

# Testspecification Audio for CAT-iq 2.0 Devices

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## Introduction

The audio specification for CAT-iq 2.0 is based on the ETSI EN 300 175-8 and EN 300 176-2.

This document describes the requirements for a DECT device to achieve the CAT-iq 2.0 and CAT-iq 2.1 label.

The reason for not only referencing to the ETSI EN 300 175-8 and EN 300 176-2 was:

- some requirements could be defined more exactly which is important for the approval process.
- only products in the focus of CAT-iq 2.0 and CAT-iq 2.1 are described;
- one single document with the requirements and the measurement description is available;
- some minor errors could be corrected without starting a change request process.

## 1.1 Types of Devices

For CAT-iq the following groups of requirements are defined:

PP Group A: Handset and headset in narrowband mode

PP Group B: Handset and headset in wideband mode

PP Group C: Handsfree in narrowband mode

PP Group D: Handsfree in wideband mode

FP Group A: VoIP narrowband mode

FP Group B: VoIP wideband mode

The correspondence between the audio types defined in ETSI EN 300 175-8 [8] and the audio groups is listed in the following table:

Applicable to:	Type nr.	Type name	CAT-iq 2.0 Audio Group:
PP	0	Reference PP (RePP) narrowband	-
	1a	"Classic" GAP handset narrowband	-
	1b	"Improved" GAP handset narrowband	-
	1c	HATS tested, "standard" narrowband handset	PP Group A
	1d	HATS tested, "improved" narrowband handset	PP Group A
	3a	HATS tested, "standard" narrowband loudspeaking and handsfree feature	PP Group C
	3b	HATS tested, "improved" narrowband loudspeaking and handsfree feature	PP Group C
	2a	ITU-T P.311 tested, wideband handset or headset	-
	2b	HATS tested, "standard" wideband handset or headset	PP Group B
	2c	HATS tested, "improved" wideband handset or headset	PP Group B
	4a	HATS tested, "standard" wideband loudspeaking and handsfree feature	PP Group D
	4b	HATS tested, "improved" wideband loudspeaking and handsfree feature	PP Group D
	5a	Superwideband 14 kHz handset or headset	-
	5b	Superwideband 14 kHz handsfree	-
FP	6	PPs with external 2 wire, 3,1 kHz telephony interface	-
	0	Reference FP (ReFP)	-
	1a	"classic" Fixed Part for ISDN network	-
	1b	"new" Fixed Part for ISDN Network	-
	2	FP with analog 2-wire interface, 3,1 kHz service	-
	3	VoIP narrowband Fixed Part	FP Group A
	4	ISDN wideband Fixed Part	-
	5	VoIP wideband Fixed Part	FP Group B
	6a	FP handling an Internal call inside a DECT FP (any service)	-
	6b	FP handling an n-party conference inside a DECT FP (any service)	-
REP	7	DECT Repeater part (REP)	-

Note: PP Group A to D relies mostly on "standard" Types and includes some quality aspects of "improved" ETSI Types.

**Table 1: Types of devices**

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## 1.2 Mandatory Requirements

A separate table (not included in this document) defines for each requirement a number of points which can be archived if this requirement is fulfilled. The minimum number of points necessary to archive the overall certification is defined by the DECT-Forum and can be adapted over the time. Additionally some measurements are defined in this document as mandatory.

The following table gives an overview about the applicable categories for different product types:

	PP-A	PP-B	PP-C	PP-D	FP-A	FP-B
CAT-iq 2.0 PP	x	x	x	x	n.a.	n.a.
CAT-iq 2.0 Headset	x	x	n.a.	n.a.	n.a.	n.a.
CAT-iq 2.0 FP	n.a.	n.a.	n.a.	n.a.	n.a.	n.a.
CAT-iq 2.1 PP	x	x	x	x	n.a.	n.a.
CAT-iq 2.1 Headset	x	x	n.a.	n.a.	n.a.	n.a.
CAT-iq 2.1 FP	n.a.	n.a.	n.a.	n.a.	x	x

In CAT-iq 2.0 there are no mandatory requirements for the FP.

If any kind of headset could be applied to the PP (e.g. corded headset or Bluetooth headset) there are no requirements for this interface or headset.

Note: The requirements for CAT-iq 2.1 are still under discussion and might be changed in future versions.

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## 2 Setup

### 2.1 General

If a PP or a FP provides an additional adaptive volume control beside of the manual volume control, this feature shall be switched off for all measurements. This requirement is not valid for any kind of adaptive gain control, which can not be influenced by the user.

If a supplier does not meet the requirements due to an optional additional audio processing (such as WB speech extension or noise reduction) then the supplier shall be able to pass the test when switching off the optional processing.

All acoustic measurements shall be performed with the HATS (Head and Torso Simulator). The HATS shall conform to ITU-T Recommendation P.58 [32].

The supplier shall give a declaration about the nominal volume settings for each mode. This volume level is used for all measurements in nominal volume. The nominal volume setting for receive and send direction shall be the same.

### 2.2 Position and calibration of HATS

#### 2.2.1 Handsets

All the sending and receive characteristics shall be tested with the HATS. For handsets if not stated otherwise 8 N application force shall be used.

The horizontal positioning of the HATS reference plane shall be guaranteed within  $\pm 2^\circ$ .

The HATS shall be equipped with a type 3.3 artificial ear for handsets. For binaural headsets two artificial ears are required. The type 3.3 artificial ear as specified in ITU-T Recommendation P.57 [31] shall be used. The artificial ear shall be positioned on HATS according to ITU-T Recommendation P.58 [32].

The exact calibration and equalization can be found in ITU-T Recommendation P.581 [40].

For send measurements, unless specified otherwise, the test signal level shall be -4.7 dBPa at the MRP.

For receive measurements, unless specified otherwise, the applied test signal level at the digital input shall be -16 dBm0.

##### 2.2.1.1 Send

Unless specified otherwise, the test signal level shall be -4.7 dBPa at the MRP.

The following procedure shall be used to perform the calibration of the artificial mouth of the HATS:

The input signal from the artificial mouth is first calibrated under free-field conditions at the MRP. The total level on the frequency range is set to -4.7 dBPa.

The spectrum at MRP is recorded. Then the level is adjusted to the level given further in this text (depending of type of terminal tested).

	At MRP	30cm from MRP
Total Level:	-4.7 dBPa	-24,3 dBPa
HFRP Calibration:	0 dB	19,6 dB

The level at MRP (measured in third octave bands) adjusted at the first step (with total level of -4.7 dBPa) is used as the reference for send characteristics.

The test setup shall be in conformance with figure 1 but, depending on the type of terminal, the appropriate distance and level will be used. When using this calibration method, send sensitivity must be calculated as follows:

$$SmJ = 20 \log Vs - 20 \log PMRP$$

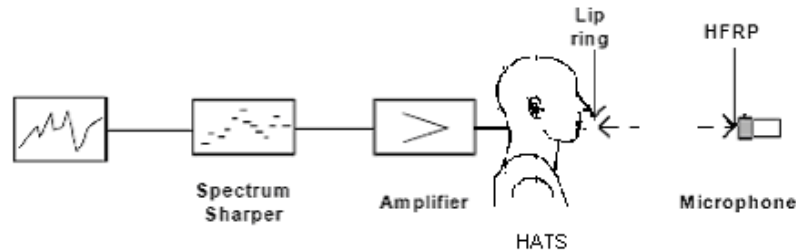
where:

Vs is the measured voltage across the appropriate termination (unless stated otherwise, a 600  $\Omega$  termination).

PMRP is the applied sound pressure at the MRP during the first step of calibration.

NOTE: Reason for this procedure of calibration in two steps is to take into account the different variation of signal with distance by using different implementations of HATS.





**Figure 1: Calibration at HFRP, the signal level for a handheld terminal in 30 cm distance is 19, 6 dB attenuated compared to the signal level at the mouth**

The distance used for level calibration corresponds to the following value:

Handheld terminal: 30 cm with -19.6dB.

### 2.2.1.2 Receive

Unless specified otherwise, the applied test signal level at the digital input shall be -16 dBm0.

All measurement values produced by HATS for handset mode are intended to be diffuse field equalized except of the loudness ratings (RLR, STMR).

### 2.2.2 Positioning of handset or headset

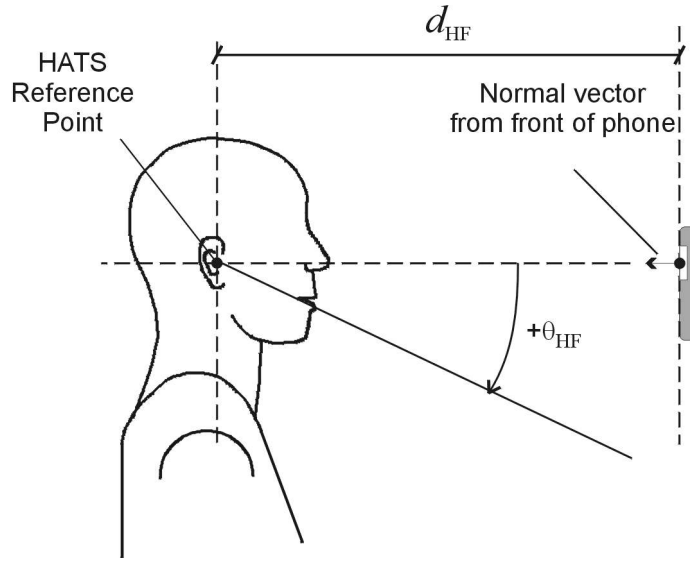
The handset is positioned on the HATS as described in ITU-T Recommendation P.64 [33]. The artificial mouth shall comply with ITU-T Recommendation P.58 [32]. The artificial ear shall comply with ITU-T Recommendation P.57 [31], type 3.3 shall be used.

Recommendations for positioning headsets are given in ITU-T Recommendation P.380 [37]. If not stated otherwise headsets shall be placed in their recommended wearing position. Further information about setup and the use of HATS can be found in ITU-T Recommendation P.380 [37].

Unless stated otherwise if a volume control is provided the setting is chosen such that the nominal RLR is met as close as possible.

### 2.2.3 Positioning of handheld hands-free devices

All the sending and receive characteristics shall be tested with the HATS acc to P.581 (artificial head with torso). The PP should be placed in according to figure 2. The HATS should be positioned so that the HATS Reference Point is at a distance  $d_{HF}$  from the centre point of the visual display of the Mobile Station. A vertical angle  $\theta_{HF}$  may be specified by the manufacturer. If no vertical angle is specified, a value of  $\theta_{HF} = 0$  is used for all handsfree measurements



**Figure 2: Configuration of Hand-Held loudspeaker relative to the HATS side view**  
**Configuration of Hand-Held loudspeaker relative to the HATS side view**

The HATS reference point should be located at a distance  $d_{HF}$  from the centre of the visual display of the Mobile Station. The distance  $d_{HF}$  is specified by the manufacturer,  $d_{HFR} = d_{HF}$ ,  $d_{HFS} = d_{HF} - d_{EM}$ , where  $d_{HFR}$  is the distance for receive measurement,  $d_{HFS}$  is the distance for send measurement, and  $d_{EM}$  is the distance from ERP to MRP. When no operating distance is specified by manufacturer, value for  $d_{HFS}$  will be 30 cm. A calculation of  $d_{EM}$  for HATS gives 12 cm.

A value of 42 cm will be taken for  $d_{HF}$ .

The measurements shall be made with a  $d_{HF} = 42$  cm.

All measurement values produced by HATS for handsfree mode are intended to be free field equalized

## 2.3 Measurement Setup

The measurement shall be done separately for the FP and the PP to achieve interoperability for the signal levels on the air interface.

### 2.3.1 Measurement Setup for PP

The DUT-PP shall be paired with a reference fixed part (Ref FP). The Ref-FP for narrowband shall only introduce a transparent transcoding from G.726 (Signal on air interface) to uniform PCM in the digital domain. In wideband mode the transcoding shall be according to G.722. There shall be no other signal processing blocks between transcoding and the PCM interface (e.g. volume adaption, error concealment or equalizing). The Ref FP should support CAT-iq 1.0 as a minimum.

The acoustic interface to the measurement system shall be the HATS.

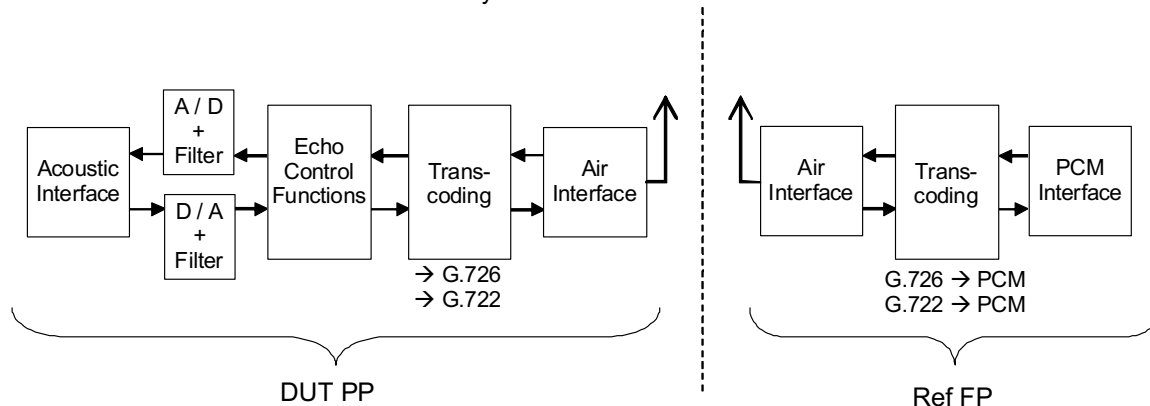


Figure 3: Measurement Setup for PP

### 2.3.2 Measurement Setup for FP

The DUT-FP shall be paired with a reference portable part (RefPP). The measurement system is connected to the VoIP interface and to the PCM interface of the RefPP. In narrowband mode the used codec on the VoIP interface shall be G.711, in wideband mode the G.722.

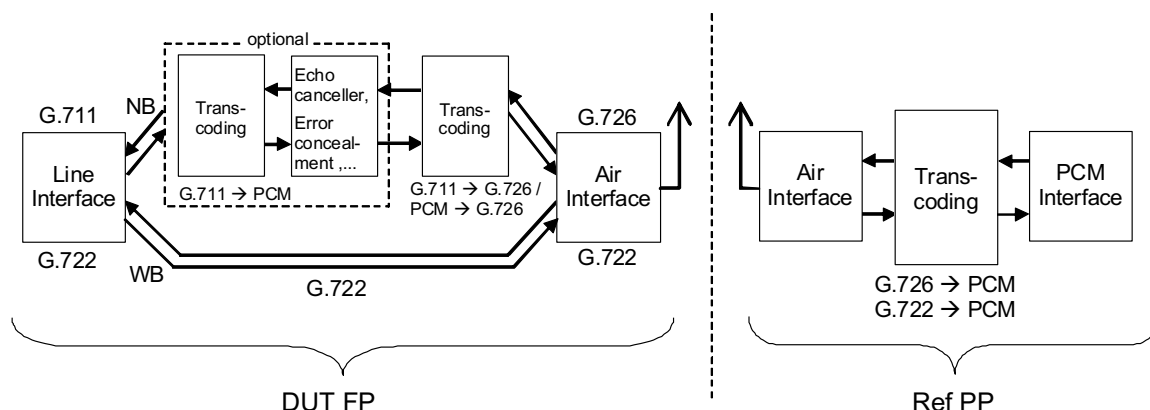


Figure 4: Measurement Setup for FP

### 2.3.3 Supplier Declaration for Test Laboratory

The supplier shall give the necessary information for the measurement to the test laboratory. This declaration should contain the following information:

- 
- Position of PP at the artificial ear ( $X_e$ ,  $Z_e$ ,  $Y_e$ ) for handset measurement
  - Position of PP for handsfree measurement
  - Nominal volume setting for all modes

If the supplier gives no information about the nominal volumes the test laboratory shall select the nominal volume setting.

## 2.4 PP group A: Handset and headset in narrowband mode

Used codec for measurement: G.726 at air interface.

### 2.4.1 Measurement Overview

Chapter	Name	Position	Appl. Force	Correction	Vol.	Nbr. of Measurements
2.4.3	PP send frequency response	HATS	8N	n.r.	nom	1
2.4.4	PP receive frequency response	HATS	8N	diffuse field	nom	1
2.4.5	PP Nominal Send Loudness Rating	HATS	8N	n.r.	nom	1
2.4.6	PP Receive Loudness Rating	HATS	8N	DRP-ERP (P.57)	nom, max	2
2.4.7	Talker sidetone	HATS	8N	DRP-ERP (P.57)	nom	1
2.4.8	Sidetone delay	HATS	8N	diffuse field	nom	1
2.4.9	TCLw	HATS	8N	n.r.	nom, max	2
2.4.10	Stability loss	Plate	n.r.	n.r.	max	1
2.4.11	Send Distortion – Average line level	HATS	8N	n.r.	nom	1
2.4.12	Send Distortion – High line level	HATS	8N	n.r.	nom	1
2.4.13	Receive Distortion - Average line level	HATS	8N	diffuse field	max	1
2.4.14	Receive Distortion - High line level	HATS	8N	diffuse field	max	1
2.4.15	Out-of-Band Signals in Send direction	HATS	8N	n.r.	nom	1
2.4.16	Out-of-band signals in receive direction	HATS	8N	diffuse field	nom	1
2.4.17	Noise in send direction	HATS	8N	n.r.	nom	1
2.4.18	Noise in receive direction	HATS	8N	diffuse field	nom, max	2
2.4.19	Acoustic Shock	HATS	13N	diffuse field	max	1
2.4.20	Delay	HATS	8N	diffuse field	nom	2
2.4.21.1	Attenuation Range in Send Direction during Double Talk $A_{H,S,dt}$	HATS	8N	n.r.	nom	1
2.4.21.2	Attenuation Range in Receive Direction during Double Talk $A_{H,R,dt}$	HATS	8N	diffuse field	nom	1
2.4.21.3	Detection of Echo Components during Double Talk	HATS	8N	diffuse field	nom	1
2.4.22.1	Activation in Send Direction	HATS	8N	n.r.	nom	1
2.4.22.2	Activation in Receive Direction	HATS	8N	diffuse field	nom	1
2.4.23.1	Temporal echo effects	HATS	8N	diffuse field	nom	1
2.4.23.2	Spectral Echo Attenuation	HATS	8N	diffuse field	nom	1
					Total:	28

Note: Some measurements counts depending on the numbers of volume levels, assumption here is 3 levels.

**Table 2: Measurement overview for PP group A**

## 2.4.2 General specification

The volume level which is closest to the RLR requirement shall be used for all measurements in nominal volume setting. The nominal volume level shall be declared by the supplier. This volume setting shall also be the nominal setting for send direction.

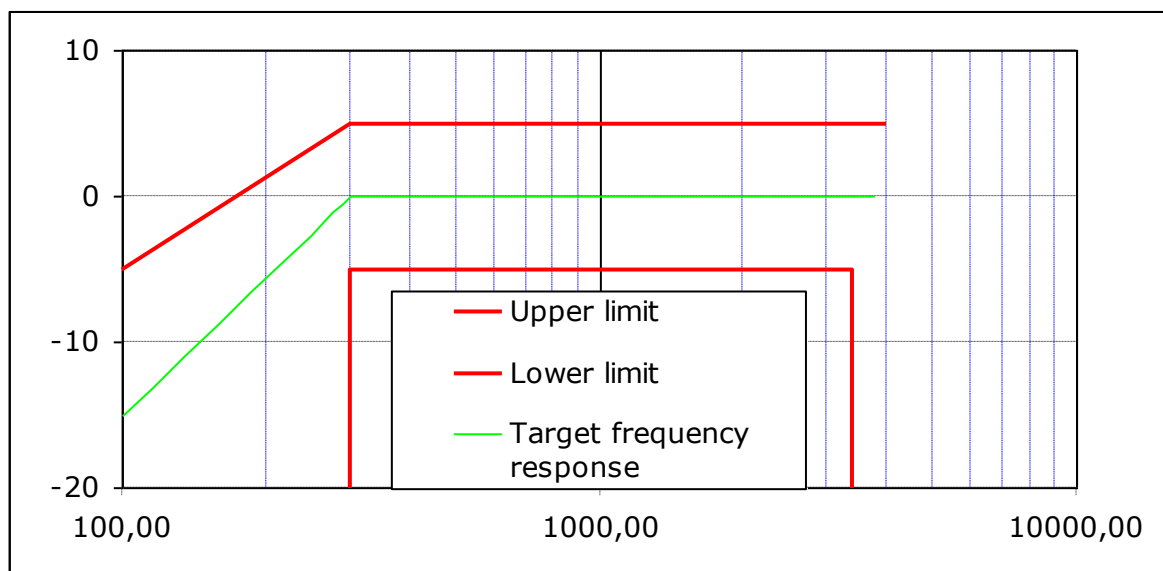
## 2.4.3 PP send frequency response

Requirement: The send frequency response of the handset shall be within a mask as defined in table 3. If the position of the microphone is not fixed (e.g. DECT headset), the supplier of the device shall inform about the position for intended use. This position shall be used for the measurement.

Frequency	Upper Limit [dB]	Lower Limit [dB]
100 Hz	-5	-
300 Hz	5	-5
3400 Hz	5	-5
3758 Hz	5	-
4000 Hz	5	-

Note: The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale is a floating or "best fit" mask.

**Table 3: Send frequency response**



**Figure 5: Send frequency response**

Measurement method: The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, duration 20 s (10 s female, 10 s male voice), measured at the MRP. The test signal level is averaged over the complete test signal sequence. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application force used to apply the handset against the artificial ear shall be 8N

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [37].

