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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 handset telephones**

ITU-T Recommendation P.311

(Previously CCITT Recommendation)

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ITU-T RECOMMENDATION P.311

TRANSMISSION CHARACTERISTICS FOR WIDEBAND (150-7000 Hz) DIGITAL HANDSET TELEPHONES

Summary

This revised Recommendation provides audio performance requirements for wideband audio (7 kHz) handset 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, sidetone, echo path and delay. 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 include the effects of a receiving volume control.

Source

ITU-T Recommendation P.311 was revised by ITU-T Study Group 12 (1997-2000) and was approved under the WTSC Resolution No. 1 procedure on the 27th of February 1998.

FOREWORD

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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.

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CONTENTS

	Page
1	Scope 1
2	Normative references..... 1
3	Definitions and abbreviations..... 2
4	Sending characteristics 2
4.1	Loudness rating 2
4.2	Sensitivity/frequency characteristics 2
4.3	Noise..... 3
4.4	Distortion..... 3
4.5	Discrimination against out-of-band input signals 4
5	Receiving characteristics 4
5.1	Loudness rating 4
5.2	Sensitivity/frequency characteristics 4
5.3	Noise..... 5
5.4	Distortion..... 5
5.5	Spurious out-of-band receiving signals 6
6	Sidetone characteristics 6
6.1	Talker sidetone 6
6.2	Sidetone distortion..... 6
7	Echo path loss characteristics..... 6
7.1	Weighted terminal coupling loss..... 6
7.2	Stability loss 6
8	Delay..... 7
Annex A	– Objective measurement methods for wideband handset telephones..... 7
A.1	Introduction 7
A.2	Electrical interface specifications..... 7
A.2.1	Digital interface..... 7
A.2.2	Wideband reference codec specification..... 7
A.2.3	Analogue interface 7
A.2.4	Definition of 0 dBr point..... 8
A.3	Electro-acoustic measurement considerations..... 8
A.3.1	Artificial mouths and ears 8
A.3.2	Stimulus signals 8
A.3.3	Accuracy of calibrations..... 8

	Page
A.4 Sending measurements.....	9
A.4.1 Loudness rating.....	9
A.4.2 Sensitivity/frequency response.....	9
A.4.3 Noise.....	9
A.4.4 Distortion.....	9
A.4.5 Discrimination against out-of-band signals.....	10
A.5 Receiving measurements.....	10
A.5.1 Loudness rating.....	10
A.5.2 Sensitivity/frequency response.....	10
A.5.3 Noise.....	10
A.5.4 Distortion.....	11
A.5.5 Spurious out-of-band receiving signals.....	11
A.6 Sidetone measurements.....	11
A.6.1 Talker sidetone.....	11
A.6.2 Distortion.....	12
A.7 Echo path loss measurements.....	12
A.7.1 Terminal coupling loss.....	12
A.7.2 Stability loss.....	12
A.8 Delay measurements.....	13

Recommendation P.311

TRANSMISSION CHARACTERISTICS FOR WIDEBAND (150-7000 Hz) DIGITAL HANDSET TELEPHONES

(Geneva, 1995; revised in 1998)

1 Scope

This Recommendation provides audio performance requirements and test methods for handset 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 [1]. 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 [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 need to have wideband loudness ratings is still under study.

Conventional telephone band (300-3400 Hz) digital handset telephones using encoding according to Recommendations G.711 [10] and G.726 [11] are covered by Recommendation P.310 [2].

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] CCITT Recommendation G.722 (1988), *7 kHz audio coding within 64 kbit/s*.
- [2] ITU-T Recommendation P.310 (1996), *Transmission characteristics for telephone band (300-3400 Hz) digital telephones*.
- [3] ITU-T Recommendation P.51 (1996), *Artificial mouth*.
- [4] ITU-T Recommendation P.57 (1996), *Artificial ears*.
- [5] ITU-T Recommendation P.64 (1997), *Determination of sensitivity/frequency characteristics of local telephone systems*.
- [6] ITU-T Recommendation P.340 (1996), *Transmission characteristics of handsfree telephones*.
- [7] ITU-T Recommendation P.79 (1993), *Calculation of loudness ratings for telephone sets*.
- [8] IEC Publication 60651 (1979), *Sound level meters*.
- [9] ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections*.

