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SERIES P: TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS

Voice terminal characteristics

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# Technical requirements and test methods for the universal wired headset or headphone interface of digital mobile terminals

Recommendation ITU-T P.381

ITU-T P-SERIES RECOMMENDATIONS

**TERMINALS AND SUBJECTIVE AND OBJECTIVE ASSESSMENT METHODS**

| | | |
|---|---|---|
| Vocabulary and effects of transmission parameters on customer opinion of transmission quality | Series | P.10 |
| **Voice terminal characteristics** | **Series** | **P.30**<br>**P.300** |
| Reference systems | Series | P.40 |
| Objective measuring apparatus | Series | P.50<br>P.500 |
| Objective electro-acoustical measurements | Series | P.60 |
| Measurements related to speech loudness | Series | P.70 |
| Methods for objective and subjective assessment of speech quality | Series | P.80<br>P.800 |
| Audiovisual quality in multimedia services | Series | P.900 |
| Transmission performance and QoS aspects of IP end-points | Series | P.1000 |
| Communications involving vehicles | Series | P.1100 |
| Models and tools for quality assessment of streamed media | Series | P.1200 |
| Telemeeting assessment | Series | P.1300 |
| Statistical analysis, evaluation and reporting guidelines of quality measurements | Series | P.1400 |

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# Recommendation ITU-T P.381

## Technical requirements and test methods for the universal wired headset or headphone interface of digital mobile terminals


### Summary

3.5 mm and 2.5 mm diameter headset/headphone interfaces have been widely used in digital mobile terminals in recent years. Nowadays, the consumer is free to choose either the headset/headphone originally provided by the terminal manufacturer or other ones that are offered separately. However, the QoS/QoE perceived by users is influenced by both the electrical performance of the interface and the compatibility between the terminal and the connected headset/headphone. Recommendation ITU-T P.381 specifies critical physical and electrical-acoustical characteristics for the universal headset interface and provides corresponding test methods.


### History

| Edition | Recommendation | Approval | Study Group |
|---|---|---|---|
| 1.0 | ITU-T P.381 | 2012-08-22 | 12 |


### Keywords

Connector, headphone, headset, interface.

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## Table of Contents

**Page**

# Recommendation ITU-T P.381

## Technical requirements and test methods for the universal wired headset or headphone interface of digital mobile terminals

### 1 Scope

This Recommendation specifies electrical requirements and test methods for the universal analogue headset/headphone interface used in digital mobile terminals.

The principle of this document is to ensure adequate compatibility between the digital mobile terminal and the wired analogue headset/headphone, and to have better user experience. The universality of the headset/headphone interface will facilitate the separation of sales between digital mobile terminals and headsets/headphones. One of the benefits is that the user can be free to choose his or her favourite type of headset or headphone that is available on the market. In the long run, it will reduce e-waste. Furthermore, the universal interface can be used as the electric coupling design in hands-free systems and hearing aids for wider harmonization.

In order to provide instructions to manufacturers and encourage them to adopt the universal headset interface, the mechanical dimensions are shown in Appendices I, II and III.

This Recommendation is applicable to digital mobile terminals with a physical analogue audio output/input interface. Other similar ICT equipment may also refer to this Recommendation.

This Recommendation is not applicable for terminals designed solely for digital headset/headphone usage.

### 2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.


[ITU-T P.10] Recommendation ITU-T P.10/G.100 (2006), *Vocabulary for performance and quality of service.*

[ITU-T P.50] Recommendation ITU-T P.50 (1999), *Artificial voices*.

[ITU-T P.56] Recommendation ITU-T P.56 (2011), *Objective measurement of active speech level*.

[ITU-T P.57] Recommendation ITU-T P.57 (2011), *Artificial ears*.

[ITU-T P.58] Recommendation ITU-T P.58 (2011), *Head and torso simulator for telephonometry*.

[ITU-T P.79] Recommendation ITU-T P.79 (2007), *Calculation of loudness ratings for telephone sets.*

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

[ITU-T P.380] Recommendation ITU-T P.380 (2003), *Electro-acoustic measurements on headsets.*

[ITU-T P.501] Recommendation ITU-T P.501 (2009), *Test signals for use in telephonometry*.

[ITU-T P.502] Recommendation ITU-T P.502 (2000), *Objective test methods for speech communication systems using complex test signals.*

[ITU-T P.581] Recommendation ITU-T P.581 (2009), *Use of head and torso simulator (HATS) for hands-free and handset terminal testing*.

[ITU-T P.863] Recommendation ITU-T P.863 (2011), *Perceptual objective listening quality assessment*.

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

[IEC 60268-7] IEC 60268-7 (2010), *Sound system equipment – Part 7: Headphones and earphones*.

[IEC 61260] IEC 61260 (1995), *Electroacoustics – Octave-band and fractional-octave-band filters*.


# 3 Definitions

## 3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

**3.1.1 composite source signal (CSS)** [ITU-T P.10]: A signal composed in time by various signal elements.

**3.1.2 ear-drum reference point (DRP)** [ITU-T P.10]: A point located at the end of the ear canal, corresponding to the ear-drum position.

**3.1.3 earphone** [IEC 60268-7]: An electroacoustic transducer by which acoustic oscillations are obtained from electric signals and intended to be closely coupled acoustically to the ear.

**3.1.4 headset** [ITU-T P.10]: A device which includes a telephone receiver and transmitter which is typically secured to the head or the ear of the wearer.

**3.1.5 mouth reference point (MRP)** [ITU-T P.10]: Point 25 mm in front of and on the axis of the lip plane of the artificial mouth or a typical human mouth (see Figure 1 of ITU-T P.64).

## 3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

**3.2.1 artificial ear**: A device which incorporates an acoustic coupler and a calibrated microphone for measuring sound pressure, and which has an overall acoustic impedance similar to that of the average adult ear over a given frequency band (based on the definition in [ITU-T P.10]).

**3.2.2 codec**: A combination of an analogue-to-digital encoder and a digital-to-analogue decoder operating in opposite directions of transmission in the same equipment.

**3.2.3 head and torso simulator (HATS) for telephonometry**: A manikin that extends downwards from the top of the head to the waist. It is designed to simulate the sound pick-up characteristics and the acoustic diffraction produced by the average adult, and to reproduce the acoustic field generated by the human mouth (based on the definition in [ITU-T P.10]).

**3.2.4 headphone**: An object based on the assembly of one or two earphones on a headband or chinband, the use of which may be optional (e.g., with intra-concha earphones) (based on the definition in [IEC 60268-7]).

**3.2.5 mean opinion score – listening-only quality objective (MOS-LQO)**: The score is calculated by means of an objective model which aims at predicting the quality for a listening-only test situation. Objective measurements made using the model given in [ITU-T P.863] give results in terms of MOS-LQO.

**3.2.6 receive**: The receiving direction of the signal transmission, usually from the measurement system to the DUT.

**3.2.7 send**: The sending direction of the signal transmission, usually from the DUT to the measurement system.

## 4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

| | |
|---|---|
| ABT | Audio Breakthrough Testing |
| AGC | Automatic Gain Control |
| $A_{H,R,dt}$ | Attenuation range in the receiving direction during double talk |
| $A_{H,S,dt}$ | Attenuation range in the sending direction during double talk |
| dBFS | decibels relative to Full Scale. |
| dBV | decibels relative to 1 Volt |
| dBVp | decibels relative to 1 Volt, psophometrically weighted |
| DRP | Drum Reference Position |
| DUT | Device Under Test |
| EC | Echo Canceller |
| ECM | Electret condenser microphone |
| EL | Echo Loss |
| EMI | Electromagnetic Interference |
| ESD | Electrostatic Discharge |
| G or GND | Ground |
| HATS | Head and Torso Simulator |
| JFET | Junction Field Effect Transistor |
| L | Left audio channel |
| $L_{S,min}$ | minimum activation Level in the sending direction |
| $L_{R,min}$ | minimum activation Level in the receiving direction |
| LQ | Listening Quality |
| LQO | Listening Quality Objective |
| $LQ_R$ | Listening Quality in the receiving direction |
| $LQ_S$ | Listening Quality in the sending direction |
| LQOn | Listening Quality Objective, narrowband |
| $LQO_{SW}$ | Listening Quality Objective superwideband |
| LSB | Lowest Significant Bit |

| | |
|---|---|
| MIC | Microphone |
| MINR | Music to Idle Noise Ratio |
| MONO | Mono Audio Channel |
| MOS | Mean Opinion Score |
| MRP | Mouth Reference Point |
| NLP | Non-linear Processing |
| POI | Point of Interconnection |
| R | Right audio channel |
| $R_{bias}$ | bias Resistance |
| SINR | Signal to Idle Noise Ratio |
| STMR | Sidetone Masking Rating |
| TCLw | Weighted Terminal Coupling Loss |
| TELRdt | Talker Echo Loudness Rating |
| $T_S$ | delay in the sending direction |
| $T_{r,S,min}$ | Minimum built-up time in Send |
| $T_R$ | delay in the receiving direction |
| $T_{TES}$ | system delay |
| UE | User Equipment |
| $V_m$ | maximum output voltage |

## 5 General description

Generally, if a headset or a headphone is used, the overall user experience during a call highly depends on both the terminal and the connected headset/headphone. Although the acoustic quality of the headset/headphone is usually the weak link, more consideration with regard to the physical and electrical performance of the universal interface should be given.

This Recommendation specifies the universal concentric connector interface for successful interconnection between the digital mobile terminal and the headset/headphone, including the plug connector and the socket connector. Normally, the socket connector is fixed inside the terminal, with the outside rim level with the surrounding shell of the terminal. Particularly, if the outside rim is lower than the surrounding shell, the dimension of the plug hand grip shall not influence the precise mating of the connection.

## 6 Physical characteristics

### 6.1 General rules

Two types of concentric socket connectors are recommended for use, the 2.5 mm and the 3.5 mm diameter socket connectors. The isometric view of a plug and socket connector are shown in Figure 1:

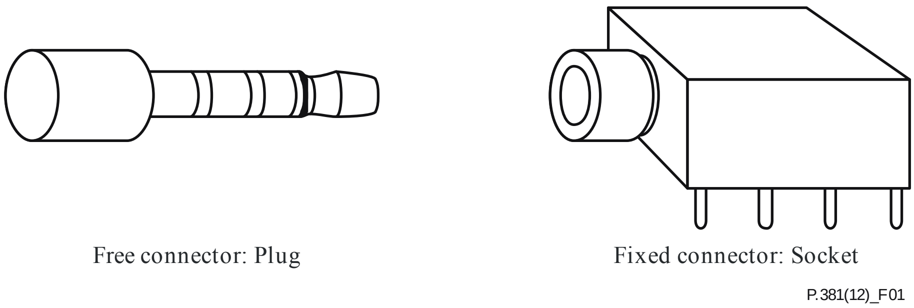



**Figure 1 – Isometric view of a plug and socket connector**

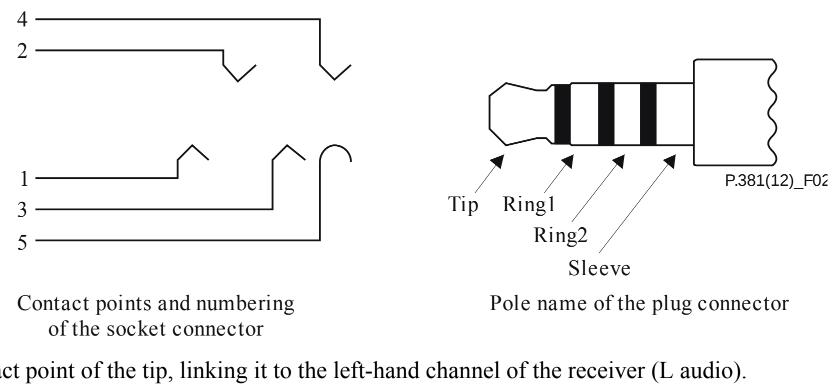
If the terminal is equipped with a headset interface and designed for both communication and audio playing, the fixed connector shall be a 3.5 mm diameter or 2.5 mm diameter concentric socket connector with four contact points. Detailed dimension information about the socket connector with four contact points is given in Appendix I. Some terminals with curved edges may not work well with the dimensions given in Appendix I. For these cases, optional dimensions are given in Appendix II, which is fully compatible with connectors designed to the dimensions of Appendix I.

A socket connector with three contact points is no longer recommended for digital mobile terminals.

NOTE – The contact points here do not include special points reserved for other functions.

### 6.2 Pin assignments

This clause gives an illustration of pin assignments of the socket connector with four contact points and those of the mated plug, as shown in Figure 2. Point 1 of the socket is to be connected to the tip of the plug, linking it to the left-hand channel of the receiver (L audio). Point 2 is to be connected to Ring 1, linking it to the right-hand channel of the receiver (R audio). Point 3 is to be connected to Ring 2, linking it to the transducer (MIC+). Points 4 and 5 are to be connected to the sleeve, linking it to GND.




1 is the contact point of the tip, linking it to the left-hand channel of the receiver (L audio).
2 is the contact point of Ring 1, linking it to the right-hand channel of the receiver (R audio).
3 is the contact point of Ring 2, linking it to the transducer (MIC+).
4 is the contact point of the sleeve, linking it to the GND.
5 is the bushing of the socket, linking it to the GND when it is made of conductive material.

**Figure 2 – Pin assignments of a socket connector with four contact points**

The physical pinout order of the universal interface is important and should coordinate with the connected headset/headphone. The headset pole order from the tip to the sleeve is recommended to be L/R/MIC/GND.

A socket connector with four contact points shall be compatible with both the plug connector with three poles and the plug connector with four poles specified in this document.

NOTE – It is agreed that the pinout order of L/R/MIC/GND has an advantage in electrostatic discharges (ESD) protection and allows for both plastic and metallic convertors. However, there is another type of headset which has different pin assignments which is more commonly used in North America. The pinout order for the latter headset plug is L/R/GND/MIC, from the tip to the sleeve. Depending on the status, it was recommended that terminals should be able to identify both plugs intelligently and automatically.

# 7 Electrical interface specification

## 7.1 Communication mode

### 7.1.1 Test set-up

Test set-ups are shown in Figures 3 and 4.