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

## 2.4.4 PP receive frequency response

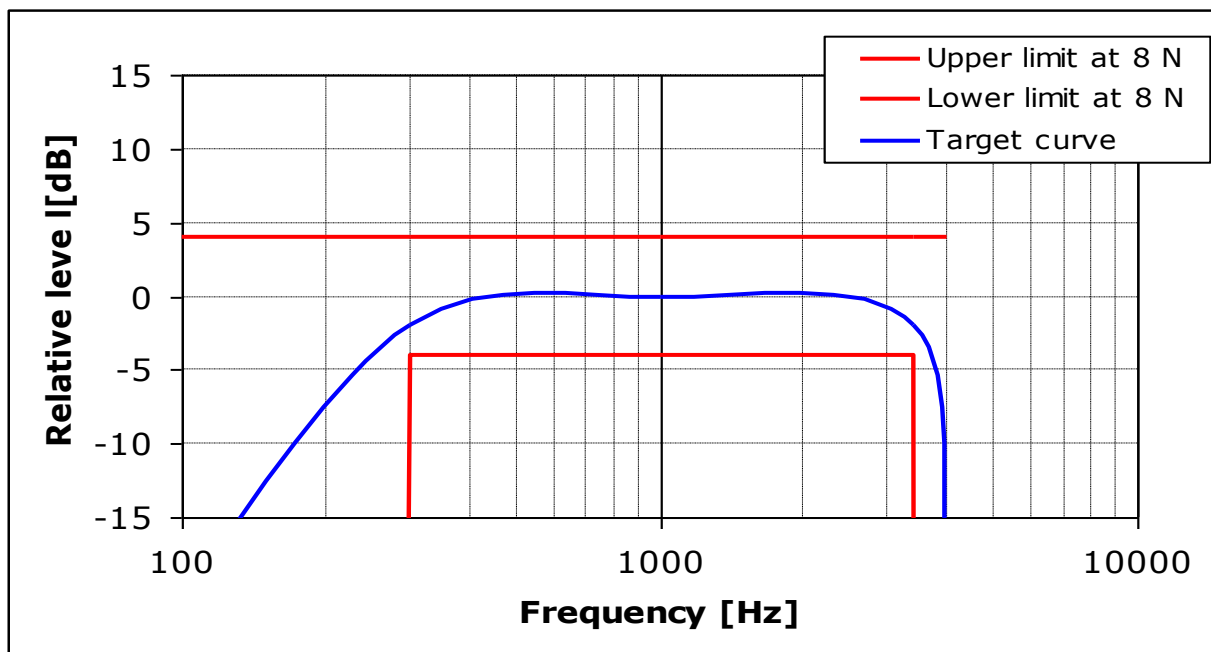
The receive frequency response for handsets is only measured for an application force of 8N, there is no differentiation between standard or improved class.

The tolerance mask is described in table 4:

Frequency [Hz]	Upper Limit	Lower Limit
100	4	
300	4	-4
1500	4	-4
3000	4	-4
3400	4	-4
4000	4	

Note: The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale is a floating or "best fit" mask.

**Table 4: Receive frequency response**



**Figure 6: Receive frequency response**

Measurement method:

Receive frequency response is the ratio of the measured sound pressure and the input level.

(dB relative Pa/V)

$$S_{\text{Jeff}} = 20 \log (p_{\text{eff}} / v_{\text{RCV}}) \text{ dB rel } 1 \text{ Pa} / \text{V}$$

$S_{\text{Jeff}}$ : Receive Sensitivity; Junction to HATS Ear with diffuse field correction.

$p_{\text{eff}}$ : DRP Sound pressure measured by ear simulator Measurement data are converted from the Drum Reference Point to diffuse field.

$v_{\text{RCV}}$ : Equivalent RMS input voltage.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28], duration 20 s (10 s female, 10 s male voice). The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [30] at the digital reference point or the equivalent analogue point.

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The handset terminal or the headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application forces used to apply the handset against the artificial ear is 8N.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [37]. In case of binaural headsets each receiver is measured separately.

The HATS is diffuse field equalized as described in ITU-T Recommendation P.581 [40]. The equalized output signal is power-averaged on the total time of analysis. The 1/12 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

## 2.4.5 PP Nominal Send Loudness Rating

The nominal value of Send Loudness Rating (SLR) shall be:

$SLR(set) = 8 \text{ dB} \pm 3.5 \text{ dB}$ .

If the device offers a user-controlled volume control, there shall be at least one volume setting which fulfills the requirement for nominal SLR.

Measurement method:

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28], duration 20 s (10 s female, 10 s male voice). The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application force used to apply the handset against the artificial ear is 8N.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [37].

The send sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [34], bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [34], formula 5-1, over bands 4 to 17, using  $m = 0.175$  and the send weighting factors from ITU-T Recommendation P.79 [34], table 1.

## 2.4.6 PP Receive Loudness Rating

The nominal value of Receive Loudness Rating (RLR) shall be:

$RLR(set) = 2 \text{ dB} \pm 3.5 \text{ dB}$ .

$RLR(\text{binaural headset}) = 8 \text{ dB} \pm 3.5 \text{ dB}$  for each earphone.

If the device offers a user-controlled volume control, there shall be at least one volume setting which fulfills the requirement for nominal RLR.

The minimum difference between nominal RLR and minimum (loudest) RLR shall be 6 dB.

Measurement method:

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28], duration 20 s (10 s female, 10 s male voice). The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application force used to apply the handset against the artificial ear is 8N. The HATS is **NOT** diffuse-field equalized as described in ITU-T Recommendation P.581 [40].

The DRP-ERP correction as defined in ITU-T Recommendation P.57 [31] is applied.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [37]. In case of binaural headsets each receiver is measured separately.

The receive sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T



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Recommendation P.79 [34], bands 4 to 17. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [34], formula 5-1, over bands 4 to 17, using  $m = 0.175$  and the receive weighting factors from table 1 of ITU-T Recommendation P.79 [34]. No leakage correction shall be applied for the measurement.

## 2.4.7 Talker sidetone

The STMR shall be  $18 \text{ dB} \pm 5 \text{ dB}$  for nominal RLR setting of the volume control.

Measurement method:

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be  $-4.7 \text{ dBPa}$ , measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]) and the application force shall be  $8 \text{ N}$  on the artificial ear type 3.3 The HATS is **NOT** diffuse-field equalized as described in ITU-T Recommendation P.581 [40].

The DRP-ERP correction as defined in ITU-T Recommendation P.57 [31] is applied.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from  $100 \text{ Hz}$  to  $8 \text{ kHz}$  inclusive. For the calculation the averaged measured level at each frequency band (ITU-T Recommendation P.79 [34], table 4, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

The Sidetone path loss (LmeST), as expressed in dB, and the Side Tone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of ITU-T Recommendation P.79 [34], using  $m = 0.225$  and the weighting factors of in table 3 of ITU-T Recommendation P.79 [34].

## 2.4.8 Sidetone delay

The maximum side-tone-round-trip delay shall be  $\leq 5 \text{ ms}$

Measurement method

The handset or the headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]).

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a pn-sequence with a length of 65535 points (for the  $48 \text{ kHz}$  sampling rate) which equals to the period  $T$ . The duration of the complete test signal is as specified in ITU-T Recommendation P.501 [38]. The level of the signal shall be  $-4.7 \text{ dBPa}$  at the MRP.

The cross-correlation function  $\Phi_{xy}(\tau)$  between the input signal  $S_x(t)$  generated by the test system in send direction and the output signal  $S_y(t)$  measured at the artificial ear is calculated in the time domain:

$$\Phi_{xy}(\tau) = \lim_{T \rightarrow \infty} \sum_{t=-T/2}^{T/2} S_x(t) S_y(t + \tau)$$

The measurement window  $T$  shall be exactly identical with 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 direct sound produced by the artificial mouth, 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  $H\{\Phi_{xy}(\tau)\}$  of the cross-correlation:

$$H\{\Phi_{xy}(\tau)\} = \sum_{-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)}$$
$$E(\tau) = \sqrt{[\Phi_{xy}(\tau)]^2 + \{H[\Phi_{xy}(\tau)]\}^2}$$

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

Note: The calculation formulas for the sidetone delay are currently in evaluation and might be changed in later versions.

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## 2.4.9 TCLw

The TCLw shall be  $\geq 55$  dB in nominal volume.  
With the volume control set to maximum TCLw shall be  $\geq 46$  dB.

Measurement method:

The PP or the headset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]) and the application force shall be 8N on the artificial ear type 3.3 as specified in ITU-T Recommendation P.57 [31]. The ambient noise level shall be less than -64 dBPa(A) for handset and headset terminals. The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4096 points (for the 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. The low crest factor is achieved by random alternation of the phase between -180 ° and 180 °.

The TCLw is calculated according to ITU-T Recommendation G.122 [14], clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1.0 s). For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

## 2.4.10 Stability loss

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 200 Hz to 4 kHz in maximum volume setting. In case of headsets the requirement applies for the closest possible position between microphone and headset receiver.

NOTE: Depending on the type of headset it may be necessary to repeat the measurement in different positions.

Measurement method:

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from digital input to digital output shall be measured for frequencies from 200 Hz to 4 kHz under the following conditions:

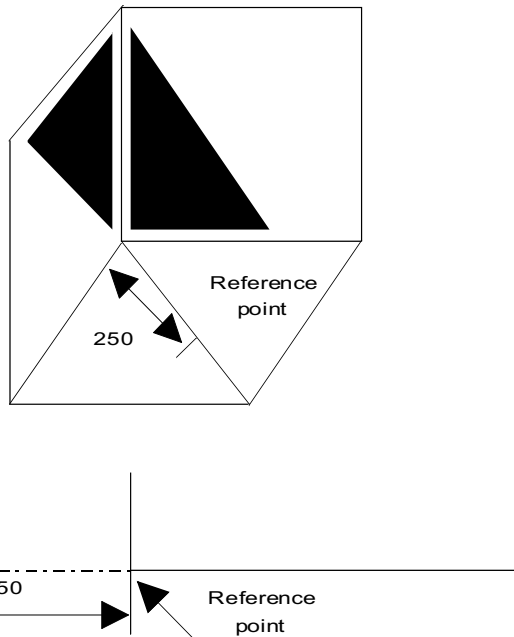
The handset or the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure 15.

The handset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

- the mouthpiece and ear cup shall face towards the surface;
- the handset shall be placed centrally, the diagonal line with the ear cup nearer to the apex of the corner;
- the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 15.

The headset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

- the microphone and the receiver shall face towards the surface;
- the headset receiver shall be placed centrally at the reference point as shown in figure 15;
- the headset microphone is positioned as close as possible to the receiver.



**Figure 7: Stability loss position**

### 2.4.11 Send Distortion - Average level

The psophometric weighted ratio of signal to total distortion (THD+N) shall be above the following mask (table 5):

Frequency [Hz]	Ratio [dB(p)]
315	26
400	30
500	30
800	30
1000	30
Note: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

**Table 5: Send distortion mask for average level**

#### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to -4.7 dBPa at the MRP.

The signal to distortion ratio is measured up to 3.15 kHz.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP.

### 2.4.12 Send Distortion - High level

The psophometric weighted ratio of signal to total distortion (THD+N) shall be above the following mask (table 6):

Frequency [Hz]	Ratio [dB(p)]
315	26
400	30
500	30
800	30
1000	30
Note: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

**Table 6: Send distortion mask for high level**

**Measurement method**

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to 1.3 dBPa at the MRP.

The signal to total distortion ratio is measured up to 3.15 kHz.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation. Level of this activation signal will be

-4.7 dBPa at the MRP.

## 2.4.13 Receive Distortion - Average line level

The A-weighted ratio of signal to total distortion (THD+N) in the max. volume setting shall be above the following mask (Table 7):

Frequency [Hz]	Ratio[dB(A)]
315	26
400	30
500	30
800	30
1000	30
Note: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

**Table 7: Receive distortion mask for average line level**

**Measurement method**

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz, with a level of -16 dBm0.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation.

The ratio of signal to total distortion shall be measured up to 10 kHz at the DRP of the artificial ear with the diffuse field equalization active.

## 2.4.14 Receive Distortion - High line level

The A-weighted ratio of signal to total distortion (THD+N) in the max. volume setting shall be above the following mask (Table 8):

Frequency [Hz]	Ratio [dB(A)]
315	26
400	30
500	30
800	30
1000	30
Note: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

**Table 8: Receive distortion mask for high line level**

**Measurement method**

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz, with a level of -10 dBm0.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation.

The ratio of signal to total distortion shall be measured up to 10 kHz at the DRP of the artificial ear with the diffuse field equalization active.

## 2.4.15 Out-of-Band Signals in Send direction

With any signal above 4.6 kHz and up to 8 kHz applied at the MRP at a level of -4.7 dBPa, the level of any image frequency shall be below the level obtained for the reference signal by at least the amount (in dB) specified in table 9.

Frequency [Hz]	Signal limit [dB]
4600	30
8000	40
Note: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

**Table 9: Out of band signal limit, send**

Measurement method:

For a correct activation of the system, an artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP.

For the test, an out-of-band signal shall be provided as a frequency band signal centered on 4.65 kHz, 5 kHz, 6 kHz, 6.5 kHz, 7 kHz and 7.5 kHz respectively. The level of any image frequencies at the digital interface shall be measured. The levels of these signals shall be -4.7 dBPa at the MRP.

The complete test signal is constituted by t1 ms of in-band signal (reference signal), t2 ms of out-of-band signal and another time t1 ms of in-band signal (reference signal).

The observation of the output signal on the first and second in-band signals permits control if the set is correctly activated during the out-of-band measurement. This measurement shall be performed during t2 period.

A value of 250 ms is suggested for t1.

T2 depends on the integration time of the analyzer, typically less than 150 ms.

## 2.4.16 Out-of-band signals in receive direction

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3 400 Hz at a level of -10 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4.6 kHz to 8 kHz measured selectively at DRP of the artificial ear with the diffuse field equalization active shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 10.

Image Signal Frequency [Hz]	Equivalent input level [dB]
4600	-35
8000	-45
Note: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

**Table 10: Discrimination levels - receive**

Measurement method:

The signal used is an activation signal followed by a sine-wave signal. For input signals at the frequencies 500 Hz, 1 000 Hz, 2000 Hz and 3 150 Hz applied at the level of -10 dBm0, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively at measurement point.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. The level of this activation signal is -10 dBm0. The out of band signal shall be measured at DRP of the artificial ear with the diffuse field equalization active.

## 2.4.17 Noise in send direction

The maximum noise level produced by the PP under silent conditions in the send direction shall not exceed -64 dBm0p.

The narrow-band noise (due to TDMA) produced by the apparatus in the sending direction, and contained within any 10 Hz bandwidth between the frequency limits 300 Hz to 3 400 Hz, shall not exceed -73 dBm0.

Measurement method:

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals

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(CSS) as described in ITU-T Recommendation P.501 [38]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The activation signal level shall be -4.7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [33]). The send noise is measured at the POI in the frequency range from 100 Hz to 4 kHz. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 second. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0p.

## **2.4.18 Noise in receive direction**

Telephone sets with adjustable receive levels shall be adjusted so that the RLR is as close as possible to the nominal RLR.

The receive noise shall be less than -57 dBPa(A).

The measured noise shall not be greater than -54 dBPa(A) at the maximum setting of the volume control.

Measurement method:

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [38]. The activation signal level shall be -16 dBm0. The activation signal level is averaged over the complete activation signal sequence.

Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation.

The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [33]).

The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 second. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The A-weighted noise level shall be measured at DRP of the artificial ear with the diffuse field equalization active.

## **2.4.19 Acoustic Shock**

### **2.4.19.1 Continuous signal**

Requirement

With a digitally encoded signal representing the maximum possible signal level at the digital interface, the sound pressure level at the ERP shall not exceed 22 dBPa (rms unweighted).

With an application force of 13N, the sound pressure level shall not exceed 22 dBPa.

Measurement Method:

See EN 300 176-2 [10].

Note: With some HATS it is not possible to make the measurements with an application force of 18N as described in ETSI EG 202 518.

## **2.4.20 Delay**

The sum of send and receive delay shall below 50 ms

The ETSI EN 300175-8 and EN 300176-2 gives requirements regarding the round trip delay from the MRP to the air interface and from the air interface to the ERP. It should not exceed 19.5 ms and includes 5 ms for looping back the signal in the reference FP.

Due to the higher TCLw requirements it is necessary to allow an additional delay for signal processing (echo cancellation).

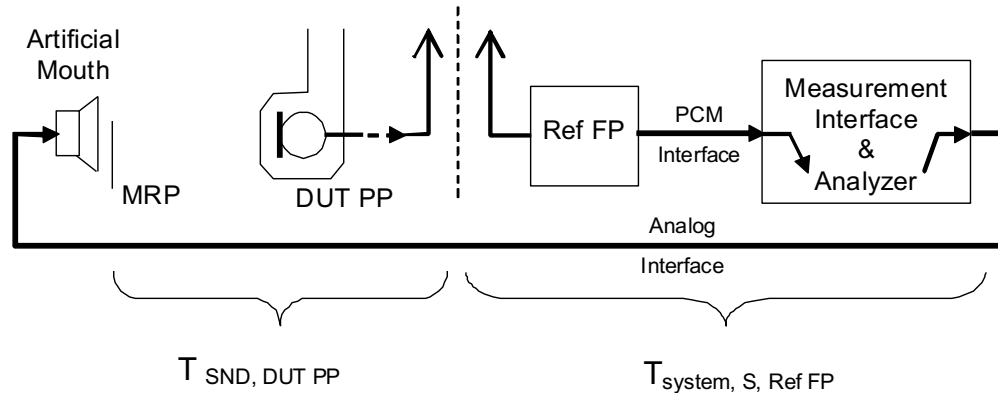
### **2.4.20.1 Send Delay**

For a Portable Part the send delay is defined as the one-way delay from the acoustic interface (MRP) to the air interface

**Measurement Method:**

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -4.7 dBPa.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the PCM interface of the FP reference interface (see figure 8) The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP sending delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 8: Measurement of PP send delay**

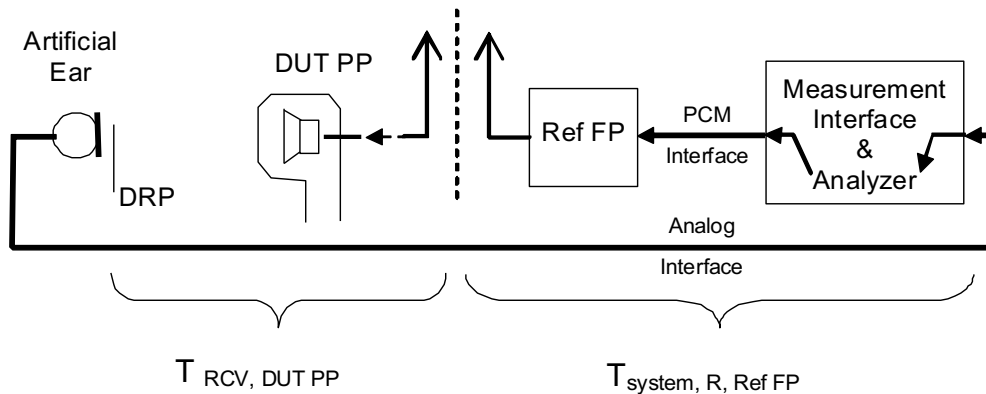
## 2.4.20.2 Receive Delay

For a Portable Part the receive delay is defined as the one-way delay from the air interface to the DRP.

**Measurement method:**

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -16 dBm0.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the DRP (see figure 9) The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP receive delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 9: Measurement of PP receive delay**

## 2.4.21 Double Talk Performance

For the measurements of double talk performance it is important the send and receive signals are synchronous. The test laboratory shall ensure this for e.g. by compensating one signal with the measured delay in send or receive direction.

### 2.4.21.1 Attenuation Range in Send Direction during Double Talk $A_{H,S,dt}$

Based on the level variation in send direction during double talk  $A_{H,S,dt}$  the behavior of the terminal can be classified according to table 11.

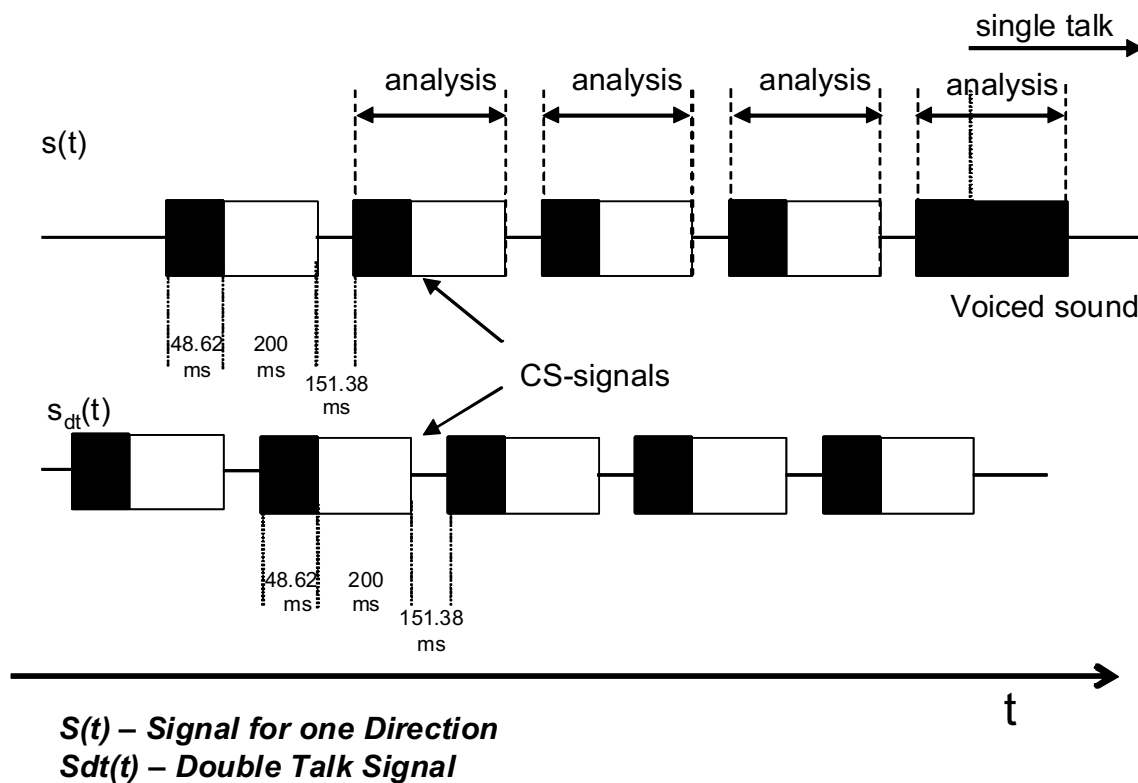
Category (according to ITU-T Recommen dation P.340 [36])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	$\leq 3$	$\leq 6$	$\leq 9$	$\leq 12$	$>12$

**Table 11: Category regarding "duplex capability" depending on  $A_{H,S,dt}$**

The category regarding duplex capability in send direction shall be at least 2b.

Measurement method:

The test signal to determine the attenuation range during double talk is shown in figure 10. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction.



**Figure 10: Measurement signal for double talk**

Figure 10 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 send and receive direction. The analysis times are shown in figure 10 as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.



The settings for the test signals are as follows:

	Receive Direction	Send Direction
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	-4.7 dBP <sub>a</sub>
Active Signal Parts	-14.7 dBm0	-3 dBP <sub>a</sub>

**Table 12: Settings for the test signals**

When determining the attenuation range in send direction the signal measured at the electrical reference point is referred to the test signal inserted.

