Global Test Specification for Terminals for Performance Measurements (Performance TST)

Acoustical Quality Evaluations of Terminals

VFTST_1.0_Audio Quality_V1.0

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Classification

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Since the purpose of the document is to define a measurement procedure for performance figures, it is herby granted to unveil this document to such terminal suppliers, in case an appropriate NDA is in place and a business relation is existing or planned to become established in respect to launch a terminal in one or more of Vodafone operational companies.

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1 Document Information

1.1 Scope

Scope of the paper is to describe measurement methods for determining terminal audio quality.

This specification defines acoustic test procedures and requirements for mobile handset terminals, headsets and hands-free telephones for quality evaluations in connection with the Vodafone network.

The measurements are conducted with 2G mobiles as well as with 3G mobiles. The measurements are carried out in a simulated GSM and WCDMA network respectively as well as in the real Vodafone D2 GSM and UMTS network respectively. In addition for both types of mobiles tests are conducted using the GSM AMR codec.
In order to reproduce realistic conditions for the users the tests are carried out via the acoustical interface. This guarantees results with high correlation to end to end speech quality assessment. In addition to standard parameters in telephonometry advanced tests are described to cover the subjectively relevant parameters double talk performance and background noise transmission.

The test description for hand-free telephones in this specification refers to vehicle mounted devices. All tests are designed for narrow-band signals.

(Note1: In general the tests can also be applied for desktop hands-free telephones or handheld hand-free telephones but the exact definition of specific tests, measurement conditions and procedures is not yet covered by this specification)

(Note 2: In general all tests can also be applied for are wide-band realisations but the exact definition of specific tests and procedures is not covered by this specification)

1.2 Document History

Version	Date	Editor	Remarks
1.0	15.08.2005	Thomas Ziolkowsky	Initial draft out of Test description Test Procedures for Acoustical Quality Evaluations of Terminals Rel. 3.0

1.3 Acceptance Criteria

The complete set of Vodafone global acceptance criteria is described in a separate document. Below an excerpt is given in respect to criteria related to performance KPI's:

Common Acceptance level for Technical Acceptance is:
- Zero (0) severity 1 faults
- Less than 50 severity 2 faults

Definition of Severity 1 Faults

A critical problem that makes the product unusable or unable to achieve service based on one or more of the following conditions:
- KPI under specification by more than 25%

Definition of Severity 2 Faults

Any fault which has less severe impact on service usability can be accepted for launch, but requiring resolution in the next maintenance release.

- KPI under specification by more than 10%

2 Common Settings and Set Up

2.1 General Procedures

2.1.1 Test setup / environment

The measurement should be done by qualified laboratories.

The general access to terminals, handsets, headsets and hands-free telephones is realised via the acoustical interface by using HATS (head and torso simulator as described through ITU-T Recommendation P.58 [1]). This acoustical interface is the most realistic simulation of the "average" subscriber and therefore suitable to be used for the tests.

During the tests with handsets and hands-free telephones the HATS is equipped with an artificial mouth according to ITU-T Recommendation P.58 [1] and a type 3.4 artificial ear as described in ITU-T Recommendation P.57 [2]. An appropriate positioning system for handset terminals guarantees a reproducible, realistic and pressure force dependent position of the handset relative to the artificial ear. The use of different pressure forces during the measurements between the handset and the artificial ear is specified in the detailed test descriptions below. This set-up guarantees the realistic influence of the acoustical leakage between handset and artificial ear.

During the tests of headsets (mainly Bluetooth headsets) the HATS is equipped with a type 3.3 artificial ear according to ITU-T Recommendation P.57 [2]. The headset is positioned as described in its manufacture's manual. For documentation purposes a photo is taken which shows the positioned headset.

All kinds of terminals are connected to a system simulator during the tests. Exactly defined settings and access points for the simulator are described below (chapter 4.7).

Depending on the type of mobile the measurements are either done with the simulated GSM or the simulated UMTS (WCDMA) network.

In addition two tests are conducted via the real Vodafone D2 network (GSM or UMTS accordingly). The measurement access for this tests is described below (chapter 4.8).

The test sequences are fed in either, electrically using different reference codecs or acoustically using the artificial mouth of HATS [1].

2.1.2 Set-up for Handset Terminals

Test environment: The handsets are measured in an anechoic room. A 4 loudspeaker or 8 loudspeaker arrangement is used for background noise playback for specific tests.

Position: The handset is placed in the HATS position as described in ITU-T Recommendation P.64 [7]. For flat handsets the "alternative position" according to ITU-T Recommendation P.64 is used.

Artificial mouth: The artificial mouth shall conform with ITU-T Recommendation P.58. The artificial mouth is equalised at the MRP according to [1].

Artificial ear: The artificial ear shall conform with ITU-T Recommendation P.57 [2] type 3.4 ear. According to ITU-T Recommendation P.57 DRP-ERP correction is applied [2].

2.1.3 Set-up for Headset Terminals

Test environment: The headsets recommended to be used with a specific phone are measured with this phone in an anechoic room. All Bluetooth headsets are paired with a reference-mobile (Sony Ericsson K700i). A 4 loudspeaker or 8 loudspeaker arrangement is used for background noise playback for specific tests.

Position: The Headsets are mounted to a type 3.3 artificial ear according to ITU-T Recommendation P.58. The exact positioning is according to normal use as recommended by the manufacturer.

For Bluetooth headsets the appropriate mobile is positioned in a distance of 1 m beside the HATS at the same height as the artificial ear, see figure 1.

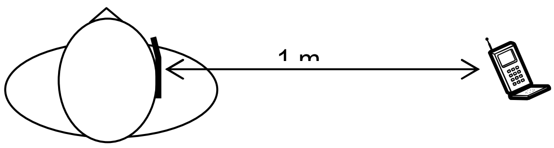


Fig. 1: Test set-up for Bluetooth headsets

In addition some measurements are repeated with the reference mobile positioned on the other side of the HATS, see figure 2.

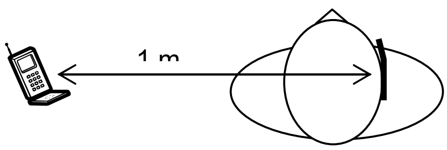


Fig. 2: Additional test set-up for Bluetooth headsets ("shadowed")

In the report the measurements with this setup are marked as "shadowed".

Artificial mouth: The artificial mouth shall conform with ITU-T Recommendation P.58. The artificial mouth is equalised at the MRP according to [1].

Artificial ear: The artificial ear shall conform with ITU-T Recommendation P.57 [2] type 3.3 ear. According to ITU-T Recommendation P.57 the DRP-ERP correction is applied [2].

2.1.4 Set-up for Vehicle Mounted Hands-free Terminals

Test environment: The transmission characteristics for vehicle mounted hands-free telephones are determined in a real vehicle cabin. To guarantee a reproducible and realistic playback of typical vehicle interior background noise a sound simulation system of a driving simulator is used.

Position: The hands-free telephone with its microphone (or microphones / microphone array) and loudspeaker (or loudspeakers) is positioned in the vehicle cabin according to manufactures guidelines. If no specification of manufactures is given a standard position is used. In this case the microphone is fixed left beside of the review mirror and the loudspeaker in the co-drivers footwell. The loudspeaker is aligned to the HATS on the driver's seat. In case this standard position is used it has to be guaranteed that the position is reproducible for all devices under test.

For hands-free telephones using the vehicle mounted loudspeaker for playback (usually the right front loudspeaker) an additional high quality external loudspeaker (e.g. CANTON PLUS S) is connected. In this case the loudspeaker is positioned in the co-drivers footwell and aligned to the HATS on the driver's seat.

The HATS is fixed on the driver's seat representing an "average" users position. It has to be ensured that the exact position is reproducible for all devices under test.

Artificial mouth: The artificial mouth shall conform with P.58. The artificial mouth is equalised at the MRP according to [1]. In addition an HATS-HFRP correction shall be applied in order to guarantee the correct test signal level at this point in a distance of 50 cm under free-field conditions [13].

Artificial ear: The artificial ears of the HATS shall be free-field equalised for the tests of hands-free telephones.

Measurement microphone: In order to guarantee a higher accuracy for recordings in receiving direction under double talk conditions an additional measurement microphone is used for some tests and positioned near the loudspeaker of the hands-free telephone. The use of this measurement microphone is mentioned in the corresponding tests.

2.1.5 Set-up for Desktop Operated Hands-free Terminals

Measurement methods for desktop operated hands-free telephones are not yet covered by this specification.

2.1.6 Set-up for Handheld Hands-free Terminals

Measurement methods for handheld hands-free telephones are not yet covered by this specification.