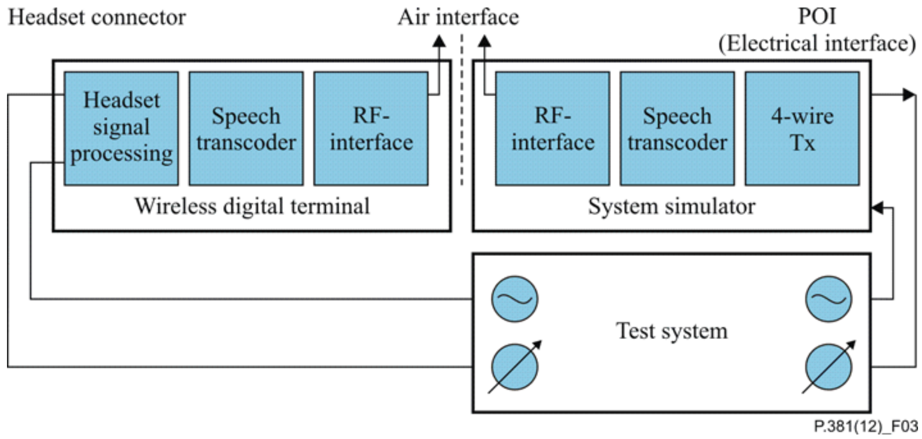



**Figure 3 – Test arrangement for testing the electrical headset interface**

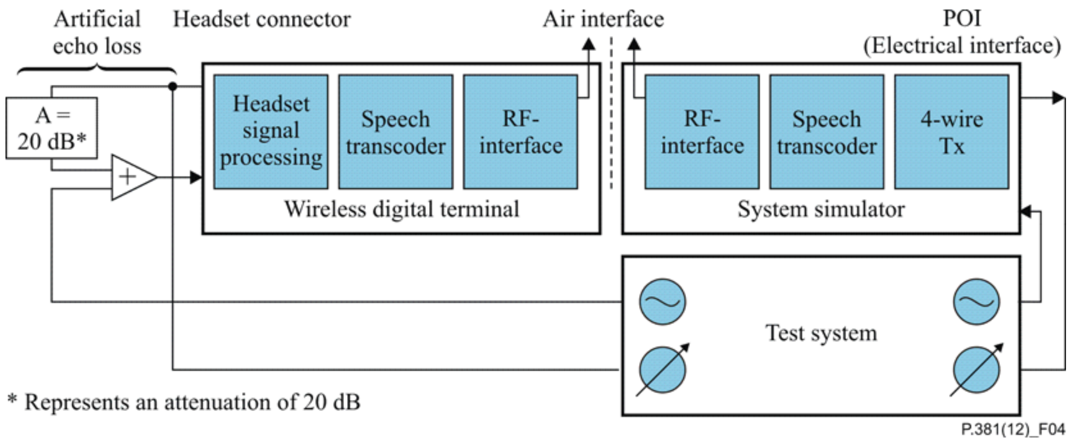



**Figure 4 – Test set-up with artificial echo loss for echo and double talk testing**

#### 7.1.1.1 Input and output characteristics of the test system for connecting to the headset connector

The output of the test system connected to the send interface of the headset connector must be DC resistant. The output impedance shall be less than 1 kΩ. The dynamic range shall be consistent with the level range provided by headset microphones.

The input of the test system connected to the receiving interfaces of the headset connectors shall have an input impedance of 32 Ω. The dynamic range shall be consistent with the output level range provided by the electrical output of digital mobile terminals' headset outputs.

#### 7.1.1.2 Test signals and test signal levels

Speech-like signals are used for the measurements which can be found in [ITU-T P.50] and [ITU-T P.501]. Detailed information about the test signal used is to be found in the corresponding clause of this Recommendation.

All test signals – which are used in Receive – have to be band-limited. The band limitation is achieved by bandpass filtering in the frequency range between 200 Hz and 4 kHz using a bandpass filter providing greater than 24 dB/octave for narrowband digital mobile terminals. In wideband mode, the band limitation is achieved by bandpass filtering in the frequency range between 50 Hz and 8 kHz, using a bandpass filter providing greater than 24 dB/octave. In Send, the test signals are used without band limitation.

All test signal levels are referred to the average level of the test signals, averaged over the complete test sequence length, if not described otherwise.

The nominal average signal levels for the measurements are as follows:

–16 dBm0 in Receive (typical signal level in networks).

–60 dBV in Send (typical equivalent microphone signal level corresponding to –4.7 dBPa at the MRP).

NOTE – If different networks' signal levels are to be used in tests, this is stated in the individual test. The "Lombard effect" (increased talker speech level due to high background noise) is considered in the background noise tests.

Some tests require exact synchronization of test signals in the time domain. Therefore, it is required to take into account the delays of the terminals. When analysing signals, any delay introduced by the test system, codecs and terminals have to be taken into account accordingly.

### 7.1.2 Delay

For further study.

### 7.1.3 Sensitivity in Send

#### 7.1.3.1 Requirement

The sending sensitivity is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulators).

The sending sensitivity shall be –16 dBm0 ±6 dB when inserting the send signal at the nominal level as described in clause 7.1.1.2.

#### 7.1.3.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level, the level is averaged over the complete test signal.

   The measured power density spectrum at the electrical reference point (POI) is used as the reference power-density spectrum for determining the sending sensitivity.

3) For wideband, the sending sensitivity is calculated from 50 Hz to 8 kHz. For narrowband, the frequency range from 100 Hz to 4 kHz is used for calculation.

   For the calculation, the average measured level at the electrical reference interface is used.

4) The sensitivity is expressed in dBm0.

### 7.1.4 Sensitivity in Receive

#### 7.1.4.1 Requirement

The receiving sensitivity is measured from the POI (output of the reference speech coder of the system simulators) to the receiving output of the headset interface.

The receiving sensitivity shall be –32 dBV ±6 dB at the maximum volume setting when inserting the receive signal at the nominal level as described in clause 7.1.1.2.

NOTE – The value –32 dBV ±6 dB is for further study.

#### 7.1.4.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.
2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level.
3) For the calculation, the averaged level at the receiving output of the headset interface is used. For wideband, the receiving sensitivity is determined from 50 Hz to 8 kHz. For narrowband, the frequency range from 100 Hz to 4 kHz is used for calculation.
4) The sensitivity is expressed in dBV.

The measurement is repeated for the second channel.

### 7.1.5 Linearity in Send

#### 7.1.5.1 Requirement (provisional, for further study)

The linearity in Send is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulators).

The test is aimed to detect any amplitude non-linearities including AGC or companding.

The sensitivity difference measured for the level range from –20 dB to 5 dB relative to the nominal signal level shall be 0 ± 3 dB.

#### 7.1.5.2 Test (provisional, for further study)

1) The test signal used for the measurements shall be CSS according to [ITU-T P.501]. The test signals are in the range of –20 dB to 5 dB relative to the nominal signal level as defined in clause 7.1.1.2 in steps of 5 dB, applied at the sending input of the headset interface. The test signal level is the average level of the complete test signal.

   The measured power density spectrum at the sending input of the headset interface is used as the reference power-density spectrum for determining the sending sensitivity.
2) The test arrangement is according to clause 7.1.1. In wideband, the sensitivity is determined from 50 Hz to 8 kHz. For narrowband, the frequency range from 100 Hz to 4 kHz is used for calculation.

   For the calculation, the average measured level at the electrical reference point is referred to the average test signal level measured at the headset interface.
3) The sensitivity is expressed in dBV/V.

### 7.1.6 Linearity in Receive

#### 7.1.6.1 Requirement (provisional, for further study)

The linearity in Receive is measured from the POI (output of the reference speech coder of the system simulators) to the receiving output of the headset interface.

The test is aimed to detect any amplitude non-linearities including AGC or companding.

The sensitivity difference measured for the level range from –20 dB to 5 dB relative to the nominal signal level shall be 0 ± 3 dB.

#### 7.1.6.2 Test (provisional, for further study)

1) The test arrangement is according to clause 7.1.1, Figure 3.
2) The test signal used for the measurements shall be CSS according to [ITU-T P.501]. The test signals are in the range of –20 dB to 5 dB in steps of 5 dB relative to the nominal signal level applied to the POI. The test signal level is the average level of the complete test signal.

   The measured power density spectrum at the POI is used as the reference power-density spectrum for determining the receive sensitivity.
3) In wideband, the sensitivity is determined from 50 Hz to 8 kHz. For narrowband, the frequency range from 100 Hz to 4 kHz is used for calculation.

   For the calculation, the average measured level at the test point for each frequency band is referred to the average test signal level measured in each frequency band at the electrical reference interface.
4) The sensitivity is expressed in dBV/V.

The measurement is repeated for the second channel.

### 7.1.7 Sending sensitivity frequency response

#### 7.1.7.1 Requirement (provisional, for further study)

The sending frequency response is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulators). The sending sensitivity response should be mostly flat in the entire frequency range in order to comply with a large variety of headsets, which in combination with the digital mobile terminal should comply with the relevant standards in Send.

The measured frequency response shall be within the limits as defined in Table 1 for narrowband and Table 2 for wideband.

**Table 1 – Tolerance mask for the narrowband sending sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 2 | –2 |
| 3 100 | 2 | –2 |
| 3 500 | 2 | –5 |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

**Table 2 – Tolerance mask for the wideband sending sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 2 | –2 |
| 6 200 | 2 | –2 |
| 7 000 | 2 | –5 |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

#### 7.1.7.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level, applied to the sending input of the headset interface. The level is averaged over the complete test signal.

The measured power density spectrum at the sending input of the headset interface is used as the reference power density spectrum for determining the sending sensitivity.

3) In wideband, the sending sensitivity is determined in third octave intervals, as given by [IEC 61260] for frequencies from 100 Hz to 8 kHz inclusive, measured at the POI. In narrowband, it is determined for frequencies from 200 Hz to 4 kHz. In each third octave band, the level of the measured signal is referred to the level of the reference signal averaged over the complete test sequence length.

4) The sensitivity is determined in dBV/V.

### 7.1.8 Receiving sensitivity frequency response

#### 7.1.8.1 Requirement

The receiving frequency response is measured from the POI (output of the reference speech coder of the system simulators) to the receiving output of the headset interface. The receiving sensitivity response should be mostly flat in the entire frequency range in order to comply with a large variety of headsets, which in combination with the digital mobile terminal should comply with the relevant standards in Receive.

The measured frequency response shall be within the limits as defined in Table 3 for narrowband and Table 4 for wideband.

**Table 3 – Tolerance mask for the narrowband receiving sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 2 | –2 |
| 3 100 | 2 | –2 |
| 3 500 | 2 | –5 |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

**Table 4 – Tolerance mask for the wideband receiving sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 2 | –2 |
| 6 200 | 2 | –2 |
| 7 000 | 2 | –5 |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

#### 7.1.8.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal is the nominal signal level, applied to the POI. The level is averaged over the complete test signal.

3) For wideband, the receiving sensitivity is determined in third octave intervals as given by [IEC 61260] for frequencies from 100 Hz to 8 kHz inclusive, measured at the headset interface. In narrowband, it is determined for frequencies from 200 Hz to 4 kHz. In each

third octave band, the level of the measured signal is referred to the level of the reference signal, averaged over the complete test sequence length.

4) The sensitivity is determined in dBV/V.

The measurement is repeated for the second channel.

### 7.1.9 Sidetone loss

#### 7.1.9.1 Requirement

The talker sidetone masking rating (STMR, electrical sidetone) is measured from the sending input of the headset interface to the receiving output of the headset interface.

The STMR shall be greater than 15 dB and should be less than 30 dB for the nominal setting of the volume control. For all other positions of the volume control, the STMR shall be greater than 10 dB.

NOTE 1 – Where a user-controlled receiving volume control is provided, it is recommended that the sidetone loss is independent of the volume control setting.

NOTE 2 – In general, it is recommended to provide an electrical sidetone path in the terminal for headset UE.

#### 7.1.9.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level, applied to the sending input of the headset interface. The level is averaged over the complete test signal.

   The measured power density spectrum at the sending input of the headset interface is used as the reference power density spectrum for determining the sending sensitivity.

3) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in [ISO 3] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band ([ITU-T P.79], Table 3, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

4) The measured result is corrected by the nominal sensitivities of the headset interface thus transferring the measured electrical signal levels to their equivalent acoustical signal levels when assuming an ideal headset.

5) The sidetone path loss (LmeST), as expressed in dB, and the sidetone masking rate (STMR) (in dB) shall be calculated from the formula 5-1 of [ITU-T P.79], using m = 0.225 and the weighting factors of in Table 3 of [ITU-T P.79].

### 7.1.10 Sidetone delay

#### 7.1.10.1 Requirement

The sidetone delay is measured from the sending input of the headset interface to the receiving output of the headset interface.

The maximum sidetone round-trip delay shall not exceed 5 ms.

#### 7.1.10.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The test signal is a CS-signal complying with [ITU-T P.501] using a pn sequence with a length of 4 096 points (for the 48 kHz sampling rate) which equals to the period T. The duration of the complete test signal is as specified in [ITU-T P.501]. The test signal level is the nominal signal level, applied to the sending input of the headset interface.

3) The cross-correlation function $\Phi xy(\tau)$ between the input signal $S_x(t)$ generated by the test system in Send and the output signal $S_y(t)$ measured at the receiving output of the headset interface is calculated in the time domain:

$$\Phi_{xy}(\tau) = \frac{1}{T} \int_{t=\frac{-T}{2}}^{\frac{T}{2}} S_x(t) \cdot S_y(t+\tau) \qquad (1)$$

The measurement window T shall be exactly identical to the time period T of the test signal, the measurement window is positioned to the pn-sequence of the test signal.

The sidetone delay is calculated from the envelope $E(\tau)$ of the cross-correlation function $\Phi xy(\tau)$. The first maximum of the envelope function occurs in correspondence with the test signal at the sending input of the headset interface, the second one occurs with a possible delayed sidetone signal. The difference between the two maxima corresponds to the sidetone delay. The envelope $E(\tau)$ is calculated by the Hilbert transformation H $\{xy(\tau)\}$ of the cross-correlation:

$$H\{xy(\tau)\} = \sum_{u=-\infty}^{+\infty} \frac{\Phi_{xy}(u)}{\pi(\tau - u)} \qquad (2)$$

$$E(\tau) = \sqrt{[\Phi_{xy}(\tau)]^2 + [H\{xy(\tau)\}]^2} \qquad (3)$$

It is assumed that the measured sidetone delay is less than T/2.

### 7.1.11 Idle channel noise in Send

#### 7.1.11.1 Requirement (provisional, for further study)

The idle channel noise in Send is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulators).

The signal to idle noise ratio (SINR) shall be higher than 30 dB.

Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

#### 7.1.11.2 Test (provisional, for further study)

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501] at nominal signal level as described in 7.1.1.2 is applied at the sending input of the headset interface. The test signal level is the average level of the complete test signal. The output level is measured using a speech level voltmeter according to [ITU-T P.56]. This level is the reference speech signal level.

3) For the noise measurement, no test signal is used. However, all sources which potentially contribute to noise at the output such as GSM noise, electrical noise introduced and others must be considered. In order to ensure a reliable activation, a conditioning sequence is inserted before the actual measurement. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

4) The idle channel noise is measured at the output in the frequency range between 100 Hz and 8 kHz. The measurement requires a time window which starts exactly at the end of the activation signal. Any "ringing" of filters or receivers shall be taken into account, the time window must be shifted accordingly. The length of the time window is 1 s which is the

averaging time for the idle channel noise. The test lab has to ensure the correct activation of the DUT during the measurement. If the DUT is deactivated during measurement, the measurement window has to be cut to the duration when the DUT remains activated.

The power density spectrum of the noise signal is determined using FFT (8 k samples/48 kHz sampling rate or equivalent). A Hann window is used.

If it is known that the DUT stays activated without any activation signal, no activation signal is required. In this case, a simple noise measurement is conducted.

5) The idle channel noise is determined by A-weighting and referring to the reference speech signal level as determined with the speech sequence. Spectral peaks are measured in the frequency domain. The average noise spectrum used for determining the spectral peak should be calculated as the arithmetic mean of the noise spectrum values when stated in dBV.

### 7.1.12 Idle channel noise in Receive

#### 7.1.12.1 Requirement (provisional, for further study)

The idle channel noise in Receive is measured from the POI (output of the reference speech coder of the system simulators) to the receiving output of the headset interface.

