used for specifying the electroacoustic requirements of digital telephone sets using wideband audio coders with a different overload point.

5.2 Sensitivity/frequency characteristics

The receiving sensitivity/frequency characteristics shall fall between the upper and lower limits given in Table 3, and shown in Figure 2. All sensitivities are in dB on an arbitrary scale.

Table 3/P.341 – Receiving sensitivity/frequency characteristics

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	–∞
160	6	–7
200	6	–4
250	6	–4
400	4	–4
1000	4	–4
5000	4	–4
6300	4	–7
8000	4	–∞
NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a logarithmic (frequency) – linear (dB) scale.		

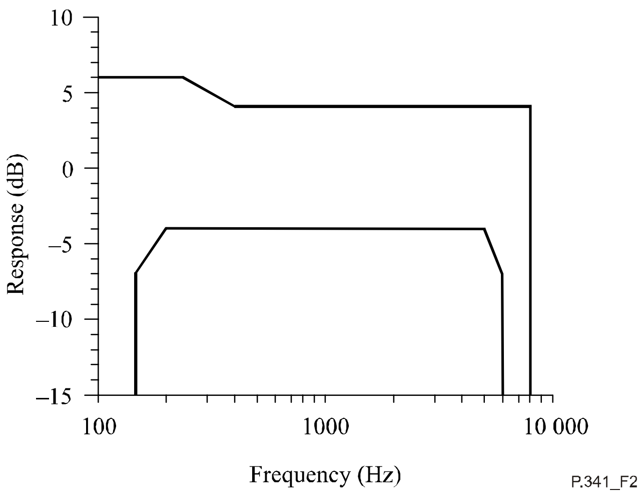


Figure 2/P.341 – Hands-free receiving characteristic

5.3 Noise

The A-weighted quiescent noise in the receiving direction shall not exceed –49 dBPa (A). If a receiving volume control is provided, the requirement applies at a setting as close as possible to the nominal value of RLR as specified in 5.1.

NOTE – The noise may be different in the active mode.

5.4 Distortion

The distortion in the receiving direction shall be measured in terms of the total distortion (harmonic and quantizing) arising from the application of 200 Hz, 1 kHz and 6 kHz tones applied separately. The limits shall be as shown in Table 4. If a receiving volume control is provided, the requirement applies at a setting as close as possible to the nominal value of RLR as specified in 5.1.

Table 4/P.341 – Distorsion in receiving direction

Receiving level at the digital interface (dBm0)	Signal-to-distortion ratio limit (dB)		
	200 Hz	1 kHz	6 kHz
+8 to –30	29.0	35.0	29.0
–40	25.0	26.5	25.0
–56	11.0	12.5	11.0
NOTE – The limits for signal-to-total distortion ratio for intermediate receiving levels lie on straight lines drawn between the given values on a linear (dB receiving level) – linear (dB ratio) scale.			

5.5 Spurious out-of-band receiving signals

The level of any spurious out-of-band signals arising from application of in-band signals at a level of 0 dBm0 shall be attenuated by at least 50 dB at 9 kHz and by at least 60 dB at 14 kHz and above, relative to the output level of a 1 kHz sine wave applied at an input of 0 dBm0.

6 Echo path loss characteristics

6.1 Weighted Terminal Coupling Loss (TCLw)

The TCLw measured from the digital input to digital output shall be at least 35 dB when corrected to the nominal values of SLR and RLR as specified in 4.1 and 5.1, respectively. If a receiving volume control is provided, the requirement applies at a setting as close as possible to the nominal value of RLR as specified in 5.1.

6.2 Stability loss

The attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range 100 Hz to 8 kHz and at all settings of the receiving volume control, if provided.

7 Delay

The total group delay of the sending and receiving parts shall be less than 10 ms. Note that this value of delay allows for the 4 ms delay inherent in the ITU-T Rec. G.722 [1] codec plus the acoustic delay to the measurement point.

NOTE – An extra delay could result from acoustic echo controller processing in the processing unit, and the total terminal delay should be no more than 16 ms.