2.1.7 Calibration and Equalisation of the Test Set-up

The following preparation has to be completed before running the tests or part of it:

Calibration:

- Acoustical calibration of measurement microphone and microphones of HATS
- Calibration and equalisation of artificial mouth at MRP
- HATS-HFRP calibration according to [13] (in case of hands-free terminal testing)

Equalisation:

- Correct equalisation of HATS:
- linear for handset terminal testing

- free-field equalisation for recordings with real speech samples and MOS-LQO analysis using TOSQA2001 and PESQ via handset terminals and headsets.
- free-field equalisation for all measurements and recordings with real speech samples for hands-free terminals

Reference measurements:

- Reference measurements in sending direction at MRP.
- Reference measurements in receiving direction with an electrical loop at the measurement frontend

2.1.8 Settings for the System Simulation

The system simulator provides the access to the terminals. The DAI is not used for the tests because this interface can not be accessed for terminals. The settings shall be chosen as follows:

- Testing 2G mobiles via GSM:
  - The GSM FR codec is used unless otherwise specified.
  - The receive level shall be higher than –90 dBm.
  - The parameter “Rx qual.” shall be 0 or 1, but shall not exceed 1 to avoid error rates that influence the transmission quality.
  - The parameter “power meas. avg.” shall be adjusted to min. –10 dBm.
  - It has to be ensured that channel 5 and 70 are not used for the tests.
  - If possible for all terminals under test the connection shall be provided by RF in/out connection via antenna (to be checked).

TCH / BCCH	Mid channel (e.g.: ARFCN for GSM 850: 190 GSM: 62, GSM 1800: 710 and GSM 1900: 660)
PLMN	HPLMN (of test system)
RX level	-82 dBm
Output Power, PCL, (TCH) (MS-TX-Lev)	PCL = 7 (29 dBm GSM 850/900) PCL = 1 (28 dBm GSM 1800/1900)
Cell Reselection	NO
Paging interval (DRX or BS_PA_MFRMS or BSP or PAGP)	5 multiframes
No of neighbour cells	0
Cell broadcast	NO
MS DTX	OFF

Tab 1: Common parameter settings for GSM

- Testing 3G mobiles via UMTS:
  - WCDMA
  - Voice Source: Speech Code Low
  - AMR: Active Codec Set: 12.2 kbit/s
  - Dedicated Channel Type: Voice
  - WCDMA Band Selection: Operating Band I

Serving Cell UARFCN (downlink)	10712 (2142.4 MHz) any carrier
Serving Cell UARFCN (downlink) – Region 2	9762 (1952.4 MHz) any carrier
Neighbour Cells on different frequencies	No
Number of neighbours, same frequency	0
Serving Cell Scrambling Code	10
Paging Interval (DRX cycle length) CS	0.640 seconds
Paging Interval (DRX cycle length) PS	1.28 seconds (K=7)
BLER (Block Error Rate) quality target UL /per RAB	Speech 1%, VT 0.3 %
RF standard (average) output power (RACH)	+10 dBm [2)]
RF Receiving Level (DL, typical, urban)	-80dBm

Tab 2: Common parameter settings for WDCMA.

- Testing 2G/3G mobiles with AMR codec:
  - For all types of mobiles (2G / 3G) the AMR codec of the GSM network is used.
  - The following AMR conditions are used (Full Rate FR, Half Rate HR):

FR – 12.2 kBit/s

FR – 4.75 k/Bit/s

HR – 7.4 kBit/s

HR – 4.75 kBit/s

2.1.9 Measurements via Real Network

To determine the objective listening speech quality in the real network the MOS-LQO and the TMOS-value are determined with the TOSQA2001 and the PESQ algorithm during a call via the real Vodafone D2 network.

The call is established between the mobile under test (either GSM or UMTS) and an ISDN-Tester. The ISDN-Tester can either be used as a regular ISDN phone or as a measurement access. For the measurements it is used as measurement access which allows – similar to the system simulator – signal feeding from and to the HEAD acoustics measurement frontend.

2.2 Test Signals and Signal Levels

Due to the coding of the speech signals, appropriate test signals are defined in ITU-T Recommendation P.50 [5] and P.501 [4]. Detailed information can be found in the test procedures described below. For narrow band terminals band limited Composite Source Signals as defined in ITU-T Recommendation P.501 [4] shall be used when measuring the receiving direction. Otherwise the test signal used shall be band limited between 200 Hz and 4 kHz with a bandpass filter providing a minimum of 24 dB/oct. filter roll off, when feeding into the receiving direction. Wide-band signals are used in sending direction fed via the artificial mouth. If a specific test signal or a test signal combination is used for a test, the sequences are described in detail in the corresponding subclause of this specification. Further information about appropriate test signals are described in ITU-T Recommendation P.50 [5] and P.501 [4] and analysis methods can be found in ITU-T Recommendation P.502 [6].

The test signal levels are referred to the average level of the band limited test signal in receiving direction, averaged over the complete test sequence (including the signal pauses for switched test signals) unless specified otherwise.

The test signals applied in sending direction via the artificial mouth are wide-band Composite Source Signals as specified through ITU-T Recommendation P.501 [4]. Unless specified otherwise the average test signal level in sending direction is averaged over the complete test sequence (including the signal pauses for switched test signals).

The average speech levels for the measurements are

- –16 dB_{m0} in receiving direction (representing average network conditions).
- -4.7 dB_{Pa} in sending direction (representing an average level in telephonometry)

if not stated otherwise.

For tests which require an exact synchronization of the test signals in time domain (e.g. tests to determine switching characteristics between both, send and receive direction channels) special care should be taken to consider the influence of the codec delay in both transmission ways. The codec delay has also to be considered for those analyses where the transmitted signals are referenced to the original test signals, in order to avoid misleading results.

Typical ambient noise scenarios for tests with background noise present at the near end are described in the following table 1. The background noise scenario which is applied during a test is described in the specific test description.

Table 1: Description of noise scenarios

terminal	kind of noise	description	level
handset terminal, headset terminal	university cafeteria	crowded cafeteria, typical noise from dishes and voice babble	63 dBA measured at the position of the HATS
vehicle mounted **hands-free terminal**	constant driving situation	middle class vehicle, 130 km/h, 5^{th} gear	77 dBA measured with the HATS at the drivers seat positions

In general these noise scenarios can also be applied for, desktop operated hands-free terminals and handheld hands-free terminals. In order to compare the test results for the different kinds of terminals it is recommended to perform the tests under comparable (background noise) conditions.

3 Measurement Parameters and Requirements

3.1 Preparation

If not stated otherwise, the levels in receiving direction of the terminals under test are adjusted close to the nominal levels. For vehicle mounted hands-free telephones the level is adjusted to nominal values if not stated otherwise.

The loudness settings of the terminals (e.g. to be read from the display) used for the tests shall be reported.

3.2 Handset Terminals

3.2.1 Delay

3.2.1.1 Delay in Sending Direction

Requirement

The delay in sending direction measured from the mouth reference point (MRP) to the POI (output of the reference speech decoder of the GSM system simulator (SS)), shall be less than: 120 ms

Test

a) The test signal to be used for the measurements shall be a composite source signal (CSS) as described in ITU-T Recommendation P.501. 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 dB_{Pa}, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [7]). The application force used to apply the handset against the artificial ear shall be 8N.

The delay is calculated using the cross correlation function between the signal at the electrical test point and the original test signal. The measurement is corrected by the delay introduced by the test equipment.

c) The delay is expressed in ms, determined from the maximum of the cross correlation function.

3.2.1.2 Delay in Receiving Direction

Requirement

The delay in receiving direction, measured from the POI (input of the reference speech coder of the GSM system simulator (SS)) to the drum reference point (DRP), shall be less than: 100 ms

Test

a) The test signal to be used for the measurements shall be a composite source signal (CSS) as described in ITU-T Recommendation P.501. The test signal level shall be –16 dB_{m0}, measured at the electrical test point. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8 N.

c) The delay is calculated using the cross correlation function between the original test signal at the electrical test point and the signal at the DRP. The measurement is corrected by the delay introduced by the test equipment.

d) The delay is expressed in ms, determined from the maximum of the cross correlation function.

3.2.2 Loudness Ratings

All SLR and RLR values apply up to the POI, measured at the reference access of the SS.

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 8 +/- 3 dB;

RLR = 2 +/- 3 dB.

Where a user controlled receiving volume control is provided, the RLR shall meet the selected nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be less than (louder than) -13 dB.