The signal to idle noise ratio (SINR) shall be higher than 30 dB.

Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

#### 7.1.12.2 Test (provisional, for further study)

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The British-English single talk sequence described in clause 7.3.2 [ITU-T P.501] at the nominal signal level as described in clause 7.1.1.2 is applied at the POI. The test signal level is the average level of the complete test signal. The output level is measured using a speech level voltmeter according to [ITU-T P.56]. This level is the reference speech signal level.

3) For the noise measurement, no test signal is used. However, all sources which potentially contribute to noise at the output such as GSM noise, electrical noise introduced and others must be considered. In order to ensure a reliable activation, a conditioning sequence is inserted before the actual measurement. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

4) The idle channel noise is measured at the output in the frequency range between 100 Hz and 8 kHz. The measurement requires a time window which starts exactly at the end of the activation signal. Any "ringing" of filters or receivers shall be taken into account, the time window must be shifted accordingly. The length of the time window is 1 s which is the averaging time for the idle channel noise. The test lab has to ensure the correct activation of the device under test (DUT) during the measurement. If the DUT is deactivated during measurement, the measurement window has to be cut to the duration when DUT remains activated.

The power density spectrum of the noise signal is determined using FFT (8 k samples/48 kHz sampling rate or equivalent). A Hann window is used.

If it is known that the DUT stays activated without any activation signal, no activation signal is required. In this case, a simple noise measurement is conducted.

5) The idle channel noise is determined by A-weighting and referring to the reference speech signal level as determined with the speech sequence. Spectral peaks are measured in the

frequency domain. The average noise spectrum used for determining the spectral peak should be calculated as the arithmetic mean of the noise spectrum values when stated in dBV.

The measurement is repeated for the second channel.

### 7.1.13 Sending distortion

#### 7.1.13.1 Requirement

The distortion in Send is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulator).

The ratio of signal to harmonic distortion shall be above the following mask.

**Table 5 – Limits for the signal to harmonic distortion**

| Frequency (Hz) | Signal to harmonic distortion ratio limit, Send (dB) |
|---|---|
| 315 | 30 |
| 400 | 40 |
| 1 000 | 40 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 7.1.13.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.
2) For the test, a sinusoidal signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is used. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be the nominal signal level. In order to ensure a reliable activation, a conditioning sequence is inserted before the actual measurement. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.
3) The signal to harmonic distortion ratio is measured selectively up to 7 kHz.
4) The test is repeated using a signal level 10 dB higher than the nominal signal level. The level of the activation signal is kept at the nominal signal level.

### 7.1.14 Receive distortion

#### 7.1.14.1 Requirement

The distortion in Receive is measured from the POI (output of the reference speech coder of the system simulator) to the receiving output of the headset interface.

The ratio of signal to harmonic distortion shall be above the following mask.

**Table 6 – Limits for the signal to harmonic distortion**

| Frequency (Hz) | Signal to harmonic distortion ratio limit, receive (dB) |
|---|---|
| 315 | 30 |
| 400 | 40 |
| 1 000 | 40 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

### 7.1.14.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 3.
2) For the test, a sinusoidal signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is used. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be the nominal signal level. In order to ensure a reliable activation, a conditioning sequence is inserted before the actual measurement. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.
3) The signal to harmonic distortion ratio is measured selectively up to 7 kHz.
4) The test is repeated using a signal level 10 dB higher than the nominal signal level. The level of the activation signal is kept at the nominal signal level.

### 7.1.15 Noise cancellation test in Send

### 7.1.15.1 Requirement (provisional, for further study)

The noise cancellation in Send is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulators).

The objective of this test is to check the performance of the noise cancellation in the Send.

When testing through the objective methodology, the terminal shall comply with the following requirements:

For narrowband terminals:

| | |
|---|---|
| N-MOS-LQOn | Average N-MOS-LQOn $\geq$ 3.0 |
| S-MOS-LQOn | Average S-MOS-LQOn $\geq$ 3.0 |
| G-MOS-LQOn | Average G-MOS-LQOn $\geq$ 3.0 |

For wideband terminals:

| | |
|---|---|
| N-MOS-LQOw | Average N-MOS-LQOw $\geq$ 3.0 |
| S-MOS-LQOw | Average S-MOS-LQOw $\geq$ 3.0 |
| G-MOS-LQOw | Average G-MOS-LQOw $\geq$ 3.0 |

### 7.1.15.2 Test (provisional, for further study)

In order to create a representative electrical test signal for the headset interface containing speech at a nominal level mixed with the amount of background noise picked up by a representative headset, the set-up in [ETSI EG 202 396-1] is used. The representative headset is connected to a reference interface providing nominal properties for the electrical interface as described in clause 7.1.1.1. The signal (speech plus noise) is recorded at this interface and inserted through the appropriate reference interfaces as described in clause 7.1.1.1 in such a way that the signal level and spectral content delivered to the terminal under test is equivalent to the one it would have seen if the headset was connected directly. Either headsets considered to be representative for the type of headset attached to the terminal are used or individual headsets are used. In addition, the unprocessed speech plus noise signal is recorded at the headset microphone position using a reference microphone positioned close to the headset microphone.

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The near-end speech signal consists of eight sentences of speech (two male and two female talkers, two sentences each). Appropriate speech samples shall be taken from [ITU-T P.501] and [ITU-T P.50]. The language shall be a mixture of American and British English. The test signal level is the level produced by the representative headset when the speech plus noise signal is applied to the headset with a speech level of –1.7 dBPa at the MRP. The exact test sequence should be as follows:

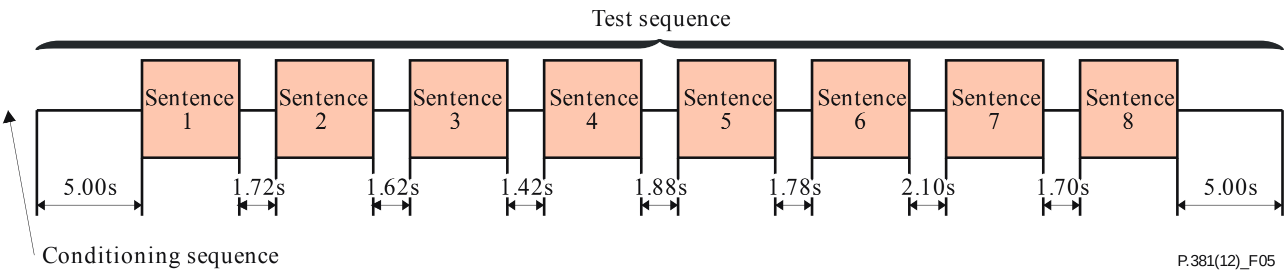



**Figure 5 – Near-end speech sequence for noise cancellation test in Send**

The following sentences shall be used:

Sentence 1 & 2: Male1, BE (ITU-T P.50, Appendix I, EN2M06)

1 He could not remember his name.

2 I never can leave you two alone.

Sentence 3 & 4: Male2, AE (ITU-T P.501, AE_Male1)

3 The shelves were bare of both jam or crackers.

4 A joy to every child is the swan boat.

Sentence 5 & 6: Female1, BE (ITU-T P.501 Version 2004, BE_Female1)

5 You must go and do it at once.

6 There were several small outhouses.

Sentence 7 & 8: Female2, AE (ITU-T P.501, AE_Female2)

7 The stems of the tall glasses cracked and broke.

8 The wall phone rang loud and often.

3) The test signal is applied to the headset interface.

4) A proper conditioning sequence should be used in advance of the measurement. Appropriate conditioning sequences can be found in [ITU-T P.501].

5) The noise types as described in Table 7 shall be used [b-ETSI EG 202 396-1].

6) The measurement over the eight noise types shall be made in the same unique and dedicated call and not in the same call as for example the one established for acoustic measurement. The noise types shall be presented according to the order specified in Table 7.

**Table 7 – Noises used for background noise simulation**

| Description | File name | Duration | Level | Type |
|---|---|---|---|---|
| Recording in pub | Pub_Noise_binaural_V2 | 30 s | L: 75,0 dB(A)<br>R: 73,0 dB(A) | Binaural |
| Recording at pavement | Outside_Traffic_Road_binaural | 30 s | L: 74,9 dB(A)<br>R: 73,9 dB(A) | Binaural |
| Recording at pavement | Outside_Traffic_Crossroads_binaural | 20 s | L: 69,1 dB(A)<br>R: 69,6 dB(A) | Binaural |
| Recording at departure platform | Train_Station_binaural | 30 s | L: 68,2 dB(A)<br>R: 69,8 dB(A) | Binaural |
| Recording at the drivers position | Fullsize_Car1_130Kmh_binaural | 30 s | L: 69,1 dB(A)<br>R: 68,1 dB(A) | Binaural |
| Recording at sales counter | Cafeteria_Noise_binaural | 30 s | L: 68,4 dB(A)<br>R: 67,3 dB(A) | Binaural |
| Recording in a cafeteria | Mensa_binaural | 22 s | L: 63,4 dB(A)<br>R: 61,9 dB(A) | Binaural |
| Recording in business office | Work_Noise_Office_Callcenter_binaural | 30 s | L: 56,6 dB(A)<br>R: 57,8 dB(A) | Binaural |

### 7.1.16 One-way speech quality in Send

#### 7.1.16.1 Requirement (provisional, for further study)

The listening speech quality in Send $LQ_S$ is measured from the sending input of the headset interface to the POI (input of the reference speech coder of the system simulators).

The listening speech quality in Send shall be

MOS-LQOn > 4.2 in narrowband

MOS-LQOsw > 4.5 in wideband

#### 7.1.16.2 Test

The tests method to be used is [ITU-T P.863].

NOTE – [ITU-T P.863] does not provide a wideband mode. Wideband systems are evaluated on a superwideband scale. Therefore the mean opinion score (MOS) requirements are given in MOS-LQOsw. See [ITU-T P.863] for more information.

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The test signals used are the English test sequences as specified in [ITU-T P.501] (two male speakers, two female speakers, two sentences each). The test signal level is the nominal signal level. The test signal level is measured as "active speech level" according to [ITU-T P.56]. The speech activity should be between 30% and 70%.

   The original speech signal is used as the reference signal for the determination of the speech quality.

3) The test arrangement is according to clause 7.1.1. For wideband systems MOS-LQOsw is determined. For narrowband systems MOS-LQOn is determined.

   The calculation is made using the signal recorded at the POI.

### 7.1.17 One-way speech quality in Receive

#### 7.1.17.1 Requirement (provisional, for further study)

The speech quality in Receive $LQ_R$ is measured from the POI (output of the reference speech coder of the system simulators) to the receiving output of the headset interface.

The listening speech quality in receive shall be

MOS-LQOn >4.2 in narrowband

MOS-LQOsw >4.5 in wideband

#### 7.1.17.2 Test

The tests method to be used is [ITU-T P.863].

NOTE – [ITU-T P.863] does not provide a wideband mode. Wideband systems are evaluated on a superwideband scale. Therefore the MOS requirements are given in MOS-LQOsw. See [ITU-T P.863] for more information.

1) The test arrangement is according to clause 7.1.1, Figure 3.
2) The test signals used are the English test sequences as specified in [ITU-T P.501] (two male speakers, two female speakers, two sentences each). The test signal level is the nominal signal level. The test signal level is measured as "active speech level" according to [ITU-T P.56]. The speech activity should be between 30% and 70%.

   The original speech signal is used as the reference signal for the determination of the speech quality.
3) The test arrangement is according to clause 7.1.1. For wideband systems MOS-LQOsw is determined, for narrowband systems, it is MOS-LQOn. The signal measured at the headset interface is used for the calculation.

The measurement is repeated for the second channel.

### 7.1.18 Terminal coupling loss TCLw

#### 7.1.18.1 Requirement

The terminal coupling loss TCLw is measured from the POI (input of the reference speech coder of the system simulator) to the POI (output of the reference speech coder of the system simulator).

The TCLw provided by the headset signal processing in conjunction with typical echo paths as described in Figure 4 shall be not less than 55 dB.

#### 7.1.18.2 Test

1) The test arrangement is according to clause 7.1.1. For the test, an artificial echo path is inserted as shown in Figure 4.
2) The attenuation between the input of the test point (POI) to the output of the test point (POI) is determined.
3) Before the actual measurement, a training sequence consisting of the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501] is inserted. The level is averaged over the complete test signal. The training sequence level is the nominal signal level.
4) The test signal is a pn sequence, according to [ITU-T P.501], with a length of 4 096 points (48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms, the test signal level is –3 dBm0 as applicable to the individual interface. The low crest factor is achieved by random alternation of the phase between –180° and +180°.
5) $TCL_W$ is calculated according to clause B.4 of [ITU-T G.122] (trapezoidal pseudo rule). For the calculation, the average measured echo level at each frequency band is referred to the average level of the test signal measured in each frequency band. For the measurement, a

time window has to be applied which is adapted to the duration of the actual test signal (250 ms).

### 7.1.19 Temporal echo effects

#### 7.1.19.1 Requirement

Temporal echo effects are measured from the POI (input of the reference speech coder of the system simulator) to the POI (output of the reference speech coder of the system simulator).

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum TCL measured during the test.

#### 7.1.19.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 4.

2) The test signal consists of the periodically repeated composite source signal according to [ITU-T P.501] with an average level of –5 dBm0 as well as an average level of –25 dBm0. The echo signal is analysed during a period of at least 2.8 s which represents eight periods of the CS signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal. The TCL variation is compared to the maximum TCL achieved in the test.

3) The measurement result is displayed as attenuation vs. time. The exact synchronization between the input and output signal has to be guaranteed.

NOTE 1 – In addition, tests with speech signals as described in [ITU-T P.501] should be carried out to see the time variant behaviour of the EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech-like signals.

NOTE 2 – The analysis is conducted only during the active signal part, the pauses between the composite source signals are not analysed. The analysis time is reduced by the integration time of the level analysis (35 ms).

### 7.1.20 Double talk performance

During double talk the speech is mainly determined by two parameters: impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the "Talker Echo Loudness Rating" should be high and the attenuation inserted should be as low as possible. Terminals which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see [ITU-T P.340] and [ITU-T P.502]):

- attenuation range in Send during double talk $A_{H,S,dt}$
- attenuation range in Receive during double talk $A_{H,R,dt}$
- echo attenuation during double talk.

The double talk performance may be highly influenced by the performance of the echo canceller, especially by the NLP implementation.

#### 7.1.20.1 Attenuation range in the sending direction during double talk $A_{H,S,dt}$

##### 7.1.20.1.1 Requirement

Based on the level variation in Send during double talk $A_{H,S,dt}$ the behaviour of the headset signal processing can be classified according to Table 8.

**Table 8 – Categorization of double talk capability according to [ITU-T P.340]**

| **Category (according to [ITU-T P.340])** | **1** | **2a** | **2b** | **2c** | **3** |
|---|---|---|---|---|---|
| | Full duplex capability | Partial duplex capability | | | No duplex capability |
| $A_{H,S,dt}$ [dB] | ≤3 | ≤6 | ≤9 | ≤12 | >12 |

In general, Table 8 provides a quality classification of the headset signal processing regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance, is of high quality concerning the overall quality as well.

### 7.1.20.1.2 Test

The test signal to determine the attenuation range during double talk is shown in Figure 6. A sequence of uncorrelated CS signals is used which is inserted in parallel in both Send and Receive.