Annex A

Objective measurement methods for wideband hands-free telephones using the reference codec approach

A.1 Introduction

This annex describes methods which may be used to measure the performance of wideband hands-free telephones, that is, telephones capable of transmitting an audio bandwidth extending beyond the conventional telephony bandwidth of 300 to 3400 Hz, to a bandwidth of approximately 150 to 7000 Hz.

A.2 Electrical interface specifications

Wideband audio will be implemented by a digital encoding scheme such as ITU-T Rec. G.722 [1], and will, therefore, require a suitable interface for test purposes. In general, there are two approaches for evaluating the transmission performance of a wideband digital telephone: the direct approach and the reference codec approach. The direct approach is in principle the most accurate, although the use of the reference codec approach may sometimes be advantageous. Detailed requirements for the direct approach are not yet available so, for the time being, the same approach may be followed as for making measurements on narrow-band digital telephones according to ITU-T Rec. P.310 [2] (see Figure A.1).

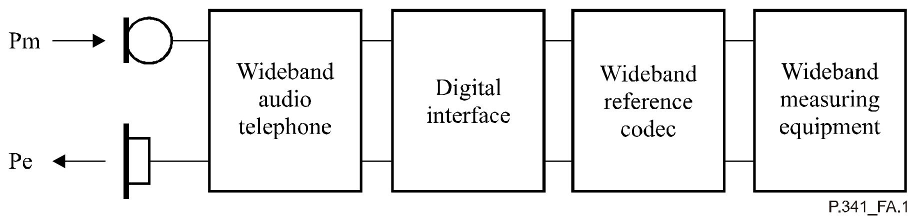


Figure A.1/P.341 – Test set-up

A.2.1 Digital interface

The interface of the test equipment connected to the terminal under test shall be capable of providing the signalling and supervision necessary for the terminal to be working in all test modes.

A.2.2 Wideband reference codec specifications

The reference codec and its audio parts shall comply with ITU-T Rec. G.722 [1]. Tests shall be carried out with the codec operating in Mode 1.

A.2.3 Analogue interface

Measurements shall be carried out by connecting the measurement instrumentation to the test points A and B of the reference codec (see Figure 2/G.722). For compatibility with existing telephone instrumentation, 600-ohm balanced electrical interfaces shall be implemented.

A.2.4 Definition of 0 dBr point

A/D conversion: A 0 dBm0 signal generated by a 600-ohm source will give the digital sequence whose equivalent analogue level is 9 dB below the maximum full-load capacity of the codec [1].

D/A conversion: A digital sequence whose equivalent analogue level is 9 dB below the maximum full-load capacity of the codec will generate 0 dBm across a 600-ohm termination.

NOTE – This definition is based on the current definition of the overload point in [1]. Should this overload point be changed in future revisions of [1], then the 0 dBr point definition in this Recommendation shall also be changed accordingly.