- [10] CCITT Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies*.
- [11] CCITT Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*.
- [12] ITU-T Recommendation P.10 (1993), *Vocabulary of terms on telephone transmission quality and telephone sets*.
- [13] ITU-T Recommendation P.501 (1996), *Test signals for use in telephony*.

3 Definitions and abbreviations

This Recommendation defines the following term:

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

Relevant abbreviations in Recommendation P.10 [12] will apply:

CSS	Composite Source Signal
DRP	Eardrum Reference Point
ERP	Ear Reference Point
LRGP	Loudness Rating Guard Ring Position
LSTR	Listener Sidetone Rating
MRP	Mouth Reference Point
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
STMR	Sidetone Masking Rating
TCL	Terminal Coupling Loss
TCL _w	Weighted Terminal Coupling Loss

4 Sending characteristics

4.1 Loudness rating

The electro-acoustic gain in the sending direction should be adjusted in terms of a narrow-band loudness rating calculated according to Recommendation P.79 [7] (calculated over the range 200 to 4000 Hz). When measured in this manner, the Sending Loudness Rating (SLR) shall be +8 dB (consistent with Recommendation P.310 [2]).

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

4.2 Sensitivity/frequency characteristics

The sending sensitivity/frequency characteristic 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.311

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	$-\infty$
125	4	-7
200	4	-4
1000	4	-4
5000	(Note)	-4
6300	9	-7
8000	9	$-\infty$

NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (frequency) – linear (dB) scale.

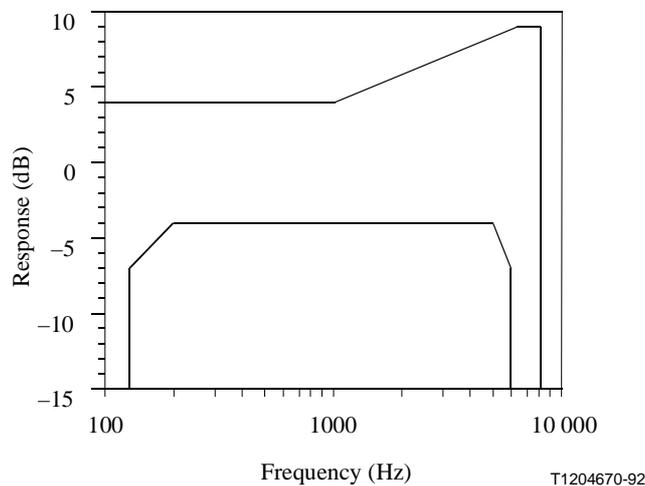


Figure 1/P.311 – Handset sending characteristic

4.3 Noise

With the microphone muted acoustically (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.311

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 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 ratio 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 at the output 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

The electro-acoustic gain in the receiving direction should be adjusted in terms of a narrow-band loudness rating calculated according to Recommendation P.79 [7] (calculated over the range 200 to 4000 Hz). When measured this way, the Receiving Loudness Rating (RLR) shall be +7 dB.

NOTE – This is derived from the RLR of +2 dB specified in Recommendation P.310 [2], with 3 dB loss added to account for the effective loudness gain going from narrow-band to wideband, and a further 2 dB loss to account for the loudness loss introduced by the Type 3.2 artificial ear specified in Recommendation P.57 [4].

If a receiving volume control is provided, the above requirement is taken to apply at the nominal volume level.

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

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.

NOTE – It is recognized that it may be difficult for telephone sets equipped with receivers designed according to the current technology to meet the recommended requirement, and still comply with the recommended TCLw limit.