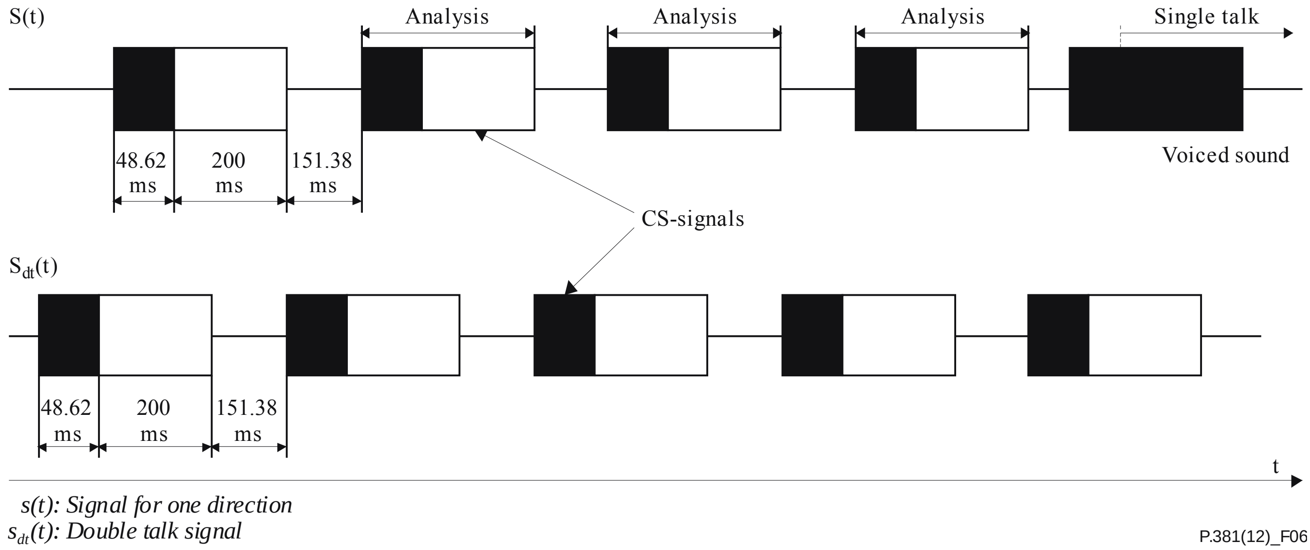



**Figure 6 – Double talk test sequence with overlapping CS signals in Send and Receive**

Figure 6 indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the pn-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in the sending and receiving directions. The analysis times are shown in Figure 6 as well. The test signals are synchronized in time at the headset interface. The delay of the test arrangement should be constant during the measurement.

NOTE – The length of voiced sound of the double talk signal is achieved by repeating one period of the voiced sound for double talk according to [ITU-T P.501] 10 times and cutting off the initial 3,3 ms of the period of the first voiced sound.

The settings for the test signals are as follows.

**Table 9 – Signal levels for double talk tests in Send and Receive**

| | **Receive (sdt(t))** | **Send (s(t))** |
|---|---|---|
| Pause length between two signal bursts | 151.38 ms | 151.38 ms |
| Average signal level (Assuming an original pause length of 101,38 ms) | –16 dBm0 | –60 dBV |
| Active signal parts | –14.7 dBm0 | –58.7 dBV |

1) The test arrangement is according to clause 7.1.1, Figure 4; the test signal is shown in Figure 8.

2) When determining the attenuation range in Send, the signal measured at the receiving interface (POI) is referred to the test signal inserted.

3) The level is determined as level vs time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in Send until its complete activation (during the pause in the receiving channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

4) The categorization is made according to Appendix III of [ITU-T P.502].

### 7.1.20.2 Attenuation range in Receive during double talk $A_{H,R,dt}$

#### 7.1.20.2.1 Requirement

Based on the level variation in Receive during double talk $A_{H,R,dt}$ the behaviour of the headset signal processing can be classified according to Table 10.

**Table 10 – Categorization of double talk capability according to [ITU-T P.340]**

| **Category (according to [ITU-T P.340])** | **1** | **2a** | **2b** | **2c** | **3** |
|---|---|---|---|---|---|
| | Full duplex capability | Partial duplex capability | | | No duplex capability |
| $A_{H,R,dt}$ [dB] | ≤3 | ≤5 | ≤8 | ≤10 | >10 |

In general, this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance, is of high quality concerning the overall quality as well.

#### 7.1.20.2.2 Test

The test signal to determine the attenuation range during double talk is shown in Figure 4. A sequence of uncorrelated CS signals is used which is inserted in parallel in Send and Receive. The test signals are synchronized in time at the POI. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows:

**Table 11 – Signal levels for double talk tests in Send and Receive**

| | **Receive (s(t))** | **Send (s(t))** |
|---|---|---|
| Pause length between two signal bursts | 151.38 ms | 151.38 ms |
| Average signal level<br>(Assuming an original pause length of 101,38 ms) | –16 dBm0 | –60 dBV |
| Active signal parts | –14.7 dBm0 | –58.7 dBV |

1) The test arrangement is according to clause 7.1.1, Figure 4; the test signal is shown in Figure 8.
2) When determining the attenuation range in Receive the signal measured at the sending interface referred to the test signal inserted.
3) The level is determined as level vs time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in Receive until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.
4) The categorization is made according to [ITU-T P.502], Appendix III.

### 7.1.20.3 Detection of echo components during double talk

#### 7.1.20.3.1 Requirement

Echo loss (EL) during double talk is the echo suppression provided by the headset signal processing during double talk measured at the receiving interface.

NOTE – The echo attenuation during double talk is based on the parameter "Talker echo loudness rating" (TELRdt). It is assumed that the terminal at the opposite end of the connection provides the nominal loudness rating (SLR + RLR = 10 dB).

Under these conditions the requirements given in Table 12 are applicable (more information can be found in Annex A of [ITU-T P.340].

**Table 12 – Categorization of double talk capability according to [ITU-T P.340]**

| **Category (according to [ITU-T P.340])** | **1** | **2a** | **2b** | **2c** | **3** |
|---|---|---|---|---|---|
| | Full duplex capability | Partial duplex capability | | | No duplex capability |
| Echo loss [dB] | ≥27 | ≥23 | ≥17 | ≥11 | <11 |

#### 7.1.20.3.2 Test

1) The test arrangement is according to clause 7.1.1, Figure 4.
2) The double talk signal consists of a sequence of orthogonal signals which are realized by voice-like modulated sine waves spectrally shaped, similar to speech. A detailed description can be found in [ITU-T P.501]. For narrowband, the narrowband test signals are used; for wideband, the wideband test signals as described in [ITU-T P.501] are used.

3) The signals are fed simultaneously in Send and Receive. The level in Send at the headset interface is –60 dBV (nominal level), the level in Receive at the POI is –16 dBm0 (nominal level).

4) The test signal is measured at the receiving interface. The measured signal consists of the double talk signal which was fed in at the sending interface and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and a bandwidth according to the signal components of the signal in Receive (see [ITU-T P.501]).The filter will suppress frequency components of the double talk signal.

5) For each frequency band which is used in Receive the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if in any frequency band the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on Table 12. The echo attenuation is to be achieved for each individual frequency band according to the different categories.

### 7.1.21 Activation in Send

The activation in Send is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level in Send ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in Send during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described below is always referred to the test signal level at the headset interface.

#### 7.1.21.1 Requirements (provisional, for further study)

The minimum activation level $L_{S,min}$ should be not greater than –75 dBV.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should not be longer than 50 ms.

#### 7.1.21.2 Test

The structure of the test signal is shown in Figure 9. The test signal consists of CSS components according to [ITU-T P.501] with increasing levels for each CSS burst.

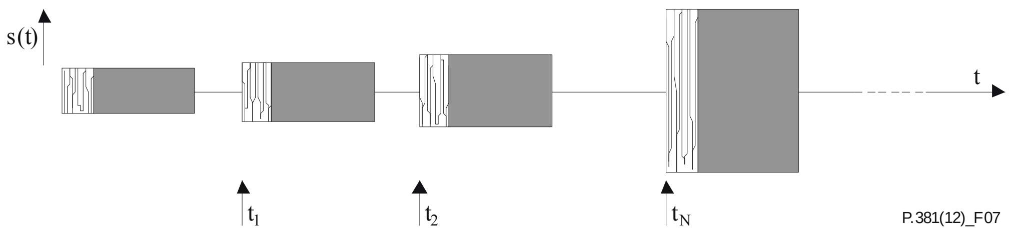



**Figure 7 – Test signal to determine the minimum activation level and the built-up time**

The settings of the test signal are as follows.

**Table 13 – Settings of the CSS in Send**

| | CSS duration/ pause duration | Level of the first CS signal (active signal part at the MRP) | Level difference between two periods of the test signal |
|---|---|---|---|
| CSS to determine switching characteristic in Send | 248.62 ms/451.38 ms | –78.3 dBV (Note 1) | 1 dB |
| NOTE 1 – The level of the active signal part corresponds typically to an average level of –80 dBV at the headset interface for the CSS according to [ITU-T P.501], assuming a pause of 101.38 ms. | | | |

It is assumed that the pause length of 451.38 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

1) The test arrangement is according to clause 7.1.1, Figure 3.

2) The level of the transmitted signal is measured at the POI. The measured signal level is referred to the test signal level and displayed versus time. The levels are calculated from the time domain using an integration time of 5 ms.

3) The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE – If the measurement using the CS signal does not allow the minimum activation level to be clearly identified, the measurement may be repeated by using the one syllable word as described in [ITU-T P.501] instead of the CS signal. The word used should be of similar duration, the average level of the word must be adapted to the CS signal level of the according CSS burst.

## 7.2 Multimedia playback (provisional, for further study)

### 7.2.1 Test set-up

The test set-up is shown in Figure 8.

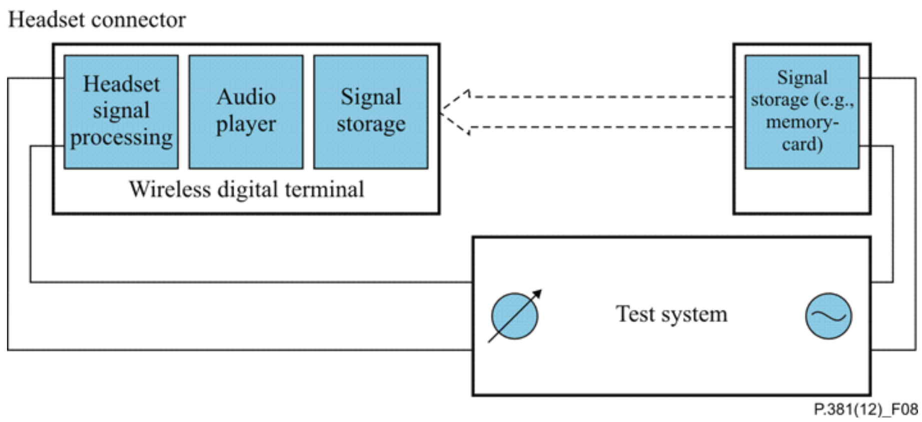



**Figure 8 – Test arrangement for testing the electrical headset interface**

#### 7.2.1.1 Input and output characteristics of the test system for connecting to the headset connector

The input of the test system connected to the Receive interfaces of the headset connectors shall have an input impedance of 32 Ohm. The dynamic range shall be consistent with the output level range provided by the electrical output of digital mobile terminals' headset outputs.

In case the microphone stays connected during the tests the output of the test system connected to the sending interface of the headset connector must be DC resistant. The output impedance shall be < 1 kOhm. The dynamic range shall be consistent with the level range provided by headset microphones.

#### 7.2.1.2 Test signals and test signal levels

For multimedia playback, the test signals have to be downloaded in the appropriate format (e.g., *.wav, *.mp3, *.aac, ….) for the phone under test. All test signals used are in the 16 bit *.pcm format and then coded into the appropriate format. All signal levels stated in this section are relative to dBFS.

Music signals or music simulating noise are used for the measurements. Detailed information about the test signal used is to be found in the corresponding clause of this Recommendation.

All test signals – which are used in receive – have to be band-limited. The band limitation is achieved by bandpass filtering in the frequency range between 20 Hz and 20 kHz using a bandpass filter providing ≥ 24 dB/octave.

All test signal levels are referred to the average level of the test signals, averaged over the complete test sequence length, if not described otherwise.

The nominal average signal level for the measurements is –26 dBFS.

Some tests require exact synchronization of test signals in the time domain. Therefore, it is required to take into account the delays of the terminals. When analysing signals, any delay introduced by the test system, codecs and terminals have to be taken into account accordingly.

### 7.2.2 Sensitivity in multimedia playback mode

#### 7.2.2.1 Requirement

The sensitivity is measured as the output level generated by the output of the mobile terminal player when playing back the pre-recorded music signal at the output of the headset interface.

The sensitivity shall be –42dBV ±6 dB at a maximum volume setting when inserting the signal at the nominal level as described in clause 7.2.1.2.

#### 7.2.2.2 Test

1) The test arrangement is according to clause 7.2.1, Figure 8.
2) The test signal used for the measurements shall be a full-band music signal providing sufficient signal energy from 20 Hz to 20 kHz. The test signal level is the nominal signal level.
3) For the calculation, the averaged level at the output of the headset interface is used. The sensitivity is determined from 20 Hz to 20 kHz.

   For the calculation, the average signal level measured at the output of the headset interface is used.
4) The sensitivity is expressed in dBV.

The measurement is repeated for the second channel.

### 7.2.3 Sensitivity frequency response in multimedia playback mode

#### 7.2.3.1 Requirement

The frequency response is measured as the output level generated by the output of the mobile terminal player when playing back the pre-recorded music signal at the output of the headset interface. The receiving sensitivity response should be mostly flat in the entire frequency range in order to comply with a large variety of headsets which in combination with the digital mobile terminal should comply with the relevant standards in Receive.

The measured frequency response shall be within the limits as defined in Table 14.

**Table 14 – Tolerance mask for the wideband receiving sensitivity frequency response in multimedia playback mode**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 50 | 2 | –2 |
| 12 000 | 2 | –2 |
| 16 000 | 2 | –5 |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

#### 7.2.3.2 Test

1) The test arrangement is according to clause 7.2.1, Figure 8.
2) The test signal used for the measurements shall be a full-band music signal providing sufficient signal energy from 20 Hz to 20 kHz. The test signal is the nominal signal level. The level is averaged over the complete test signal.
3) The sensitivity frequency response is determined in third octave intervals as given by [IEC 61260] for frequencies between 50 Hz and 16 kHz, inclusive. In each third octave band, the level of the measured signal is referred to the level of the reference signal (downloaded to the signal storage of the mobile terminal), averaged over the complete test sequence length.
4) The sensitivity is determined in dBV/V.

The measurement is repeated for the second channel.

### 7.2.4 Idle channel noise in multimedia playback mode

#### 7.2.4.1 Requirement

The idle channel noise is measured as the output level generated by the output of the mobile terminal player when playing back the pre-recorded music signal at the output of the headset interface and referring this to the idle channel noise produced when playing back a dithering noise signal.

The music to idle noise ratio MINR shall be greater than 40 dB.

Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

#### 7.2.4.2 Test

1) The test arrangement is according to clause 7.2.1, Figure 8.
2) A full-band music signal providing sufficient signal energy from 20 Hz - 20 kHz at the nominal signal level is used for playback. The test signal level is the A-weighted average level of the complete test signal. The output level is measured as an unweighted broadband signal level between 20 Hz and 20 kHz. This level is the reference speech signal level.
3) For the noise measurement, a dithering noise signal with a stochastically varying LSB is used for playback.

4) The idle channel noise is measured at the output in the frequency range between 20 Hz and 20 kHz. The length of the time window is 1 s which is the averaging time for the idle channel noise. The test lab has to ensure the correct activation of the DUT during the measurement.

   The power density spectrum of the noise signal is determined using FFT (8 k samples/48 kHz sampling rate or equivalent). A Hann window is used.

5) The idle channel noise is determined by A-weighting and referring to the reference signal level as determined with the music signal. Spectral peaks are measured in the frequency domain. The average noise spectrum used for determining the spectral peak should be calculated as the arithmetic mean of the noise spectrum values when stated in dBV.

The measurement is repeated for the second channel.

### 7.2.5 Distortion in multimedia playback mode

#### 7.2.5.1 Requirement

The distortion is measured as harmonic distortion generated by the output of the mobile terminal player when playing back the pre-recorded sinusoidal signal at the output of the headset interface.

The ratio of signal to harmonic distortion shall be above the following mask.

**Table 15 – Limits for the signal to harmonic distortion**

| Frequency (Hz) | Signal to harmonic distortion ratio limit, Send (dB) |
|---|---|
| 100 | 40 |
| 315 | 50 |
| 5 000 | 50 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 7.2.5.2 Test

1) The test arrangement is according to clause 7.2.1, Figure 8.
2) For the test, a sinusoidal signal at frequencies of 100 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz and 5 000 Hz is used. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be the nominal signal level.
3) The signal to harmonic distortion ratio is measured selectively up to 20 kHz.
4) The test is repeated using a signal level 20 dB higher than the nominal signal level.

The measurement is repeated for the second channel.

### 7.2.6 Receiving crosstalk

#### 7.2.6.1 Requirement

The receiving crosstalk is measured as the L-R crosstalk and the R-L crosstalk generated by the output of the mobile terminal player when playing back the pre-recorded pink noise at the output of the left output of the headset interface and measuring the resulting level at the right output (L-R crosstalk) of the headset interface and vice versa (R-L crosstalk).

The attenuation measured at the right output, referenced to the spectrum generated at the left output shall be above the following mask. The attenuation measured at the left output, referenced to the spectrum generated at the right output shall be above the following mask.

**Table 16 – Limits for the narrowband L-R and R-L crosstalk in Receive**

| Frequency (Hz) | R-L and L-R crosstalk limit, Receive (dB) |
|---|---|
| 50 | >30 |
| 16 000 | >30 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 7.2.6.2 Test

1) The test arrangement is according to clause 7.2.1, Figure 8.
2) The test signal used for the measurements shall be a pink noise from 20 Hz to 20 kHz. The test signal is the nominal signal level. The level is averaged over the complete test signal. The test signal is generated on the left channel only.
3) The crosstalk is determined in third octave intervals as given by [IEC 61260] for frequencies from 50 Hz to 16 kHz inclusive, by analysing the signal at the right channel of the headset interface output. In each third octave band, the level of the measured signal is referred to the left channel of the headset interface output (generated by the signal downloaded to the storage of the mobile terminal), averaged over the complete test sequence length.
4) The crosstalk is determined in dBV/V.

The measurement is repeated by reversing the channels (generating the output signal at the right channel output and measuring the crosstalk signal at the left channel output).

## 8 Headset specification

### 8.1 Communication mode

#### 8.1.1 Test set-up

The test set-up is shown in Figure 9.

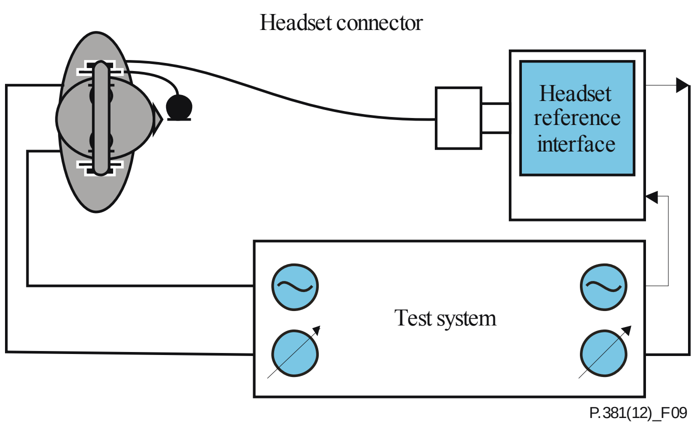



**Figure 9 – Test arrangement for testing the headset**

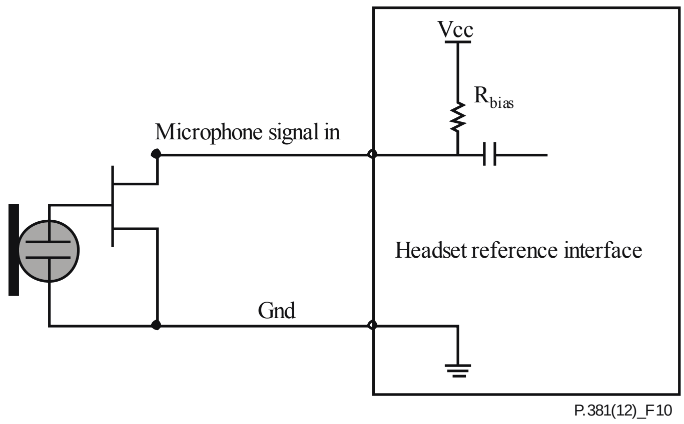



**Figure 10 – Input connection of a headset reference interface**

### 8.1.1.1 Input and output characteristics of the test system for connecting the headset

The output of the test system must fulfil the requirements. The output impedance shall be < 2 Ω. The maximum output voltage shall be 150 mV ± 1mV when loaded with a 32 Ω resistor. This is defined as 0 dBFS.

The bias voltage provided by the test system shall be 2.6 V ±1%. $R_{bias}$ is the bias resistance inside the input of the test system. The bias resistance shall be 2.2 kΩ ± 2%. The nominal sensitivity shall be –60 dBV (expected from a headset with a nominal sensitivity of –55 dBV/Pa).

### 8.1.1.2 Test signals and test signal levels

Speech-like signals are used for the measurements which can be found in [ITU-T P.50] and [ITU-T P.501]. Detailed information about the test signal used is to be found in the corresponding clause of this Recommendation.

All test signals – which are used in Receive – have to be band-limited. The band limitation is achieved by bandpass filtering in the frequency range between 200 Hz and 4 kHz using a bandpass filter providing higher than 24 dB/octave for narrowband digital mobile terminals. In wideband mode, the band limitation is achieved by bandpass filtering in the frequency range between 50 Hz and 8 kHz using a bandpass filter providing high than 24 dB/octave. In Send, the test signals are used without band limitation.

All test signal levels are referred to the average level of the test signals, averaged over the complete test sequence length, if not described otherwise. In Receive, the band-limited test signal is measured; in Send, no band-limitation is applied.

The nominal average signal levels for the measurements are as follows:

–26 dBFS in receive

–4.7 dBPa at the MRP.

Some tests require exact synchronization of test signals in the time domain. Therefore, it is required to take into account the delays of the headsets. When analysing signals, any delay introduced by the test system and the headset has to be taken into account accordingly.

### 8.1.1.3 Positioning of the headsets

Recommendations for the set-up and positioning of headsets are given in [ITU-T P.380]. If not stated otherwise, headsets shall be placed in their recommended wearing position. Some insert earphones might not fit properly in Type 3.3 ear simulators. For such insert type headsets, an [ITU-T P.57] Type 2 ear simulator may be used in conjunction with the HATS mouth simulator. The HATS should be equipped with two artificial ears as specified in [ITU-T P.57]. For binaural headsets two artificial ears are required.

If not stated otherwise the measurements in the receive are repeated five times and averaged. The averaged result is used.

#### 8.1.1.4 Position and calibration of HATS

The HATS shall be equipped with a Type 3.3 artificial ear. For the measurement of binaural headsets the HATS shall be equipped with two artificial ears. The pinnae are specified in [ITU-T P.57] for Type 3.3 artificial ears. The pinnae shall be positioned on HATS according to [ITU-T P.58].

The exact calibration and equalization procedures as well as how to combine the two ear signals for the purpose of measurements, can be found in [ITU-T P.581]. If not stated otherwise, the HATS shall be diffuse-field equalized. The reverse nominal diffuse field curve as found in Table 3 of [ITU-T P.58] shall be used. For measurements requiring diffuse-field correction values for closer frequency spacing than that which is specified in [ITU-T P.58], the interpolation method found in Annex A shall be used.

### 8.1.2 Sensitivity in Send

#### 8.1.2.1 Requirement

The sending sensitivity is measured from MRP to the sending input of the headset reference interface input.

The sending sensitivity shall be –55 dBV/Pa ±6 dB when inserting the sending signal at the nominal level as described in clause 8.1.1.2.

#### 8.1.2.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level, the level is averaged over the complete test signal.

   The measured power density spectrum at MRP is used as the reference power-density spectrum for determining the sending sensitivity.

3) For wideband, the sending sensitivity is calculated from 50 Hz to 8 kHz. For narrowband, the frequency range from 100 Hz to 4 kHz is used for calculation.

   For the calculation, the average measured level at the headset reference interface is used.

4) The sensitivity is expressed in dBV/Pa.

### 8.1.3 Sensitivity in Receive

#### 8.1.3.1 Requirement

The receiving sensitivity is measured from the receiving output of the headset reference interface to the DRP.

The receiving sensitivity shall be 26 dBPa/V with a tolerance of –6 dB when inserting the receiving signal at the nominal level as described in clause 8.1.1.2.

#### 8.1.3.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level, the level is averaged over the complete test signal.

   The measured power density spectrum at the headset reference interface is used as the reference power-density spectrum for determining the receiving sensitivity.

3) For wideband, the sending sensitivity is calculated from 50 Hz to 8 kHz. For narrowband, the frequency range from 100 Hz to 4 kHz is used for calculation.

For the calculation, the average measured level at the DRP is used.

4) The sensitivity is expressed in dBPa/V.

### 8.1.4 Sending sensitivity frequency response

#### 8.1.4.1 Requirement

The sending frequency response is measured from MRP to the sending input of the headset reference interface input.

The measured frequency response shall be within the limits defined in Table 17.

**Table 17 – Tolerance mask for the narrowband sending sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | –12 | |
| 200 | 0 | |
| 300 | 0 | –12 |
| 1 000 | 0 | –6 |
| 2 000 | 4 | –6 |
| 3 000 | 4 | –6 |
| 3 400 | 4 | –9 |
| 4 000 | 0 | |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

**Table 18 – Tolerance mask for the wideband sending sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 0 | |
| 200 | 5 | –5 |
| 5 000 | 5 | –5 |
| 6 300 | 5 | –10 |
| 8 000 | 5 | |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

#### 8.1.4.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal level is the nominal signal level, applied at the MRP.

The measured power density spectrum at the MRP is used as the reference power density spectrum for determining the sending sensitivity.

3) In wideband, the sending sensitivity is determined in third octave intervals, as given by [IEC 61260] for frequencies between 100 Hz and 8 kHz inclusive, measured at the POI. In narrowband, it is determined for frequencies from 200 Hz to 4 kHz. In each third octave band, the level of the measured signal is referred to the level of the reference signal averaged over the complete test sequence length.

4) The sensitivity is determined in dBV/Pa.

### 8.1.5 Receiving sensitivity frequency response

#### 8.1.5.1 Requirement

The receiving frequency response is measured from the receiving output of the headset reference interface to the DRP.

The measured frequency response shall be within the limits as defined in Table 19.

**Table 19 – Tolerance mask for the narrowband receiving sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 6 | |
| 300 | 6 | –6 |
| 3 400 | 6 | –6 |
| 4 000 | 6 | |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

**Table 20 – Tolerance mask for the wideband receiving sensitivity frequency response**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 6 | |
| 200 | 6 | –10 |
| 300 | 6 | –6 |
| 1 000 | 6 | –6 |
| 2 000 | 8 | –6 |
| 5 000 | 8 | –6 |
| 6 300 | 8 | –12 |
| 8 000 | 8 | |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

#### 8.1.5.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) The test signal used for the measurements shall be the British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501]. The test signal is the nominal signal level, applied to the headset reference interface. The level is averaged over the complete test signal.

3) For wideband, the receiving sensitivity is determined in third octave intervals as given by [IEC 61260] for frequencies between 100 Hz and 8 kHz inclusive, measured at the headset interface. In narrowband, it is determined for frequencies from 200 Hz to 4 kHz. In each third octave band, the level of the measured signal is referred to the level of the reference signal, averaged over the complete test sequence length.

4) The sensitivity is determined in dBPa/V.

The measurement is repeated for the second channel.

#### 8.1.6 Idle channel noise in Send

##### 8.1.6.1 Requirement

The idle channel noise in Send is measured from MRP to the sending input of the headset reference interface.

The signal to idle noise ratio (SINR) shall be > 30 dB.

Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

##### 8.1.6.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) The British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501] at the nominal signal level as described in clause 8.1.1.2 is applied at the MRP. The test signal level is the average level of the complete test signal. The output level is measured using a speech level voltmeter according to [ITU-T P.56]. This level is the reference speech signal level.

3) For the noise measurement, no test signal is used. However, all sources which potentially contribute to noise at the output such as GSM noise, electrical noise introduced and others must be considered. In order to ensure a reliable activation for active headsets, a conditioning sequence is inserted before the actual measurement for such headsets. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

4) The idle channel noise is measured at the output in the frequency range between 100 Hz and 8 kHz. The measurement requires a time window which starts exactly at the end of the activation signal. Any "ringing" of filters or receivers shall be taken into account, the time window must be shifted accordingly. The length of the time window is 1 s which is the averaging time for the idle channel noise. The test lab has to ensure the correct activation of the headset during the measurement. If the headset is deactivated during measurement, the measurement window has to be cut to the duration when the headset remains activated.

   The power density spectrum of the noise signal is determined using FFT (8 k samples/48 kHz sampling rate or equivalent). A Hann window is used.

   If it is known that the headset stays activated without any activation signal, no activation signal is required. In this case, a simple noise measurement is conducted.

5) The idle channel noise is determined by A-weighting and referring to the reference speech signal level as determined with the speech sequence. Spectral peaks are measured in the frequency domain. The average noise spectrum used for determining the spectral peak should be calculated as the arithmetic mean of the noise spectrum values when stated in dBV.

#### 8.1.7 Idle channel noise in Receive

##### 8.1.7.1 Requirement

The idle channel noise in Receive is measured from the receiving output of the headset reference interface to the DRP.

The signal to idle noise ratio (SINR) shall be higher than 30 dB.

Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

#### 8.1.7.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) The British-English single talk sequence described in clause 7.3.2 of [ITU-T P.501] at the nominal signal level as described in clause 8.1.1.2 is applied at the POI. The test signal level is the average level of the complete test signal. The output level is measured using a speech level voltmeter according to [ITU-T P.56]. This level is the reference speech signal level.