A.3 Electro-acoustic measurement considerations

A.3.1 Testing environment

A.3.1.1 Test room

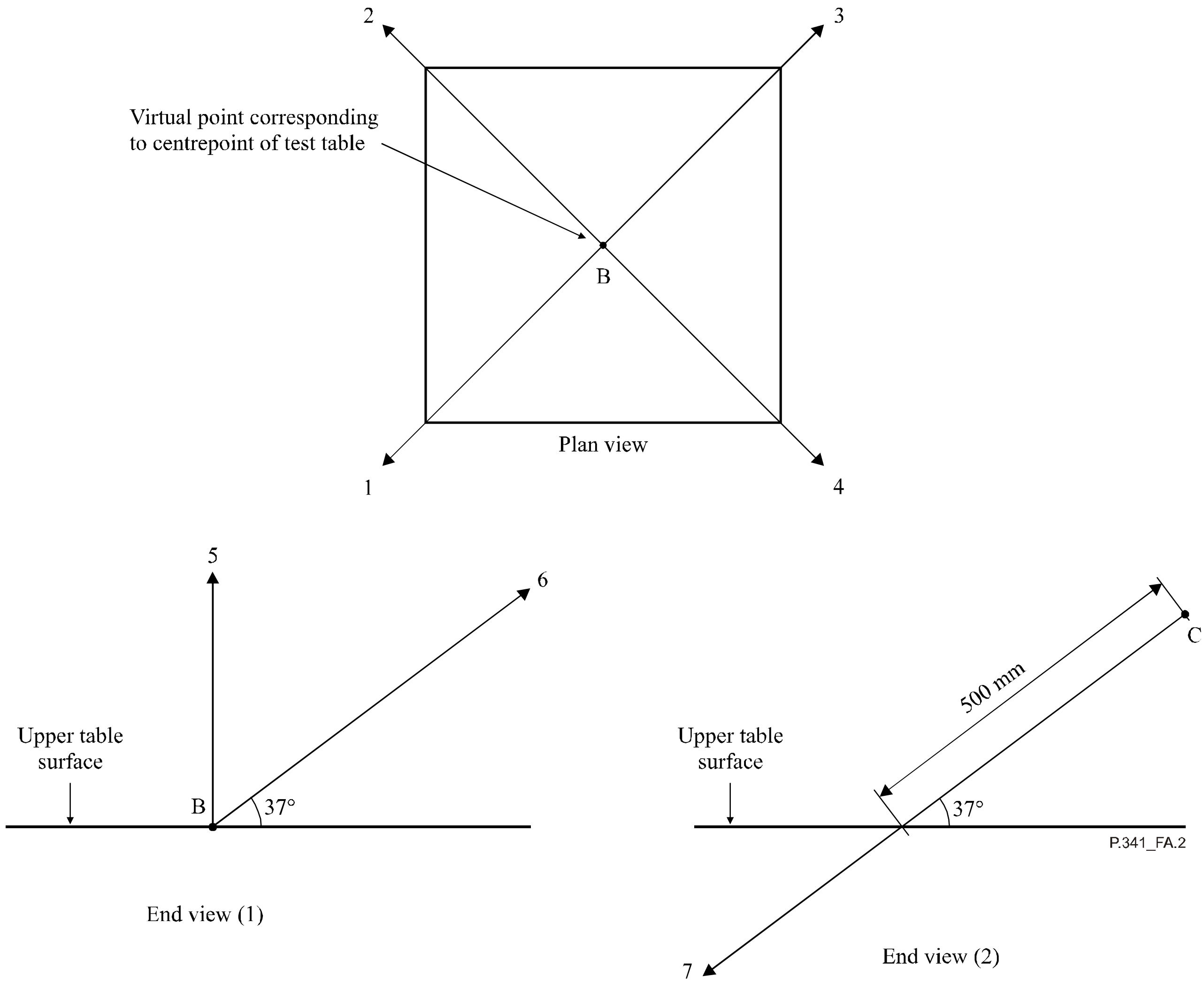
To ensure repeatability of tests, the environment for most of the measurements shall be free-field (anechoic) down to the lowest frequency of the 1/3 octave band centred at 125 Hz [10].

Satisfactory free-field conditions are deemed to exist where errors due to the departure from ideal conditions do not exceed the limits reported in Table A.1, inside a sphere centred at point B (see Figure 3/P.340 [3]), with one-meter radius, in the absence of the table.

Table A.1/P.341 – Allowable departure from ideal conditions

1/3 Octave centre frequency (Hz)	Allowable departure (dB)
≤630	±1.5
800 to 5000	±1.0
≥6300	±1.5

The test signal used for the verification of free-field conditions shall be –20 dBPa at the HFRP. A wideband noise signal shall be used and third octave spectrum measurements shall be carried out at the measurement points. Measurements shall be made along the seven axes numbered 1 to 7 in Figure A.2. The sound source (the artificial mouth [4]) shall be placed at positions equivalent to B or C as appropriate. When placed at point B, the artificial mouth axis shall be perpendicular to the test table surface. When placed at point C, the artificial mouth axis shall be coincident with the axis 7. Measurement points along each axis, taken from the lip plane of the artificial mouth, shall be at distances of 315 mm, 400 mm, 500 mm, 630 mm, 800 mm and 1000 mm.