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 direction until its complete activation (during the pause in the receive 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.

### 2.4.21.2 Attenuation Range in Receive Direction during Double Talk $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [36])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,R,dt}$ [dB]	$\leq 3$	$\leq 5$	$\leq 8$	$\leq 10$	$> 10$

**Table 13: Category regarding "duplex capability" depending on  $A_{H,R,dt}$**

The category regarding duplex capability in receive direction shall be at least 2b.

#### Measurement method

The test signal to determine the attenuation range during double talk is shown in Figure 11. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are described in table 12.

When determining the attenuation range in receive direction the signal measured at the artificial ear referred to the test signal inserted.

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 direction until its complete activation (during the pause in the send 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.

### 2.4.21.3 Detection of Echo Components during Double Talk

Echo Loss during double talk is the echo suppression provided by the terminal during double talk measured at the electrical reference point.

	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	$\geq 27$	$\geq 23$	$\geq 17$	$\geq 11$	$< 11$

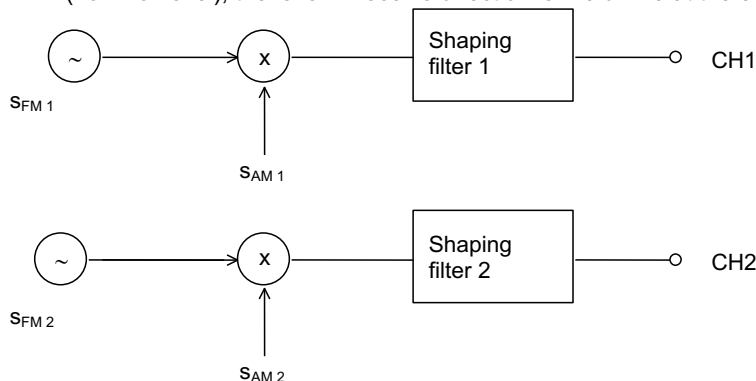
**Table 14: Category regarding "duplex capability" depending on  $A_{H,R,dt}$**

The category regarding echo components in double talk shall be at least 2b.

#### Measurement method

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. The measurement signals used are shown in figure 10. A detailed description can be found in ITU-T Recommendation P.501 [38].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4.7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).



**Figure 11: Measurement signals**

$$s_{FM1,2}(t) = \sum A_{FM1,2} \cdot \cos(2\pi t n \cdot F_{01,2}) ; n = 1, 2, \text{ etc.}$$

$$s_{AM1,2}(t) = A_{AM1,2} \cdot \cos(2\pi t F_{AM1,2});$$

Note: The calculation formulas are currently in evaluation and might be changed in later versions.

The settings for the signals are as follows.

Receive Direction			Send Direction		
$f_m$ [Hz]	fmod(fm) [Hz]	$F_{am}$ [Hz]	$f_m$ [Hz]	fmod(fm)[Hz]	$F_{am}$ [Hz]
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2650	±35	3
2 750	±40	3	2 900	±35	3
3 000	±40	3	3 150	±35	3
3 250	±40	3	3 400	±35	3
3 500	±40	3	3 650	±35	3
3 750	±40	3	3 900	±35	3

NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.

**Table 15: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves**

The test signal is measured at the electrical reference point (send direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receive direction (see ITU-T Recommendation P.501 [38]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receive direction 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 the table 14. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

## 2.4.22 Switching characteristics

### 2.4.22.1 Activation in Send Direction

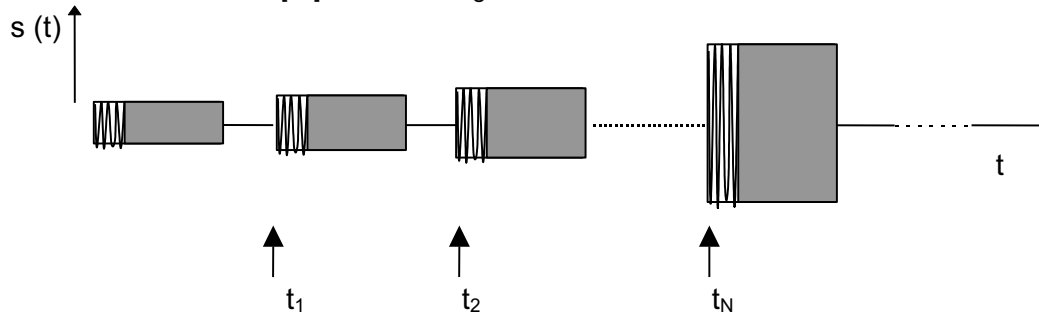
The activation in send direction is mainly determined by the built-up time  $T_{r,S,min}$  and the minimum activation level ( $L_{S,min}$ ). The minimum activation level is the level required to remove the inserted attenuation in send direction during idle mode. The built-up time is determined for the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

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

The minimum activation level  $L_{S,min}$  shall be  $\leq -20$  dBPa and the built-up time  $T_{r,S,min}$  (measured with minimum activation level) should be  $\leq 15$  ms.

#### Measurement method

The structure of the test signal is shown in figure 12. The test signal consists of CSS components according to ITU-T Recommendation P.501 [38] with increasing level for each CSS burst.



**Figure 12: Test Signal to Determine the Minimum Activation Level and the Built-up Time**

The settings of the test signal are as follows.

	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 Direction	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24.7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [38] assuming a pause of about 100 ms.			

**Table 16: Settings for test signals**

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

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.

The measured delay in send direction (see chapter 2.4.20) shall be subtracted from the measured value.

### 2.4.22.2 Activation in Receive Direction

The activation in receive direction is mainly determined by the built-up time  $T_{r,R,min}$  and the minimum activation level ( $LR_{min}$ ). The minimum activation level is the level required to remove the inserted attenuation in receiving direction 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 in the following is always referred to the test signal level at the electrical reference point (POI).

The minimum activation level  $LR_{min}$  shall be  $\leq -35.7$  dBm0 (measured during the active signal part) and the built-up time  $T_{r,R,min}$  (measured with minimum activation level) shall be  $< 15$  ms.

#### Measurement method

The test signal to determine the attenuation range during double talk is shown in Figure 10. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

	Receiving Direction	Sending Direction
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	-4.7 dBPa
Active Signal Parts	-14.7 dBm0	-3 dBPa

**Table 17: Settings for test signals**

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

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 receiving direction 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.

## 2.4.23 Quality of echo cancellation

### 2.4.23.1 Temporal echo effects

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 measured during the TCLw test.

#### Measurement method

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [38] with an average level of -5 dBm0. The echo signal is analyzed during a period of at least 2.8 s which represents 8 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 measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analyzed.

### 2.4.23.2 Spectral Echo Attenuation

The echo attenuation vs. frequency shall be below the tolerance mask given in table 18.

100	-20
200	-30
300	-38
800	-34

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1 500	-33
2 600	-24
3500	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale.	
NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

**Table 18: Mask for echo attenuation vs. frequency**

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in send direction in order to mask the echo signal.

#### Measurement method

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [38]. The level of the training sequence is -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1.4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window). The spectral echo attenuation is analyzed in the frequency domain in dB.

## 2.5 PP group B: Handset and headset in wideband mode

Used codec for measurement: G.722

### 2.5.1 Measurement Overview

Chapter	Name	Position	Appl. Force	Correction	Vol.	Nrb. of Measurements
2.5.3	PP send frequency response	HATS	8N	n.r.	nom	1
2.5.4	PP receive frequency response	HATS	8N	diffuse field	nom	1
2.5.5	PP Send Loudness Rating	HATS	8N	n.r.	nom,	1
2.5.6	PP Receive Loudness Rating	HATS	8N	DRP-ERP (P.57)	nom,max	2
2.5.7	Talker sidetone	HATS	8N	DRP-ERP (P.57)	nom	1
2.5.8	Sidetone delay	HATS	8N	diffuse field	nom	1
2.5.9	TCLw	HATS	8N	n.r.	nom, max	2
2.5.10	Stability loss	HATS	n.r.	n.r.	max	1
2.5.11	Send Distortion	HATS	8N	n.r.	nom	1
2.5.13	Receive Distortion - Average line level	HATS	8N	diffuse field	max	1
2.5.14	Receive Distortion - High line level	HATS	8N	diffuse field	max	1
2.5.15	Noise in send direction	HATS	8N	n.r.	nom	1
2.5.16	Noise in receive direction	HATS	8N	diffuse field	nom, max	2
2.5.17	Acoustic Shock	HATS	13N	tbd	max	1
2.5.18	Delay	HATS	8N	diffuse field	nom	2
2.5.19.1	Attenuation Range in Send Direction during Double Talk AH,S,dt	HATS	8N	diffuse field	nom	1
2.5.19.2	Attenuation Range in Receive Direction during Double Talk AH,S,dt	HATS	8N	diffuse field	nom	1
2.5.19.3	Detection of Echo Components during Double Talk	HATS	8N	diffuse field	nom	1
2.5.20.1	Activation in Send Direction	HATS	8N	diffuse field	nom	1
2.5.20.2	Activation in Receiving Direction	HATS	8N	diffuse field	nom	1
2.5.21.1	Temporal echo effects	HATS	8N	diffuse field	nom	1
2.5.21.2	Spectral Echo Attenuation	HATS	8N	diffuse field	nom	1
					Total:	26

Note: Some measurements counts depending on the numbers of volume levels, assumption here is 3 levels.

**Table 19: Measurement overview for PP group B**

### 2.5.2 General specification

The volume level which is closest to the RLR requirement shall be used for all measurements in nominal volume setting. This volume setting shall also be the nominal setting for send direction.

### 2.5.3 PP send frequency response

The send frequency response of the handset or the headset shall be within a mask as defined in table 20. This mask shall be applicable for all types of handsets and headsets.

Frequency (Hz)	Upper Limit	Lower Limit
100	0	
200	5	-5
5 000	5	-5
6 300	5	-10

8 000	5	
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Table 20: Send frequency response

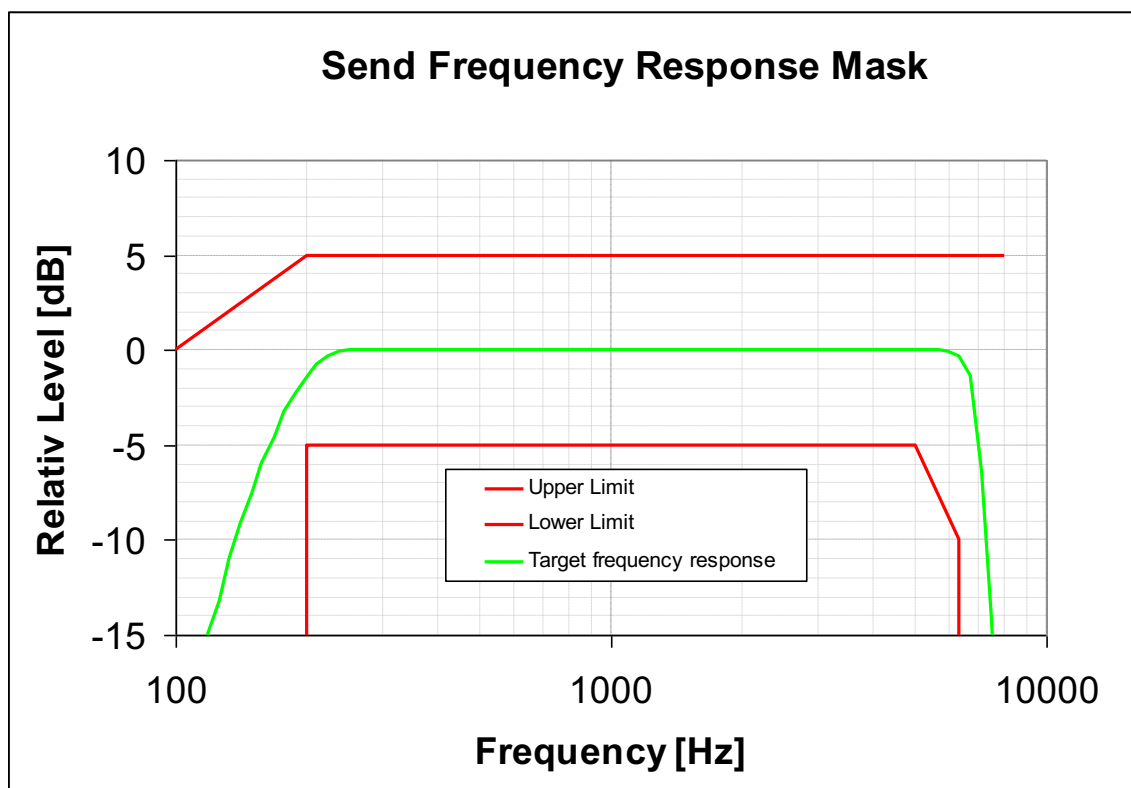


Figure 13: Send frequency response

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. is a floating or "best fit" mask.

#### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501[38] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, duration 20 s (10 s female, 10 s male voice), measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application force used to apply the handset against the artificial ear is 8N.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [37]. The results are averaged (averaged value in dB, for each frequency).

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

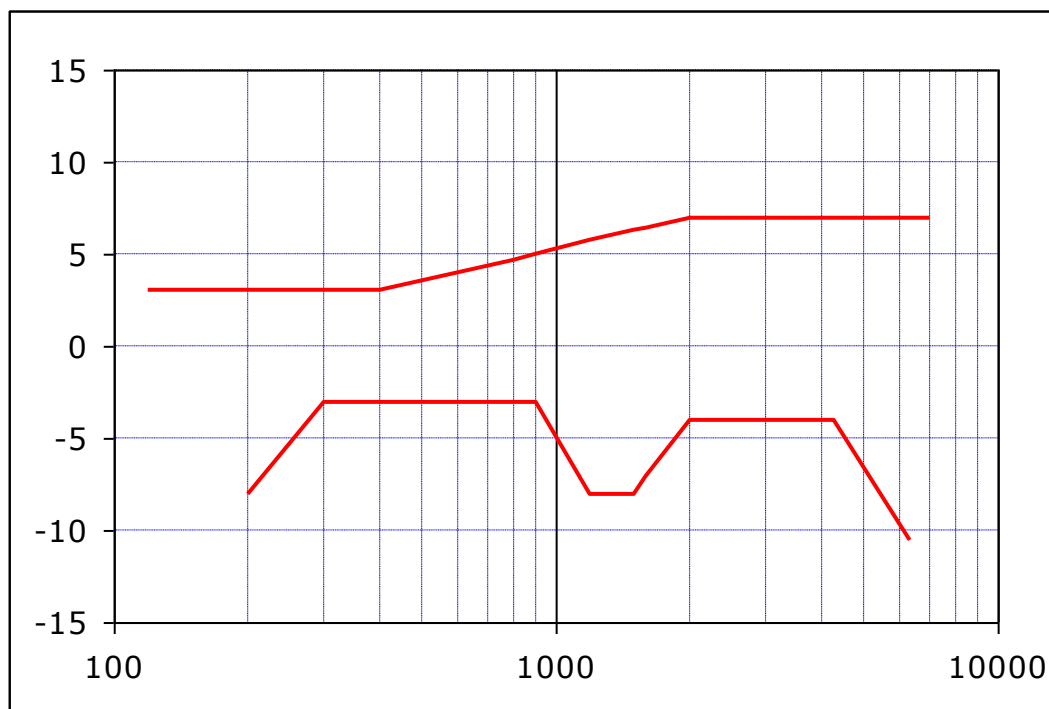
Note: Within the frequency range of 3 400 Hz to 7 000 Hz, it is allowed that the frequency response of the device under test falls once below the lower limit of the frequency mask. The width of this gap where the frequency response is outside the defined mask shall not be more than  $0.08 \cdot f_{gap\_center}$ , where  $f_{gap\_center}$  is the center frequency of the allowed gap.

## 2.5.4 PP receive frequency response

The receive frequency response of the handset or the headset shall be within a mask as defined in table 21 The application force for handsets is 8N. This mask shall be applicable for all types of headsets.

Frequency (Hz)	Upper Limit 8N	Lower Limit 8N
120	3	
200	3	-8
300	3	-3
400	3	-3
900	4.7	-3
1 200	5.7	-8
1 500	6.3	-8
1 600	6.4	-7.1
2 000	7	-4
3 000	7	-4
3 500	7	-4
4 250	7	-4
5 000	7	-6.7
6 300	7	-10.5
7 000	7	
8 000	7	

**Table 21: Receive Frequency Response Mask**



**Figure 14: Receive frequency response**

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. is a floating or "best fit" mask.

Measurement method

Receive frequency response is the ratio of the measured sound pressure and the input level. (dB relative Pa/V).

$$S_{\text{Jeff}} = 20 \log (p_{\text{eff}} / v_{\text{RCV}}) \text{ dB rel 1 Pa / V}$$

$S_{\text{Jeff}}$  Receive Sensitivity; Junction to HATS Ear with diffuse field correction.



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peff      DRP Sound pressure measured by ear simulator Measurement data are converted from the Drum Reference Point to diffuse field.  
 $V_{RCV}$       Equivalent RMS input voltage.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28], duration 20 s (10 s female, 10 s male voice). If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [38] shall be used. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [3030] at the digital reference point or the equivalent analogue point.

The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application forces used to apply the handset against the artificial ear is 8N.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [37]. The results are averaged (averaged value in dB, for each frequency).

The HATS is diffuse field equalized as described in ITU-T Recommendation P.581 [40]. The equalized output signal is power-averaged on the total time of analysis. The 1/12 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

Note: Within the frequency range of 3 400 Hz to 7 000 Hz, it is allowed that the frequency response of the device under test falls once below the lower limit of the frequency mask. The width of this gap where the frequency response is outside the defined mask shall not be more than  $0.08 \cdot f_{gap\_center}$ , where  $f_{gap\_center}$  is the center frequency of the allowed gap.

## 2.5.5 PP Send Loudness Rating

The nominal value of Send Loudness Rating (SLR) shall be:

$$SLR(set) = +8 \text{ dB} \pm 3 \text{ dB}.$$

If the device offers a user-controlled volume control, there shall be at least one volume setting which fulfills the requirement.

### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28], duration 20 s (10 s female, 10 s male voice). If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [38] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The handset or headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application force used to apply the handset against the artificial ear is noted in the test report.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [37]. The results are averaged (averaged value in dB, for each frequency).

The send sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 [34], bands 1 to 20. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [34], see annex A.

## 2.5.6 PP Receive Loudness Rating

The nominal value of Receive Loudness Rating (RLR) shall be:

$$RLR(set) = +2 \text{ dB} \pm 3 \text{ dB}.$$

$$RLR(\text{binaural headset}) = +8 \text{ dB} \pm 3 \text{ dB for each earphone}.$$

If the device offers a user-controlled volume control, there shall be at least one volume setting which fulfills the requirement.

The minimum difference between nominal RLR and minimum (loudest) RLR shall be 6 dB.

#### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28], duration 20 s (10 s female, 10 s male voice). If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [38] shall be used. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The handset terminal or the headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]). The application force used to apply the handset against the artificial ear is 8N. The HATS is *NOT* diffuse field equalized as described in ITU-T Recommendation P.581 [40]. The DRP-ERP correction as defined in ITU-T Recommendation P.57 [31] is applied.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [37]. The results are averaged (averaged value in dB, for each frequency).

The receive sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 [34], bands 1 to 20. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [34], annex A. No leakage correction shall be applied for the measurement.

## 2.5.7 Talker sidetone

The STMR shall be +18 dB ± 5 dB for nominal setting of the volume control.

#### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset or the headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]) and the application force shall be 8N on the artificial ear type 3.3. Where a user operated volume control is provided, the measurements shall be carried out the nominal setting of the volume control.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (ITU-T Recommendation P.79 [34], table 3, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

The Sidetone path loss (LmeST), as expressed in dB, and the Side Tone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of ITU-T Recommendation P.79 [34], using  $m = 0.225$  and the weighting factors of in table 3 of ITU-T Recommendation P.79 [34].

## 2.5.8 Sidetone delay

The maximum sidetone-round-trip delay shall be ≤ 5 ms.

#### Measurement method

The handset or the headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]).

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a pn sequence with a length of 65535 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 Recommendation P.501[38]. The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function  $\Phi_{xy}(\tau)$  between the input signal  $S_x(t)$  generated by the test system in send direction and the output signal  $S_y(t)$  measured at the artificial ear is calculated in the time domain:

$$\Phi_{xy}(\tau) = \lim_{T \rightarrow \infty} \frac{1}{T} \sum_{t=-T/2}^{T/2} S_x(t) S_y(t + \tau)$$

The measurement window T shall be exactly identical with 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 direct sound produced by the artificial mouth, the

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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:

$$H\{xy(\tau)\} = \sum_{-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)}$$
$$E(\tau) = \sqrt{[\Phi_{xy}(\tau)]^2 + \{H[\Phi_{xy}(\tau)]\}^2}$$

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

Note: The calculation formulas for the sidetone delay are currently in evaluation and might be changed in later versions.

## 2.5.9 TCLw

The TCLw shall be  $\geq 55$  dB.

With the volume control set to maximum TCLw shall be  $\geq 46$  dB.

### Measurement method

The handset or headset terminal is setup as described in clause 2.2.1. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [33]) and the application force shall be 8N on the artificial ear type 3.3 as specified in ITU-T Recommendation P.57 [31]. The ambient noise level shall be less than -64 dBPa(A) for handset and headset terminals. The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4096 points (for the 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. The low crest factor is achieved by random alternation of the phase between  $-180^\circ$  and  $180^\circ$ .

The TCLw is calculated according to ITU-T Recommendation G.122 [14], clause B.4 (trapezoidal rule) but using the frequency range of 300 Hz to 6 700 Hz (instead of 300 Hz to 3 400 Hz). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

## 2.5.10 Stability loss

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 100 Hz to 8 kHz. In case of headsets the requirement applies for the closest possible position between microphone and headset receiver.

### Measurement method

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from digital input to digital output shall be measured for frequencies from 100 Hz to 8 kHz under the following conditions:

The handset or the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure 15.

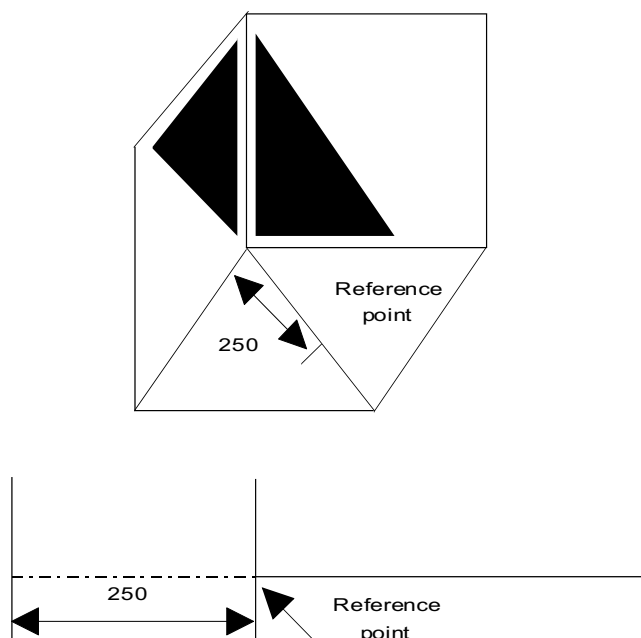
The handset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

the mouthpiece and ear cup shall face towards the surface;

the handset shall be placed centrally, the diagonal line with the ear cup nearer to the apex of the corner;  
the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 15.