Table 3/P.311

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	$-\infty$
160	4	-7
200	4	-4
1000	4	-4
5000	4	-4
6300	4	-7
8000	4	$-\infty$

NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a logarithmic (frequency) – linear (dB) scale.

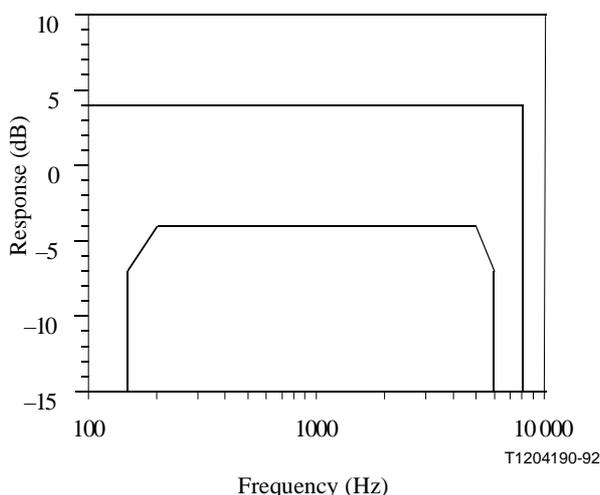


Figure 2/P.311 – Handset receiving characteristic

5.3 Noise

The noise in the receiving direction shall not exceed -59 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.

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 requirements apply at a setting as close as possible to the nominal value of RLR as specified in 5.1.

Table 4/P.311

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 Sidetone characteristics

6.1 Talker sidetone

The nominal value of Sidetone Masking Rating (STMR) shall be 13 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.

NOTE – At present, there is no value for LSTR and this requires further study.

6.2 Sidetone distortion

The signal to third harmonic distortion ratio generated by the sidetone path shall be greater than 20 dB. 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.

7 Echo path loss characteristics

7.1 Weighted terminal coupling loss

With the handset suspended in the free air, the value of TCLw 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.

7.2 Stability loss

With the handset lying on a hard surface with the transducers facing that surface, the attenuation from digital input to 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.

8 Delay

The total audio group delay of the sending and receiving parts shall be less than 7 ms. Note that this value allows for the 4 ms delay inherent in G.722 codec [1].

ANNEX A

Objective measurement methods for wideband handset telephones

A.1 Introduction

This annex describes methods which may be used to measure the performance of wideband handset 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 [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 Recommendation P.310 [2], (see Figure A.1).

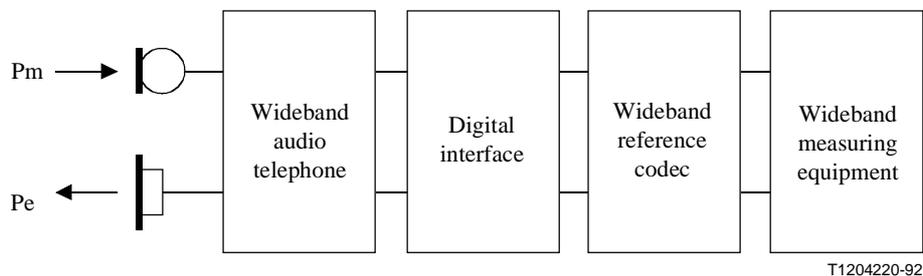


Figure A.1/P.311

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 specification

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 [1]). 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 Artificial mouths and ears

The Mouth Reference Point (MRP) and Ear Reference Point (ERP) used for wideband audio measurements are defined in Annex A/P.64 [5].

The Loudness Rating Guarding Position (LRGP) is defined in Annex C/P.64 [5].

The artificial mouth specified in Recommendation P.51 [3] shall be used for making wideband sending measurements.

NOTE – If the Brüel and Kjaer type 4227 artificial mouth is used, the rounded face plate is recommended.

For making handset receiving measurements, the Type 3.2 artificial ear described in Recommendation P.57 [4] is recommended. The low leakage option should be used.