NOTE 1 – Axes 1 to 7 are used in the determination of free-field conditions for 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 the free-field sound pressure are made in the absence of the test table.

Figure A.2/P.341 – Verification of the free-field conditions

The broadband noise level shall not exceed –70 dBPa (A). Furthermore, the octave band noise level shall not exceed the limits given in Table A.2.

NOTE (informative) – A room fulfilling the following requirements probably meets the anechoic conditions:

$$\text{Room height} \leq 2.2 \text{ m, Volume} \geq 30 \text{ m}^3$$

The test table shall be placed horizontally in the centre of the test room and there shall be an inclination of about 30 degrees between the table and the ceiling. The reverberation time T, measured at points B and C, shall satisfy the following inequality:

$$T(\text{s}) \leq 0.0033 \ V (\text{m}^3)$$

Table A.2/P.341 – Octave band noice level limits

Octave centre frequency (Hz)	Octave band noise level (dBPa)
63	–45
125	–60
250	–65
500	–65
1000	–65
2000	–65
4000	–65
8000	–65
16 000	–65

A.3.1.2 Testing arrangements

The HFT is placed on a test table according to 5.1/P.340 (Test table) and 5.2/P.340 [3] (Test arrangements).

The artificial mouth and the microphone, respectively, is placed at a position equivalent to C in Figure A.3. The artificial mouth axis and the microphone axis shall be coincident with the straight line drawn between point C and point B.

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

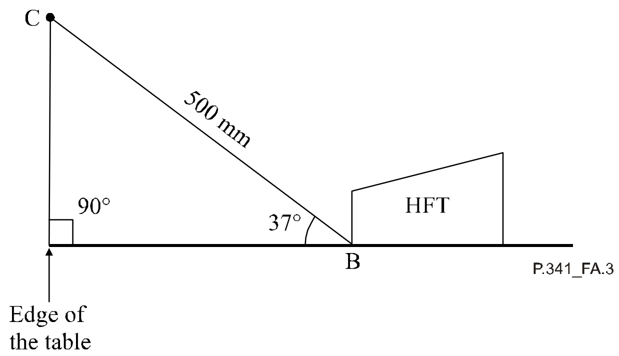


Figure A.3/P.341 – Measurement set-up

A.3.2 Electro-acoustic equipment

Artificial mouth – The artificial mouth shall conform to ITU-T Rec. P.51 [4].

NOTE – If the B&K 4227 artificial mouth is used, it shall be fitted with the original round adaptor.

Sound level meter – The sound level measurement equipment shall conform to IEC 61672-2 [9], Type 1.

A.3.3 Test signals

The test signal shall preferably be either sinusoidal or pink noise, as specified for the different measurements. The pink noise shall be band limited to the frequency range 100 Hz-8 kHz, with a band-pass filter with at least 24 dB/octave slopes and 25 dB out-of-band attenuation. The third octave spectrum of the electrically-generated pink noise shall be equalized to within ±1 dB, while the acoustically-generated pink noise shall be equalized at the MRP within ±3 dB. The crest factor of the (continuous) pink noise signal shall be indicated in the test report.

An on/off modulation (250 ms (±5 ms) "ON" and 150 ms (±5 ms) "OFF")) [3] shall be applied both for noise and for sinusoidal measurements. The excitation levels are referred to the ON component of the signals.

For noise excitation, measurements shall be made by 1/3rd octave filters, at centre frequencies as given by the R10-Series of preferred numbers specified in ISO 3 [10], in the range from 100 Hz to 8 kHz.

If correct activation of the terminal cannot be achieved by the above signals, an alternative test signal providing the correct activation of the terminal should be used. An alternative signal could be as described in ITU-T Rec. P.501 [17].