3.2.2.1 Sending Loudness Rating

Test

a) The test signal to be used for the measurements shall be the Composite Source Signal (CSS) as described in ITU-T Recommendation P.501. 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 dB_{Pa}, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8 N.

c) The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [8], 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.

d) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula 2.1, over bands 4 to 17, using m = 0.175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

3.2.2.2 Receiving Loudness Rating (RLR)

Test

a) The test signal to be used for the measurements shall be the Composite Source Signal (CSS) as described in ITU-T Recommendation P.501. The test signal level shall be –16 dB_{m0}, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 2N, 8N, 13N.

c) The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, 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.

d) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79, formula 2.1, over bands 4 to 17, using m = 0.175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

e) No leakage correction shall be applied.

3.2.3 Sensitivity/Frequency Characteristics

3.2.3.1 Sending Frequency Response

Requirement

The sensitivity/frequency characteristics shall be as follows:

The sending sensitivity frequency response, measured either from the mouth reference point (MRP) to digital interface or from the MRP to the SS audio output (digital output of the reference speech decoder of the SS), shall be within a mask, which can be drawn between the points given in table 2. The mask is drawn with straight lines between the breaking points in table 2 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 2: Sending sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-12	-
200	0	-
300	0	-12
1 000	0	-6
2 000	4	-6
3 000	4	-6
3 400	4	-9
4 000	0	

NOTE: All sensitivity values are expressed in dB on an arbitrary scale.

Test

a) The test signal to be used for the measurements shall be the Composite Source Signal (CSS) as described in ITU-T Recommendation P.501. 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 dB_{Pa}, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8N.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [9] 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.

c) The sensitivity is expressed in terms of dBV/Pa.

3.2.3.2 Receiving Frequency Response

Requirement

The receiving sensitivity frequency response, measured either from the digital interface to the ERP or from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the

ERP, shall be within a mask, which can be drawn with straight lines between the breaking points in table 3 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 3: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-10	-
200	2	-
300		-9
1 000	(see note 2)	-7
3 400		-12
4 000	2	-

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.

Test

a) The test signal to be used for the measurements shall be the Composite Source Signal (CSS) as described in ITU-T Recommendation P.501. The test signal level shall be -16 dB_{m0}, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 2N, 8N, 13N.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [9] 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.

c) The sensitivity is expressed in terms of dBPa/V, referred to the ERP. Information about correction factors are available in ITU-T Recommendation P.57.

3.2.4 Listening Speech Quality - TMOS (TOSQA2001)

3.2.4.1 TMOS in Sending Direction

Requirement

The TMOS, calculated by the analysis method TOSQA2001, shall be ≥ 3.2.

Values higher or equal to 3.5 are explicitly indicated as a “good point” in the report.

Values lower than 3.0 are explicitly indicated as a “bad point” in the report.

Test

a) The test signal used for the TOSQA2001 analysis is real speech (German, wideband). The sequence consists of 8 sentences from two male and two female voices (2 sentences each). The test signal level shall be –4.7 dB_{Pa}, measured at the MRP. The test signal level is determined over the complete sequence as active speech level.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8N.

c) The TMOS is calculated using the TOSQA2001 analysis method. The settings are: “Search fix delay < 250 ms”, “Search variable delay < 62 ms”, “Measurement: electrical” “Compare to: High quality handset”.

d) The TMOS in sending direction is determined for the following test conditions:
- simulated GSM or UMTS network
- GSM network, AMR FR 12.2 kBit/s
- GSM network, AMR FR 4.75 kBit/s
- GSM network, AMR HR 7.4 kBit/s
- GSM network, AMR HR 4.75 kBit/s
- Vodafone GSM network (to ISDN tester)

e) In order to verify if overmodulations or distortions occur with high speech levels the TMOS value in the simulated GSM or UMTS network is also carried out with a 3 dB, 6 dB and 9 dB higher speech level at the MRP.

3.2.4.2 TMOS in Receiving Direction

Requirement

The TMOS, calculated by the analysis method TOSQA2001, shall be ≥ 2.5.

Values higher or equal to 3.0 are explicitly indicated as a “good point” in the report.

Values lower than 2.5 are explicitly indicated as a “bad point” in the report.

Test

a) The test signal used for the TOSQA2001 analysis is real speech (German, IRS send filtered as described in ITU-T Recommendation P.48 [16]). The sequence consists of 8 sentences from two male and two female voices (2 sentences each). The test signal level shall be -16 dB_{m0}, measured at the digital reference point or the equivalent analogue point. The test signal level is determined over the complete sequence as active speech level.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 2N, 8N and 13N. The artificial ear shall be free field equalized as described in subclause “Calibration and Equalisation of the Test Set-up”. The calibration shall be set to 114 dB.

c) For each application force the TMOS is calculated using the TOSQA2001 analysis method. The settings are: “Search fix delay < 250 ms”, “Search variable delay < 62 ms”, “Measurement: acoustical”, “Compare to: High quality handset”.

d) The TMOS in receiving direction is determined for the following test conditions (8N application force):
- simulated GSM or UMTS network
- GSM network, AMR FR 12.2 kBit/s
- GSM network, AMR FR 4.75 kBit/s
- GSM network, AMR HR 7.4 kBit/s
- GSM network, AMR HR 4.75 kBit/s
- Vodafone GSM network (to ISDN tester)

e) In order to verify if overmodulations or distortions occur with high signal levels in the network the TMOS value in the simulated GSM or UMTS network is also measured with a 6 dB higher speech level at the POI (8N application force).

3.2.5 Listening Speech Quality – MOS-LQO (ITU-T Rec. P.800.1) using PESQ (ITU-T Rec. P.862)

NOTE:

The PESQ algorithm is valid only for measurements at electrical interfaces. PESQ-values are determined in receiving direction but do *not* lead to valid results.

3.2.5.1 MOS-LQO in Sending Direction

Test

a) The test signal used for the PESQ analysis is real speech (German, wideband). The sequence consists of 8 sentences from two male and two female voices (2 sentences each). The test signal level shall be –4.7 dB_{Pa}, measured at the MRP. The test signal level is determined over the complete sequence as active speech level.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8N.

Headsets are mounted to a type 3.3 artificial ear. The exact positioning is according to normal use as recommended by the manufacturer.

c) The MOS-LQO is calculated using the PESQ analysis method.

d) The MOS-LQO in sending direction is determined for the following test conditions.
- simulated GSM / UMTS network
- GSM network, AMR FR 12.2 kBit/s
- GSM network, AMR FR 4.75 kBit/s
- GSM network, AMR HR 7.4 kBit/s
- GSM network, AMR HR 4.75 kBit/s
- Vodafone GSM network (to ISDN tester)

3.2.5.2 MOS-LQO in Receiving Direction

In order to compare PESQ results for terminals to measurements in the Vodafone D2 network this analysis is used in this scenario.

Test

a) The test signal used for the PESQ analysis is real speech (German, IRS send filtered as described in ITU-T Recommendation P.48 [16]). The sequence consists of 8 sentences from two male and two female voices (2 sentences each). The test signal level shall be -16 dB_{m0}, measured at the digital reference point or the equivalent analogue point. The test signal level is determined over the complete sequence as active speech level.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 2N, 8N and 13N. The artificial ear shall be free field equalized as described in subclause “Calibration and Equalisation of the Test Set-up”. The calibration shall be set to 114 dB.

c) For each pressure force the MOS-LQO is calculated using the PESQ analysis method.

d) The MOS-LQO in receiving direction is determined for the following test conditions (8N application force):
- simulated GSM / UMTS network
- GSM network, AMR FR 12.2 kBit/s
- GSM network, AMR FR 4.75 kBit/s
- GSM network, AMR HR 7.4 kBit/s
- GSM network, AMR HR 4.75 kBit/s
- Vodafone GSM network (to ISDN tester)

3.2.6 Idle Channel Noise

3.2.6.1 Sending Direction

Requirement

The maximum noise level produced by the apparatus measured at the POI under silent conditions in the sending direction shall not exceed -64 dB_{m0}(P). No peaks in frequency higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: This level includes the eventual noise contribution of an acoustic echo canceller under the condition that no signal is received.

Test

a) 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. 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.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8N.

c) The idle channel noise is measured at the electrical reference point 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. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. 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 spectrum is calculated by FFT (8k/48 kHz sampling rate) using Hanning windowing.

d) The idle channel noise is determined by using psophometric weighting. Spectral peaks are measured in the frequency domain.

3.2.6.2 Receiving Direction

Requirement

The maximum (acoustic) noise level at the handset terminal when no signal (0-level) is received from the speech transcoder shall be as follows:

If no user-controlled receiving volume control is provided, or, if it is provided, using the receiving volume setting which leads to a nominal RLR value, the noise measured at the ear reference point (ERP) contributed only by the receiving equipment shall not exceed -57 dB(A). No peaks in frequency higher than 10 dB above the average noise spectrum shall occur.

