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TELEPHONE TRANSMISSION QUALITY

TRANSMISSION CHARACTERISTICS OF WIDEBAND HANDSFREE TELEPHONES

ITU-T Recommendation P.341

Superseded by a more recent version

(Previously "CCITT Recommendation")

Superseded by a more recent version

FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The 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 Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation P.341 was prepared by ITU-T Study Group 12 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 18th of April 1995.

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NOTE

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

CONTENTS

SUMMARY

This Recommendation provides preliminary audio performance requirements for wideband audio (7 kHz) handsfree 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. However, since this is a new technical area, the requirements in this Recommendation are not yet complete, and studies are ongoing in ITU-T Study Group 12.

Recommendation P.341

TRANSMISSION CHARACTERISTICS OF WIDEBAND HANDSFREE TELEPHONES

(Geneva, 1995)

1 Scope

This Recommendation provides audio performance requirements and test methods for handsfree 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 Recommendation G.722. Wideband audio telephones are expected to be used in new services such as high quality audio conferencing, video conferencing and multimedia applications.

The requirements listed in this Recommendation are primarily applicable to telephones using G.722 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.

2 Normative References

The following 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] CCITT Recommendation G.722 (1988), *7 kHz audio coding within 64 kbit/s*.

[2] ITU-T Recommendation P.31 (1993), *Transmission characteristics for digital telephones*.

[3] ITU-T Recommendation P.34 (1993), *Transmission characteristics of handsfree telephones*.

[4] ITU-T Recommendation P.51 (1993), *Artificial mouth*.

[5] ITU-T Recommendation P.57 (1993), *Artificial ears*.

[6] ITU-T Recommendation P.64 (1993), *Determination of sensitivity/frequency characteristics of local telephone systems*.

[7] ITU-T Recommendation P.66 (1993), *Methods for evaluating the transmission performance of digital telephone sets*.

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

[9] IEC Publication 651, *Sound level meters*.

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

[11] CCITT Recommendation G.122 (1988), *Influence of national systems on stability, talker echo and listener echo in international connections*.

3 Abbreviations and definitions

For the purposes of this Recommendation, the following definitions and abbreviations are used:

acoustic reference level (ARL): The acoustic level at MRP which gives –10 dBm0 at the digital interface.

handsfree reference point (HFRP): A point located on the axis of the Artificial Mouth, at 50 cm from the lip plane, where the level calibration is made under free-field conditions. It corresponds to measurement point n.11 defined in Recommendation P.51 [4].

HFT	Handsfree 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 Levels

Following the approach used for narrow-band handsfree telephones in Recommendation P.34, the levels in the handsfree send direction are related to those in wideband handset mode (see Recommendation P.311) with an allowance of 5 dB for higher talking levels and the difference in speaking position. The provisional value of SLR is therefore +13 dB, measured in terms of a narrow-band loudness rating according to Recommendation P.79 [8].

NOTE – The overload point for wideband audio is defined as +9 dBm0.

4.2 Sensitivity/frequency characteristics

The sending sensitivity/frequency characteristics from mouth reference point to the digital interface shall fall within a mask which can be drawn between the points given in Table 1, also shown in Figure 1. All sensitivities are dB on an arbitrary scale.

TABLE 1/P.341

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 limit lies on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.		

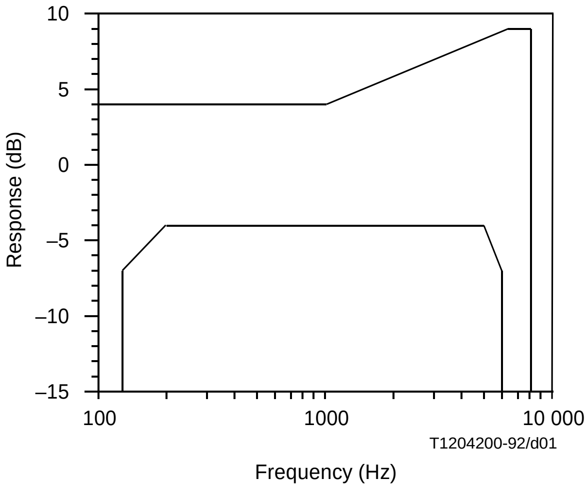


FIGURE 1/P.341

Handsfree send characteristic

4.3 Noise

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

4.4 Distortion

The distortion of the apparatus 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

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

4.5 Discrimination against out-of-band input signals

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

5 Receiving characteristics

5.1 Levels

Following the approach used for narrow-band handsfree telephones in Recommendation P.34 [3], the levels in the handsfree receive direction are in principle the same as those in wideband handset mode, with a correction factor of 14 dB applied during testing to take into account head diffraction and binaural listening. The nominal value of RLR is +5 dB. This value is based on the value RLR = 2 dB as specified in Recommendation P.31 with a correction factor of 3 dB taking into account the difference between narrow-band and wideband.