The headset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

the microphone and the receiver shall face towards the surface;  
the headset receiver shall be placed centrally at the reference point as shown in figure 15;  
the headset microphone is positioned as close as possible to the receiver.



**Figure 15: Stability loss position**

## 2.5.11 Send Distortion - Average level

The A-weighted ratio of signal to total distortion (THD+N) shall be above the following mask.

Frequency [Hz]	Ratio [dB(A)]
315	26
400	30
500	30
630	30
800	30
1000	30
2000	30

**Table 22: Ratio of signal to total distortion (send)**

### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to -4.7 dBPa at the MRP.

The signal to total distortion ratio is measured up to 6.3 kHz.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP.

## 2.5.12 Send Distortion - High level

The A-weighted ratio of signal to total distortion (THD+N) shall be above the following mask.

Frequency [Hz]	Ratio [dB(A)]
315	26
400	30
500	30
630	30
800	30
1000	30
2000	30

**Table 23: Ratio of signal to total distortion (send)**

#### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to 1.3 dBPa at the MRP.

The signal to total distortion ratio is measured up to 6,3 kHz.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP.

Note: Usually the psophometric weighting is used for distortion measurements in send direction. Due to the limited bandwidth of this weighting function the A-weighting was used here also for send direction. This might be changed in a later version after evaluation.

### 2.5.13 Receive Distortion - Average line level

The A-weighted ratio of signal to total distortion (THD+N) in the max. volume setting shall be above the following mask.

Frequency [Hz]	Signal to distortion ratio limit, receive[dB(A)]
315	26
400	30
500	30
630	30
800	30
1000	30
2000	30

**Table 24: Ratio of signal to total distortion (receive)**

#### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000Hz.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation.

The signal level shall be -16 dBm0.

Measurement are made at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz.

The signal to total distortion ratio is measured up to 10kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with the diffuse field equalization active.

### 2.5.14 Receive Distortion - High line level

The A-weighted ratio of signal to total distortion (THD+N) in the max. volume setting shall be above the following mask.

Frequency [Hz]	Signal to distortion ratio limit, receive [dB(A)]
315	26
400	30
500	30
630	30
800	30
1000	30

2000	30
------	----

**Table 25: Ratio of signal to total distortion (receive)**

#### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000Hz.

An artificial voice according to ITU-Recommendation P.50 [1228] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation.

The signal level shall be -6 dBm0.

Measurement are made at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz.

The signal to total distortion ratio is measured up to 10 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with the diffuse field equalization active.

## 2.5.15 Noise in send direction

The maximum noise level produced by the PP at the POI under silent conditions in the send direction shall not exceed -64 dBm0(A).

#### Measurement method

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [38]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The activation signal level shall be -4.7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence.

Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [33]).

The send noise is measured at the POI in the frequency range from 100 Hz to 8 kHz. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 s. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0(A).

## 2.5.16 Noise in receive direction

Telephone sets with adjustable receive levels shall be adjusted so that the RLR is as close as possible to the nominal RLR.

The receive noise shall be less than -57 dBPa(A).

Where a volume control is provided, the measured noise shall not be greater than -54 dBPa(A) at the maximum setting of the volume control.

#### Measurement method:

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [38]. The activation signal level shall be -16 dBm0. The activation signal level is averaged over the complete activation signal sequence.

Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [33]).

The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 second. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The A-weighted noise level shall be measured at DRP of the artificial ear with the diffuse field equalization active.

## 2.5.17 Acoustic Shock

### 2.5.17.1 Continuous signal

#### Requirement

With a digitally encoded signal representing the maximum possible signal level at the digital interface, the sound pressure level at the ERP shall not exceed 22 dBPa (rms unweighted).

With an application force of 13N, the sound pressure level shall not exceed 22 dBPa

#### Measurement Method:

See EN 300 176-2 [1049].

Note: With some HATS it is not possible to make the measurements with an application force of 18N as described in ETSI EG 202 518.

## 2.5.18 Delay

The ETSI EN 300175-8 and EN 300176-2 gives requirements regarding the round trip delay from the MRP to the air interface and from the air interface to the ERP. It should not exceed 19.5 ms and includes 5 ms for looping back the signal in the reference FP.

Due to the higher TCLw requirements it is necessary to allow an additional delay for signal processing (echo cancellation).

The sum of send and receive delay shall be below 50 ms.

### 2.5.18.1 Send Delay

For a Portable Part the send delay is defined as the one-way delay from the acoustic interface (MRP) to the air interface

#### Measurement Method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -4.7 dBPa.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the PCM interface of the FP reference interface (see figure 16). The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP sending delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.

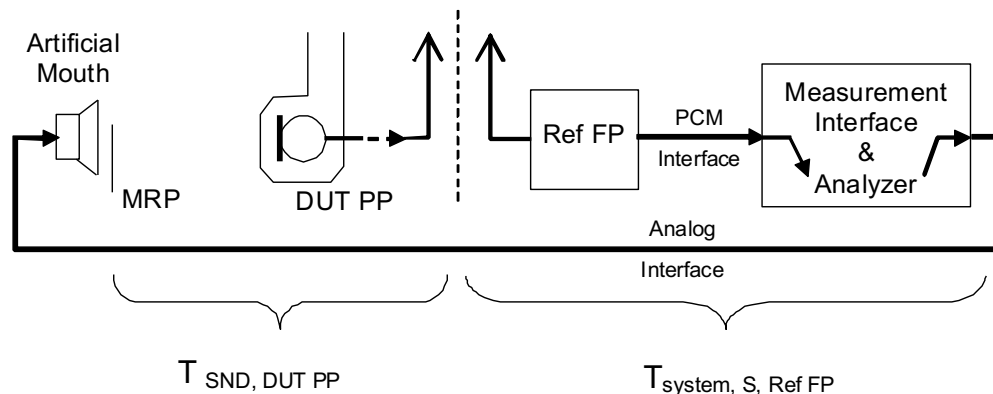


Figure 16: Measurement of PP send delay

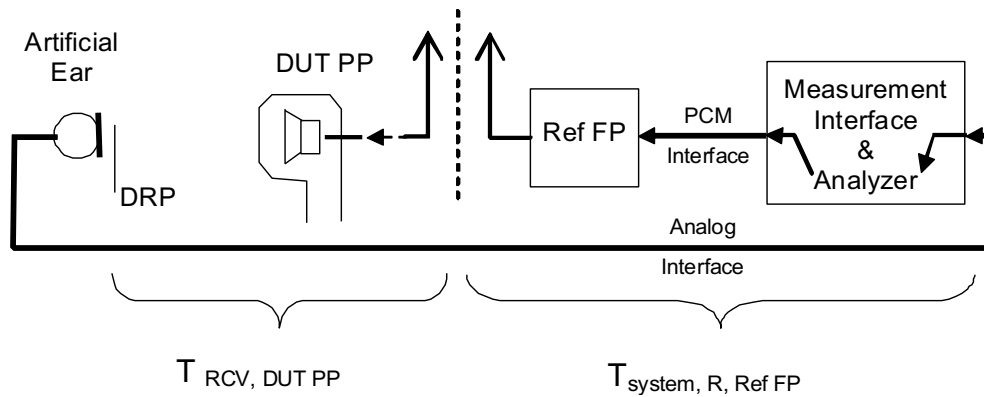
### 2.5.18.2 Receive Delay

For a Portable Part the receive delay is defined as the one-way delay from the air interface to the DRP.

#### Measurement method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -16 dBm0.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the DRP (see figure 17). The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP receive delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 17: Measurement of PP receive delay**

## 2.5.19 Double Talk Performance

For the measurements of double talk performance it is important the send and receive signals are synchronous. The test laboratory shall ensure this for e.g. by compensating one signal with the measured delay in send or receive direction.

### 2.5.19.1 Attenuation Range in Send Direction during Double Talk $A_{H,S,dt}$

Based on the level variation in send direction during double talk  $A_{H,S,dt}$  the behavior of the terminal can be classified according to table 26.

Category (according to ITU-T Recommendation P.340 [36])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	$\leq 3$	$\leq 6$	$\leq 9$	$\leq 12$	$> 12$

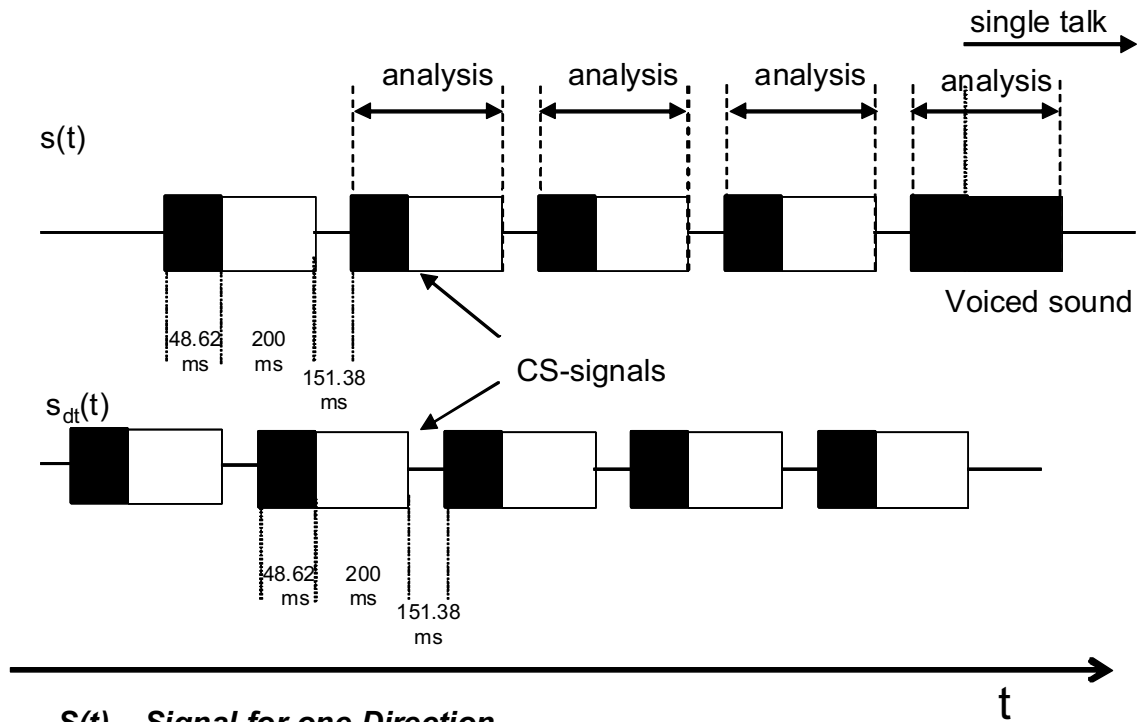
**Table 26: Category regarding "duplex capability" depending on  $A_{H,S,dt}$**

The category regarding duplex capability in send direction shall be at least 2b.

#### Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 18. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction.





**$S(t)$  – Signal for one Direction**

**$S_{dt}(t)$  – Double Talk Signal**

**Figure 18: Double Talk Test Sequence with overlapping CS signals in send and receive direction**

Figure 18 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 send and receive direction. The analysis times are shown in figure 18 as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

	Receive Direction	Send Direction
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	-4.7 dBPa
Active Signal Parts	-14.7 dBm0	-3 dBPa

**Table 27: Settings for test signal**

When determining the attenuation range in send direction the signal measured at the electrical reference point is referred to the test signal inserted.

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 direction until its complete activation (during the pause in the receive 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.

### 2.5.19.2 Attenuation Range in Receive Direction during Double Talk $A_{H,S,dt}$

Echo Loss during double talk is the echo suppression provided by the terminal during double talk measured at the electrical reference point.

Category (according to ITU-T Recommendation P.340 [36])	1	2a	2b	2c	3
	Full Duplex	Partial Duplex Capability			No Duplex

	Capability				Capability
AH,R,dt [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

**Table 28:Category regarding "duplex capability" depending on AH,R,dt**

The category regarding duplex capability in receive direction shall be at least 2b.

#### Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 18. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are described in table 28.

When determining the attenuation range in receive direction the signal measured at the artificial ear referred to the test signal inserted.

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 direction until its complete activation (during the pause in the send 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.

### 2.5.19.3 Detection of Echo Components during Double Talk

Echo Loss during double talk is the echo suppression provided by the terminal during double talk measured at the electrical reference point.

	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

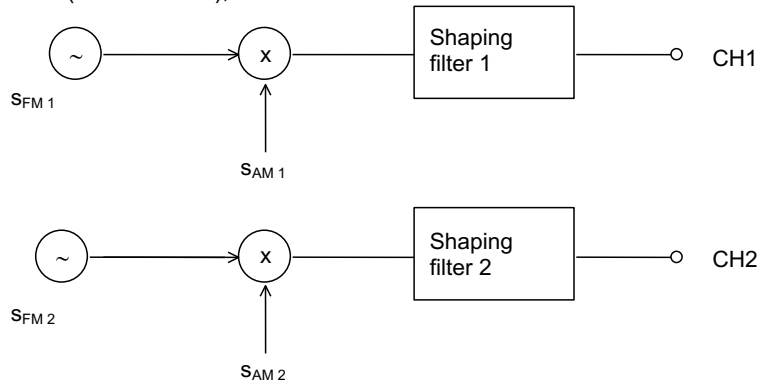
**Table 29: Category regarding "duplex capability" depending on A<sub>H,R,dt</sub>**

The category regarding echo components in double talk shall be at least 2b.

#### Measurement method

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. The measurement signals used are shown in figure 19. A detailed description can be found in ITU-T Recommendation P.501 [38].

The signals are fed simultaneously in send and receive direction. The level in send direction is -4.7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).



**Figure 19: Measurement signals**

$$s_{FM1,2}(t) = \sum A_{FM1,2} \cdot \cos(2\pi t n \cdot F_{01,2}) ; n= 1, 2, \text{ etc.}$$

$$s_{AM1,2}(t) = A_{AM1,2} \cdot \cos(2\pi t F_{AM1,2});$$

Note: The calculation formulas are currently in evaluation and might be changed in later versions.

The settings for the signals are as follows.

Receiving Direction			Sending Direction		
f <sub>m</sub> [Hz]	fmod(fm)[Hz]	F <sub>am</sub> [Hz]	f <sub>m</sub> [Hz]	fmod(fm)[Hz]	F <sub>am</sub> [Hz]
125	±2.5	3	150	±2.5	3
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 650	±35	3
2 750	±40	3	2 900	±35	3
3 000	±40	3	3 150	±35	3
3 250	±40	3	3 400	±35	3
3 500	±40	3	3 650	±35	3
3 750	±40	3	3 900	±35	3
4 000	±40	3	4 150	±35	3
4 250	±40	3	4 400	±35	3
4 500	±40	3	4 650	±35	3
4 750	±40	3	4 900	±35	3
5 000	±40	3	5 150	±35	3
5 250	±40	3	5 400	±35	3
5 500	±40	3	5 650	±35	3
5 750	±40	3	5 900	±35	3
6 000	±40	3	6 150	±35	3
6 250	±40	3	6 400	±35	3
6 500	±40	3	6 650	±35	3
6 750	±40	3	6 900	±35	3
7 000	±40	3			

NOTE: Parameters of the Shaping Filter:

f ≥ 250 Hz: Low Pass Filter, 5 dB/oct; f < 250 Hz.: High Pass Filter

**Table 30: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves**

The test signal is measured at the electrical reference point (send direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receive direction (see ITU-T Recommendation P.501 [38]). The filter will suppress frequency components of the double talk signal. In each frequency band which is used in receive direction 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 the table 14. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

## 2.5.20 Switching characteristics

### 2.5.20.1 Activation in Send Direction

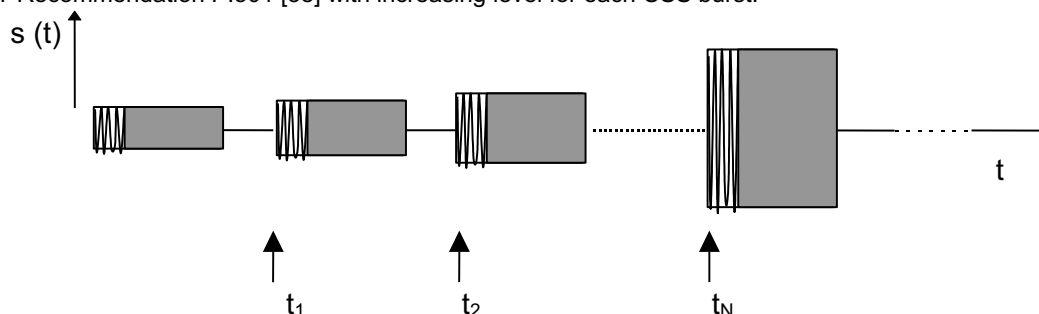
The activation in send direction is mainly determined by the built-up time  $T_{r,S,min}$  and the minimum activation level ( $L_{S,min}$ ). The minimum activation level is the level required to remove the inserted attenuation in send direction during idle mode. The built-up time is determined for the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

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

The minimum activation level  $L_{S,min}$  shall be  $\leq -20$  dBPa and the built-up time  $T_{r,S,min}$  (measured with minimum activation level) should be  $\leq 15$  ms.

Measurement method

The structure of the test signal is shown in figure 20. The test signal consists of CSS components according to ITU-T Recommendation P.501 [38] with increasing level for each CSS burst.



**Figure 20: Test Signal to Determine the Minimum Activation Level and the Built-up Time**

The settings of the test signal are as follows.

	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 Direction	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24.7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [38] assuming a pause of about 100 ms.			

**Table 31: Settings for test signals**

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

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.

The measured delay in send direction (see chapter 2.5.18) shall be subtracted from the measured value.

### 2.5.20.2 Activation in Receiving Direction

The activation in receive direction is mainly determined by the built-up time  $T_{r,R,min}$  and the minimum activation level ( $L_{R,min}$ ). The minimum activation level is the level required to remove the inserted attenuation in receiving direction 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 in the following is always referred to the test signal level at the electrical reference point (POI).

The minimum activation level  $LR_{min}$  shall be  $\leq -35.7$  dBm0 (measured during the active signal part). The built-up time  $Tr,R,min$  (measured with minimum activation level) shall be  $< 15$  ms.

#### Measurement method

The structure of the test signal is shown in Figure 20. The test signal consists of CSS components according to ITU-T Recommendation P.501 [38] with increasing level for each CSS burst.

The settings of the test signal are as follows.

	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 <b>in receive direction</b>	~250 ms / ~450 ms	-38.7 dBm0 (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of $-40$ dB <sub>mo</sub> at the POI for the CSS according to ITU-T Recommendation P.501 [38] assuming a pause of 101.38 ms.			

**Table 32: settings for test signal**

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

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.

## 2.5.21 Quality of echo cancellation

### 2.5.21.1 Temporal echo effects

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 measured during the TCLw test.

#### Measurement method

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [38] with an average level of  $-5$  dBm0. The echo signal is analyzed during a period of at least 2.8 s which represents 8 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 measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analyzed.

### 2.5.21.2 Spectral Echo Attenuation

The echo attenuation vs. frequency shall be below the tolerance mask given in Table 33.

Frequency [Hz]	Attenuation [dB]
100	-41
200	-41
300	-46
800	-46
1500	-37
2600	-37
4000	-41

---

7000	-41
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

**Table 33: Mask for echo attenuation vs. frequency**

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in send direction in order to mask the echo signal.

#### Measurement method

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [38]. The level of the training sequence is -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1.4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window). The spectral echo attenuation is analyzed in the frequency domain in dB.

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## 2.6 PP group C: Narrowband Handsfree

Used codec for measurement: G.726

### 2.6.1 Measurement Overview

Chapter	Name	Vol.	Nbr. of Measurements
2.6.3	Send Frequency Response	nom.	1
2.6.4	Receive Frequency Response	nom	1
2.6.5	Send Loudness Rating	nom	
2.6.6	Receive Loudness Rating	nom	1
2.6.7	Send Distortion	nom	1
2.6.8	Receive Distortion	max	1
2.6.9	Out-of-band signals in receive direction	nom	1
2.6.10	Send noise	nom	1
2.6.11	Receive noise	nom	1
2.6.12	Terminal Coupling Loss of PP	nom, max	2
2.6.13	Delay	nom	2
2.6.14.1	Detection of Echo Components during Double Talk	nom	1
2.6.15.1	Activation in Send Direction	nom	1
2.6.15.2	Activation in Receive Direction	nom	1
2.6.16.1	Temporal echo effects	nom	1
2.6.16.2	Spectral Echo Attenuation	nom	1
		Total:	17

Note: Some measurements counts depending on the numbers of volume levels, assumption here is 3 levels.

**Table 34: Measurement overview for PP group C**

## 2.6.2 General specification

The volume level which is closest to the RLR requirement shall be used for all measurements in nominal volume setting. This volume setting shall also be the nominal setting for send direction.

## 2.6.3 Send Frequency Response

The send sensitivity/frequency response shall be within the limits given in table 35.