3) For the noise measurement, no test signal is used. However, all sources which potentially contribute to noise at the output, such as GSM noise, electrical noise introduced and others must be considered. In order to ensure a reliable activation in case of active headsets, a conditioning sequence is inserted before the actual measurement for such headsets. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

4) The idle channel noise is measured at the DRP in the frequency range between 100 Hz and 8 kHz. The measurement requires a time window which starts exactly at the end of the activation signal. Any "ringing" of filters or receivers shall be taken into account, the time window must be shifted accordingly. The length of the time window is 1 s which is the averaging time for the idle channel noise. The test lab has to ensure the correct activation of the headset during the measurement. If the headset is deactivated during measurement, the measurement window has to be cut to the duration when the headset remains activated.

   The power density spectrum of the noise signal is determined using FFT (8 k samples/48 kHz sampling rate or equivalent). A Hann window is used.

   If it is known that the headset stays activated without any activation signal, no activation signal is required. In this case, a simple noise measurement is conducted.

5) The idle channel noise is determined by A-weighting and referring to the reference speech signal level as determined with the speech sequence. Spectral peaks are measured in the frequency domain. The average noise spectrum used for determining the spectral peak should be calculated as the arithmetic mean of the noise spectrum values when stated in dBV.

The measurement is repeated for the second channel.

### 8.1.8 Distortion in Send

#### 8.1.8.1 Requirement

The distortion in Send is measured from MRP to the sending input of the headset reference interface.

The ratio of signal to harmonic distortion shall be above the following mask.

**Table 21 – Limits for the signal to harmonic distortion**

| Frequency (Hz) | Signal to harmonic distortion ratio limit, Send (dB) |
|---|---|
| 315 | 40 |
| 400 | 50 |
| 1 000 | 50 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 8.1.8.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) For the test, a sinusoidal signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is used. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be the nominal signal level. In order to ensure a reliable activation, a conditioning sequence is inserted before the actual measurement, in case of active headsets. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

3) The signal to harmonic distortion ratio is measured selectively up to 7 kHz.

4) The test is repeated using a signal level 10 dB higher than the nominal signal level. The level of the activation signal is kept at the nominal signal level.

### 8.1.9 Receiving distortion

#### 8.1.9.1 Requirement

The distortion in Receive is measured from the receiving output of the headset reference interface to the DRP.

The ratio of signal to harmonic distortion shall be above the following mask.

**Table 22 – Limits for signal to harmonic distortion**

| Frequency (Hz) | Signal to harmonic distortion ratio limit, Receive (dB) |
|---|---|
| 315 | 40 |
| 400 | 50 |
| 1 000 | 50 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 8.1.9.2 Test

1) The test arrangement is according to clause 8.1.1, Figure 9.

2) For the test, a sinusoidal signal at frequencies of 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz is used. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be the nominal signal level. In order to ensure a reliable activation, a conditioning sequence is inserted before the actual measurement, in case of active headsets. The conditioning sequence is according to [ITU-T P.501], clause 7.3.7. The short conditioning sequences (either male or female) should be used. The level of the activation signal is the nominal signal level.

3) The signal to harmonic distortion ratio is measured selectively up to 7 kHz.

4) The test is repeated using a signal level 10 dB higher than the nominal signal level. The level of the activation signal is kept at the nominal signal level.

NOTE – This measurement is not repeated 5 times.

## 8.2 Multimedia playback mode (provisional, for further study)

### 8.2.1 Test set-up

The test set-up is shown in Figure 11.

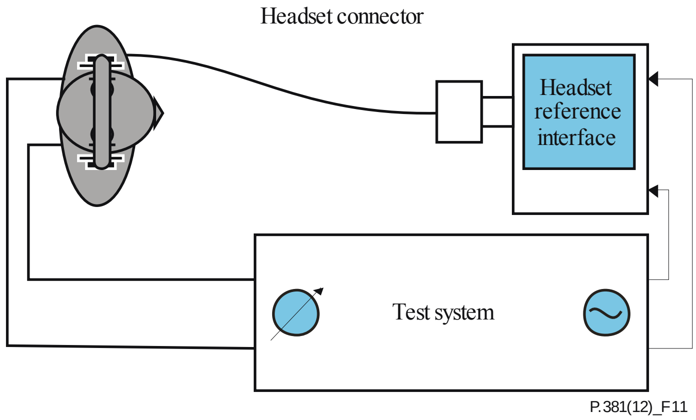



**Figure 11 – Test arrangement for testing the headset**

#### 8.2.1.1 Input and output characteristics of the test system for connecting the headset

The output impedance shall be < 2 Ω. The maximum output voltage shall be 150 mV ± 1mV when loaded with a 32 Ω resistor. This is defined as 0 dBFS.

#### 8.2.1.2 Test signals and test signal levels

Music signals or music simulating noise are used for the measurements. Detailed information about the test signal used is to be found in the corresponding clause of this Recommendation.

All test signals – which are used in Receive – have to be band-limited. The band limitation is achieved by bandpass filtering in the frequency range between 20 Hz and 20 kHz using a bandpass filter providing higher than 24 dB/octave.

All test signal levels are referred to the average level of the test signals, averaged over the complete test sequence length, if not described otherwise.

The nominal average signal level for the measurements is –26 dBFS.

Some tests require exact synchronization of test signals in the time domain. Therefore, it is required to take into account the delays of the terminals. When analysing signals, any delay introduced by the test system, codecs and terminals has to be taken into account accordingly.

#### 8.2.1.3 Positioning of the headsets

See clause 8.1.1.3 of this Recommendation.

#### 8.2.1.4 Position and calibration of HATS

See clause 8.1.1.4 of this Recommendation.

### 8.2.2 Sensitivity in multimedia playback mode

#### 8.2.2.1 Requirement

The receiving sensitivity is measured from the receiving output of the headset reference interface to the DRP.

The receiving sensitivity shall be 26 dBPa/V with a tolerance of –6 dB when inserting the sending signal at the nominal level as described in clause 8.2.1.2.

#### 8.2.2.2 Test

1) The test arrangement is according to clause 8.2.1, Figure 11.
2) The test signal used for the measurements shall be a full-band music signal providing sufficient signal energy from 20 Hz to 20 kHz. The test signal level is the nominal signal level.

3) For the calculation, the averaged level at the output of the headset interface is used. The sensitivity is determined from 20 Hz to 20 kHz.

4) The sensitivity is expressed in dBPa/V.

The measurement is repeated for the second channel.

### 8.2.3 Sensitivity frequency response in multimedia playback mode

#### 8.2.3.1 Requirement

The frequency response is measured from the receiving output of the headset reference interface to the DRP.

The measured frequency response shall be within the limits defined in Table 23.

**Table 23 – Tolerance mask for the wideband sensitivity frequency response in multimedia playback mode**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | 6 | |
| 200 | 6 | –10 |
| 300 | 6 | –6 |
| 1 000 | 6 | –6 |
| 2 000 | 8 | –6 |
| 5 000 | 8 | –6 |
| 8 000 | 8 | –12 |
| 10 000 | 8 | |
| NOTE – All sensitivity values are expressed in dB on an arbitrary scale. | | |

#### 8.2.3.2 Test

1) The test arrangement is according to clause 8.2.1, Figure 11.

2) The test signal used for the measurements shall be a full-band music signal providing sufficient signal energy from 20 Hz to 20 kHz. The test signal is the nominal signal level. The level is averaged over the complete test signal.

3) The sensitivity frequency response is determined in third octave intervals as given by [IEC 61260] for frequencies between 50 Hz and 10 kHz, inclusive. In each third octave band, the level of the measured signal is referred to the level of the reference signal (downloaded to the signal storage of the mobile terminal), averaged over the complete test sequence length.

4) The sensitivity is determined in dBPa/V.

The measurement is repeated for the second channel.

### 8.2.4 Idle channel noise in multimedia playback mode

#### 8.2.4.1 Requirement

The idle channel noise is measured from the receiving output of the headset reference interface to the DRP.

The music to idle noise ratio MINR shall be > 40 dB.

Spectral peaks in the frequency domain shall not exceed the averaged spectrum by more than 10 dB.

#### 8.2.4.2 Test

1) The test arrangement is according to clause 8.2.1, Figure 11.

2) A full-band music signal providing sufficient signal energy from 20 Hz-20 kHz at the nominal signal level is used for playback. The test signal level is the A-weighted average level of the complete test signal. The output level is measured as an unweighted broadband signal level between 20 Hz and 20 kHz. This level is the reference speech signal level.

3) For the noise measurement, no signal is applied to the headset.

4) The idle channel noise is measured at the output in the frequency range between 20 Hz and 20 kHz. The length of the time window is 1 s which is the averaging time for the idle channel noise. The test lab has to ensure the correct activation of the headset during the measurement.

   The power density spectrum of the noise signal is determined using FFT (8 k samples/48 kHz sampling rate or equivalent). A Hann window is used.

5) The idle channel noise is determined by A-weighting and referring to the reference speech signal level as determined with the music signal. Spectral peaks are measured in the frequency domain. The average noise spectrum used for determining the spectral peak should be calculated as the arithmetic mean of the noise spectrum values when stated in dBV.

The measurement is repeated for the second channel.

### 8.2.5 Distortion in multimedia playback mode

#### 8.2.5.1 Requirement

The distortion is measured from the receiving output of the headset reference interface to the DRP.

The ratio of signal to harmonic distortion shall be above the following mask.

**Table 24 – Limits for the signal to harmonic distortion**

| Frequency (Hz) | Signal to harmonic distortion ratio limit, Send (dB) |
|---|---|
| 100 | 40 |
| 315 | 50 |
| 5 000 | 50 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 8.2.5.2 Test

1) The test arrangement is according to clause 8.2.1, Figure 11.

2) For the test, a sinusoidal signal at frequencies of 100 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz and 5 000 Hz is used. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be the nominal signal level.

3) The signal to harmonic distortion ratio is measured selectively up to 20 kHz.

4) The test is repeated using a signal level 20 dB higher than the nominal signal level.

The measurement is repeated for the second channel.

The measurement is not repeated 5 times.

### 8.2.6 Receiving crosstalk

#### 8.2.6.1 Requirement

The receiving crosstalk is measured as the L-R crosstalk and the R-L crosstalk generated by a headset playing a pink noise at the output of the left output of the headset interface and measuring the resulting level at the right output (L-R crosstalk) of the headset interface and vice versa (R-L crosstalk).

The attenuation measured at the right ear, referenced to the spectrum generated at the left ear shall be above the following mask. The attenuation measured at the ear output, referenced to the spectrum generated at the right ear shall be above the following mask.

**Table 25 – Limits for the narrowband L-R and R-L crosstalk in receive**

| Frequency (Hz) | R-L and L-R crosstalk limit, Receive (dB) |
|---|---|
| 50 | >20 |
| 16 000 | >20 |
| NOTE – The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) – logarithmic (Hz) scale. | |

#### 8.2.6.2 Test

1) The test arrangement is according to clause 8.2.1, Figure 11.
2) The test signal used for the measurements shall be a pink noise from 20 Hz-20 kHz. The test signal is the nominal signal level. The level is averaged over the complete test signal. The test signal is generated on the left channel only.
3) The crosstalk is determined in third octave intervals as given by [IEC 61260] for frequencies from 50 Hz to 16 kHz inclusive, by analysing the signal at the right ear. In each third octave band, the level of the measured signal is referred to the left ear, averaged over the complete test sequence length.
4) The crosstalk is determined in dBPa/Pa.

The measurement is repeated by reversing the channels (generating the output signal at the right channel output and measuring the crosstalk signal at the left channel output).

The measurement is not repeated 5 times.

## 9 Function requirements for terminals with the universal headset interface

The terminal shall provide the intelligent detection mechanism.

1) The terminal should be capable of detecting any plug-in action automatically and then activate the corresponding function according to the current state.
2) The terminal should be capable of detecting any plug-out action automatically and then activate the corresponding function according to the current state.
3) The terminal should be capable of determining whether the inserted plug has three poles or four poles.
4) Some headsets have a Send/End button for submitting the Send/End signal to the terminal. Figure 12 is an example illustrating how the signal is submitted through the MIC input pin. The resistance to GND is less than 2 Ω.

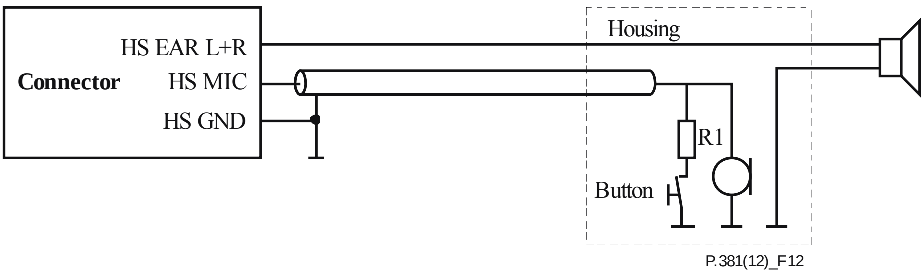



**Figure 12 – Button-control headset function**

NOTE – For harmonization with some already-in-market headsets and the better user experience, it has been suggested that the terminal with a universal interface be able to detect headsets of different pole orders automatically and adjust the speech or audio stream accordingly.

# Annex A

## Interpolation method for diffuse-field correction

(This annex forms an integral part of this Recommendation.)

Interpolated values for 1/12-octave bands shall be calculated from 1/3-octave band values using Table A.1.

For measurements requiring diffuse-field correction values for closer frequency spacing than 1/12-octave bands, linear interpolation on a log scale from the 1/12-octave band interpolated values in Table A.1 shall be used.