NOTE – The 2 dB offset with respect to the handset level requirement results from the lack of need in the handsfree case to apply the 2 dB correction embedded in the handset requirement due to the Type 3.2 artificial ear.

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

For sets only equipped 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 –5 dB.

Note that the overload point for wideband audio is defined as +9 dBm0.

5.2 Sensitivity/frequency characteristics

The receiving sensitivity/frequency characteristics from the digital interface to the measurement point C shall fall within a mask which can be drawn between the points given in Table 3, also shown in Figure 2. All sensitivities are dB on an arbitrary scale.

TABLE 3/P.341

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	–∞

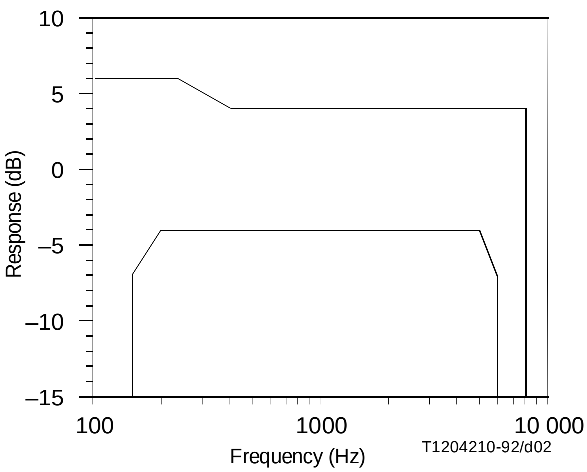


FIGURE 2/P.341

Handsfree receive characteristic

5.3 Noise

The A-weighted noise produced at the measurement point C shall not exceed –49 dBPa (A).

5.4 Distortion

The distortion of the apparatus 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.

TABLE 4/P.341

Receive level (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

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 the following amounts relative to the output level of a 1 kHz sine wave applied at an input of 0 dBm0:

9 kHz	50 dB
14 kHz and above	60 dB

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.

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.

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 Recommendation G.722 [1] codec plus the acoustic delay to the measurement point.

Annex A

Objective measurement methods for wideband handsfree telephones

(This annex forms an integral part of this Recommendation)

A.1 Introduction

This annex describes methods which may be used to measure the performance of wideband handsfree 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 Recommendation G.722, 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 Recommendation P.66 [7], see Figure A.1.

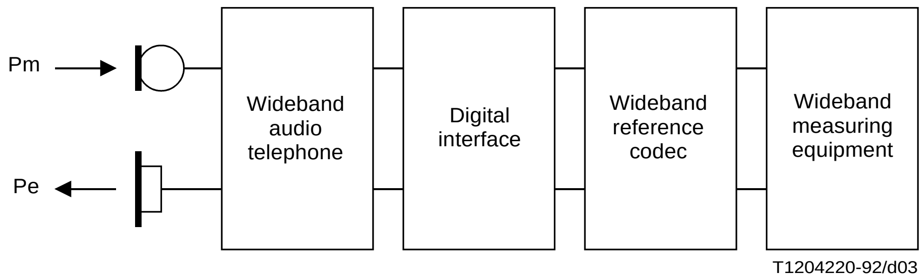


FIGURE A.1/P.341

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 Recommendation 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.

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.

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.

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 on point B (Figure 3/P.34 [3]), with one meter radius, in the absence of the table.

TABLE A.1/P.341

1/3 Octave centre frequency (Hz)	Allowable departure (dB)
$\leq$ 630	$\pm$ 1.5
800 to 5000	$\pm$ 1.0
$\geq$ 6300	$\pm$ 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 shall be placed at positions equivalent to B or C as appropriate. 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.

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:

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

The 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:

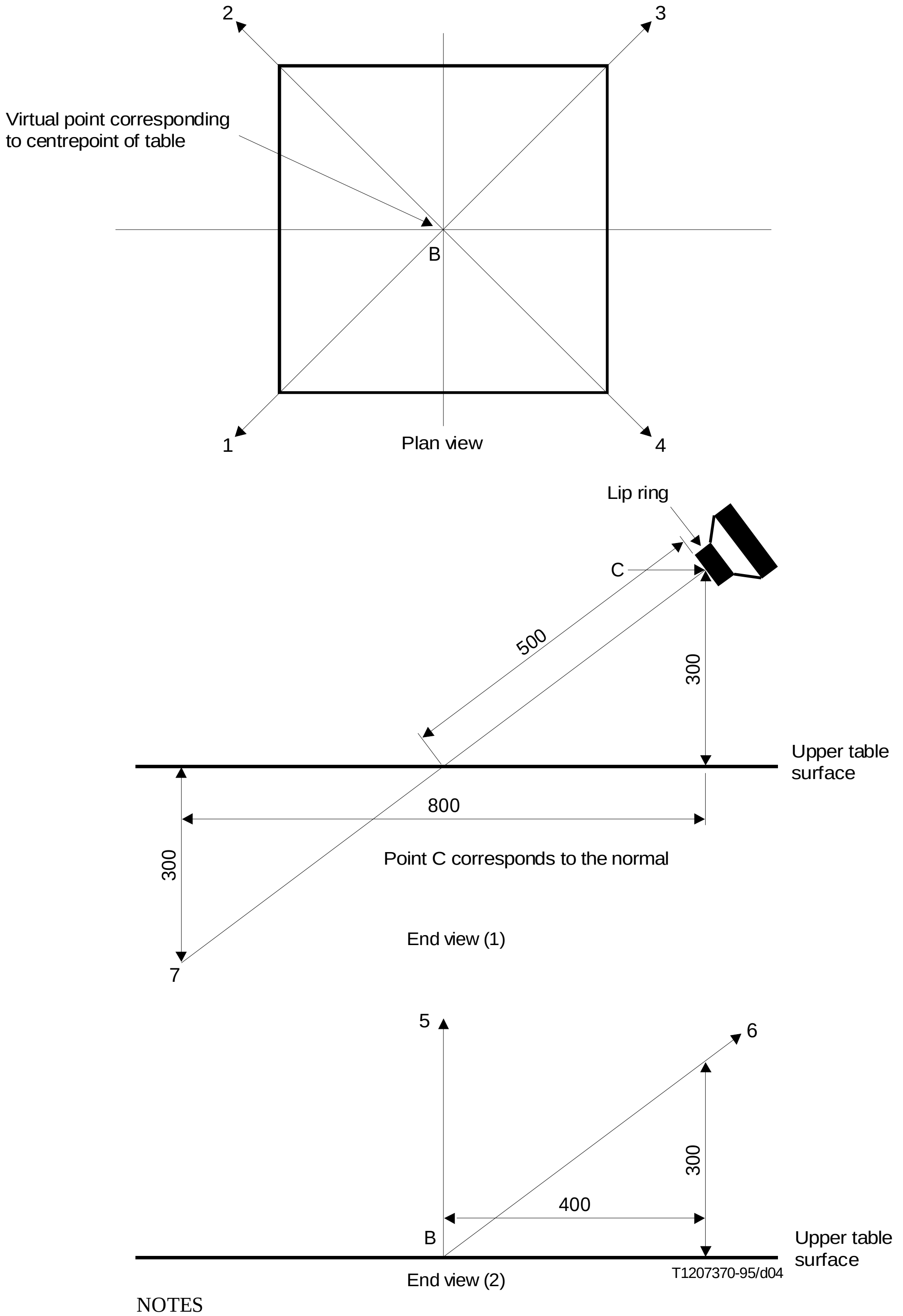
$$T(s) \leq 0.0033\ V\ (m^3)$$

A.3.1.2 Testing arrangements

The HFT is placed on a table according to 6.1/P.34 (Test table) and 6.2/P.34 [3] (Test arrangements).

The artificial mouth axis and the microphone axis are coincident with the straight line drawn between point C and point B (see Figure A.3).

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 from each others, but without modifying the normal use configuration of the HFT.



NOTES

1 Dimensions in millimetres.

2 Points 1, 2, 3 and 4 are in the horizontal plane normally occupied by the table surface.

3 Measurement of free-field sound pressure are made in the absence of the table.

4 Axes used in the determination of free-field conditions for 1 m radius sphere.

FIGURE A.2/P.341

Calibration of the free-field conditions

TABLE A.2/P.341

Octave centre frequency (Hz)	Octave band pressure level (dBPa)
63	–45
125	–60
250	–65
500	–65
1 000	–65
2 000	–65
4 000	–65
8 000	–65
16 000	–65

A.3.2 Electro-acoustic equipment

Artificial Mouth – The artificial mouth shall conform to Recommendation 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 Publication 651 [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/oct slopes and 25 dB out-of-band attenuation. The third octave spectrum of the electrically-generated pink noise shall be equalized to within ± 1dB, while the acoustically-generated pink noise shall be equalized at the MRP within ± 3dB. 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") 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 center frequencies as specified by ISO 3 (1973) [10], in the range from 100 Hz to 8 kHz.