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	5	-
200	5	-
250	5	-
315	5	-∞
315	5	-9
400	5	-8
500	5	-7
630	5	-6
800	5	-4
1 000	5	-3
1 300	7	-3
1 600	8	-3
2 000	9	-3
2 500	9	-3
3 100	9	-3
4 000	5	-∞

Table 35: Send frequency response

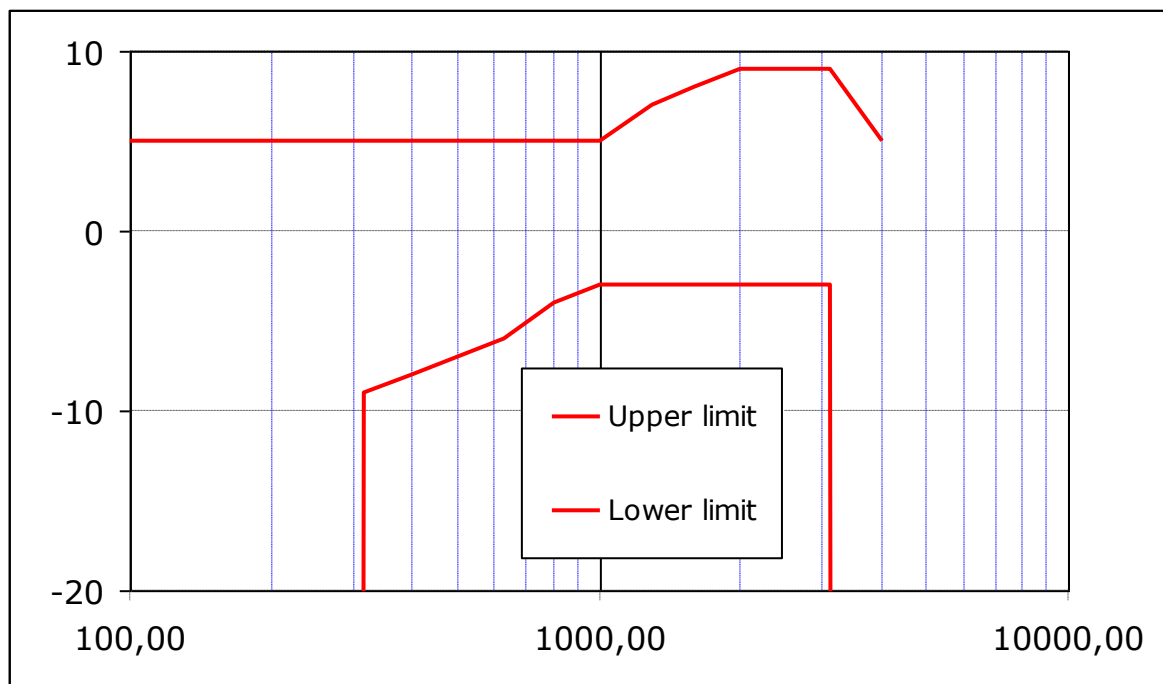


Figure 21: Send frequency response

### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501[38] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, duration



20 s (10 s female, 10 s male voice), measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

## 2.6.4 Receive Frequency Response

The following masks are required for handsfree and loudspeaking terminals. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) - linear (dB sensitivity) scale.

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
400	6	-
500	6	-∞
500	6	-9
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
3 100	6	-∞
4 000	6	-

Table 36: : Receive frequency response handheld handsfree PP

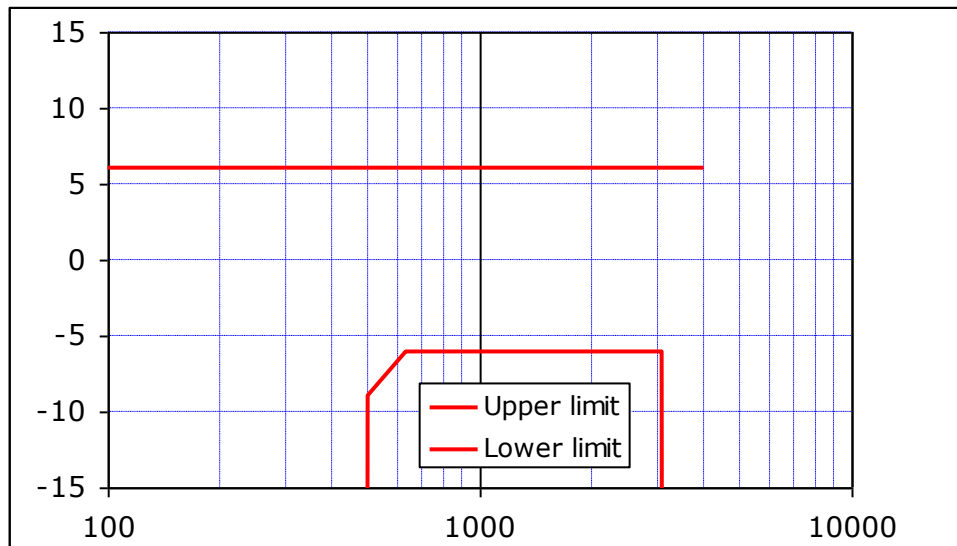


Figure 22: Receive frequency response

Measurement method

Measurement is operated at nominal value of volume control.

Receive frequency response is the ratio of the measured sound pressure and the input level. (dB relative Pa/V).

$$S_{\text{Jeff}} = 20 \log (p_{\text{eff}} / v_{\text{RCV}}) \text{ dB rel 1 Pa / V}$$

$S_{Jeff}$  Receive Sensitivity; Junction to HATS Ear with free field correction.  
 $p_{eff}$  DRP Sound pressure measured by ear simulator Measurement data are converted from the Drum Reference Point to free field.  
 $V_{RCV}$  Equivalent RMS input voltage.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [2828]. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [30] at the digital reference point or the equivalent analogue point.  
The HATS is free field equalized as described in ITU-T Recommendation P.581 [40]. The equalized output signal is power-averaged on the total time of analysis.  
Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.  
The sensitivity is expressed in terms of dBPa/V.

## 2.6.5 Send Loudness Rating

The value of SLR shall be +13 dB  $\pm$  3 dB.

This value is derived from Handset SLR. According to ITU-T Recommendation P.340 [36] the SLR of a hands-free telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

### Measurement method

An artificial voice according to ITU-T Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used to test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Calibration is realized as explained in clause 2.2.3.

SLR shall be calculated according ITU-T Recommendation P.79 [34].

## 2.6.6 Receive Loudness Rating

Nominal value of RLR = +9  $\pm$  3 dB. This value has to be fulfilled for one position of volume range.

### Measurement method:

The RLR shall be calculated according to ITU-T Recommendation P.79 [34]. The HATS is free field equalized as described in ITU-T Recommendation P.581 [40].

The receive sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [3434], bands 4 to 17. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dB Pa/V and the RLR(cal) shall be calculated according to the formula 5-1 of ITU-T Recommendation P.79 [34], using the receive weighting factors from table 1 and according to clause 6, of ITU-T Recommendation P.79 [34]; The RLR shall then be computed as RLR(cal) minus 14 dB according to ITU-T Recommendation P.340 [36], and without the LE factor.

## 2.6.7 Send Distortion

The psophometric weighted ratio of signal to total distortion shall be above the following mask.

Frequency [Hz]	Ratio [dB(p)]
315	26
400	30
500	30
630	30
800	30
1000	30

**Table 37: Ratio of signal to total distortion (send)**

### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz, the duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be calibrated to -4.7 dBPa at the MRP.

The signal to distortion ratio is measured up to 3.15 kHz.

An artificial voice according to ITU-T Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP.

## 2.6.8 Receive Distortion

The A-weighted ratio of signal to total distortion shall be above the following mask.

Frequency [Hz]	Ratio [dB(A)]
500	20
630	30
800	30
1000	30

**Table 38: Ratio of signal to total distortion (receive)**

Measurement method

Test setup is described in clause 2.2.3.

The signal used is an activation signal followed by a series sine-wave signal with a frequency at 500 Hz, 630 Hz, 800 Hz and 1000 Hz, The duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be calibrated to -16 dBm0.

An artificial voice according to ITU-T Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -16 dBm0.

The signal to total distortion ratio is measured up to 10 kHz.

## 2.6.9 Out-of-band signals in receive direction

Any spurious out-of-band image signals in the frequency range from 4.6 kHz to 8 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in table 39.

Frequency (kHz)	Signal limit (dB)
4.6	35
8	45
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

**Table 39: Out-of-band signal limit (receive)**

Measurement method

Measurement is operated at nominal value of volume control.

The signal used is an activation signal followed by a sine-wave signal. For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz applied at the level of -16 dBm0, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively at measurement point.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -16 dBm0.

## 2.6.10 Send noise

The limit for the maximum send noise level shall be -64 dBm0p.

Measurement method

For a correct activation of the system, an artificial voice according to ITU-T Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation. Level of this activation signal shall be -4.7 dBPa at the MRP. The analyze window shall start 100 ms after the activation signal to avoid any influences caused by reverberations and shall be at least 1s.

The psophometric noise level at the output of the test setup is measured. The psophometric filter is described in ITU-T Recommendation O.41 [25]

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### 2.6.11 Receive noise

The noise level shall not exceed -57 dBPa(A) at nominal setting of the volume control.

#### Measurement method

Test setup is described in clause 2.2.3.

A signal is applied to input of test system in order to ensure correct activation of receive state. An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -16 dBm0.

The analyze window shall start 100 ms after the activation signal to avoid any influences caused by reverberations and shall be at least 1s.

### 2.6.12 Terminal Coupling Loss of PP

TCLw shall be greater than 42 dB when measured under free field conditions at nominal setting of volume control. TCLwst shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.

#### Measurement method

The setup for terminal is described in clause 2.2.3.

For hands-free measurement, HATS is positioned but not used.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4096 points (for the 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. The low crest factor is achieved by random alternation of the phase between -180 ° and 180 °.

The TCLw is calculated according to ITU-T Recommendation G.122 [14], clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window (e.g. 200 ms) has to be applied adapted to the duration of the actual test signal.

### 2.6.13 Delay

The ETSI EN 300175-8 [8] and EN 300176-2 [10] gives requirements regarding the round trip delay from the MRP to the air interface and from the air interface to the ERP. It should not exceed 19.5 ms and includes 5 ms for looping back the signal in the reference FP.

Due to the higher TCLw requirements it is necessary to allow an additional delay for signal processing (echo cancellation).

The sum of send and receive delay shall below 60 ms.

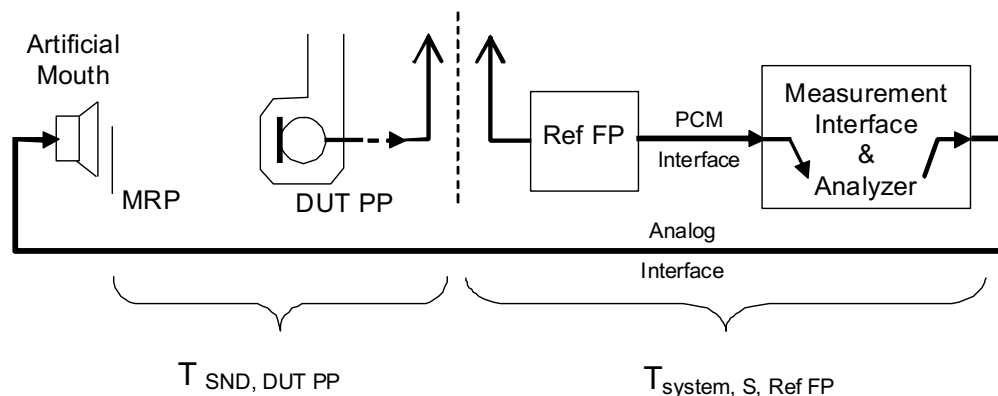
#### 2.6.13.1 Send Delay

For a Portable Part the send delay is defined as the one-way delay from the acoustic interface (MRP) to the air interface

#### Measurement Method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -4.7 dBPa.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the PCM interface of the FP reference interface (see23) The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP sending delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 23: Measurement of PP send delay**

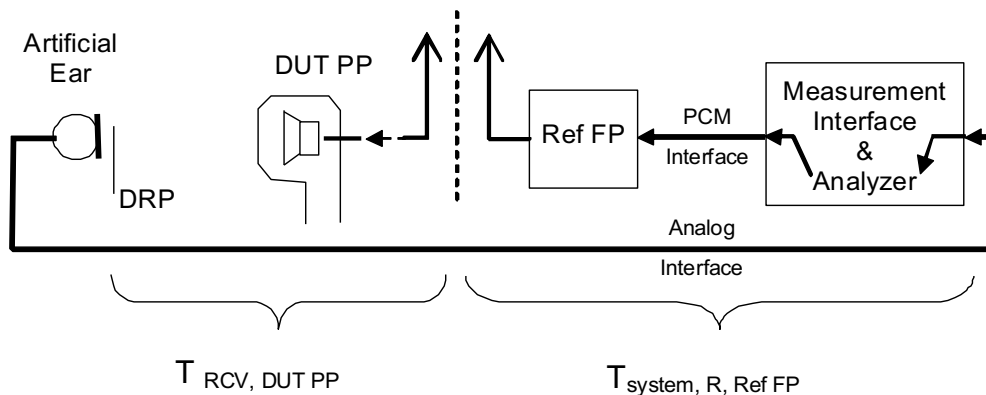
### 2.6.13.2 Receive Delay

For a Portable Part the receive delay is defined as the one-way delay from the air interface to the DRP.

Measurement method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -16 dBm0.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the DRP (see 24). The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP receive delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 24: Measurement of PP receive delay**

## 2.6.14 Double Talk Performance

For the measurements of double talk performance it is important the send and receive signals are synchronous. The test laboratory shall ensure this for e.g. by compensating one signal with the measured delay in send or receive direction.

### 2.6.14.1 Detection of Echo Components during Double Talk

"Echo Loss" is the echo suppression provided by the terminal measured at the electrical reference point. Under these conditions the requirements given in the table below are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [36]).

Category	1	2a	2b	2c	3
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(according to ITU-T Recommendation P.340 [36])					
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

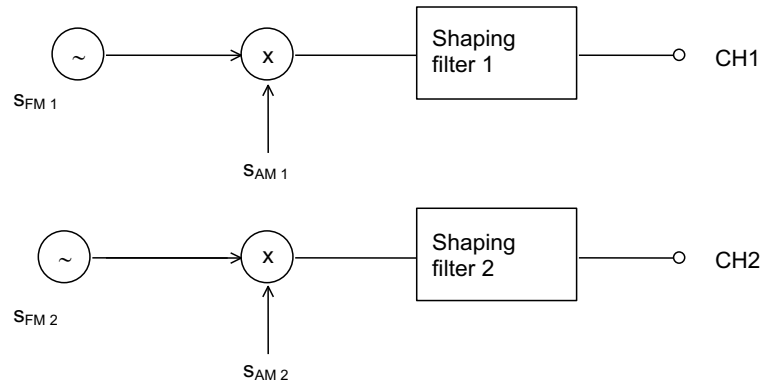
**Table 40: Category regarding "duplex capability" depending on Echo Loss**

The category regarding duplex capability in send direction shall be at least 2b.

#### Measurement method

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. The measurement signals used are shown in Figure 25. A detailed description can be found in ITU-T Recommendation P.501 [38].

The signals are fed simultaneously in send and receive direction. The level in send direction shall be -4.7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).



**Figure 25: Measurement signals**

$$s_{FM1,2}(t) = \sum A_{FM1,2} \cdot \cos(2\pi t n \cdot F_{01,2}) ; n=1, 2, \text{ etc.}$$

$$s_{AM1,2}(t) = A_{AM1,2} \cdot \cos(2\pi t F_{AM1,2});$$

Note: The calculation formulas are currently in evaluation and might be changed in later versions.

The settings for the signals are as follows.

Receive Direction			Send Direction		
$f_m$ [Hz]	fmod(fm)[Hz]	$F_{am}$ [Hz]	$f_m$ [Hz]	fmod(fm)[Hz]	$F_{am}$ [Hz]
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 650	±35	3
2 750	±40	3	2 900	±35	3
3 000	±40	3	3 150	±35	3
3 250	±40	3	3 400	±35	3

3 500	±40	3	3 650	±35	3
3 750	±40	3	3 900	±35	3

NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.

**Table 41: Settings for the signal**

Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves  
The test signal is measured at the electrical reference point (send direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receive direction (see ITU-T Recommendation P.501 [38]). The filter will suppress frequency components of the double talk signal. In each frequency band which is used in receive direction 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 the Table 40. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

## 2.6.15 Switching characteristics

### 2.6.15.1 Activation in Send Direction

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

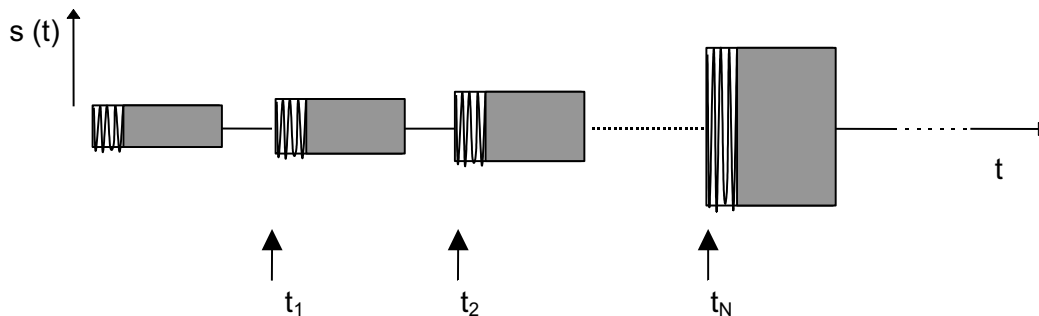
The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

The minimum activation level  $L_{S,min}$  shall be  $\leq -20$  dBPa.

The built-up time  $T_{r,S,min}$  (measured with minimum activation level) should be  $\leq 15$  ms.

Measurement method

The structure of the test signal is shown in Figure 26: Test Signal to Determine the Minimum Activation Level and the Built-up Time. The test signal consists of CSS components according to ITU-T Recommendation P.501 [38] with increasing level for each CSS burst.



**Figure 26: Test Signal to Determine the Minimum Activation Level and the Built-up Time**

The settings of the test signal are as follows.

	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 Direction	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB

NOTE: The level of the active signal part corresponds to an average level of -24.7 dBPa at the MRP for

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the CSS according to ITU-T Recommendation P.501 [38] assuming a pause of about 100 ms.

**Table 42: Settings for test signals**

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

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.

### 2.6.15.2 Activation in Receive Direction

The activation in receive direction is mainly determined by the built-up time  $T_{r,R,min}$  and the minimum activation level ( $L_{R,min}$ ). The minimum activation level is the level required to remove the inserted attenuation in receive direction 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 in the following is always referred to the test signal level at the electrical reference point (POL).

The minimum activation level  $L_{R,min}$  shall be  $\leq -35.7$  dBm0 (measured during the active signal part).

The built-up time  $T_{r,R,min}$  (measured with minimum activation level) shall be  $\leq 15$  ms.

#### Measurement method

The test signal to determine the attenuation range during double talk is shown in Figure 26. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

	Receive Direction	Send Direction
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	-4.7 dBPa
Active Signal Parts	-14.7 dBm0	-3 dBPa

**Table 43: Settings for test signals**

When determining the attenuation range in receive direction the signal measured at the artificial ear referred to the test signal inserted.

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 direction until its complete activation (during the pause in the send 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.

## 2.6.16 Quality of echo cancellation

### 2.6.16.1 Temporal echo effects

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 measured during the TCLw test.

#### Measurement method

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [38] with an average level of -5 dBm0. The echo signal is analyzed during a period of at least 2,8 s which represents 8 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 measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.



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### 2.6.16.2 Spectral Echo Attenuation

The echo attenuation vs. frequency shall be below the tolerance mask given in table 44

Frequency [Hz]	Upper Limit [dB]
100	-41
200	-41
300	-46
800	-46
1 500	-37
2 600	-37
3500	-37
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale.	
NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

**Table 44. Mask for echo attenuation vs. frequency**

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in send direction in order to mask the echo signal.

#### Measurement method

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [38]. The level of the training sequence shall be -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1.4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window).

The spectral echo attenuation is analyzed in the frequency domain in dB.

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## 2.7 PP group D: Wideband Handsfree

### 2.7.1 Measurement Overview

Chapter	Name	Vol.	Nbr. of Measurements
2.7.3	Send Frequency Response	nom	1
2.7.4	Receive Frequency Response	nom.	1
2.7.5	Send Loudness Rating	nom.	1
2.7.6	Receive Loudness Rating	nom.	1
2.7.7	Send Distortion	nom.	1
2.7.8	Receive Distortion	max.	1
2.7.9	Send noise	nom.	1
2.7.10	Receive noise	nom	1
2.7.11	Terminal Coupling Loss of PP	nom, max	2
2.7.12	Delay	nom	2
2.7.13.1	Detection of Echo Components during Double Talk	nom.	1
2.7.14.1	Activation in Send Direction	nom	1
2.7.14.2	Activaton in Receive Direction	nom	1
2.7.15.1	Temporal echo effects	nom.	1
2.7.15.2	Spectral Echo Attenuation	nom.	1
		Total:	17

**Table 45: Measurement overview for PP group D**

## 2.7.2 General specification

It is useful to start with measuring the RLR and SLR in different volume levels to find the settings which are closest to the nominal setting. The volume level which is closest to the RLR requirement shall be used for all measurements in nominal volume setting. This volume setting shall also be the nominal setting for send direction.

## 2.7.3 Send Frequency Response

The send sensitivity/frequency response shall be within the limits given in table 46:

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	$-\infty$
125	4	-10
200	4	-4
1 000	4	-4
5 000	(see note)	-4
6 300	9	-7
8 000	9	$-\infty$

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

Table 46. Send frequency response

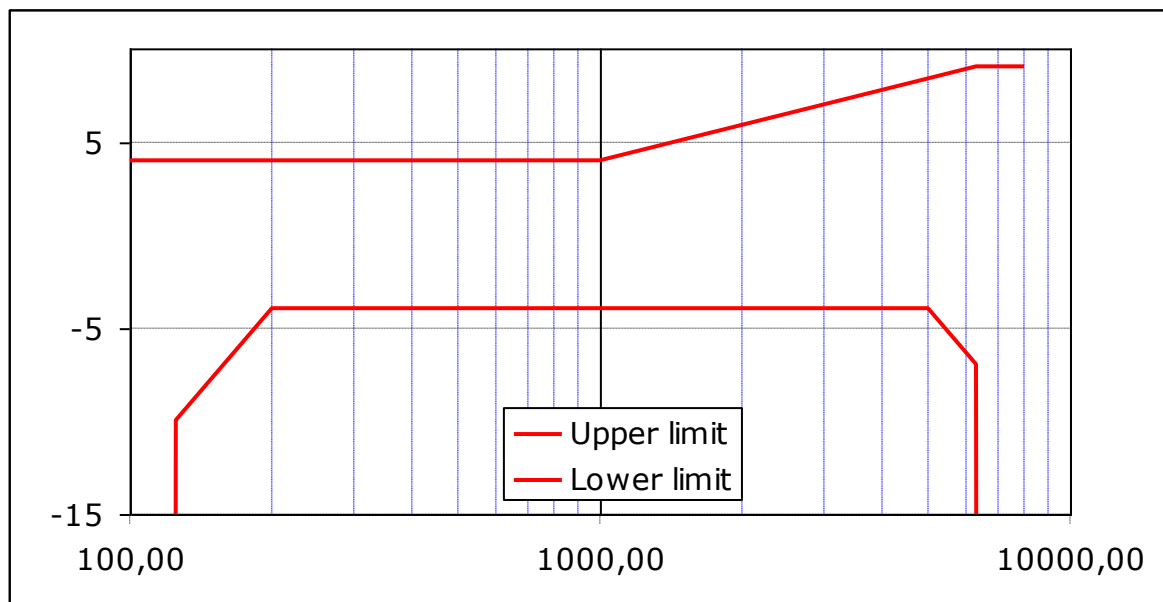


Figure 27: Send frequency response

### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501[38] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, duration 20 s (10 s female, 10 s male voice), measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

Note: Within the frequency range of 3 400 Hz to 7 000 Hz, it is allowed that the frequency response of the device under test falls once below the lower limit of the frequency mask . The width of this gap where the frequency response is outside the defined mask shall not be more than 0.08\* fgap\_center, where fgap\_center is the center frequency of the allowed gap.