If the geometry of the handset does not allow the use of the Type 3.2 artificial ear, then either Type 3.3 or Type 3.4 artificial ears shall be used. The force against the ear shall be as specified in Recommendation P.57 [4].

Sound pressure levels should be referred to ERP using the correction factors given in Tables 2a/P.57 and 2b/P.57.

A.3.2 Stimulus signals

In general, a speech-like stimulus signal is preferred for testing, but care should be taken that the signal contains sufficient high-frequency components to obtain an adequate signal-to-noise ratio for the measurement. If pink noise is used, it should be band-limited to the range 100 Hz to 8 kHz, using a band-pass filter with at least 24 dB/octave slope in the transition band and a minimum of 25 dB out-of-band attenuation. A selection of various test signals, including speech-like signals, can be found in Recommendation P.501 [13].

An on/off modulation (250 ms "ON" and 150 ms "OFF") [6] shall be applied if echo control or automatic noise detection mechanisms are involved. If modulated signals are used, excitation levels are referred to the ON component of the signals. Composite Source Signals (CSS) as described in Recommendation P.501 [13] or switched pink noise are signals which provide the desired ON/OFF modulation.

NOTE – Care should be taken since this type of test signal may influence the result of the measurements.

A.3.3 Accuracy of calibrations

Unless specified otherwise, the accuracy of measurements made by test equipment shall not exceed the limits given in Table A.1.

Table A.1/P.311

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	$\pm 5\%$
Frequency	$\pm 0.2\%$

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall not exceed the limits given in Table A.2.

Table A.2/P.311

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	$\pm 2\%$ (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 $\pm 2\%$ 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 Sending Loudness Rating (SLR) is calculated according to clause 3/P.79 [7], using the sending sensitivity response of A.4.2.

A.4.2 Sensitivity/frequency response

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear. The testing signal is applied to the MRP as described in Recommendation P.64 [5] at a sound pressure level of -4.7 dBPa. The output signal is measured at the output of the reference codec.

A.4.3 Noise

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. In a quiet environment [ambient noise less than -64 dBPa (A)], the noise level at the output of the reference codec is measured with an apparatus including A-weighting according to IEC Publication 60651 [8].

A.4.4 Distortion

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. A sinusoidal signal at the measurement frequency is applied at the MRP. The level of this signal is

adjusted until the output signal at the output of the reference codec is -10 dBm0. This acoustic level at the MRP [5] is by definition the Acoustic Reference Level (ARL). The test signal is applied at the following levels, with the proviso that the sound pressure level at the MRP shall not exceed $+6$ dBPa:

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

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

A.4.5 Discrimination against out-of-band signals

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. A reference signal at 1 kHz is applied at the MRP as described in Recommendation P.64 [5] at a sound pressure level of -4.7 dBPa. The level is measured at the output of the reference codec.

The handset is then placed in a free field where an acoustic signal at 8 kHz, 9 kHz, 10 kHz, 12 kHz, 13 kHz and 15 kHz is generated in turn.

The generated field shall approximate a plane acoustic wave, parallel to the earphone reference plane with a propagation direction towards the microphone of the handset.

The microphone input signal is measured by a calibrated probe microphone (diameter < 3.2 mm) placed near the centre of the acoustic input port of the handset closest to the centre of the artificial mouth opening [3] when the handset is mounted at LRGP [5].

The level of the out-of-band signals is -4.7 dBPa, measured with the probe microphone.

The level of each image frequency is measured at the output interface of the reference codec.

NOTE – Plane wave propagation conditions are considered to be adequately reproduced for the purposes of this measurement by positioning the acoustic centre of the sound source at least 500 mm from the earphone reference plane and on the perpendicular to the plane through the monitored acoustic input port.

A.5 Receiving measurements

A.5.1 Loudness rating

The Receiving Loudness Rating (RLR) is calculated according to clause 3/P.79 [7] using the receiving sensitivity response of A.5.2. The correction for Ear Leakage (L_E) shall **not** be applied.