The measurement is made during the time when the terminal is correctly activated. The correct activation needs to be verified.

A.3.4 Test signal levels

A.3.4.1 Sending

Unless specified otherwise, the test signal level shall be –4.7 dBPa at the MRP defined in ITU-T Rec. P.64 [6]. The characteristics of the artificial mouth shall be in accordance with ITU-T Rec. P.51 [4].

The signal generated by the artificial mouth is equalized at the MRP under free-field conditions in order to obtain the spectrum specified in A.3.3, at a level of –4.7 dBPa in the frequency range corresponding to the third octave bands [10] from 100 Hz to 8 kHz. The spectrum at the MRP [6] is then recorded and the level is adjusted in order to obtain –28.7 dBPa at the HFRP. The spectrum recorded at the MRP [6] is used as a reference for calculating SLR and response characteristics.

A.3.4.2 Receiving

Unless specified otherwise, the test signal shall be –30 dBm0 when measurements with the volume control at its maximum position are carried out. For measurements with the volume control at its minimum position, a test signal level of –15 dBm0 shall be used.

A.3.5 Accuracy of calibrations

Unless specified otherwise, the accuracy of measurements made by test equipment shall not exceed the limits given in Table A.3.

Table A.3/P.341 – Accuracy of measurements

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 not exceed the limits given in Table A.4.

Table A.4/P.341 – Accuracy of the signals

Quantity	Accuracy
Sound pressure level at the MRP	±3 dB (100 Hz to 200 Hz) ±1 dB (200 Hz to 8 kHz) ±3 dB (8 kHz to 16 kHz)
Electrical excitation level	±0.4 dB (Note 1)
Frequency generation	±2% (Note 2)
NOTE 1 – Across the whole frequency range. NOTE 2 – When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ±2% on the generated frequencies, which may be used to avoid this problem, except for 8 kHz, where only –2% tolerance may be used.	

The measurement results shall be corrected for the measured deviations from the nominal level.

A.4 Sending measurements

A.4.1 Loudness rating

The SLR shall be calculated according to Formula A-23b/P.79, over bands 4 to 17 and using the sending weighting factors from Table A.2/P.79, adjusted by subtracting 0.3 dB from each value, using the sending sensitivity response of A.4.2.

A.4.2 Sensitivity/frequency response

The HFT is placed on the test table as specified in A.3.1.2. The noise signal is generated by the mouth, placed at point C, at the level specified in A.3.4.1. The spectrum of the output signal is measured at the output of the reference codec. The sending sensitivity is calculated as follows:

The sending sensitivity is given by the difference between the electrical spectrum and the acoustic spectrum at the MRP [6]:

$$Smj = 20 \log Vs - 20 \log Pm + Corr - 24$$

where:

20 log Vs	is the electrical spectrum
20 log Pm	is the acoustic spectrum at MRP [6]
Corr	is the correction factor (20 log Pmrp/Phfrp) of the artificial mouth

A.4.3 Noise

The HFT is placed on the test table as specified in A.3.1.2. The noise level at the output of the reference codec is measured with an apparatus including A-weighting according to IEC 61672-2 [9].

A.4.4 Distortion

The HFT is placed on the test table as specified in A.3.1.2. A pulsed sine tone at the measurement frequency is generated by the mouth, placed at point C. The level of this signal is adjusted until the output of the terminal is –10 dBm0 (ON periods). The level of the signal at the MRP as defined in ITU-T Rec. P.64 [6] is then the ARL.

The test signal is applied at the following levels:

–46, –40, –35, –30, –24, –20, –17, –10, –5, 0, 5, 10, 15, 18 dB relative to ARL.

The ratio of the signal to total distortion power of the signal at the output of the reference codec is measured.

The sound pressure level at the MRP [6] shall never exceed the rated maximum output level of the artificial mouth [4] (i.e., +6 dBPa according to ITU-T Rec. P.51 [4]). In case the specified measurement range cannot be completely covered, this shall be stated in the measurement report.