## 2.7.4 Receive Frequency Response

The following mask is required for handsfree terminals. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) - linear (dB sensitivity) scale.

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
250	6	-
315	6	-
400	6	-∞
400	6	-12
500	6	-6
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
4 000	6	-6
5 000	6	-9
6 300	6	-12
6 300	6	-∞
7 000	6	-
8 000	6	-

Table 47: Receive frequency response handheld handsfree PP standard

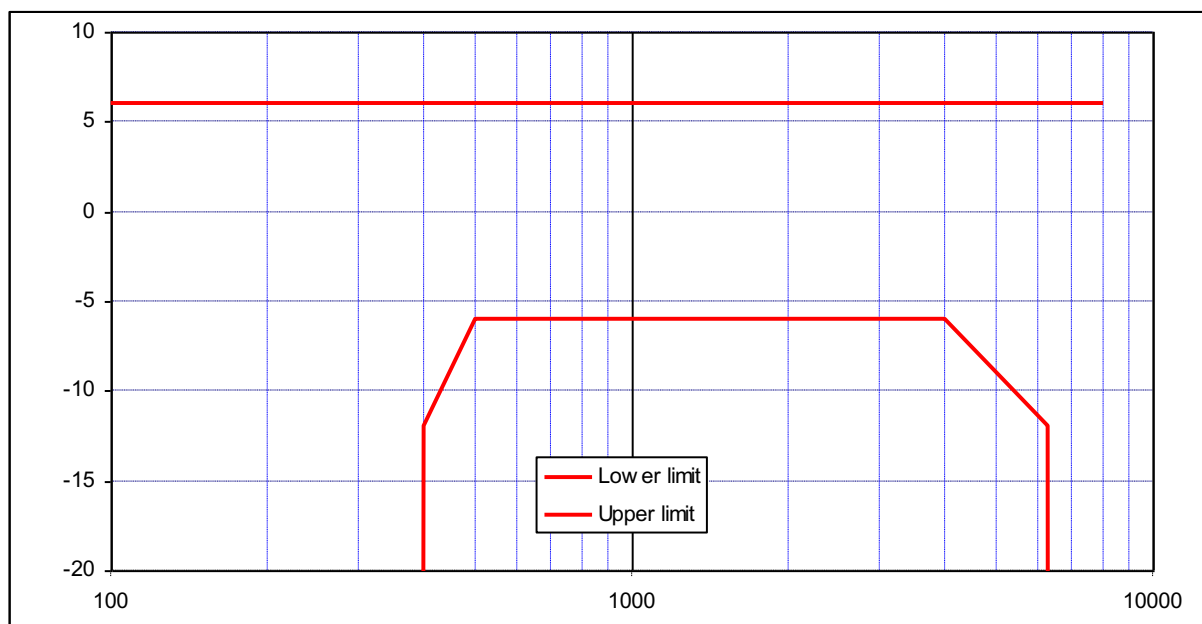


Figure 28: Receive frequency response

Measurement method  
Measurement is operated at nominal value of volume control.

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Receive frequency response is the ratio of the measured sound pressure and the input level.  
(dB relative Pa/V)

$$S_{\text{Jeff}} = 20 \log (p_{\text{eff}} / v_{\text{RCV}}) \text{ dB rel 1 Pa / V} \quad (1)$$

$S_{\text{Jeff}}$	Receive Sensitivity; Junction to HATS Ear with free field correction.
$p_{\text{eff}}$	DRP Sound pressure measured by ear simulator Measurement data are converted from the Drum Reference Point to free field
$v_{\text{RCV}}$	Equivalent RMS input voltage

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [38] shall be used. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [30] at the digital reference point or the equivalent analogue point.

The HATS is free field equalized as described in ITU-T Recommendation P.581 [40]. The equalized output signal is power-averaged on the total time of analysis.

Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [43] for frequencies from 100 Hz to 10 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

Note: Within the frequency range of 3 400 Hz to 7 000 Hz, it is allowed that the frequency response of the device under test falls once below the lower limit of the frequency mask. The width of this gap where the frequency response is outside the defined mask shall not be more than  $0.08 \cdot f_{\text{gap\_center}}$ , where  $f_{\text{gap\_center}}$  is the center frequency of the allowed gap.

## 2.7.5 Send Loudness Rating

The value of SLR shall be  $+13 \text{ dB} \pm 3 \text{ dB}$ .

### Measurement method

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used to test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4.7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Calibration is realized as explained in clause 2.2.3.

The send sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 [34], bands 1 to 20. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [34], annex A.

## 2.7.6 Receive Loudness Rating

The nominal value of RLR is  $+9 \pm 3 \text{ dB}$ .

This value has to be fulfilled for at least one position of volume range.

### Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [28]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [38] shall be used. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [30] at the digital reference point or the equivalent analogue point.

The receive sensitivity shall be calculated from each band of the 20 frequencies given in table A.2 of ITU-T Recommendation P.79 [34], bands 1 to 20. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [34], annex A. The RLR shall then be computed as  $\text{RLR}(\text{cal})$  minus 14 dB according to ITU-T Recommendation P.340 [36]), and without the LE factor.

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## 2.7.7 Send Distortion

The A-weighted ratio of signal to total distortion shall be above the following mask.

Frequency [Hz]	Ratio [dB(A)]
200	25
315	26
400	30
1 000	30
3 000	30

**Table 48: Ratio of signal to total distortion (send)**

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

### Measurement method

The signal used is an activation signal followed by a series sine-wave signal with a frequency at 200 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 kHz. The duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be calibrated to -4.7 dBPa at the MRP.

The signal to distortion ratio is measured up to 10 kHz.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP.

Note: Depending on the type of codec the test signal used may need to be adapted.

Note: Usually the psophometric weighting is used for distortion measurements in send direction. Due to the limited bandwidth of this weighting function the A-weighting was used here also for send direction. This might be changed in a later version after evaluation.

## 2.7.8 Receive Distortion

The A-weighted ratio of signal to total distortion shall be above the following mask.

Frequency [Hz]	Signal to distortion ratio limit [dB]
630	30
800	30
1 000	30
2 000	30
3 000	30

**Table 49: Ratio of signal to total distortion (receive)**

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

### Measurement method

The signal used is an activation signal followed by a sine-wave signal with a frequency at , 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz and 3 000 Hz. The duration of the sine-wave shall be of less than 1 s. Appropriate signals for activation and signal combinations can be found in ITU-T Recommendation P.501 [38]. The sinusoidal signal level shall be calibrated to -16 dBm0.

An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -16 dBm0.

The signal to total distortion ratio is measured up to 10 kHz.

## 2.7.9 Send noise

The limit for the maximum send noise level shall be -64 dBm0(A).

### Measurement method

---

For a correct activation of the system, an artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] shall be used for activation. Level of this activation signal will be -4.7 dBPa at the MRP. The analyze window shall start 100 ms after the activation signal to avoid any influences caused by reverberations and shall be at least 1s. The level at the output of the test setup is measured with A filtering.

### **2.7.10 Receive noise**

The noise level shall not exceed -54 dBPa(A) at nominal setting of the volume control.

#### **Measurement method**

A signal is applied to input of test system in order to ensure correct activation of receive state. An artificial voice according to ITU-Recommendation P.50 [28] or a speech like test signal as described in ITU-T Recommendation P.501 [38] can be used for activation. Level of this activation signal will be -16 dBm0.

The analyze window shall start 100 ms after the activation signal to avoid any influences caused by reverberations and shall be at least 1s.

### **2.7.11 Terminal Coupling Loss of PP**

TCLw shall be greater than 42 dB when measured under free field conditions at nominal setting of volume control. TCLw shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.

#### **Measurement method**

The setup for PP is described in subclause clause 2.2.3.

For hands-free measurement, HATS is positioned but not used.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order to not overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4096 points (for the 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. The low crest factor is achieved by random alternation of the phase between -180 ° and 180 °.

The TCLw is calculated according to ITU-T Recommendation G.122 [14], clause B.4 (trapezoidal rule), but using the frequency range of 300 Hz to 6 700 Hz instead of 300 Hz to 3 400 Hz. For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

### **2.7.12 Delay**

The ETSI EN 300175-8 [8] and EN 300176-2 [10] gives requirements regarding the round trip delay from the MRP to the air interface and from the air interface to the ERP. It should not exceed 19.5 ms and includes 5 ms for looping back the signal in the reference FP.

Due to the higher TCLw requirements it is necessary to allow an additional delay for signal processing (echo cancellation).

The sum of send and receive delay shall below 60 ms.

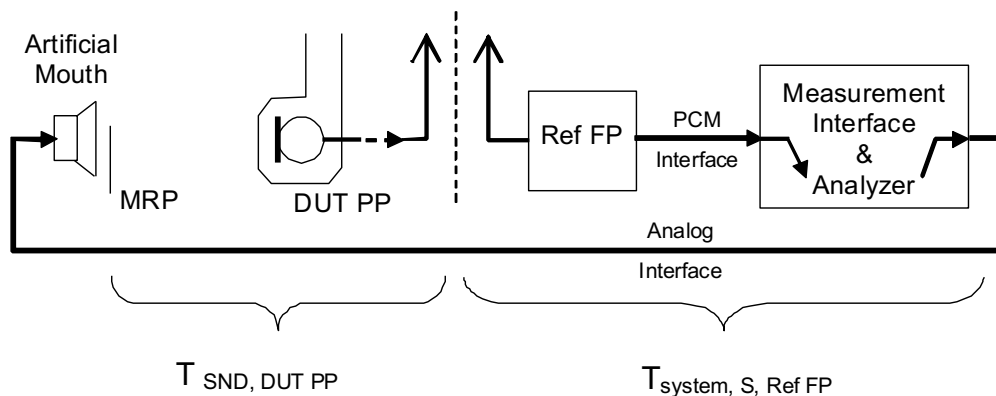
#### **2.7.12.1 Send Delay**

For a Portable Part the send delay is defined as the one-way delay from the acoustic interface (MRP) to the air interface

#### **Measurement Method:**

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -4.7 dBPa.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the PCM interface of the FP reference interface (see Figure 29) The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP sending delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 29: Measurement of PP send delay**

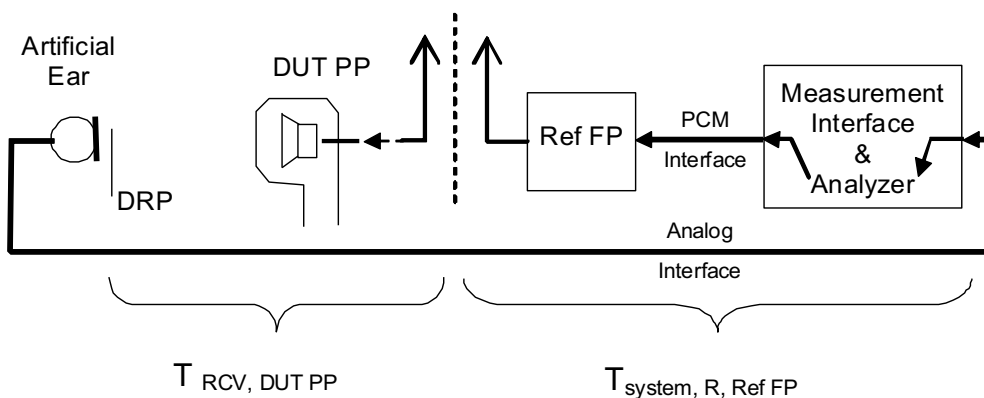
### 2.7.12.2 Receive Delay

For a Portable Part the receive delay is defined as the one-way delay from the air interface to the DRP.

Measurement method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -16 dBm0.

The delay is determined by calculating the cross-correlation function between the signal applied to the PP and the signal measured at the DRP (see Figure 30). The parameters of the cross-correlation analysis need to ensure that a delay of 100 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the PP receive delay, it shall be compensated by the delay introduced by the reference FP and the measurement interface.



**Figure 30: Measurement of PP receive delay**

## 2.7.13 Double Talk Performance

For the measurements of double talk performance it is important the send and receive signals are synchronous. The test laboratory shall ensure this for e.g. by compensating one signal with the measured delay in send or receive direction.

### 2.7.13.1 Detection of Echo Components during Double Talk

"Echo Loss" is the echo suppression provided by the terminal measured at the electrical reference point. Under these conditions the requirements given in the table below are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [36]).



Category (according to ITU-T Recommendation P.340 [36])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

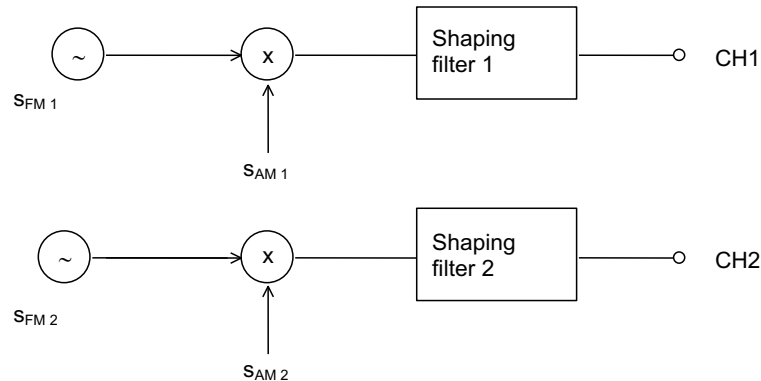
**Table 50: Category regarding "duplex capability" depending on Echo Loss**

The category regarding duplex capability in send direction shall be at least 2b.

#### Measurement method

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. The measurement signals used are shown in Figure 31. A detailed description can be found in ITU-T Recommendation P.501 [38].

The signals are fed simultaneously in send and receive direction. The level in send direction shall be -4.7 dBPa at the MRP (nominal level), the level in receive direction is -16 dBm0 at the electrical reference point (nominal level).



**Figure 31: Measurement signals**

$$s_{FM1,2}(t) = \sum A_{FM1,2} \cdot \cos(2\pi t n \cdot F_{01,2}); n = 1, 2, \text{ etc.}$$

$$s_{AM1,2}(t) = A_{AM1,2} \cdot \cos(2\pi t F_{AM1,2});$$

Note: The calculation formulas are currently in evaluation and might be changed in later versions.

The settings for the signals are as follows.

Receiving Direction			Sending Direction		
$f_m$ [Hz]	$f_{mod(fm)}$ [Hz]	$F_{am}$ [Hz]	$f_m$ [Hz]	$f_{mod(fm)}$ [Hz]	$F_{am}$ [Hz]
125	±2.5	3	150	±2.5	3
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 650	±35	3
2 750	±40	3	2 900	±35	3
3 000	±40	3	3 150	±35	3
3 250	±40	3	3 400	±35	3
3 500	±40	3	3 650	±35	3

3 750	±40	3		3 900	±35	3
4 000	±40	3		4 150	±35	3
4 250	±40	3		4 400	±35	3
4 500	±40	3		4 650	±35	3
4 750	±40	3		4 900	±35	3
5 000	±40	3		5 150	±35	3
5 250	±40	3		5 400	±35	3
5 500	±40	3		5 650	±35	3
5 750	±40	3		5 900	±35	3
6 000	±40	3		6 150	±35	3
6 250	±40	3		6 400	±35	3
6 500	±40	3		6 650	±35	3
6 750	±40	3		6 900	±35	3
7 000	±40	3				

NOTE: Parameters of the Shaping Filter:  
 $f \geq 250$  Hz: Low Pass Filter, 5 dB/oct;  $f < 250$  Hz.: High Pass Filter

**Table 51: Settings for the signal**

Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves  
The test signal is measured at the electrical reference point (send direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receive direction (see ITU-T Recommendation P.501 [38]). The filter will suppress frequency components of the double talk signal. In each frequency band which is used in receive direction 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 the Table 51. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

## 2.7.14 Switching characteristics

### 2.7.14.1 Activation in Send Direction

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

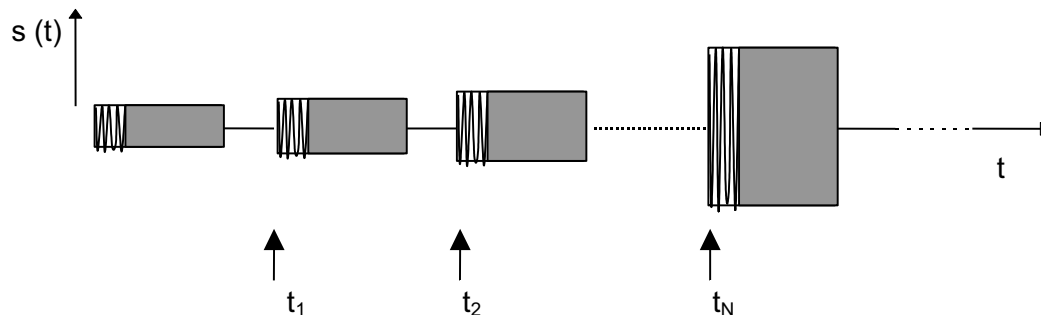
The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

The minimum activation level  $L_{S,min}$  shall be  $\leq -20$  dBPa.

The built-up time  $T_{r,S,min}$  (measured with minimum activation level) should be  $\leq 15$  ms.

Measurement method

The structure of the test signal is shown in Figure 32. The test signal consists of CSS components according to ITU-T Recommendation P.501 [38] with increasing level for each CSS burst.



**Figure 32: Test Signal to Determine the Minimum Activation Level and the Built-up Time**

The settings of the test signal are as follows.

	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 Direction	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24.7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [38] assuming a pause of about 100 ms.			

**Table 52: Settings for test signals**

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

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.

### 2.7.14.2 Activation in Receive Direction

The activation in receive direction is mainly determined by the built-up time  $T_{r,R,min}$  and the minimum activation level ( $L_{R,min}$ ). The minimum activation level is the level required to remove the inserted attenuation in receive direction 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 in the following is always referred to the test signal level at the electrical reference point (POI).

The minimum activation level  $L_{R,min}$  shall be  $\leq -35.7$  dBm0 (measured during the active signal part).

The built-up time  $T_{r,R,min}$  (measured with minimum activation level) shall be  $\leq 15$  ms.

#### Measurement method

The test signal to determine the attenuation range during double talk is shown in Figure 26. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

	Receive Direction	Send Direction
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	-4.7 dBPa
Active Signal Parts	-14.7 dBm0	-3 dBPa

**Table 53: Settings for test signals**

When determining the attenuation range in receive direction the signal measured at the artificial ear referred to the test signal inserted.

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 direction until its complete activation (during the pause in the send 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.

## 2.7.15 Quality of echo cancellation

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### 2.7.15.1 Temporal echo effects

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 measured during the TCLw test.

#### Measurement method

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [38] with an average level of -5 dBm0. The echo signal is analyzed during a period of at least 2.8 s which represents 8 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 measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

### 2.7.15.2 Spectral Echo Attenuation

The echo attenuation vs. frequency shall be below the tolerance mask given in table 44

Frequency (Hz)	Upper Limit (dB)
100	-20
200	-30
300	-38
800	-34
1 500	-33
2 600	-24
4 000	-24
7 000	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale.	
NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

**Table 54: Mask for echo attenuation vs. frequency**

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in send direction in order to mask the echo signal.

#### Measurement method

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [38]. The level of the training sequence shall be -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1.4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window). The spectral echo attenuation is analyzed in the frequency domain in dB.

## 2.8 FP group A: VoIP Narrowband mode

The measurements for FP are mandatory for CAT-iq 2.1, for CAT-iq 2.0 they are optional.

If not stated otherwise, all measurements of the FP shall be made in conjunction with a reference PP with a TCLw of 55 dB TCLw.

### 2.8.1 Measurement Overview

Chapter	Name	Nbr. of Measurements
2.8.2	PP type detection	0
2.8.3	Echo canceller for PP	0
2.8.4	Echo suppressor for PP	1
2.8.5	Send delay	1
2.8.6	Receive delay	1
2.8.7	Quality of Packet loss concealment	6
2.8.8	Clock accuracy	1
2.8.9	Send Jitter	1
	Total:	11

Table 55: Measurement overview for FP group A

### 2.8.2 PP type detection

The fixed part shall detect the terminal capabilities of the handset and interpret as described in table 56:

	Echo parameters (octet 3b):		
	001	010	011
GAP-PP	>34 dB	>46 dB	>55 dB
NG-DECT PP	>42 dB	>46 dB	>55 dB

Table 56: Interpretation of echo capabilities

### 2.8.3 Echo canceller for PP

t.b.d.

### 2.8.4 Echo suppressor for PP

Depending on the echo parameter delivered by the PP the FP shall introduce an additional echo suppressor to ensure that the overall TCLw of the system FP and PP is not below 55 dB.

The test shall be made with a reference PP emulating a TCLw of 34 dB (GAP-PP) connected to the FP.

Measurement method:

The FP is connected to the reference PP. In the digital domain of the reference PP an artificial echo according to the echo parameters described in the octet 3b is inserted. The delay of the echo generated by the reference PP shall be the maximum allowed round trip delay for the PP.

The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal. Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [28] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [38] with a length of 4096 points (for the 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. The low crest factor is achieved by random alternation of the phase between -180 ° and 180 °.

The TCLw is calculated according to ITU-T Recommendation G.122 [14], clause B.4 (trapezoidal rule) but using the frequency range of 300 Hz to 3400 Hz. For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

Note: The handling of the Echo suppression is also described in ETSI EN 300 176-2 Annex E.1[9]

## 2.8.5 Send delay

For a VoIP fixed part send delay is defined as the one-way delay from the air interface of this VoIP Fixed Part to its interface to the packet-based network. The send delay shall be <35 ms.

It is desirable that the FP keeps this delay as low as possible, preferably < 30 ms.

If the FP has to insert echo cancellation or echo suppression for PP with low TCLw the maximum send delay shall be 65 ms.

Measurement Method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -16 dB<sub>m0</sub> (nominal network level).

The delay is determined by calculating the cross-correlation function between the signal applied to the reference PP and the signal measured at the PCM interface of the IP reference interface (see figure 33) The parameters of the cross-correlation analysis need to ensure that a delay of 500 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the FP sending delay, it shall be compensated by the delay introduced by the reference PP, the IP reference interface and the measurement interface.