A.5.2 Sensitivity/frequency response

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. A test signal source is connected to the input of the reference codec at a level of -20 dBm0. Sound pressure measurements made in the artificial ear shall be referred to the ERP using the correction method specified in Recommendation P.57 [4].

A.5.3 Noise

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. The input of the reference codec is terminated in 600 ohms. The 1/3 octave noise spectrum is measured in the artificial ear and the correction factors of Table 2a/P.57 [4] is used for referring to ERP. The A-weighted noise level at ERP is then calculated.

A.5.4 Distortion

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. A sinusoidal signal at the measurement frequency is applied to the 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 ratio of signal-to-total distortion power measured at 1 kHz is incremented by 6 dB.

NOTE – This is measured at the DRP and the measurements need to be referred to the ERP, which requires a correction of 6 dB due to the ear canal transmission characteristic.

A.5.5 Spurious out-of-band receiving signals

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. For input signals at frequencies of 200 Hz, 350 Hz, 500 Hz, 1000 Hz, 2000 Hz, 3500 Hz, 5000 Hz, and 7000 Hz, applied at 0 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 in the artificial ear and corrected to ERP [5].

The correction factors for referring the sound pressure level at ERP [5] in the frequency range between 8 kHz and 16 kHz are given in Table A.3.

Table A.3/P.311

Frequency (kHz)	Correction (dB)
9.0	14.0
9.5	21.0
10.0	18.0
10.7	14.0
11.3	13.0
12.0	11.0
12.7	5.0
13.5	2.0
14.3	4.0
15.1	0.0
16.0	–2.0

A.6 Sidetone measurements

For the sidetone tests, the input of the reference codec shall be terminated in 600 ohms.

A.6.1 Talker sidetone

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. The test signal is applied at the MRP [5] at a sound pressure level of –4.7 dBPa. If a receiving volume control is provided, the measurements shall be carried out with the control at a setting as close as possible to the nominal value of RLR as specified in 5.1.

Measurements of the sound pressure in the artificial ear [4] are made at 1/3 octave points at the frequencies specified in Table 3/P.79 [7], corrected to sound pressure levels at ERP [5], and STMR

is then calculated according to clause 4/P.79. The correction for Ear Leakage (L_E) shall **not** be applied.

A.6.2 Distortion

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. A sinusoidal tone of -4.7 dBPa is applied at the MRP [5] at frequencies of 200 Hz, 315 Hz, 500 Hz, 1000 Hz and 2000 Hz. At each frequency, the third harmonic distortion of the acoustic signal in the artificial ear is measured.

The measured sound pressure levels are algebraically added to the correction factors below (see Table A.4).

Table A.4/P.311

Frequency (Hz)	Correction factor (dB)
200	+1
315	+2
500	+3
1000	+8
2000	-3

NOTE – This is assumed measuring at the DRP, and the measurements need to be referred to the ERP which require corrections as shown in Table A.4 due to the ear canal transmission characteristic.

A.7 Echo path loss measurements

A.7.1 Terminal coupling loss

The handset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not affected. The test space shall be practically free-field (anechoic) down to a lowest frequency of 100 Hz, and be such that the test handset lies totally within the free-field volume. This is met where the reverberation distance is greater than 50 cm. The ambient noise level shall be less than -64 dBPa (A).

The attenuation from the input to the output of the reference codec is measured at $1/12^{\text{th}}$ octave frequencies from 100 Hz to 8 kHz. The input signal shall be 0 dBm0. The weighted terminal coupling loss (TCLw) is calculated according to the method in Annex B.4/G.122 [9], (trapezoidal rule) over the frequency range 100 Hz to 8 kHz.