**Table A.1 – Interpolation parameters on 1/12-octave bands**

| Frequency (Hz) | Interpolated value (dB) | Frequency (Hz) | Interpolated value (dB) |
|---|---|---|---|
| 95 | 0.000 | 1 000 | 5.000 |
| 100 | 0.000 | 1 060 | 5.375 |
| 106 | 0.000 | 1 120 | 5.750 |
| 112 | 0.000 | 1 180 | 6.125 |
| 118 | 0.000 | 1 250 | 6.500 |
| 125 | 0.000 | 1 320 | 6.800 |
| 132 | 0.000 | 1 400 | 7.150 |
| 140 | 0.000 | 1 500 | 7.550 |
| 150 | 0.000 | 1 600 | 8.000 |
| 160 | 0.000 | 1 700 | 8.550 |
| 170 | 0.000 | 1 800 | 9.175 |
| 180 | 0.000 | 1 900 | 9.850 |
| 190 | 0.000 | 2 000 | 10.500 |
| 200 | 0.000 | 2 120 | 11.500 |
| 212 | 0.125 | 2 240 | 12.550 |
| 224 | 0.250 | 2 360 | 13.500 |
| 236 | 0.390 | 2 500 | 14.050 |
| 250 | 0.500 | 2 650 | 13.850 |
| 265 | 0.525 | 2 800 | 13.250 |
| 280 | 0.500 | 3 000 | 12.400 |
| 300 | 0.480 | 3 150 | 12.000 |
| 315 | 0.500 | 3 350 | 11.750 |
| 335 | 0.600 | 3 550 | 11.650 |
| 355 | 0.725 | 3 750 | 11.600 |
| 375 | 0.875 | 4 000 | 11.500 |
| 400 | 1.000 | 4 250 | 11.425 |
| 425 | 1.135 | 4 500 | 11.375 |
| 450 | 1.275 | 4 750 | 11.275 |
| 475 | 1.375 | 5 000 | 11.000 |

**Table A.1 – Interpolation parameters on 1/12-octave bands**

| Frequency (Hz) | Interpolated value (dB) | Frequency (Hz) | Interpolated value (dB) |
|---|---|---|---|
| 500 | 1.500 | 5 300 | 10.400 |
| 530 | 1.625 | 5 600 | 9.550 |
| 560 | 1.650 | 6 000 | 8.600 |
| 600 | 1.800 | 6 300 | 8.000 |
| 630 | 2.000 | 6 700 | 7.375 |
| 670 | 2.450 | 7 100 | 6.800 |
| 710 | 3.000 | 7 500 | 6.450 |
| 750 | 3.500 | 8 000 | 6.500 |
| 800 | 4.000 | 8 500 | 7.150 |
| 850 | 4.325 | 9 000 | 8.250 |
| 900 | 4.550 | 9 500 | 9.450 |
| 950 | 4.750 | 10 000 | 10.450 |

# Appendix I

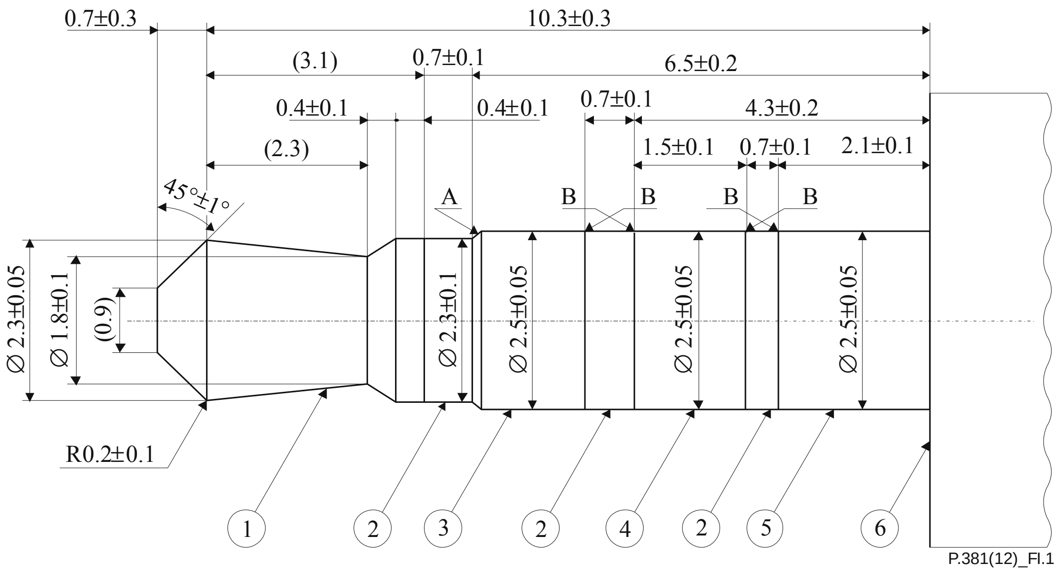
## Audio connectivity for sockets with four contact points

(This appendix does not form an integral part of this Recommendation.)

This appendix illustrates the dimensions of the concentric plug and socket connector with four contact points.

### I.1 2.5 mm diameter plug connector with four poles

Figure I.1 shows the shape and dimensions of the 2.5 mm diameter plug connector with four poles. The width of strip A along the axial direction is 0.1 mm. Junction B should be free of burr or fash.




NOTE 1 – ① is the Tip and made of conductive material; ② is the insulating ring; ③ is Ring 1 and made of conductive material; ④ is Ring 2 and made of conductive material; ⑤ is the sleeve and made of conductive material; ⑥ is an illustration of the hand grip at the end of a plug.

NOTE 2 – "Fash" here refers to a rough edge or ridge on the surface.

**Figure I.1 – Shape and dimensions of the 2.5 mm diameter plug connector with four poles**

### I.2 2.5 mm diameter socket connector with four contact points

The socket should be able to mate and cooperate with the plug reliably. The dimensions and positioning for each contact spring are illustrated in Figure I.2. Considering the tolerance of the plug dimension and positioning of the socket contact spring, in addition to the shift of the practical contact point location caused by the width of the spring, the minimum distance between the contact point of the Ring 2 spring and that of the sleeve spring is recommended to be longer than 1.6 mm. If bushing of the socket is made of conductive material, the contact area of the sleeve spring may exceed the given range indicated in Figure I.2, so bushing of the socket should not be longer than 2.0 mm.

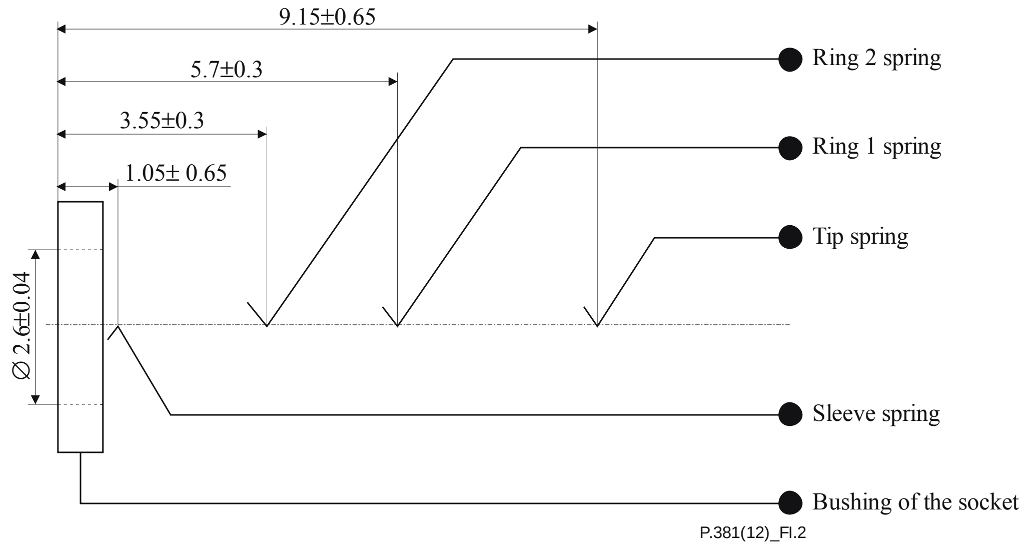
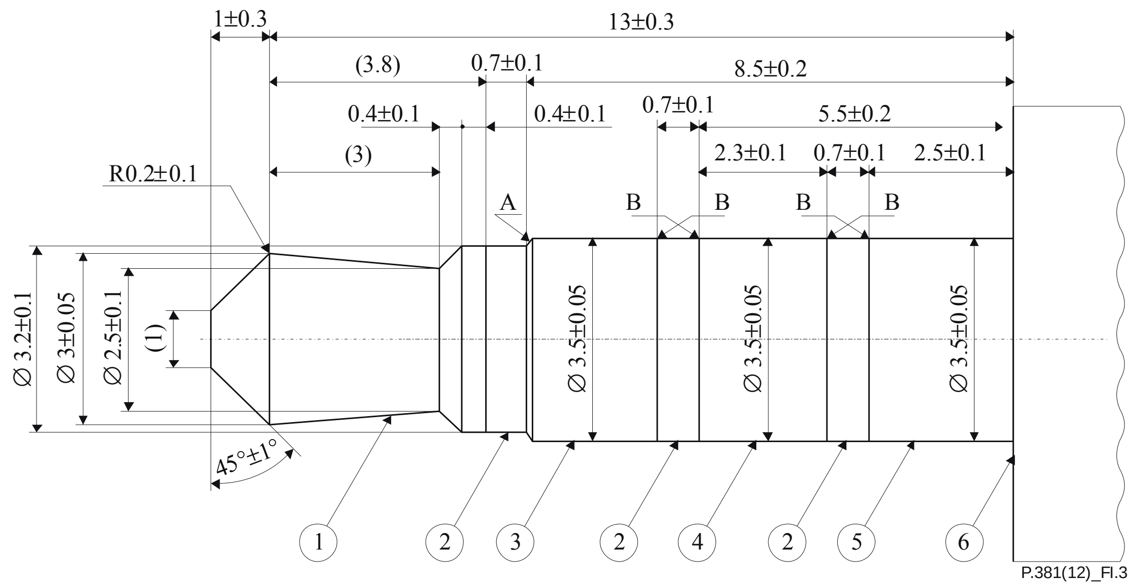



**Figure I.2 – Dimensions of the 2.5 mm diameter socket with four contact points and positioning of each contact spring**

## I.3 3.5 mm diameter plug connector with four poles

Figure I.3 shows the shape and dimensions of the 3.5 mm diameter plug connector with four poles. The width of strip A along the axial direction is 0.15 mm. Junction B should be free of burr or fash.




NOTE 1 – ① is the Tip and made of conductive material;② is the insulating ring; ③ is Ring 1 and made of conductive material; ④ is Ring 2 and made of conductive material; ⑤ is the sleeve and made of conductive material; ⑥ is an illustration of the hand grip at the end of a plug.

NOTE 2 – "Fash" here refers to a rough edge or ridge on the surface.

**Figure I.3 – Shape and dimensions of the 3.5 mm diameter plug connector with four poles**

## I.4 3.5 mm diameter socket connector with four contact points

The socket should be able to mate and cooperate with the plug reliably. The dimensions and the positioning for each contact spring are illustrated in Figure I.4. Considering the tolerance of the plug dimension and positioning of the socket contact spring, in addition to the shift of the practical contact point location caused by the width of the spring, the minimum distance between the contact point of the Ring 2 spring and that of the sleeve spring is recommended to be more than 2.4 mm. If bushing of the socket is made of conductive material, the contact area of the sleeve spring may exceed the given range indicated in Figure I.4, so bushing of the socket should not be longer than 2.4 mm.

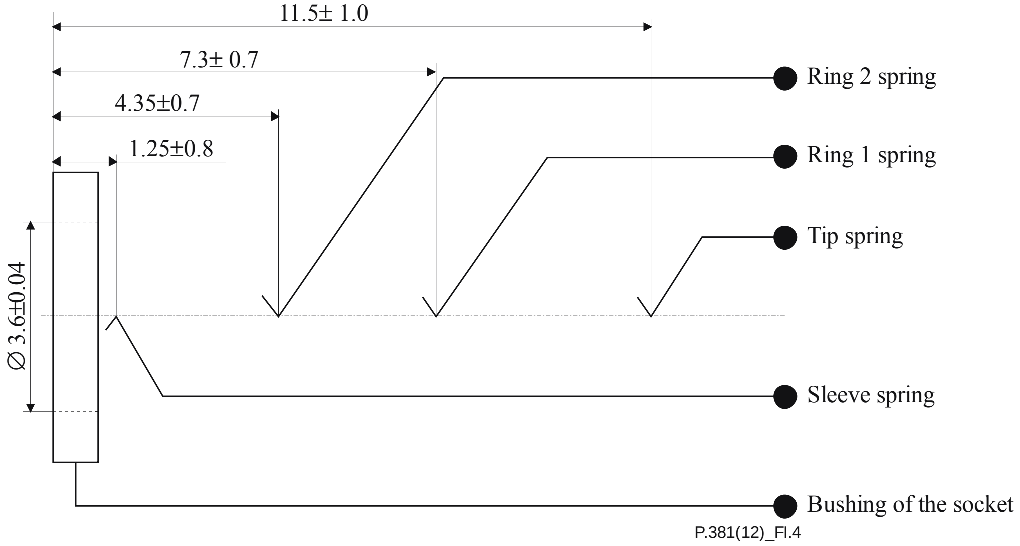



**Figure I.4 – Dimensions of the 3.5 mm diameter socket with four contact points and positioning of each contact spring**

# Appendix II

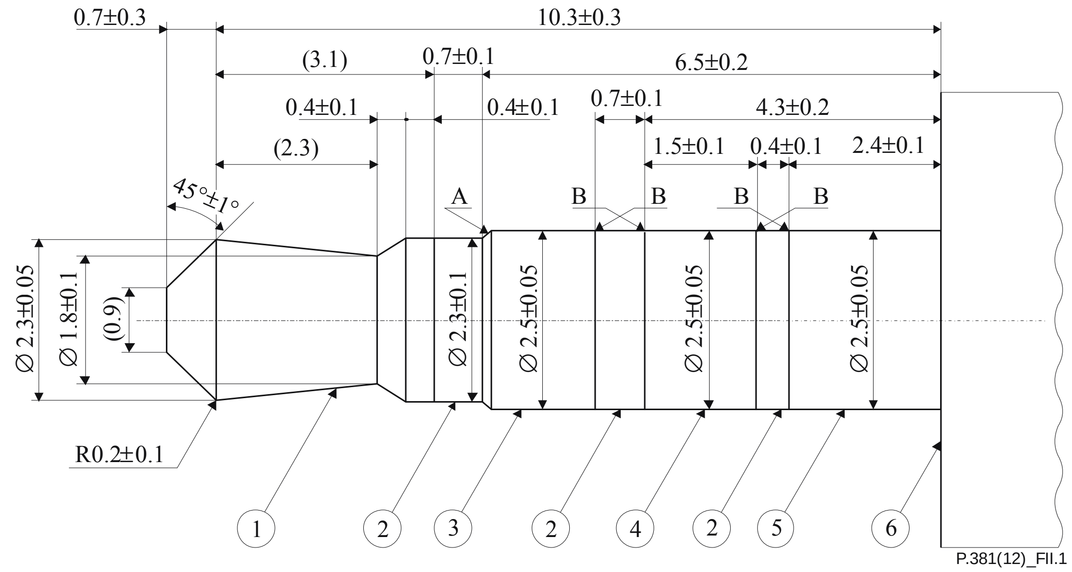
## Audio connectivity for sockets with four contact points (optional dimensions to accommodate terminal designs with curved edges)

(This appendix does not form an integral part of this Recommendation.)

This appendix illustrates the optional dimensions of the concentric plug and socket connector with four contact points.

### II.1 2.5 mm diameter plug connector with four poles

Figure II.1 shows the shape and dimensions of the 2.5 mm diameter plug connector with four poles. The width of strip A along the axial direction is 0.1 mm. Junction B should be free of burr or fash.