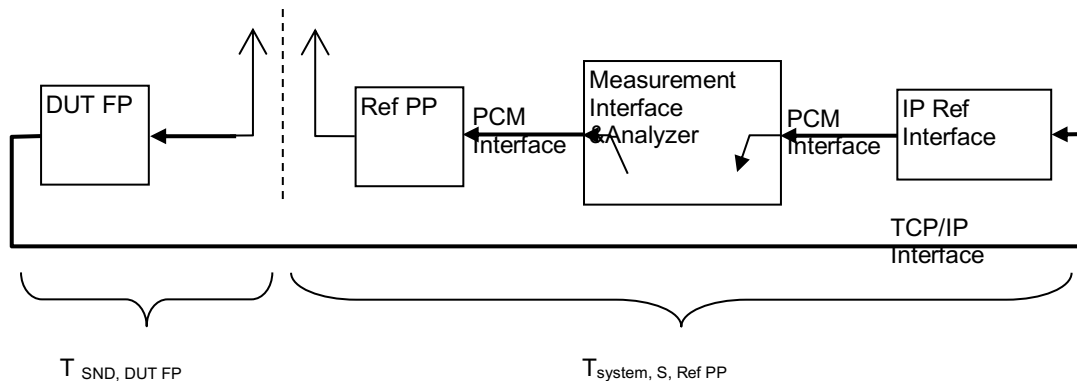


Figure 33: Sending delay of narrowband FP

## 2.8.6 Receive delay

For a VoIP fixed part, receive delay is defined as the one-way delay from the interface of the packet based network of this VoIP terminal to its air interface. The receive delay shall be <45 ms.

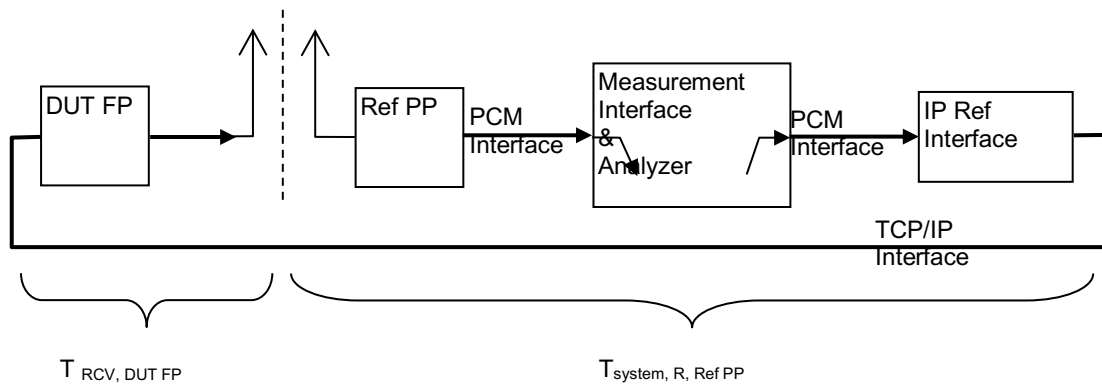
It is desirable that the FP keeps this delay as low as possible, preferably < 40 ms.

Measurement Method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the IP reference interface (see figure 34) with a level of -16 dB<sub>m0</sub> (nominal network level).

To ensure that there is no negative effect from the adaption of the jitter buffer or other adaptive signal processing blocks in front of each measurement could be an additional adaption phase with real speech signal.

The delay is determined by calculating the cross-correlation function between the signal applied to the IP reference interface and the signal measured at the PCM interface of the reference PP. The parameters of the cross-correlation analysis need to ensure that a delay of 500 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. It shall be compensated by the delay introduced by the IP reference interface, the reference PP and the measurement interface in order to derive the receiving delay of only the FP.



**Figure 34: Receive delay of narrowband FP**

Note: In receive direction it is necessary to insert a jitter buffer and packet loss concealment. Due to this additional buffers the maximum allowed delay is longer than for the send direction.

## 2.8.7 Quality of Packet loss concealment

The narrowband one-way listening speech quality of the FP in receiving direction - expressed by the MOS-LQON - shall be determined for the network conditions listed in table 57. The MOS-LQON for non-ideal network conditions shall not deviate more than specified in table 57 from the MOS-LQON determined under ideal network conditions.

Condition	Packet Loss (Equal)	Delay Variation	$\Delta$ MOS-LQON requirement (see Note 3)
1	0	No	---
2	1%	No	$\leq 0.3$
3	2%	No	$\leq 0.5$
4	3%	No	$\leq 0.6$
5	5%	No	$\leq 0.9$
6	1%	20 ms (see note 1)	$\leq 0.6$

NOTE 1: Delay Variation produced with a Pareto-Distribution and  $r = 0.5$ .

NOTE 2: For some network emulation tools, it is necessary to introduce a constant delay to offer the possibility to generate a delay variation distribution. This delay has to be subtracted from the measured delay before interpreting the results.

NOTE 3: The  $\Delta$ MOS-LQO requirements are derived from the average MOS-LQO results determined for VoIP gateways during the 5<sup>th</sup> ETSI Speech Quality Test Event in 2008. 4x8 German test sentences were used as test signals (2 sentences of 8 male and 8 female speakers, averaging per speaker).

**Table 57: receiving listening speech quality for impaired network conditions**

### Measurement Method:

A real speech signal consisting of eight sentences (1 language, 2 male and 2 female speaker each uttering 2 sentences) and providing a speech activity of approximately 50% is applied to the PCM interface of the IP reference interface. Appropriate speech signals can be found in ITU-T Recommendation P.501[38]. The active speech level acc. to ITU-T Rec. P.56 [30] shall be -16 dB<sub>m0</sub>, measured at the PCM interface.

To ensure that there is no negative effect from the adaption of the jitter buffer or other adaptive signal processing blocks in front of each measurement could be an additional adaption phase with real speech signal.

For each network condition given in table 57 the one-way listening speech quality is calculated from the original test signal and the signal received at the PCM interface of the reference PP and is expressed as MOS-LQON determined by the PESQ (narrowband) objective model acc. to ITU-T Rec. P.862 [41,].

For the packet loss conditions the test laboratory shall ensure that the loss does not deviate by more than  $\pm 10 \%$  from the required packet loss rate during a test.

Furthermore a sufficient randomization of the lost packets shall be guaranteed. This can be achieved e.g. by averaging at least four results of the transmitted speech sequence described above. The averaging should be speaker based in this case. Any averaging process shall be documented.

Note: This measurement is currently under evaluation and improvement at ETSI STQ. So it might be changed in a future release of this document.

## 2.8.8 Clock accuracy

The clock drift between the DUT FP and the IP reference interface shall be less than 40 ppm under ideal network conditions.

Measurement Method:

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyze clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1.2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level at the PCM interface of the IP reference interface shall be -16 dBm0. The transmitted signal is measured at the PCM interface of the reference PP analyzing the receiving behavior of the FP under test.

A cross correlation analysis versus time is carried out over the whole 120s sequence between the transmitted and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock drift within an analysis time range of at least 60 s is calculated as follows:

$$clock\ drift\ [ppm] = \frac{delay\ drift\ [ms]}{analysis\ duration\ [s]} \cdot 1 \cdot 10^6$$

## 2.8.9 Send Jitter

The measured interarrival jitter in sending direction of the FP under test should be less than 20 ms. Note: Any jitter introduced in sending direction will lead to potentially increased delay due to increased de-jitter buffer at the far end terminal.

Measurement Method:

The RTP data stream in sending direction should be monitored with a tap or a switch providing a monitoring port, see figure 35.

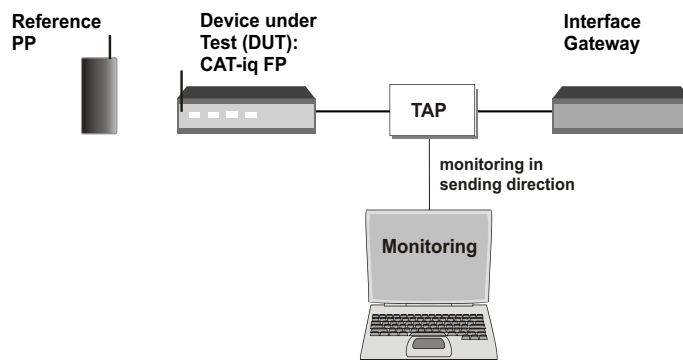


Figure 35: Monitoring of send jitter

The monitoring time should be greater than 60s. A speech signal according P.50 [28] should be played back in sending direction using a nominal network level of -16 dBm0 at the PCM interface of the measurement interface. The estimate of the statistical variance of the RTP data packet interarrival time is measured in timestamp units.

The interarrival jitter J is defined to be the mean deviation of the difference D in packet spacing at the receiver (IP reference interface) compared to the sender (DUT FP) for a pair of packets.

The mean deviation of the difference in packet spacing at the receiver is

$$D(i,j) = (R_j - S_j) - (R_i - S_i)$$



---

With  $S_i$  = the RTP timestamp for packet  $i$   
 $R_i$  = the time of arrival

The interarrival jitter is calculated continuously as each data packet  $i$  is received from the source (DUT FP) using this difference  $D$  for that packet and the previous packet  $i-1$  in order of arrival, according to the formula

$$J(i) = J(i-1) + (|D(i-1,i)| - J(i-1)) / 16 \text{ and } J(0)=0\text{ms}$$

---

## 2.9 FP group B: VoIP Wideband mode

The measurements for FP are mandatory for CAT-iq 2.1, for CAT-iq 2.0 they are optional.

If not stated otherwise, all measurements of the FP shall be made in conjunction with a reference PP with a TCLw of 55 dB TCLw.

### 2.9.1 Measurement Overview

Chapter	Name	Nbr. of Measurements
2.9.2	PP type detection	0
2.9.3	Send delay	1
2.9.4	Receive delay	1
2.9.5	Quality of Packet loss concealment	6
2.9.6	Clock accuracy	1
2.9.7	Send Jitter	1
	Total:	10

**Table 58: Measurement overview for FP group B**

### 2.9.2 PP type detection

The fixed part shall detect the terminal capabilities of the handset and interpret as described in table 59:

	Echo parameters (octet 3b):		
	001	010	011
NG-DECT PP	>42 dB	>46 dB	>55 dB

**Table 59: Interpretation of echo capabilities**

### 2.9.3 Send delay

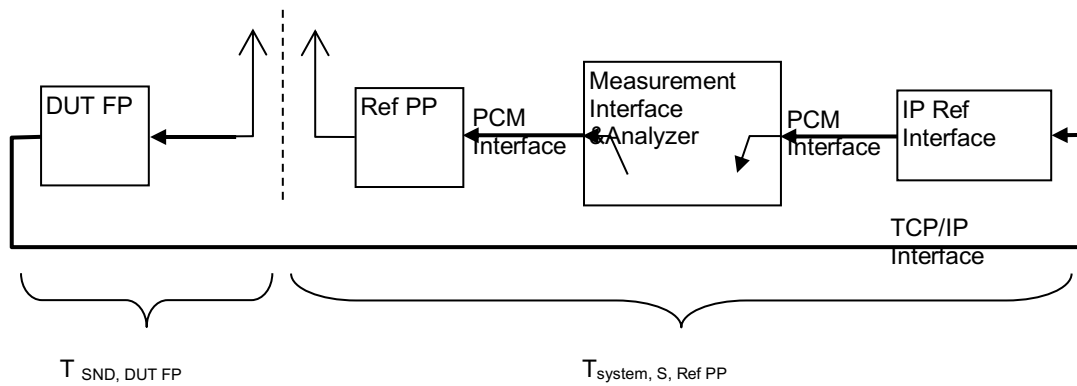
For a VoIP Fixed Part, send delay is defined as the one-way delay from the air interface of this VoIP Fixed Part to its interface to the packet-based network. The send delay shall be <35 ms.

It is desirable that the FP keeps this delay as low as possible, preferably < 30 ms.

Measurement Method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the reference PP with a level of -16 dB<sub>m0</sub> (nominal network level).

The delay is determined by calculating the cross-correlation function between the signal applied to the reference PP and the signal measured at the PCM interface of the IP reference interface (see Figure 36: Send delay of wideband FP) The parameters of the cross-correlation analysis need to ensure that a delay of 500 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. In order to determine the FP sending delay, it shall be compensated by the delay introduced by the reference PP, the IP reference interface and the measurement interface.



**Figure 36: Send delay of wideband FP**

## 2.9.4 Receive delay

For a VoIP fixed part, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its air interface. The receive delay shall be <45 ms.

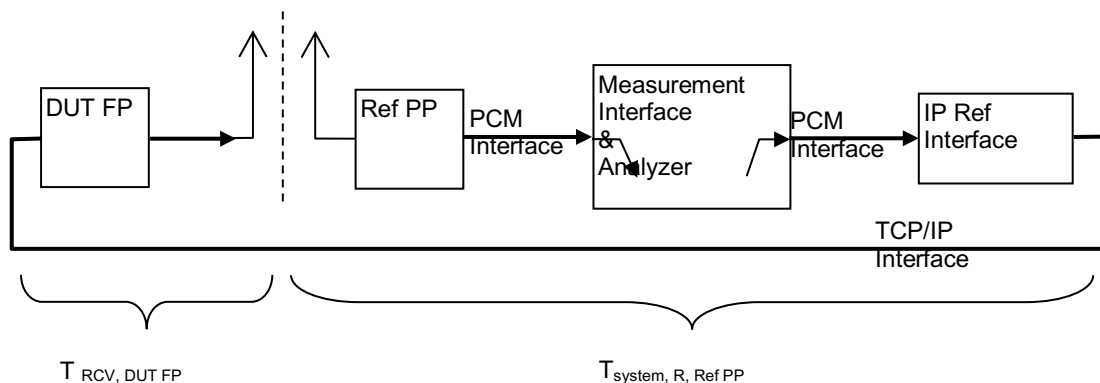
It is desirable that the FP keeps this delay as low as possible, preferably < 40 ms.

Measurement Method:

The test signal is a CS-signal complying with ITU-T Recommendation P.501[38] using a random noise part with a length of at least 500 ms and shall be applied to the PCM interface of the IP reference interface (see figure 37) with a level of -16 dB<sub>m0</sub> (nominal network level).

To ensure that there is no negative effect from the adaption of the jitter buffer or other adaptive signal processing blocks in front of each measurement could be an additional adaption phase with real speech signal.

The delay is determined by calculating the cross-correlation function between the signal applied to the IP reference interface and the signal measured at the PCM interface of the reference PP. The parameters of the cross-correlation analysis need to ensure that a delay of 500 ms can be determined. The delay is expressed in ms and determined from the maximum of the cross-correlation function. It shall be compensated by the delay introduced by the IP reference interface, the reference PP and the measurement interface in order to derive the receiving delay of only the FP.



**Figure 37: Receive delay of narrowband FP**

Note: In receive direction it is necessary to insert an jitter buffer and packet loss concealment. Due to this additional buffers the maximum allowed delay is longer than for the send direction.

## 2.9.5 Quality of Packet loss concealment

The wideband one-way listening speech quality of the FP in receiving direction - expressed by the MOS-LQOW - shall be determined for the network conditions listed in table 60. The MOS-LQOW for non-ideal network conditions shall not deviate more than specified in table 60 from the MOS-LQOW determined under ideal network conditions.

Condition	Packet Loss (Equal)	Delay Variation	$\Delta$ MOS-LQOW requirement (see Note 3)
1	0	No	---
2	1%	No	$\leq 0.3$
3	2%	No	$\leq 0.5$
4	3%	No	$\leq 0.6$
5	5%	No	$\leq 0.9$
6	1%	20 ms (see note 1)	$\leq 0.6$
<p>NOTE 1: Delay Variation produced with a Pareto-Distribution and <math>r = 0.5</math>.</p> <p>NOTE 2: For some network emulation tools, it is necessary to introduce a constant delay to offer the possibility to generate a delay variation distribution. This delay has to be subtracted from the measured delay before interpreting the results.</p> <p>NOTE 3: The <math>\Delta</math>MOS-LQO requirements are derived from the average MOS-LQO results determined for VoIP gateways during the 5<sup>th</sup> ETSI Speech Quality Test Event in 2008. 4x8 German test sentences were used as test signals (2 sentences of 8 male and 8 female speakers, averaging per speaker).</p>			

**Table 60: Wideband receiving listening speech quality for impaired network conditions**

#### Measurement Method:

A real speech signal consisting of eight sentences (1 language, 2 male and 2 female speaker each uttering 2 sentences) and providing a speech activity of approximately 50% is applied to the PCM interface of the IP reference interface. Appropriate speech signals can be found in ITU-T Recommendation P.501 [38]. The active speech level acc. to ITU-T Rec. P.56 [30] shall be -16 dB<sub>m0</sub>, measured at the PCM interface.

To ensure that there is no negative effect from the adaption of the jitter buffer or other adaptive signal processing blocks in front of each measurement could be an additional adaption phase with real speech signal.

For each network condition given in table 60 the one-way listening speech quality is calculated from the original test signal and the signal received at the PCM interface of the reference PP and is expressed as MOS-LQOW determined by the PESQ (wideband) objective model acc. to ITU-T Rec. P.862.2 [41,42].

For the packet loss conditions the test laboratory shall ensure that the loss does not deviate by more than  $\pm 10$  % from the required packet loss rate during a test.

Furthermore a sufficient randomization of the lost packets shall be guaranteed. This can be achieved e.g. by averaging at least four results of the transmitted speech sequence described above. The averaging should be speaker based in this case. Any averaging process shall be documented.

Note: This measurement is currently under evaluation and improvement at ETSI STQ. So it might be changed in a future release of this document.

Note: There are two possible approaches to handle the Packet Loss Concealment, either in FP or in the PP. Here the assumption is that the PLC is implemented in the FP, a general strategy is currently in evaluation, so the measurement may be changed in later versions.

## 2.9.6 Clock accuracy

The clock drift between the DUT FP and the IP reference interface shall be less than 40 ppm under ideal network conditions.

#### Measurement Method:

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyze clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1.2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level at the PCM interface of the IP reference interface shall be -16 dBm0. The transmitted signal is measured at the PCM interface of the reference PP analyzing the receiving behavior of the FP under test.

A cross correlation analysis versus time is carried out over the whole 120s sequence between the transmitted and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock drift within an analysis time range of at least 60 s is calculated as follows:

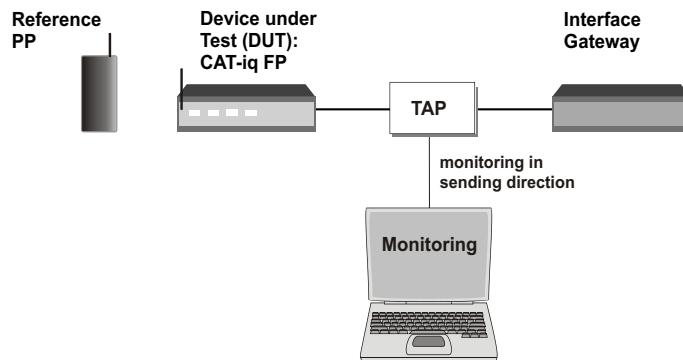
$$\text{clock drift [ppm]} = \frac{\text{delay drift [ms]}}{\text{analysis duration [s]}} \cdot 1 \cdot 10^6$$

## 2.9.7 Send Jitter

The measured interarrival jitter in sending direction of the FP under test should be less than 20 ms. Note: Any jitter introduced in sending direction will lead to potentially increased delay due to increased de-jitter buffer at the far end terminal.

Measurement Method:

The RTP data stream in sending direction should be monitored with a tap or a switch providing a monitoring port, see figure 38.



**Figure 38: Monitoring of send jitter**

The monitoring time should be greater than 60s. A speech signal according P.50 [28] should be played back in sending direction using a nominal network level of -16 dB<sub>m0</sub> at the PCM interface of the measurement interface. The estimate of the statistical variance of the RTP data packet interarrival time is measured in timestamp units.

The interarrival jitter J is defined to be the mean deviation of the difference D in packet spacing at the receiver (IP reference interface) compared to the sender (DUT FP) for a pair of packets.

The mean deviation of the difference in packet spacing at the receiver is

$$D(i,j) = (R_j - S_j) - (R_i - S_i)$$

With  $S_i$  = the RTP timestamp for packet i  
 $R_i$  = the time of arrival

The interarrival jitter is calculated continuously as each data packet i is received from the source (DUT FP) using this difference D for that packet and the previous packet i-1 in order of arrival, according to the formula

$$J(i) = J(i-1) + (|D(i-1,i)| - J(i-1)) / 16 \text{ and } J(0)=0 \text{ ms}$$

## 3 Annex A

### 3.1 Measurement Summary

	Audio profile in ETSI EN 300 176-2	Nb of measurements	Testing time	List of measurements	Status	Comments
PP Group A : Handset and headset in narrowband mode	PP Type1c, 1d	31	TBC	Table 2	Mandatory for handset & headsets	Some measurements were removed from 176-2
PP Group B: Handset and headset in wideband mode	PP Type 2b and 2c	29	TBC	Table 19	Mandatory for handset & headsets	Some measurements were removed from 176-2
PP Group C: Narrowband handsfree	PP Type 3a and 3b	23	TBC	Table 34	Mandatory for handset	Some measurements were removed from 176-2
PP Group D: Wideband handsfree	PP Type 4a and 4b	21	TBC	Table 45	Mandatory for handset	Some measurements were removed from 176-2
FP Group A: VoIP narrow band mode	FP Type 3	11	TBC	Table 55	Mandatory for base station for CAT-iq 2.1, not for 2.0	Some measurements not specified in EN300 176-2.
FP Group B: VoIP wideband mode	FP Type 5	10	TBC	Table 58	Mandatory for base station for CAT-iq 2.1, not for 2.0	Some measurements not specified in EN300 176-2.
TOTAL :		125				

### 3.2 Accuracy of test equipment

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

Item	Accuracy
Electrical Signal Power	$\pm 0.2$ dB for levels $\geq -50$ dBm
Electrical Signal Power	$\pm 0.4$ dB for levels $< -50$ dBm
Sound pressure	$\pm 0.7$ dB
Time	$\pm 0.2$ %
Frequency	$\pm 0.2$ %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Quantity	Accuracy
Sound pressure level at MRP	$\pm 3$ dB for 100 Hz to 200 Hz $\pm 1$ dB for 200 Hz to 4 kHz $\pm 3$ dB for 4 kHz to 8 kHz
Electrical excitation levels	$\pm 0.4$ dB across the whole frequency range
Frequency generation	$\pm 2$ % (see note)
Time	$\pm 0.2$ %
NOTE : This tolerance may be used to avoid measurements at critical frequencies, e.g. those due to sampling and coding operations within the terminal under test.	