Where a volume control is provided, the measured noise shall also not exceed -54 dB(A) at the maximum setting of the volume control. Again no peaks in frequency higher than 10 dB above the average noise spectrum shall occur.

Test

a) 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. The activation signal level shall be –16 dBm0, measured at the electrical access point. The activation signal level is averaged over the complete activation signal sequence.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”. The handset is mounted at the HATS position (see ITU-T Recommendation P.64). The application force used to apply the handset against the artificial ear shall be 8N.

c) The idle channel noise is measured at the ear reference point (ERP) in the frequency range from 50 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 second. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. 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.

d) The noise spectrum is calculated by FFT (8k/48 kHz sampling rate) using Hanning windowing.

e) The idle channel noise is determined by using A-weighting. Spectral peaks in the idle channel noise are measured in the frequency domain.

3.2.7 Ambient Noise Rejection

Requirement

The nature of mobile telephony is such that the terminal will typically be operated in high ambient acoustic noise. Due to the adverse interaction of noise signals with speech codecs operating at lower rates, for example 8kbit/s or less, a minimum noise rejection specification is required.

The terminal ambient noise rejection, calculated as D-value from DELSM (ITU-T Recommendation G.111) shall be greater than or equal to 0 dB. For good performance, it is recommended that a figure of +3 dB should be achieved.

Two tests are carried out using two different kind of ambient noise signals. For Test 1 a pink random noise signal is used and the sound field is calibrated in the absence of any local obstacles. The result is named “D” value. This procedure is often used in standards, but the test signal (noise) is not very realistic for real life use of mobile terminals. Therefore Test 2 is carried out using a more realistic background noise signal (university cafeteria noise as described in table 1) and a different calibration. The sound field is calibrated at the MRP with a mobile terminal mounted to the HATS. In order to guarantee reproducibility for all tests the same mobile terminal has to be used for sound field calibration. This result is named “D real”.

Test

a) Sound field calibration (identical for Test 1 and 2): The diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within ±3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands according to IEC 225 [12] from 100 Hz to 8 kHz (bands 1 to 20). The loudspeakers for generating the noise field are equalized within ±3 dB from 100 Hz to 8 kHz (bands 1 to 20).

NOTE 1: The pressure intensity index, as defined in ISO 9614, may prove to be a suitable method for assessing the diffuse field.

b) NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers may require to be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.Test signal for Test 1 is pink random noise signal. The average level shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance for this level is ±1 dB. The reference measurement is carried out in the absence of any local obstacles.

Test signal for Test 2 is a binaurally recorded background noise signal representing noises at typical user locations. The average level shall be adjusted to 63 dB(A) for the university cafeteria noise (see table 1). The tolerance for this level is ±1 dB. The reference measurement is carried out at the MRP with a mobile terminal mounted to the HATS. This handset is mounted in the same way as the handsets under test as described in subclause “Set-up for Handset Terminals”

c) The handset is mounted as described in subclause "Set-up for Handset Terminals". Measurements are made on one-third octave bands according to IEC 225 [12] for the 14 bands centred at 200 Hz to 4 kHz (bands 4 to 17). For each band the diffuse sound sensitivity S_{si}(diff) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

d) The direct sound sensitivity shall be measured using the test set-up specified in subclause 4.1 (Set-up for Handset Terminals) and a speech like test signal as defined in ITU-T Recommendation P.50 or P.501. The type of test signal used shall be stated in the test report. The direct sound sensitivity is measured in one-third octave bands according to IEC 225 [12] for the 14 bands centred at 200 Hz to 4 kHz (bands 4 to 17). For each band the direct sound sensitivity S_{si}(direct) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

e) The value of the D-factor and the "D real"-factor respectively shall be calculated according to ITU-T Recommendation P.79, annex E, formulas E2 and E3, over the bands from 4 to 17, using the coefficients K_i from table E1 of ITU-T Recommendation P.79.

3.2.8 Acoustic Echo Control

General

The echo loss (EL) presented at the POI should be at least 46 dB during single talk. This value takes into account the fact that terminal is likely to be used in a wide range of noise environments.

The use of acoustic echo control is not mandated and the connection between the terminal and the POI is zero loss. Therefore the acoustic echo control provided in terminal should provide a TCLw of at least 46 dB at the POI over the likely range of acoustic end delays.

If acoustic echo control is provided by voice switching, comfort noise should be injected. This comfort noise shall operate in the same way to that used in DTX.

3.2.8.1 Terminal Coupling Loss TCLw

Requirement

The TCLw for the handset terminal for nominal and maximum speaker volume shall be 46 dB. Careful acoustic design of the handset body and selection of the mouth and ear piece transducers may facilitate the required acoustic echo loss without the need for active echo control techniques. However, the echo cancellation is designed it be capable of dealing with the variations in handset positions when in normal use.

Care should be taken that violations of this requirement are caused by echo and not by comfort noise injection. A corresponding mark shall be given in the report.

Test

The terminal is set-up as described in subclause "Set-up for Handset Terminals". The ambient noise level shall be less than -64 dBPa(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal.

a) 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 is altered. The training sequence level shall be –16 dB_{m0} in order not to overload the codec.

b) The test signal is a PN-sequence complying with ITU-T Recommendation P.501 with a length of 4096 points (for the 48 kHz sampling rate, "white" spectrum from 200 Hz – 4 kHz, crest factor 6 dB). The duration of the test signal is 1 s. The test signal level is –3 dB_{m0}. The low-crest factor is achieved by random-alternation of the phase between -180° and 180°.

c) The TCLw is calculated according to ITU-T Recommendation G.122 [11], annex B, 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 (at least 200 ms).

d) The TCLw shall be determined for a nominal receiving volume (nominal RLR). If a volume control is provided the TCLw should be as well determined for a maximum receiving volume. For the maximum receiving volume the TCLw shall not be less than 46 dB.

3.2.8.2 Echo Level versus Time

Requirement

The measured echo attenuation versus time (single talk) under steady state conditions should not vary for more than 6 dB between periods of the test signal (measured during the active signal parts). This level variation is independent of the average echo attenuation (TCLw) or the spectral echo attenuation.

Test

The test signal consists of a periodical repetition of the composite source signal. The measurement is conducted with two signal levels. The average signal levels are –5 dB_{m0} and -25 dB_{m0}. The echo signal level is analysed for at least 2.8 s, which represents 8 periods of the CSS, using a time 35 ms constant and referenced to the test signal level.

Note: The analysis is carried out only during the active signal parts. The pauses between two bursts of the test signal are not analysed. The analysis range during the active signal parts is reduced in the initial part by the length represented by the time constant.

3.2.8.3 Spectral Echo Attenuation

Requirement

The echo attenuation versus frequency expressed through the difference between the power density spectra of the test signal and the echo signal shall be below the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in the following table 6.

Table 6: Spectral echo attenuation mask

Frequency (Hz)	Upper limit
100	-20
200	-30
300	-38
800	-34
1500	-33
2600	-24
4000	-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.

Test

a) The test signal consists of a periodical repetition of the composite source signal. The measurement is carried under steady state conditions with an average test signal level of -16 dB_{m0}.

b) The power density spectrum of the echo signal is averaged over a sequence length of 1.4 s (i.e. 4 CSS bursts including the test signal pauses) and referenced to the power density spectrum of the original test signal.

c) The echo loss is expressed in dB.

d) If comfort noise injection is implemented and this implementation leads to a violation of the echo attenuation mask, a corresponding remark shall be given in the report. Tests to detect comfort noise injection are described in subclause 6.2.11 “Background noise transmission”.

3.2.9 Switching Characteristics

3.2.9.1 Activation in Sending Direction

General

The activation in sending direction is determined by the two parameters

- minimum activation level $L_{S,min}$ and
- build up time $T_{r,S,min}$.

The minimum activation level corresponds to the test signal level necessary to remove the inserted loss in sending direction in idle mode. The build up time is determined for the test signal burst which is applied with the minimum activation level.

Requirement

The minimum activation level $L_{S,min}$ shall be less or equal to -20 dB_{Pa}, i.e. 17 dB below the nominal level of -3 dB_{Pa} (active signal part).

Note: This active signal level corresponds to an average signal level of -4.7 dB_{Pa} at the MRP assuming a pause of 101.38 ms between two CSS bursts.

The build up time shall be less or equal to 50 ms.

Test

The signal structure shown in Fig. 3 represents signal parts with increasing levels. Periods of the CSS (as a simulation of speech) with increasing levels are suited for this signal.