A.4.5 Discrimination against out-of-band input signals

The HFT is placed on the test table as specified in A.3.1.2. For input signals at the frequencies of 8 kHz, 9 kHz, 10 kHz, 12 kHz, 13 kHz, 14 kHz and 15 kHz, at –28.7 dBPa at the HFRP, the level of each image frequency is measured at the output of the reference codec.

As the artificial mouth [4] is only specified up to 8 kHz, the acoustic signal can be generated by a suitable alternative loudspeaker, placed in the same position. The sound pressure developed by the loudspeaker at HFRP shall be calibrated under free-field conditions.

In order to activate the HFT in the sending direction, every second measurement burst shall be substituted by an in-band burst at 1 kHz. The correct activation shall be checked by measuring the output level of the transduced in-band bursts.

A.5 Receiving measurements

A.5.1 Loudness rating

The RLR shall be calculated according to formula A-23c/P.79, over bands 4 to 17 and using the receiving weighting factors from Table A.2/P.79, adjusted by subtracting 0.3 dB from each value, using the receiving sensitivity response of A.5.2.

The receiving sensitivity shall not be corrected by the L_e factor. The calculated RLR shall be corrected by subtracting 14 dB, according to ITU-T Rec. P.340 [3].

A.5.2 Sensitivity/frequency response

The HFT is placed on the test table as specified in A.3.1.2. The measurement microphone is placed at point C. The noise signal generator is connected to the input of the reference codec.

The sensitivity at each 1/3 octave band is calculated by subtracting the spectrum of the electric signal from the acoustic spectrum measured at point C.

The measurement is repeated at the minimum and maximum position of the (manual) volume control, changing the input level accordingly. In case of devices not provided with manual volume control, the measurement is repeated for excitation levels of –30 dBm0 and –15 dBm0.

A.5.3 Noise

The HFT is placed on the test table as specified in A.3.1.2. The input of the reference codec is terminated by a 600-ohm resistor. The A-weighted noise level is measured at point C.

A.5.4 Distortion

The HFT is placed on the test table as specified in A.3.1.2. A pulsed sine tone at the measurement frequency is applied at the electrical input of the reference codec at the following levels:

–56, –50, –45, –40, –34, –30, –27, –20, –15, –10, –5, 0, 5, 8 dBm0.

The receiving distortion shall be calculated by normalizing the levels of the distortion components according to the receiving sensitivity frequency response. This is accomplished by subtracting from each distortion component the difference between the receiving sensitivity at its frequency and the sensitivity at the measurement frequency.

A.5.5 Spurious out-of-band signals

The HFT is placed on the test table as specified in A.3.1.2. For input signals at the frequencies 200 Hz, 350 Hz, 500 Hz, 1000 Hz, 2000 Hz, 3500 Hz, 5000 Hz and 7000 Hz, applied at –30 dBm0 at the input of the reference codec, the level of spurious out-of-band image signals at frequencies up to 16 kHz is measured selectively at point C.

A.6 Echo path loss measurements

A.6.1 Weighted Terminal Coupling Loss (TCLw)

The HFT is placed on the test table as specified in A.3.1.2. The input signal shall be a pink noise with a level of –20 dBm0.

The attenuation from the input to the output of the reference codec is measured at the 1/3rd octave frequencies given by the R10 series of preferred numbers in ISO 3 [10] for frequencies from 100 Hz to 8000 Hz.

The weighted terminal coupling loss is calculated according to the method in B.4/G.122 [11] (trapezoidal rule) on the frequency band from 100 Hz to 8 kHz.

A.6.2 Stability loss

The HFT is placed on the test table as specified in A.3.1.2. The test signal shall be sinusoidal, with a level of –20 dBm0. The attenuation from the input to the output of the reference codec is measured at 1/12th octave intervals for frequencies from 100 Hz to 8 kHz.

A.7 Delay measurements

The delay in the sending and receiving directions shall be measured separately from MRP to the digital interface and from the digital interface to the measurement microphone.