### 3.3 Bandwidth for measurements in receive directions

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The receive sensitivity of the HATS is defined for the frequency range from 100Hz to 10kHz [32], so all acoustic measurements in receive direction shall be band-limited to this frequency range.

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## 4 Annex B (informative): practical guidelines for accoutic measurements

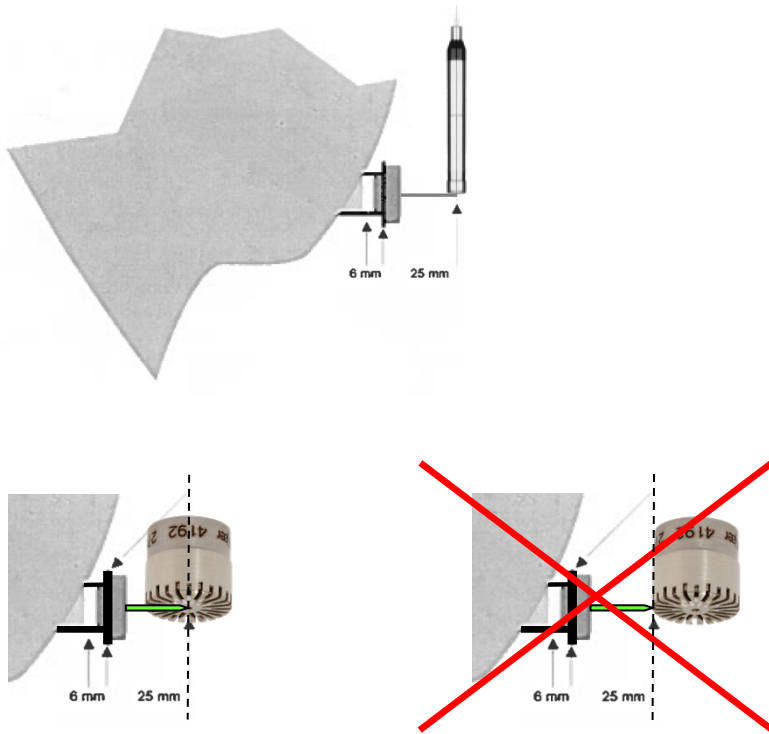
The following guidelines should be understood as good practice guidelines when performing audio measurements.

### 4.1 Handset measurements

#### 4.1.1 Mouth calibration

HATS mouth calibration is an important step before measuring devices. Care must be taken to ensure correct positioning of the reference microphone in front of the artificial mouth

The center of the Reference microphone for calibration must be located at MRP, which is 25 mm in front of the Lip Plane of the HATS :



Note : for example, if the side of a ½" Reference microphone is located at MRP instead of centre of microphone, this 6mm deviation leads to a 2,5 dB error on calibration, and thus leads to an error on level measurements results – especially SLR-.

#### 4.1.2 Mouth calibration selection on the bench

For measurements systems which support several mouth calibration profiles, this may be a possible source of mistakes. As a consequence :

- It is advised to use only one mouth calibration and definition in the system.
- If several mouth calibration profiles method is used, it is recommended to perform a “quick check” of the calibration at the beginning of the measurement session to make sure the correct calibration is used for the session. This is done via the dedicated menu of the bench. This check is based on a measurement of the mouth + equalization.

#### 4.1.3 Device positioning



If position is NOT the standard one as defined in ITU-T Recommendation P.64 [33], Supplier should declare the exact positioning of the device for measurements. The following tables may be used for this purpose :

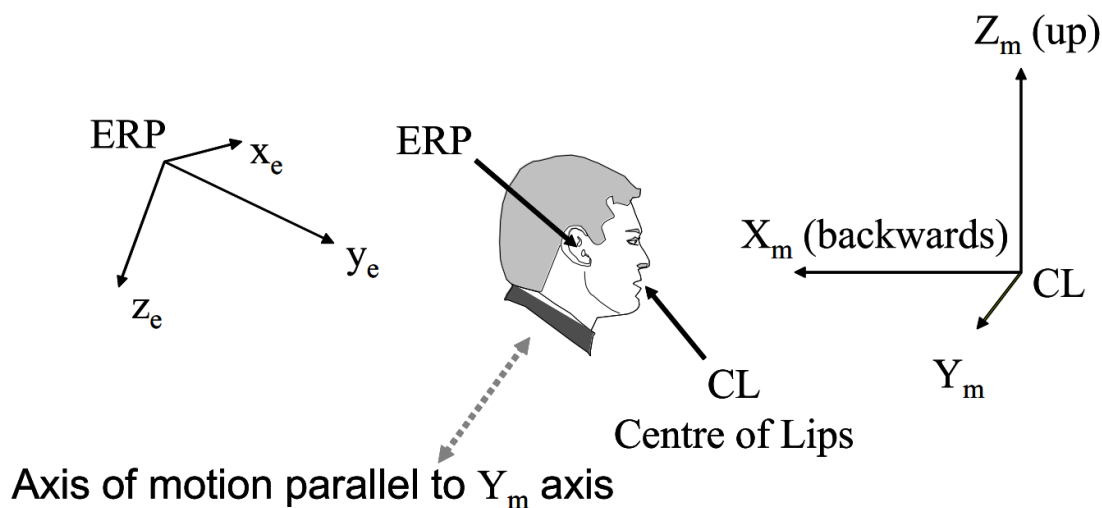


Illustration of coordinates defined in ITU-T P.64, Annex E (Source: IEEE 269)

MECRP stands for Manufacturer defined Ear Cap Reference Point

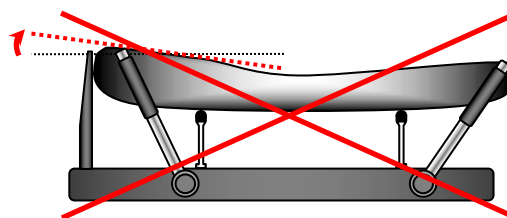
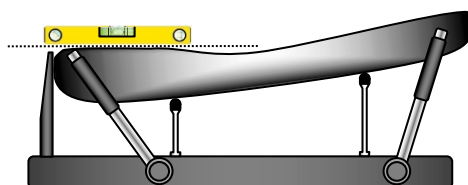
MECRP (Delta to actual ECRP)	
Axis	Delta [mm]
Ye	
Ze	

Angle Settings (Delta to Standard Position 0°)	
Angle (Axis)	Delta [°]
A ( $X_e$ )	
B ( $Z_e$ )	
C ( $Y_e$ )	

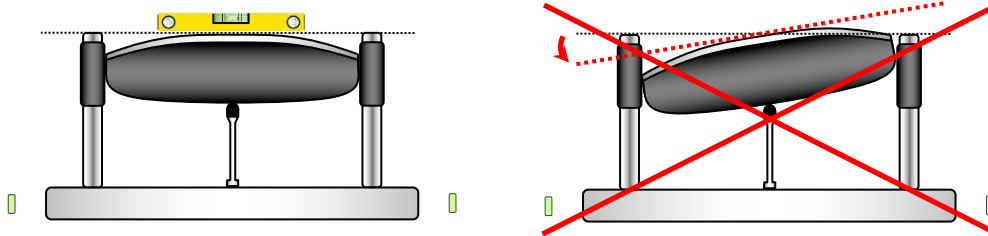
ERP Position	
Axis	Delta [mm]
Ym	

Depending of the shape of the device and acoustic components (earpiece), Receive acoustic measurement results may be very sensitive to the device positioning on HATS.

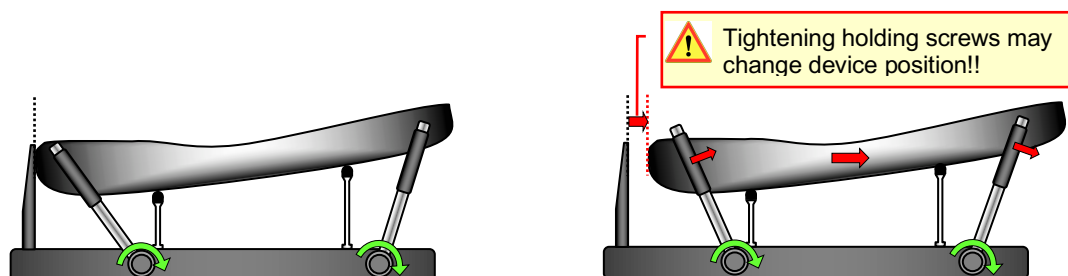
Care must be taken to ensure a correct horizontality of the device earpiece area in the holder, in the two plans :  
Length axis



Width axis :



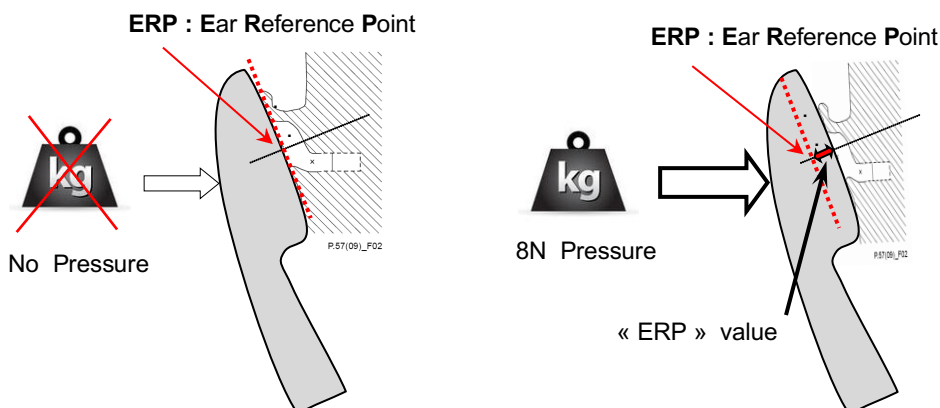
Make also sure to keep proper device positioning when tightening up holding screws



#### 4.1.4 Comparing handset measurements between two labs

When applying the standard 8N pressure on HATS, the HATS “ERP value” indicates the distance between ERP (Ear Reference Point) and device earpiece. This parameter ensures same positioning of the device whatever the HATS used.

Manufacturer should inform the certification lab about the “ERP value” along with other positioning information.



Note : as a consequence of this statement, it was observed that with different test bench manufacturers, the same ERP value was reached with +9,2 N pressure instead of 8N pressure.

#### 4.1.5 Correct handset pressure on HATS

The ear may physically re-act to the pressure applied on the handset. As a consequence, it is necessary to check that the applied pressure is still the correct one, a few seconds after the device was positioned on the HATS

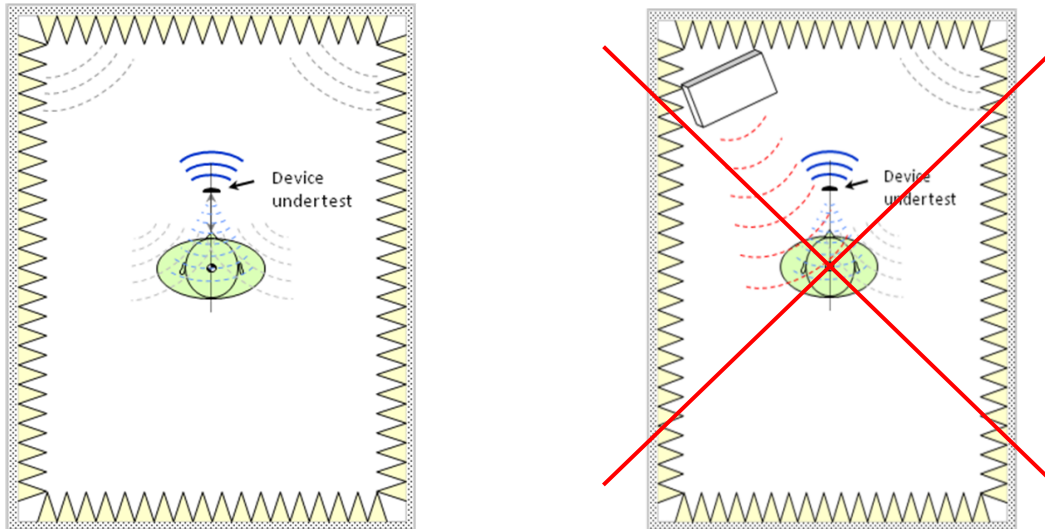
Note : for example on some benches it is necessary to set initially 8,2N in order to finally reach 8N

Consistently with the previous clause, it is a good practice to check the ERP value along with the pressure value.

## 4.2 Handsfree measurements

### 4.2.1 Room effect

Care must be taken that the measurement room keeps its anechoic characteristics i.e without too many furniture. For example superfluous tables/chairs in the room may impact measurements, especially for receiving side, where loudspeaker is playing sound on the back side of device. In this configuration, few direct sound comes to the HATS, and reflections on furniture introduces a bias on result.

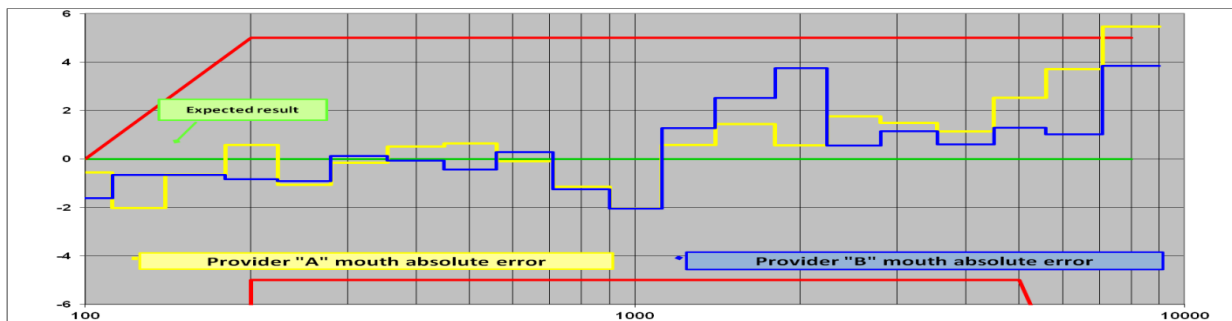


### 4.2.2 Mouth calibration

In handsfree mode, the mouth calibration is made in 2 steps, as defined in ITU-T Recommendation P.581 [40].  
Step1 : calibration at MRP. Same care as for handset measurements must be taken (see clause 4.1.1).  
Step2 : calibration at HFRP (Handsfree Reference point). Care must be taken that handsfree sending measurements are influenced by the manufacturer bench .See clause 4.2.3 for details.

### 4.2.3 Influence of the manufacturer bench (frequency response of the mouth at HFRP)

As mouth calibration at HFRP (Step 2) only consists in a global level alignment, the effective frequency response of the measurement signal (mouth frequency response at HFRP) may differ from a bench provider to another. As an example, the following figure shows the frequency response of a reference Free Field microphone measured at HFRP with two different bench providers :



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More than 3 dB difference at 2 kHz 1/3 Octave can be observed on a frequency response, and could lead to a FAIL result.

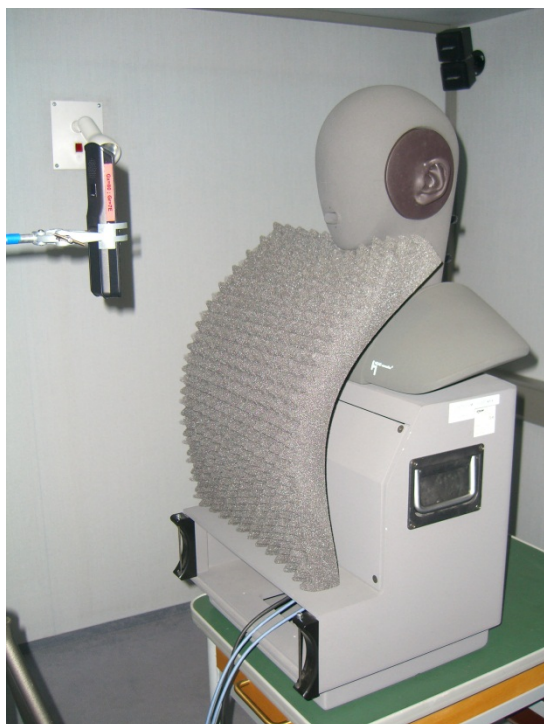
Studies have shown that the HATS chest depth could have an influence on the Frequency response of Devices under test in handsfree modes, by adding an acoustic signal component generated by reflexion on the HATS torso.

This may lead to a difference of up to 4 dB on 1 kHz and 2 kHz frequencies levels, when using min and Max HATS chest depth defined in ITU-T Recommendation P.58 [32].

As a consequence, a device can be declared as "Pass" in one case, and "False" in the other one, even if these two cases are compliant with ITU-T Recommendation P.58 [32].

This fact is under study at ITU-T, COM12, a contribution has been received (C0114 E) on this topic.

During this ITU-T study period, in order to cancel reflecting troubleshot for measurement, it is allowed to perform handheld hands free measurements by covering HATS chest with an absorber material, as shown in the following picture :



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## 5 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

- [1] ETSI EN 300 175-1: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 1: Overview".
- [2] ETSI EN 300 175-2: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 2: Physical layer (PHL)".
- [3] ETSI EN 300 175-3: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 3: Medium Access Control (MAC) layer".
- [4] ETSI EN 300 175-4: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 4: Data Link Control (DLC) layer".
- [5] ETSI EN 300 175-5: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 5: Network (NWK) layer".
- [6] ETSI EN 300 175-6: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 6: Identities and addressing".
- [7] ETSI EN 300 175-7: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 7: Security features".
- [8] ETSI EN 300 175-8: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".
- [9] ETSI EN 300 176-1: "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 1: Radio".
- [10] ETSI EN 300 176-2: Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 2: Audio and speech
- [11] ETSI EG 202 396-1: "Speech Processing, Transmission and Quality Aspects (STQ); Speech quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database".
- [12] ETSI EG 202 518: "Speech Processing, Transmission and Quality Aspects (STQ); Acoustic Output of Terminal Equipment; Maximum Levels and Test Methodology for Various Applications".
- [13] ITU-T Recommendation G.111 (1993): "Loudness ratings (LRs) in an international connection".
- [14] ITU-T Recommendation G.122 (1993): "Influence of national systems on stability and talker echo in international connections".
- [15] ITU-T Recommendation G.223 (1988): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".
- [16] Void.
- [17] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".
- [18] ITU-T Recommendation G.712: "Transmission performance characteristics of pulse code modulation channels".
- [19] ITU-T Recommendation G.722: "7 kHz audio-coding within 64 kbit/s".
- [20] ITU-T Recommendation G.722 (Appendix III): "A high quality packet loss concealment algorithm for G.722".

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- [21] ITU-T Recommendation G.722 (Appendix IV): "A low-complexity algorithm for packet loss concealment with G.722".
  - [22] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".
  - [23] ITU-T Recommendation G.729.1: "G.729 based Embedded Variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".
  - [24] ITU-T Recommendation G.1020: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".
  - [25] ITU-T Recommendation O.41: "Psophometer for use on telephone-type circuits".
  - [26] ITU-T Recommendation O.132 (1988): "Quantizing distortion measuring equipment using a sinusoidal test signal".
  - [27] ITU-T Recommendation O.133 (1993): "Equipment for measuring the performance of PCM encoders and decoders".
  - [28] ITU-T Recommendation P.50 (1999): "Artificial voices".
  - [29] ITU-T Recommendation P.51 (1996): "Artificial mouth".
  - [30] ITU-T Recommendation P.56: "Objective measurement of active speech level".
  - [31] ITU-T Recommendation P.57 (2002): "Artificial ears".
  - [32] ITU-T Recommendation P.58: "Head and torso simulator for telephony".
  - [33] ITU-T Recommendation P.64 (1999): "Determination of sensitivity/frequency characteristics of local telephone systems".
  - [34] ITU-T Recommendation P.79 (1999): "Calculation of loudness ratings for telephone sets".
  - [35] ITU-T Recommendation P.311: "Transmission characteristics for wideband (150-7000 Hz) digital handset telephones".
  - [36] ITU-T Recommendation P.340: "Transmission characteristics and speech quality parameters of hands-free terminals".
  - [37] ITU-T Recommendation P.380: "Electro-acoustic measurements on headsets".
  - [38] ITU-T Recommendation P.501: "Test signals for use in telephony".
  - [39] ITU-T Recommendation P.502: "Objective test methods for speech communication systems using complex test signals".
  - [40] ITU-T Recommendation P.581: "Use of head and torso simulator (HATS) for hands free terminal testing".
  - [41] ITU-T Recommendation P.862: "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs"
  - [42] ITU-T Recommendation P.862: "Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs"
  - [43] ISO 3 (1973): "Preferred numbers - Series of preferred numbers".
  - [44] IEC 61260: "Electroacoustics - Octave-band and fractional-octave-band filters".
  - [45] ISO 9614 (all parts): "Acoustics - Determination of sound power levels of noise sources using sound intensity".

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- [46] ISO/IEC JTC1/SC29/WG11 (MPEG): "International Standard ISO/IEC 14496-3/AMD 1:2007: "Information Technology - Coding of audio-visual objects - Part 3: Audio; AMENDMENT 1: Low Delay AAC profile".
  - [47] ISO/IEC JTC1/SC29/WG11 (MPEG): International Standard ISO/IEC 14496-3:2005: "Information Technology - Coding of audio-visual objects - Part 3: Audio".
  - [48] ETSI TBR 038: "Public Switched Telephone Network (PSTN); Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe".
  - [49] ETSI EN 300 700: "Digital Enhanced Cordless Telecommunications (DECT); Wireless Relay Station (WRS)".
  - [50] ETSI I-ETS 300 245-3: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 3: Pulse Code Modulation (PCM) A-law, loudspeaking and handsfree telephony".
  - [51] Directive 2006/95/EC of the European Parliament and of the Council of 12 December 2006 on the harmonisation of the laws of Member States relating to electrical equipment designed for use within certain voltage limits (codified version).
  - [52] ITU-T Recommendation G.191: "Software tools for speech and audio coding standardization".
  - [53] ITU-T Recommendation G.726 (Appendix II): "Digital test sequences for the verification of the G.726 40, 32, 24 and 16 kbit/s ADPCM algorithm".
  - [54] ITU-T Recommendation G.722 (Appendix II): "Digital test sequences for the verification of the G.722 64 kbit/s SB-ADPCM 7 kHz codec".
  - [55] ITU-T Recommendation G.729.1 (Amendment 1): "New Annex A on G.729.1 usage in H.245, plus corrections to the main body and updated test vectors".
  - [56] ITU-T Recommendation P.360: "Efficiency of devices for preventing the occurrence of excessive acoustic pressure by telephone receivers".