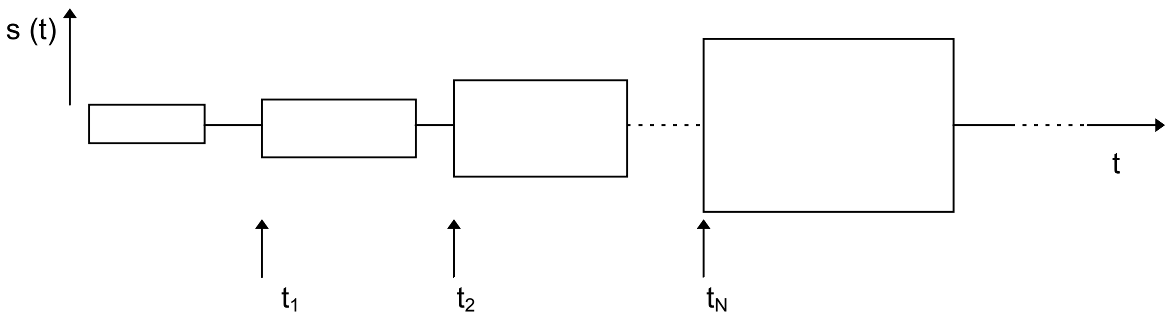


Fig. 3: Structure of test signal to determine the minimum activation level and build up time

Note:

The dotted line indicates the repetition or elongation of the test signal to achieve the suitable length for the measurement

The settings for the test signal in sending direction are as follows:

	Active duration / pause duration	level of the first period (active signal part, at the MRP)	Level difference between two periods
CSS for switching in **Sending direction**	248.62 ms / 451.38 ms	-23 dB_{Pa} *)	1 dB

*) Note: This active signal level corresponds to an average signal level of -24,7 dB_{Pa} at the MRP assuming a pause of 101,38 ms between two CSS bursts.

It is assumed that the pause length of 451.38 ms is longer than the hangover time so that the test object mode is the idle mode after each signal burst independent if it was activated or not.

The level of the transmitted signal is measured at the POI and referred to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The minimum activation level is determined from the resulting curve and the build up time is analyzed at the beginning of the corresponding signal burst (t_1, t_2, ..., t_N in Fig.3).

3.2.9.2 Attenuation range in sending direction

General

The attenuation range is determined by applying a test signal in sending direction after the terminal was activated in opposite direction (receiving direction). Two parameters can be determined

- attenuation range $a_{H,S}$ (complete inserted loss) and
- the build up time $T_{r,S}$ (until the output level reaches 3 dB below its final value)

Requirement

The attenuation range $a_{H,S}$ shall be ≤ 20 dB.

The build up time $T_{r,S}$ shall be ≤ 50 ms (recommended ≤ 15 ms except a residual value of 13 dB below the final value).

Test

The signal structure is given through Fig. 4. Periods of the CSS (as a simulation of speech) and voiced sounds are used for this signal. The signals on both channels are uncorrelated.

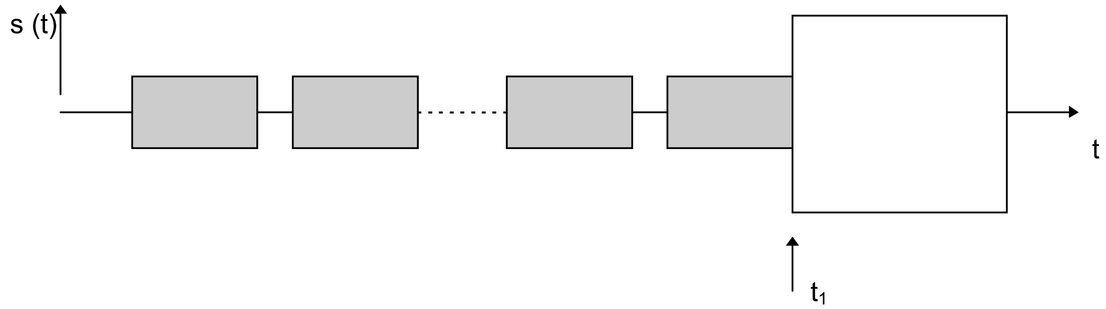


Fig. 4: Structure of test signal for attenuation range measurement

Note: The dotted line indicates the repetition or elongation of the test signal to achieve the suitable length for the measurement

A periodical repetition of CSS bursts as a simulation of speech is used to activate one transmission path (grey colour). In case of the attenuation range measurement in sending direction, the receiving direction is activated first. At the end of the voiced part of the CSS, the measurement signal is applied in the opposite path (white colour). This signal part consists of a periodical repetition of a voiced sound.

The signal settings are as follows:

	receiving direction	sending direction (at the MRP)
average signal level (incl. 101,38 ms pauses)	-16 dB_m	---
active signal parts	-14,7 dB_m	-3 dB_{Pa}

The level of the transmitted signal is measured at the POI and referred to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve and the build up time is analyzed starting from the beginning of the voiced activation signal in sending direction (t_1 in Fig. 4).

3.2.10 Double Talk Performance

General

In duplex conditions the system performance is determined by mostly two parameters: talker echo loudness rating and level variation compared to the single talk condition (attenuation range). In order to achieve a auditory perceived high quality in double talk conditions the double talk echo attenuation should be high and the inserted loss in sending and receiving direction (due to level switching) should be low. For terminals which are not intended to provide double talk capabilities the total echo loss may be achieved by a correspondingly high echo attenuation based on level switching.

The corresponding parameters which determine the double talk performance are

- attenuation range in sending direction during double talk $a_{H,S,dt}$
- attenuation range in receiving direction during double talk $a_{H,R,dt}$
- echo attenuation during double talk

In addition to these analyses speech recording under double talk conditions can be used to judge double talk performance.

3.2.10.1 Attenuation range in sending direction during double talk

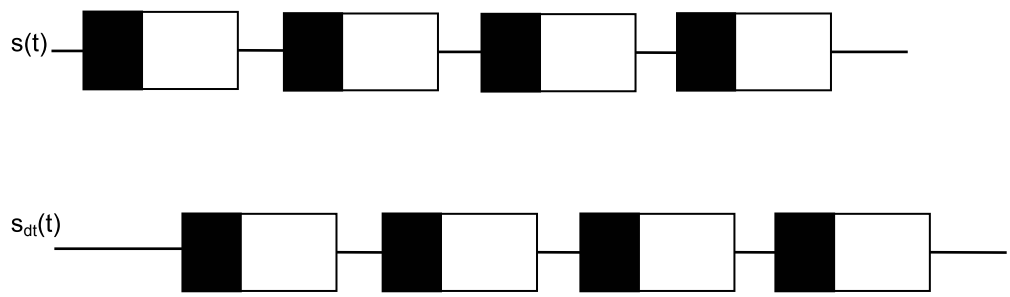
Requirement

Based on the level variation in sending direction during double talk determined by the parameter $a_{H,S,dt}$ (attenuation range in sending direction during double talk) the behaviour can be distinguished according to the following table:

Behaviour	***1***	***2a***	***2b***	***2c***	***3***
a_{HSdt} [dB]	*≤ 3*	*≤ 6*	*≤ 9*	*≤ 12*	*>12*

Test

The test signal used for the attenuation range test is shown in Fig. 5. This test signal consists of a series of uncorrelated CSS which are fed in sending and receiving direction simultaneously simulating the double talk situation.

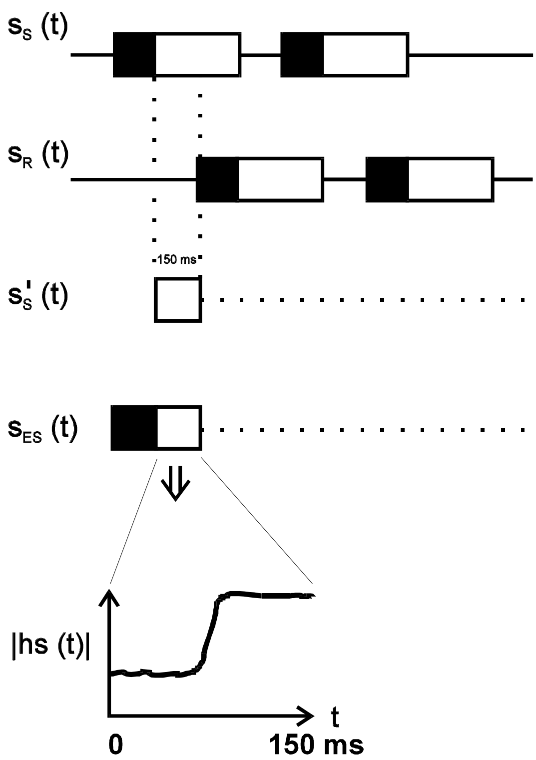


s(t) - signal in one direction

s_{dt}(t) - double talk signal

Fig. 5: Measurement sequence for double talk simulation with overlapped CSS bursts in sending and receiving direction

From Fig. 5 it can be seen that the overlap of the sequences is only partially. Always the voiced sound (black) overlaps the end of the noise sequence (white) of the opposite channel. The sequence is constructed in that way, that during the pauses in receiving direction, the sending direction can be measured; during the pauses in sending direction, the receiving direction can be evaluated. The principle of signal construction and the analysis procedure is given in Fig. 6.