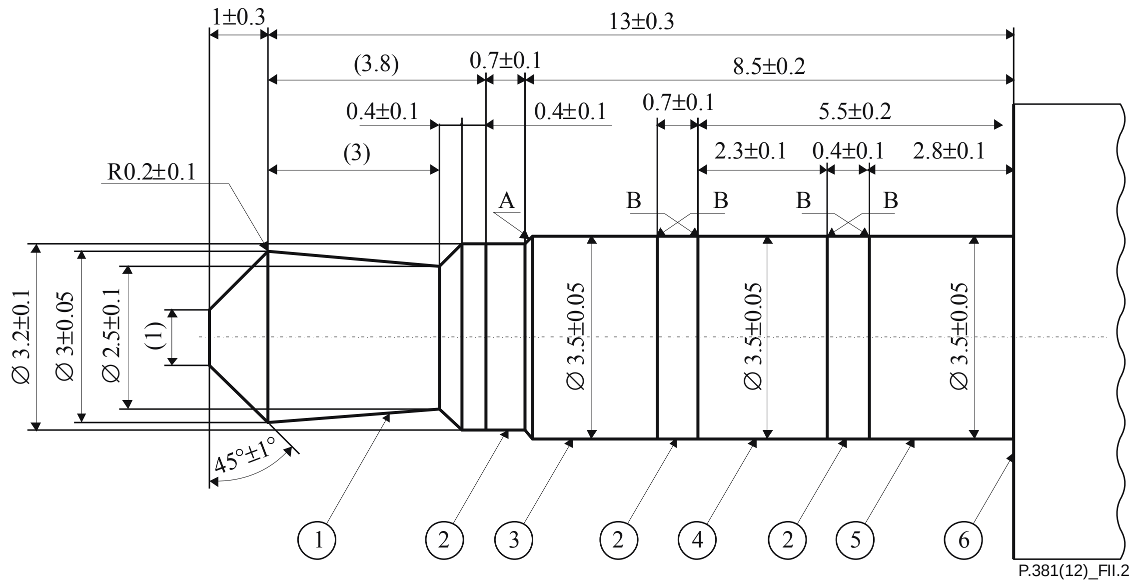

NOTE 1 – ① is the Tip and made of conductive material; ② is the insulating ring; ③ is Ring 1 and made of conductive material; ④ is Ring 2 and made of conductive material; ⑤ is the sleeve and made of conductive material; ⑥ is illustration of the hand grip at the end of a plug.

NOTE 2 – "Fash" here refers a rough edge or ridge on the surface.

**Figure II.1 – Shape and dimensions of the 2.5 mm diameter plug connector with four poles**

### II.2 3.5 mm diameter plug connector with four poles

Figure II.2 shows the shape and dimensions of the 3.5 mm diameter plug connector with four poles. The width of strip A along the axial direction is 0.15 mm. Junction B should be free of burr or fash.




NOTE 1 – ① is Tip and made of conductive material;② is the insulating ring; ③ is Ring 1 and made of conductive material; ④ is Ring 2 and made of conductive material; ⑤ is the sleeve and made of conductive material; ⑥ is an illustration of the hand grip at the end of a plug.

NOTE 2 – "Fash" here refers to a rough edge or ridge on the surface.

**Figure II.2 – Shape and dimensions of the 3.5 mm diameter plug connector with four poles**

# Appendix III

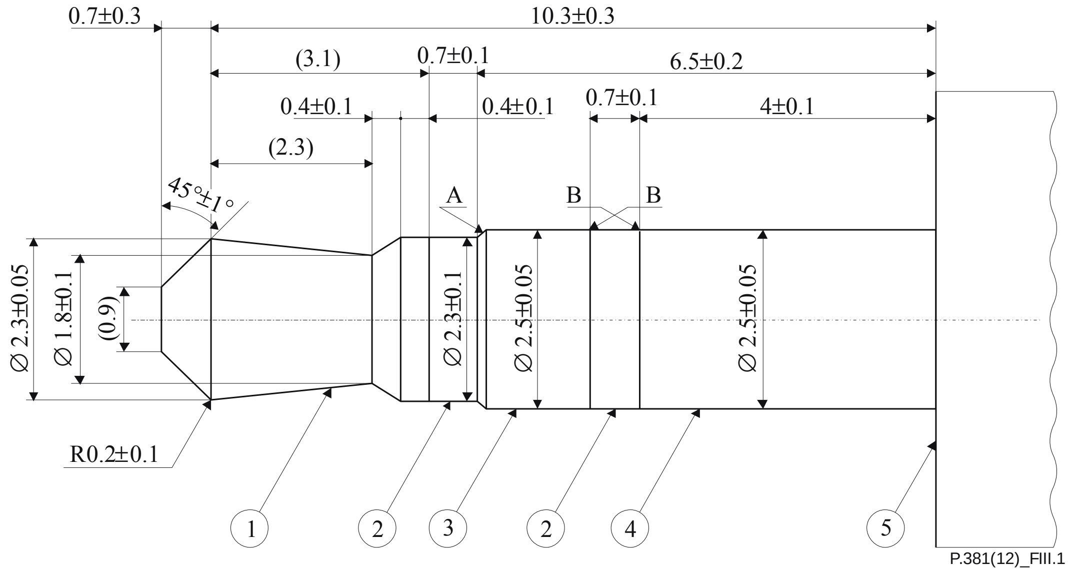
## Audio connectivity for sockets with three contact points

(This appendix does not form an integral part of this Recommendation.)

This appendix illustrates the dimensions of the concentric plug and socket connector with three contact points.

### III.1 2.5 mm diameter plug connector with three poles

Figure III.1 shows the shape and dimensions of the 2.5 mm diameter plug connector with three poles. The width of strip A along the axial direction is 0.1 mm. Junction B should be free of burr or fash.




NOTE 1 – ① is the Tip and made of conductive material; ② is the insulating ring; ③ is the middle ring and made of conductive material; ④ is the sleeve and made of conductive material; ⑤ is an illustration of the hand grip at the end of a plug.

NOTE 2 – "Fash" here refers to a rough edge or ridge on the surface.

**Figure III.1 – Shape and dimensions of the 2.5 mm diameter plug connector with three poles**

### III.2 2.5 mm diameter socket connector with three contact points

The socket should be able to mate and cooperate with the plug reliably. The dimensions and positioning for each contact spring are illustrated in Figure III.2. To achieve better compatibility with various headphones or headsets, especially for those used in digital mobile terminals, sockets with three contact points should be in line with this requirement. If bushing of the socket is made of conductive material, the contact area of the sleeve spring may exceed the given range indicated in Figure III.2, so that bushing of the socket should not be longer than 3.9 mm.

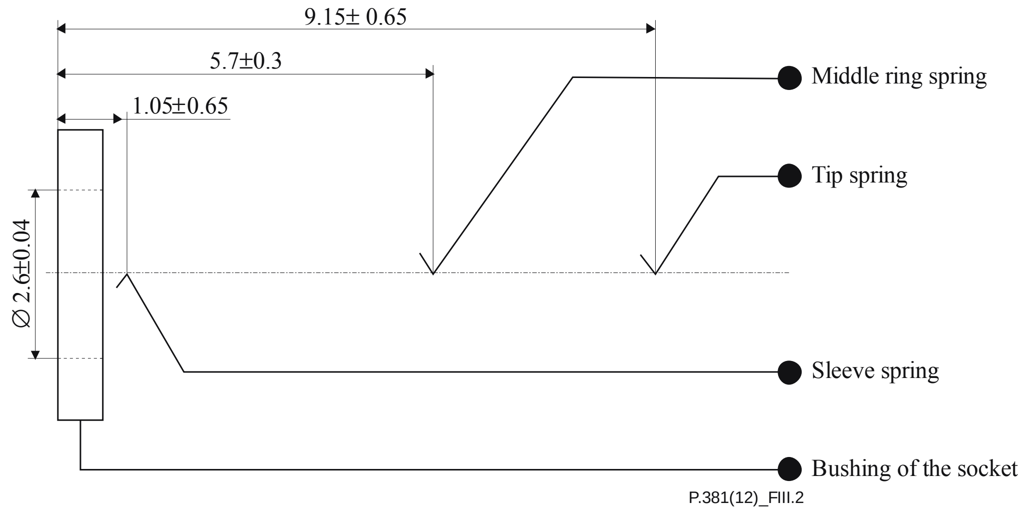
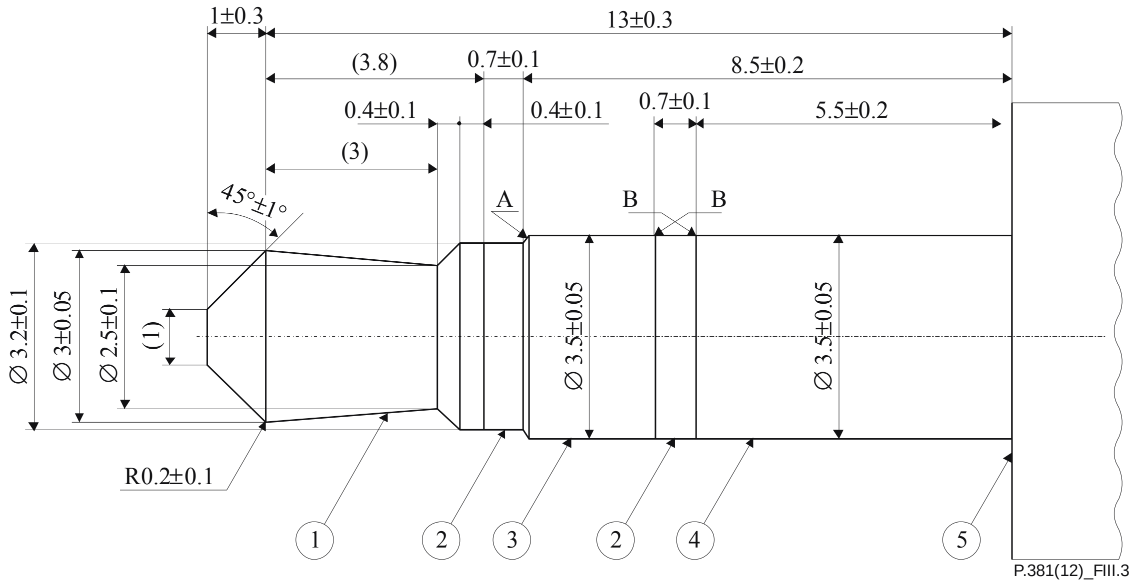



**Figure III.2 – Dimensions of the 2.5 mm diameter socket with three contact points and positioning of each contact spring**

## III.3 3.5 mm diameter plug connector with three poles

Figure III.3 shows the shape and dimensions of the 3.5 mm diameter plug connector with three poles. The width of strip A along the axial direction is 0.15 mm. Junction B should be free of burr or fash.




NOTE – ① is the Tip and made of conductive material; ② is the insulating ring; ③ is the middle ring and made of conductive material; ④ is the sleeve and made of conductive material; ⑤ is an illustration of the hand grip at the end of a plug.

**Figure III.3 – Shape and dimensions of the 3.5 mm diameter plug connector with three poles**

### III.4 3.5 mm diameter socket connector with three contact points

The socket should be able to mate and cooperate with the plug reliably. The dimensions and positioning for each contact spring are illustrated in Figure III.4. To achieve better compatibility with various headphones or headsets, the 3.5 mm diameter sockets with three contact points should be in line with this requirement. If bushing of the socket is made of conductive material, the contact area of the sleeve spring may exceed the given range indicated in Figure III.4, so bushing of the socket should not be longer than 5.3 mm.

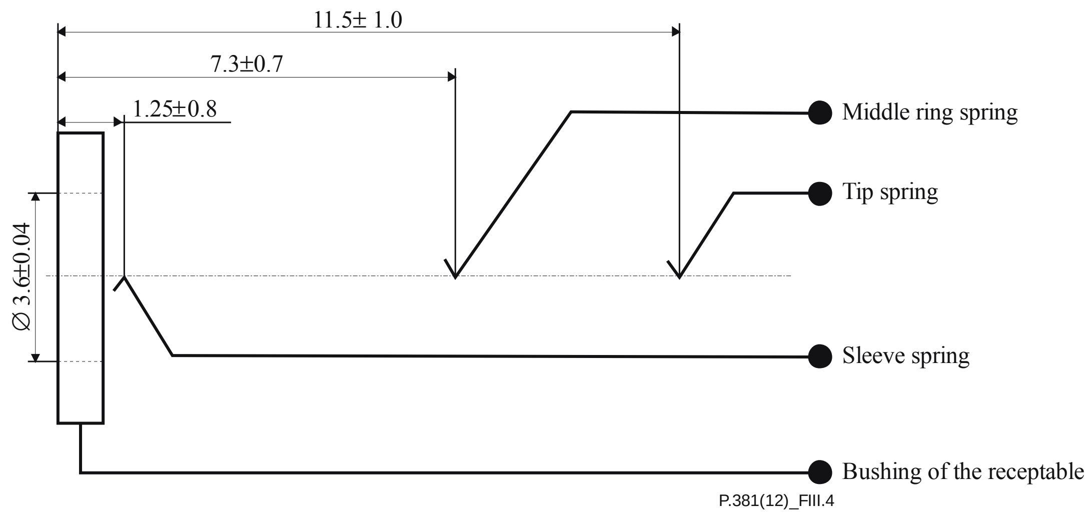



**Figure III.4 – Dimensions of the 3.5 mm diameter socket with three contact points and positioning of each contact spring**

# Appendix IV

# Other considerations

(This appendix does not form an integral part of this Recommendation.)

## IV.1 Filter recommendation

It is recommended to filter the connection to the microphone in order that the headset shall not generate any appreciable non-linear distortion as a result of interfering RF signals. This is covered by audio breakthrough testing (ABT) specifications.

## IV.2 ESD

It is recommended to protect the microphone using an ESD diode. A low value can result in poor EMI performance (this can result in ABT failures) and a high value will result in voltage spikes as seen by the microphone itself.

## IV.3 Microphone basics – background

The standard electret microphones are not voltage sources, but current sources when biased correctly. They contain an internal junction field effect transistor (JFET) that does not amplify the signal much, but changes impedance levels. When the bias voltage is low, below 0.8 V for example, the JFET will instead work as a resistor. When they operate as a current source, the output impedance is around 30 kΩ and the current is 220-250 *u*A.

Since they operate as a current source, a larger bias resistor will result in a bigger signal swing which is the desired behaviour. However, at a certain point, the voltage drop across the resistor means that the JFET inside the microphone goes out of saturation and the benefits of a large resistor are lost. The optimal resistor value depends on the bias point. Typically, it is in the range 3-6 kΩ. However, in some cases a lower value is desired, e.g., if you want to detect multiple passive button values.

Traditionally, 2.2 kΩ has been used in the past 30 years by electret manufacturers. The value comes from a max current of 500 *u*A from the old JFET design. As JFET technology has improved, manufacturers have increased the DC resistance value. Although conservative, some typical values seen have been 2.6 kΩ.

## IV.4 Vcc Voltage for microphone bias

In principle, the same goes for the biasing voltage, but due to other constraints, there may be a reason to use a range (e.g., due to EMC/ESD reasons). A range of 1.5-3.6V can be used, but the range 2.7-3.0 V is preferred, for compatibility considerations with all headsets (see note below).

A high bias voltage allows a larger bias resistor and pushes JFET further into saturation and thereby gives better S/N.

NOTE – The preferred range of 2.7 – 3.0 V for microphone bias is not in conflict with the 2.6 V specified in clause 8.1.1.1 for the headset testing system. The headset testing system voltage of 2.6 V was selected for backward compatibility with existing mobile phone sockets. The performance of headsets measured using a 2.6 V microphone bias will also be representative of performance when using a slightly higher microphone bias in the range of 2.7-3.0 V. However, the reverse may not be true, because 2.6 V is near the edge of what is needed for some microphone headsets to operate properly. Therefore, testing with a slightly lower microphone bias voltage may catch interoperability issues with existing sockets not designed to the preferred 2.7-3.0 V range (while still being representative of performance for new sockets that are designed as per the preferred voltage range). Mobile phone sockets using this preferred operating range will be backwards compatible with a wide range of existing headsets.

## IV.5 DC resistance of microphone

Microphone current should be in the range of 120-300 *u*A (i.e., the DC resistance should be such that the resulting microphone current is in this range).

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| | |
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