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. The characteristics of the Artificial Mouth shall be in accordance with Recommendation 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 from 100 Hz to 8 kHz. The spectrum at the MRP is then recorded and the level is adjusted in order to obtain –28.7 dBPa at the HFRP. The spectrum recorded at the MRP 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 be better than (see Table A.3):

TABLE A.3/P.341

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 (see Table A.4):

TABLE A.4/P.341

Quantity	Accuracy
Sound pressure level at the MRP	± 1 dB (200 Hz to 8 kHz) ± 3 dB (100 Hz à 200 Hz) and (8 kHz to 16 kHz)
Electrical excitation level	± 0.4 dB (Note 1)
Frequency generation	± 2% (Note 2)

NOTES

1 Across the whole frequency range.

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 Measurement methods

A.4.1 Sensitivity frequency response

A.4.1.1 Sending

The set is mounted on the measurement table as specified in Figure A.3. The noise signal is generated by the mouth at the level specified in A.3.4.1. The spectrum of the output signal is measured at the output interface 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:

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

where:

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

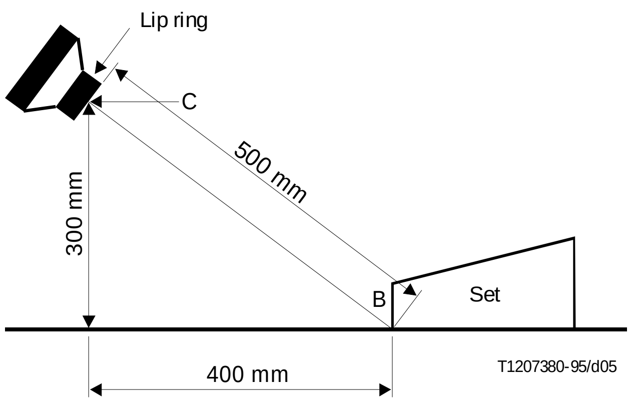


FIGURE A.3/P.341

Measurement set-up

A.4.1.2 Receiving

The telephone is placed on the measurement table as specified in Figure A.3. 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.4.2 Loudness rating

A.4.2.1 Sending loudness rating

The sending sensitivity shall be measured for each of the fourteen 1/3rd octave bands given in Table 2/P.79 [8], bands 4-17. The SLR shall be calculated according to Recommendation P.79, formula 4.19b, over bands 4 to 17 and using the sending weighting factors from Table 2/P.79, adjusted according to Table 3/P.79.

A.4.2.2 Receiving loudness rating

The receiving sensitivity shall be measured for each of the fourteen 1/3rd octave bands given in Table 2/P.79 [8], bands 4-17. The RLR shall be calculated according to Recommendation P.79, formula (4.19c), over bands 4 to 17 and using the receiving weighting factors from Table 2/P.79 [8], adjusted according to Table 3/P.79.

The receiving sensitivity shall *not* be corrected by the Le factor. The calculated RLR shall be corrected by subtracting 14 dB, according to Recommendation P.34 [3].

A.4.3 Terminal coupling loss

A.4.3.1 Weighted Terminal Coupling Loss (TCLw)

The set is placed as specified in A.3.1.2. The input signal shall be a pink noise with a level of –20 dBm0.

The attenuation from digital input to digital output is measured at the 1/3rd octave frequencies given by the R10 series of preferred numbers in ISO 3 (1973) [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.4.3.2 Stability loss

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

A.4.4 Harmonic distortion

A.4.4.1 Sending

The set is placed on the measurement table as specified in A.3.1.2. A pulsed sine tone at the measurement frequency is generated by the mouth. 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 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 reference codec output is measured.

The sound pressure level at the MRP shall never exceed the rated maximum output level of the Artificial Mouth (i.e. +6 dBPa according to Recommendation P.51). In case the specified measurement range can not be completely covered, this shall be stated in the measurement report.

A.4.4.2 Receiving

The set is placed on the measurement 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.4.5 Out-of-band signals

A.4.5.1 Discrimination against out-of-band input signals (Sending)

The set is placed on the measurement table as specified in A.3.1.2. For input signals at the frequencies of 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 interface of the reference codec.

As the artificial mouth 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 handsfree set 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.4.5.2 Spurious out-of-band (Receiving)

The set is placed on the measurement table as specified in A.3.1.2. For input signal 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 port 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.4.6 Noise

A.4.6.1 Sending

With the set placed on the measurement table as specified in A.3.1.2, the noise level at the digital output is measured with an apparatus including A-weighting according to IEC Publication 651 [9].

A.4.6.2 Receiving

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

A.4.7 Delay

Under study.