$s_S(t)$	: Measured signal sending direction
$s_R(t)$	: Measured signal receiving direction
$s'_S(t)$	: Extracted signal sending direction
$s_{ES}(t)$	: Excitation signal for sending direction
$\|h_S(t)\|$	: Level vs. time

$$\left|h_s(t)\right| = \left|\int s_s{}'(t)e^{t/10ms}\right| - \left|\int s_{ES}\ (t)e^{t/10ms}\right|$$

Note: The time constant of 10 ms in this equation is shown as one example

Fig. 6: Principle of signal extraction and analysis during double talk.

The signal settings are as follows:

	receiving direction	sending direction
pause length between two signal bursts	151,38 ms	151,38 ms
average signal level (assuming a original pause length of 101,38 ms)	-16 dB_{m0}	-4,7 dB_{Pa}
active signal parts	-14,7 dB_{m0}	-3 dB_{Pa}

The level of the transmitted signal is measured at the POI and referenced to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve.

3.2.10.2 Attenuation range in receiving direction during double talk

Requirement

Based on the level variation in receiving direction during double talk determined by the parameter $a_{H,R,dt}$ (attenuation range in receiving direction during double talk) the behaviour can be distinguished according to the following table:

Behaviour	***1***	***2a***	***2b***	***2c***	***3***
a_{HRdt} [dB]	*≤3*	*≤5*	*≤8*	*≤10*	*>12*

Test

The test signal given in Fig. 5 (subclause “Level variation in sending direction during double talk”) is also used for the test in receiving direction. The signal settings are identical as for the measurement in sending direction.

The level of the transmitted signal is measured at the acoustical interface and referenced to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve.

3.2.10.3 Detection of Echo Components during Double Talk

Requirement

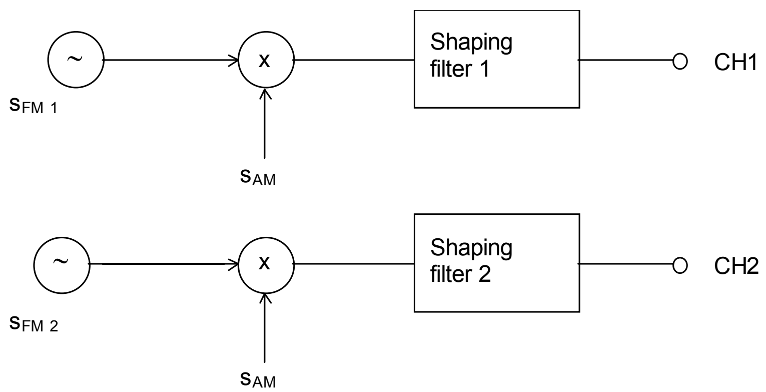
The behaviour of a terminal in a conversation based on the echo attenuation during double talk is determined by the parameter $TELR_{dt}$. Values resulting from subjective tests are given in Annex A of ITU Recommendation P.340 [3] for a complete end to end scenario. Assuming a nominal SLR + RLR of 10 dB these values can be reduced by 10 dB to set up requirements for terminals under test. The resulting values (echo loss during double talk EL_{dt}) are given in the following table:

Behaviour	***1***	***2a***	***2b***	***2c***	***3***
EL_{dt} [dB]	*≥ 27*	*≥ 23*	*≥ 17*	*≥ 11*	*< 11*

Test

The double talk test signal consists of orthogonal sequences generated by a set of voice like modulated sine waves, spectrally shaped. The general construction principle which in detail can be found in ITU-T Recommendation P.501 is shown below.

Fig. 7: Two channel test signal generation for double talk evaluations based on AM - FM signals



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

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

The settings are given in the following table:

Sending direction			**Receiving direction**		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]	f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	± 5	3	270	± 5	3
500	± 10	3	540	± 10	3
750	± 15	3	810	± 15	3
1000	± 20	3	1080	± 20	3
1250	± 25	3	1350	± 25	3
1500	± 30	3	1620	± 30	3
1750	± 35	3	1890	± 35	3
2000	± 40	3	2160	± 35	3
2250	± 40	3	2400	± 35	3
2500	± 40	3	2650	± 35	3
2750	± 40	3	2900	± 35	3
3000	± 40	3	3150	± 35	3
3250	± 40	3	3400	± 35	3
3500	± 40	3	3650	± 35	3
3750	± 40	3	3900	± 35	3

Parameters of the shaping filter: LP, 5 dB/oct.

Both signals are applied simultaneously in sending direction via the artificial mouth and in receiving direction of the terminal with nominal level. The recorded signal in sending direction consists of the measured echo signal and the double talk signal from the artificial mouth.

The echo attenuation (not weighted according to G.122) is determined for each individual frequency band which was excited in receiving direction and which therefore may cause echo. Due to the orthogonal test sequences used for sending and receiving direction the distinction between the near end signal and a potential echo in the recorded sending signal is assured.

3.2.11 Background Noise Transmission

General

Mobile terminals are often used in noisy environments. Consequently the transmission quality of the ambient background noise in sending direction is an important factor. The background noise is regarded as the test signal for the following tests (and not a disturbing factor).

3.2.11.1 Background Noise Transmission with Near End Speech

Requirement

Two different background noises are played via a 4 or 8 loudspeaker arrangement. The test is carried out by applying a simulated speech signal (composite source signal) at the artificial mouth. In presence of both background noise the signal level in sending direction should not vary more than 10 dB during and after the end of the simulated speech signal.

Test

a) Two different background noises are used: “university cafeteria” (63 dB(A), see table 1) and a Hoth noise with a level of 60 dB(A)

b) The near end speech signal is simulated with a modified composite source signal. It consists of 15 bursts. As described in ITU-T Recommendation P.501 the pause length is 101.38 ms. For this test the pauses between the bursts are extended by 100 ms after every pair of burst – starting with the regular pause length of 101.38 ms between burst #1 and #2 and burst #2 and #3. The behaviour of the noise reduction during speech pauses can be analysed with this modified composite source signal.

c) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”.

d) The test is carried out under steady state conditions, the each noise must have been applied for at least 5 s in order to adapt implemented noise reduction algorithm. For each noise the test consists of three steps: In the first step the noise is recorded in sending direction without applying a simulated speech signal at the artificial mouth. The noise level versus time is calculated with a 5 ms time constant In the second step the test is repeated but with the simulated speech signal active at the artificial mouth. Again the level versus time of the recorded sending direction signal is calculated with a 5 ms time constant. In the last step both level versus time curves are plotted in one chart. Level variations can be detected by the comparison of the two curves.

   A potentially different behaviour in presence of a stationary and an in-stationary noise can be analysed by comparing the charts determined with the two different noises.

3.2.11.2 Background Noise Transmission with Far End Speech

Requirement

Two different background noises are played via a 4 or 8 loudspeaker arrangement. The test is carried out by applying a simulated speech signal (composite source signal) in receiving direction at the POI. The level modulation in sending direction before, during and after the application of the CSS bursts in receiving direction shall be ≤ 10 dB for the cafeteria noise.

For the Hoth noise the level modulation shall not exceed 5 dB versus time.

Note that it is recommended that the level of an injected comfort noise should never exceed the background noise level, see chapter 6.2.11.3 and 6.2.11.4

Test

The realistic background noise is played back via 4 or 8 loudspeakers in the test room. The level is adjusted to 63 dB(A). Additionally the test is carried out with a stationary background noise signal (Hoth spectrum) applied with a level of 60 dB(A).

a) After applying the background noise for (at least) 5 s a periodical repetition of the Composite Source Signal is applied in receiving direction for 5.6 s (16 CSS bursts) using an average level of –16 dB_{m0}.

b) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”.

c) The signal is recorded in sending direction and analysed as level vs. time using a time constant of 35 ms. The analyses starts 2.5 s before the first CSS burst is applied in receiving direction and lasts at least 2 s longer than the last CSS burst. For each noise the test consists of three steps: In the first step the noise is recorded in sending direction without applying a simulated speech signal in receiving direction. The noise level versus time is calculated with a 35 ms time constant In the second step the test is repeated but with the simulated speech signal active in receiving direction. Again the level versus time of the recorded sending direction signal is calculated with a 35 ms time constant. In the last step both level versus time curves are plotted in one chart. Level variation can be detected by comparing the two curves.