For each of the nominal frequencies (F_0) given in Table A.5 in turn, the audio group delay at each value of F_0 is derived from the phase measurements at the corresponding frequencies F_1 and F_2.

Table A.5/P.341 – Frequencies for audio group delay measurement

F_0 (Hz)	F_1 (Hz)	F_2 (Hz)
1000	990	1010
6000	5990	6010

The measurement configuration is given in Figure A.4.

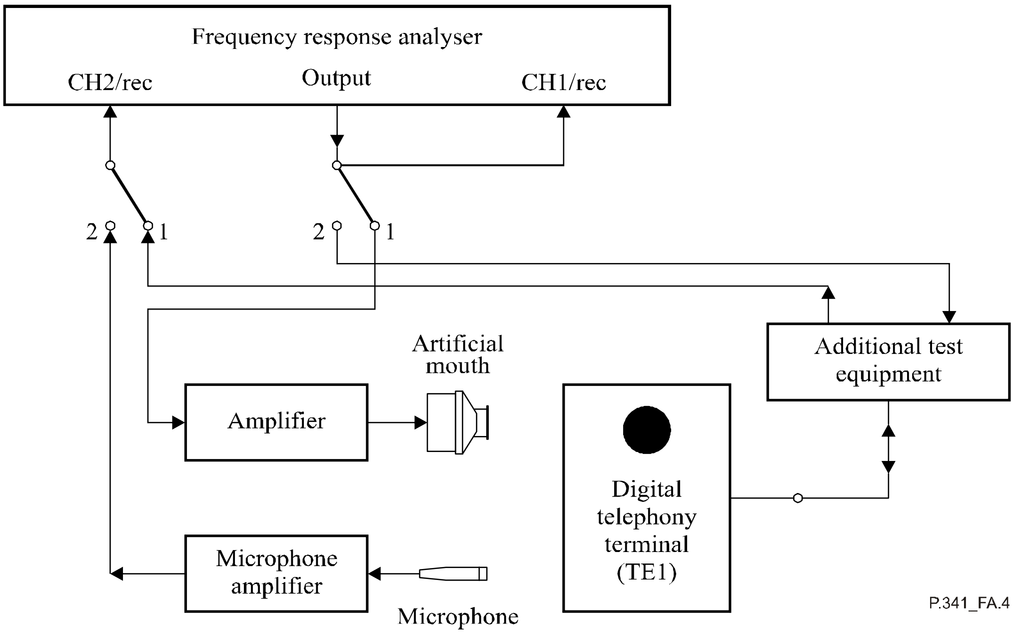


Figure A.4/P.341 – Configuration for delay measurements

For each value of F_0, the audio group delay is evaluated according to the following procedure:

1) output frequency F_1 from the frequency-response analyser;
2) measure the phase shift in degrees between CH1 and CH2 (P_1);
3) output the frequency F_2 from the frequency-response analyser;
4) measure the phase shift in degrees between CH1 and CH2 (P_2);
5) compute the audio group delay (in ms) from the formula:

$$D = \frac{1000(P_1 - P_2)}{360(F_2 - F_1)}$$

Calculate the absolute average of D (for the two values of F_0).

The measured phases P_2 and P_1 shall be used as original values. When using this formula, a negative audio group delay at individual frequencies is possible. Care shall be taken that the real effect is not confused with measurement effect caused by passing 0° or a multiple of 360°.

The audio group delay shall be measured for the sending direction (D_s) and the receiving direction (D_r) of the configuration as shown in Figure A.4.

The audio group delay introduced by the test equipment connected to the acoustic interface shall be measured by mounting the measurement microphone at the MRP and repeating the measurement described above. The audio group delay of all additional test equipment between the interface provided for the connection to a digital network and the output (CH1) and input (CH2) of the test equipment shall also be determined.

The audio group delay of the telephone is calculated from the formula:

$$D = D_s + D_r - D_e$$

where D_e is the group delay of the test equipment.

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

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