A.7.2 Stability loss

With an input signal of 0 dBm0, the attenuation from the input to the output of the reference codec is measured at $1/12^{\text{th}}$ octave intervals for frequencies from 100 Hz to 8 kHz under the following conditions:

- a) The handset, with the speech transmission circuit fully active shall be positioned on one inside surface that is part of three perpendicular plane, smooth and hard surfaces. Each surface shall extend 500 mm from the apex of the corner. One surface shall be marked with a diagonal line extending from the corner and with a reference position 250 mm from the corner formed by the three surfaces, as shown in Figure B.10/P.310 [2].

- b) The handset shall be positioned on the defined surface as follows:
- 1) the mouthpiece and earcap shall face towards the surface;
 - 2) the handset shall be placed centrally above the diagonal line, with the earcap nearer to the apex of the corner;
 - 3) the extremity of the handset shall coincide with the perpendicular to the reference point, as shown in Figure B.10/P.310 [2].

A.8 Delay measurements

The handset is mounted in the LRGP [5] and the earpiece is coupled to the artificial ear [4]. The delay in sending and receiving directions shall be measured separately from MRP [5] to the digital interface and from digital interface to ERP [5].

The audio group delay shall be measured for the sending direction (D_s) and the receiving direction (D_r) as shown in Figure A.2.

The acoustic input level shall be ARL, as defined in clause 3.

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.311 – 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.2.

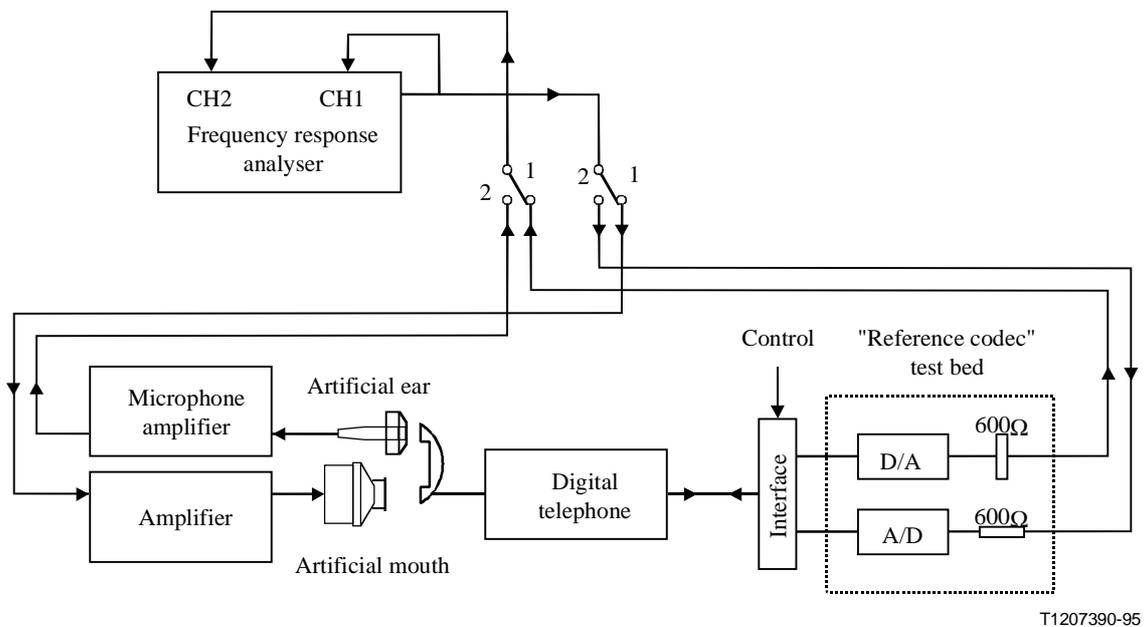


Figure A.2/P.311 – Configuration for audio group delay measurement

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

- 1) output the 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 this real effect is not confused with measurement effect caused by passing 0° or a multiple of 360° .

The audio group delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone, or equivalent, at the MRP [5]. The audio group delay of all additional test equipment shall be determined.

The audio group delay is calculated from the formula:

$$D = D_s + D_r - D_E$$

where D_E is the audio group delay of the test equipment.

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