   A potentially different behaviour in presence of a stationary and an in-stationary noise can be analysed by comparing the charts determined with the two different noises.

3.2.11.3 Comfort Noise Injection – Level

Requirement

The level of the injected comfort noise shall be in a range of +2 dB and -5 dB compared to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

Test

a) The realistic background noise (university café) is played back via 4 or 8 loudspeakers in the test room. The level is adjusted to 63 dB(A).

b) After applying the background noise for (at least) 5 s a periodical repetition of the Composite Source Signal is applied in receiving direction for 5.6 s (16 CSS bursts, using an average level of –16 dB_{m0}) to activate the comfort noise injection.

c) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”.

d) The test is done in three steps: At first the psophometric weighted level of the transmitted background noise measured in sending direction is determined without the composite source signal active in receiving direction and stored as a “reference”-level. In the second step the psophometric weighted level of the transmitted background noise measured in sending direction is determined while the composite source signal in receiving direction is active and stored as “comfort noise”-level. In the last step the deviation of the comfort noise level compared to the reference level is calculated.

3.2.11.4 Comfort Noise Injection – Spectral Comparison

Requirement

The spectral difference between comfort noise and original (transmitted) background noise shall be in between the mask given through straight lines between the breaking points on a logarithmic (frequency) – linear (dB sensitivity) scale as given in table 7.

Table 7: Requirements for Speactral Adjustment of Commfort Noise (Mask)

Frequency (Hz)	Upper limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2000	10	-10
2000	6	-6
4000	6	-6

NOTE: All sensitivity values are expressed in dB on an arbitrary scale.

Test

a) The realistic background noise (university café) is played back via 4 or 8 loudspeakers in the test room. The level is adjusted to 63 dB(A).

b) After applying the background noise for (at least) 5 s a periodical repetition of the Composite Source Signal is applied in receiving direction for 5.6 s (16 CSS bursts, using an average level of –16 dB_{m0}) to activate the comfort noise injection.

c) The handset terminal is set-up as described in subclause “Set-up for Handset Terminals”.

d) The test is done in three steps: At first the power density spectrum measured in sending direction of the transmitted background noise is determined without the composite source signal active in receiving direction and stored as a “reference”-spectrum. In the second step the power density spectrum measured in sending direction of the transmitted background noise is determined while the composite source signal in receiving direction is active and stored as “comfort noise”-spectrum. In the last step the comfort noise spectrum is referred to the reference spectrum. The tolerance mask is overlaid to verify the requirement.

3.3 Headset Terminals

Two different kinds of headsets are can be tested:
a) Headsets connected to a specific mobile (wired connection)
b) Bluetooth headsets which can be operated with any Bluetooth compatible mobile

Both types of mobiles shall be tested with the setup described in chapter 4.2, figure 1. The tests which shall be carried out are the same as for handset terminals described in chapter 6.2. The application force dependent tests will not be carried out.

With Bluetooth handsets in addition tests are carried out according to the setup shown in figure 2, chapter 4.2 where the artificial head is positioned in the direct link between mobile and headset. The following measurements – already described for handset terminal testing (number of chapter in brakets) – are conducted with this setup:

- Sending Frequency Response (6.2.3.1)
- Receiving Frequency Response (6.2.3.2)
- TMOS in Sending Direction (6.2.4.1)
- TMOS in Receiving Direction, 8N (6.2.4.2)
- Attenuation range in sending direction during double talk (6.2.10.1)
- Background Noise Transmission with Far End Speech: café and Hoth noise (6.2.11.2)
- Speech Recordings (6.5.1):
    - Sending Direction: Single Talk with and without BGN
    - Sending Direction: Double Talk with and without BGN
    - Sending Direction: Single Talk Echo with BGN

The results of those additional measurements are labelled as “shadowed” in the report.

3.4 Vehicle Mounted Hands-free Terminals

3.4.1 Delay

3.4.1.1 Delay in Sending Direction

Requirement

The delay in sending direction measured either from the mouth reference point (MRP) to the POI (output of the reference speech decoder of the GSM system simulator (SS)), shall be less than:

120 ms

Test

b) The test signal to be used for the measurements shall be a composite source signal (CSS) as described in ITU-T Recommendation P.501. 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 dB_{Pa}, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to –28.7 dB_{Pa} at the HATS-HFRP (as defined in P.581) and the spectrum is not altered.

The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals".

The delay is calculated using the cross correlation function between the signal at the electrical test point and the original test signal. The measurement is corrected by the delay introduced by the test equipment.

c) The delay is expressed in ms, determined from the maximum of the cross correlation function.

3.4.1.2 Delay in Receiving Direction

Requirement

The delay in receiving direction, measured either from the POI (input of the reference speech coder of the GSM system simulator (SS)) to the drum reference point (DRP) to, shall be less than:

100 ms

Test

a) The test signal to be used for the measurements shall be a composite source signal (CSS) as described in ITU-T Recommendation P.501. The test signal level shall be -16 dB_{m0}, measured at the electrical test point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals". The HATS is used then it is free-field equalized as described in ITU-T Recommendation P.581 [13]. The equalized output signal of right ears is used for the measurement.

c) The delay is calculated using the cross correlation function between the original test signal and the signal recorded with the artificial heads right ear. The measurement is corrected by the delay introduced by the test equipment.

d) The delay is expressed in ms, determined from the maximum of the cross correlation function.

3.4.2 Loudness Ratings

Requirement

The nominal values of SLR/RLR to/from the POI shall be:

SLR = 13 +/- 4 dB;

RLR = 2 +/- 4 dB.

Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for hands-free units intended to work in the vehicle environment. This is to allow for the increased noise volume in a moving vehicle.

Test

3.4.2.1 Sending Loudness Ratings

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 or a speech like test signal as described in ITU-T Recommendation P.501. 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 dB_{Pa}, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to -28.7 dB_{Pa} at the HATS-HFRP (as defined in P. 581) and the spectrum is not altered.

The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals". The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, 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.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79, formula 2.1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79, table 1.

3.4.2.2 Receiving Loudness Ratings

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dB_{m0}, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. The HATS is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of the right artificial ear is used for the measurement. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79, 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.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [18], formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79.

d) No leakage correction shall be applied. The hands-free correction of 14 dB as described in P.340 shall be applied.

3.4.3 Sensitivity/Frequency Characteristics

3.4.3.1 Sending Frequency Response

Requirement

The sending sensitivity frequency response from the MRP to the SS audio output (digital output of the reference speech decoder of the SS) shall be as follows:

The sending sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 7 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 7: Hands-free sending sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-14
400	0	-13
500	0	-12
630	0	-11
800	0	-10
1 000	0	-8
1 300	2	-8
1 600	3	-8
2 000	4	-8
2 500	4	-8
3 100	4	-8
4 000	0	

NOTE: All sensitivity values are expressed in dB on an arbitrary scale.

Test

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. 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 dB_{Pa}, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to -28.7 dB_{Pa} at the HATS-HFRP (as defined in P. 581) and the spectrum is not altered.

The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity S_{mJ}.

b) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [9] 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.

c) The sensitivity is expressed in terms of dBV/Pa.

3.4.3.2 Receiving Frequency Response

Requirement

The receiving sensitivity frequency response from the SS audio input (analogue or digital input of the reference speech encoder of the SS) to the ERP shall be as follows:

The receiving sensitivity frequency response shall be within the mask which can be drawn with straight lines between the breaking points in table 4 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 8: Hands-free receiving sensitivity/frequency response

Frequency (Hz)	Upper limit	Lower limit
200	0	
250	0	
315	0	-15
400	0	-12
500	0	-12
630	0	-12
800	0	-12
1 000	0	-12
1 300	0	-12
1 600	0	-12
2 000	0	-12
2 500	0	-12
3 100	0	-12
4 000	0	

NOTE: All sensitivity values are expressed in dB on an arbitrary scale.

Test

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level shall be –16 dB_{m0}, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals". If the HATS is used then it is free-field equalized as described in ITU-T Recommendation P.581. The equalized output signal of right ear is used for the measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [9] 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.

c) The sensitivity is expressed in terms of dBPa/V.

3.4.4 Idle Channel Noise

3.4.4.1 Sending

Requirement

The maximum noise level produced by the apparatus measured at the POI under silent conditions in the sending direction shall not exceed -64 dBm0p. No peaks in frequency higher than 10 dB above the average noise spectrum shall occur.

NOTE 1: This level includes the eventual noise contribution of an acoustic echo canceller under the condition that no signal is received.

Test

a) 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. 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 –28.7 dB_{Pa}, measured at the HATS-HFRP. The activation signal level is averaged over the complete activation signal sequence.

b) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”.

c) The idle channel noise is measured at the electrical reference point in the frequency range form 100 Hz to 4 kHz. The measurement window is applied directly after stopping the activation signal. The averaging time is 1 second. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. 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.

d) The noise spectrum is calculated by FFT (8k/48 kHz sampling rate) using Hanning windowing.

c) The idle channel noise is determined by using psophometric weighting. Spectral peaks are measured in the frequency domain.

3.4.4.2 Receiving

Requirement

The maximum (acoustic) noise level from the hand-free terminal when no signal (0-level) is received from the speech transcoder shall be as follows:

If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured at the artificial heads right ear contributed by the receiving equipment alone shall not exceed -53 dBA. No peaks in frequency higher than 10 dB above the average noise spectrum shall occur.

The requirement has to be checked with activation signal in order to ensure, that possible AGC devices are disabled during the actual measurement period.

Where a volume control is provided, the measured noise shall also not exceed -50 dB_{Pa}(A) at the maximum setting of the volume control. Again no peaks in frequency higher than 10 dB above the average noise spectrum shall occur.

NOTE: In a connection with the PSTN, noise conditions as described in ITU-T Recommendation G.103 can be expected at the input (POI) of the network. The characteristics of this noise may be influenced by the speech transcoding process (for further study).

Test

a) 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. The activation signal level shall be –16 dB_{m0}, measured at the electrical access point. The activation signal level is averaged over the complete activation signal sequence.

b) The hands-free terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals".

The idle channel noise is measured at the ear reference point (ERP) in the frequency range form 50 Hz to 8 kHz. The measurement window is applied directly after stopping the activation signal. The averaging time is 1 second. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. 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 spectrum is calculated by FFT (8k/48 kHz sampling rate) using Hanning windowing.

c) The idle channel noise is determined by using A-weighting. Spectral peaks are measured in the frequency domain.

3.4.5 Out-of-band Signals

3.4.5.1 Discrimination Against Out-of-band Signals

Requirement

When out-of-band signals are applied at the MRP, a range of frequencies will be transmitted to the terminal and input to the speech encoder. For the signals at the output of the speech encoder, measured at the POI the following requirements shall apply:

With a white Gaussian noise signal band-limited to 4,6 kHz up to 8 kHz applied at the MRP at a level of -4,7 dB_{Pa}, the total power in the frequency band 300 Hz to 3,4 kHz measured after decoding the output of the speech encoder shall be below the reference level by at least 40 dB. This reference level is obtained by applying an ITU-T P.50 artificial speech signal band-limited to 300 Hz and 3,4 kHz at a level of -4,7 dB_{Pa} at the MRP and measuring the average level of the signal at the speech encoder output after decoding it.

Test

a) The hands-free terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals".

b) 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. The activation signal level shall be –4,7 dB_{Pa}, measured at the mouth reference point. The activation signal level is averaged over the complete activation signal sequence.

b) Directly after the activation sequence the test signal is fed in. The test signal is inserted directly after the voiced sound of the last CSS burst of the activation sequence (instead of the pn-sequence). The duration of the test signal is 200 ms.

c) With a white Gaussian noise signal band-limited to 4.6 kHz up to 8 kHz applied at the MRP at a level specified above. The test signal level is averaged over the complete test signal sequence.

d) For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms). The total power in the frequency band 300 Hz to 3.4 kHz is measured after decoding the output of the speech encoder (electrical reference point). The reference level is obtained by applying an ITU-T P.50 artificial speech signal bandlimited to 300 Hz and 3,4 kHz at a level of -4,7 dB_{Pa} at the MRP and measuring the average level of the signal at the speech encoder output after decoding it (at the electrical reference point).

3.4.5.2 Spurious Out-of-Band Receiving Signals

Requirements

The level of out-of-band signals at the output of the head and torso simulator (HATS) shall meet the following requirements when the relevant input signals are applied in the receive direction.

With an ITU-T P.50 artificial speech signal in the frequency range of 300 Hz to 3,4 Hz and at a level of -12 dB_{m0} applied in the receive direction, the level of spurious out-of-band image signals in the frequency range of 4,6 to 8 kHz measured at the ERP shall be below the reference level by at least 45 dB. This reference level is obtained by measuring the in-band acoustic reference level produced by the same input signal.

Test

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-Recommendation P. 50 or a speech like test signal as described in ITU-T Recommendation P.501. The type of test signal used shall be stated in the test report. The test signal level is specified above, measured at the electrical access point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. The HATS is free-field equalized as described in ITU-T Recommendation P.581.The equalized output signal of the right artificial ear is used for the measurements.

c) The level of spurious out-of-band image signals is measured in the frequency range of 4,6 to 8 kHz . The reference level is obtained by measuring the in-band acoustic reference level produced by the same input signal.

3.4.6 Acoustic Echo Control

General

The echo loss (EL) presented by the 3G network at the POI should be at least 46 dB during single talk. This value takes into account the fact that UE is likely to be used in a wide range of noise environments.

The use of acoustic echo control is not mandated for 3G networks and the connection between the UE and the POI is zero loss. Therefore the acoustic echo control provided in terminal should provide a TCLw of at least 46 dB at the POI over the likely range of acoustic end delays.

If acoustic echo control is provided by voice switching, comfort noise should be injected. This comfort noise shall operate in the same way to that used in DTX.

3.4.6.1 Terminal Coupling Loss TCLw

Requirement

The TCLw for vehicle hands-free terminals shall be 40 dB at the nominal setting of the volume control in quiet background conditions and 33 dB at the maximum user selectable volume control setting. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm should be designed to cope with the expected reverberation and dispersion.

Test

For conducting the test a typical car cabin shall be used. The terminal set-up is described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. The ambient noise level shall be less than -70 dB_{Pa}(A). The attenuation from electrical reference point input to electrical reference point output shall be measured using a mostly speech like test signal providing a maximum of signal energy during the measurement.

e) 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 is altered. The training sequence level shall be –16 dB_{m0} in order not to overload the codec.

f) The test signal is a PN-sequence complying with ITU-T Recommendation P.501 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 250ms. The test signal level is –3 dB_{m0}. The low-crest factor is achieved by random-alternation of the phase between -180° and 180°.

g) The TCLw is calculated according to ITU-T Recommendation G.122 [8], annex B, 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 long enough in order to guarantee a reliable measurement. For the measurement a time window has to be applied adapted to the duration of the actual test signal (250 ms).

3.4.6.2 Echo Level versus Time

Requirement

The measured echo attenuation versus time under single talk conditions under steady state conditions should not vary for more than 6 dB between periods of the test signal (measured during the active signal parts).

Test

The test signal consists of a periodical repetition of the composite source signal. The measurement is conducted with two signal levels. The average signal levels are –5 dB_{m0} and -25 dB_{m0}. The echo signal level is analysed for at least 2.8 s, which represents 8 periods of the CSS, using a time 35 ms constant and referenced to the test signal level.

Note: The analysis is carried out only during the active signal parts. The pauses between two bursts of the test signal are not analysed. The analysis range during the active signal parts is reduced in the initial part by the length represented by the time constant.

3.4.6.3 Spectral Echo Attenuation

Requirement

The echo attenuation versus frequency expressed through the difference between the power density spectra of the test signal and the echo signal shall be below the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in the following table.

Table 9: Spectral echo attenuation mask

Frequency (Hz)	Upper limit
100	-20
200	-30
300	-38
800	-34
1500	-33
2600	-24
4000	-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.	

Test

a) The hands-free terminal is set-up as described in subclause "Set-up for Vehicle Mounted Hands-free Terminals".

b) The test signal consists of a periodical repetition of the composite source signal. The measurement is carried under steady state conditions with an average test signal level of –16 dB_{m0}. The power density spectrum of the echo signal is averaged over a sequence length of 1.4 s (i.e. 4 CSS bursts including the test signal pauses) and referenced to the power density spectrum of the original test signal. The analysis is conducted using 4k-FFT (48 kHz sampling rate).

c) The echo loss is expressed in dB.

e) If comfort noise injection is implemented and this implementation leads to a violation of the echo attenuation mask, a corresponding remark shall be given in the report. Tests to detect comfort noise injection are described in subclause 6.2.12 “Background noise transmission”.

3.4.7 Switching Characteristics

3.4.7.1 Activation in Sending Direction

General

The activation in sending direction is determined by the two parameters

- minimum activation level $L_{S,min}$ and
- build up time $T_{r,s,min}$.

The minimum activation level corresponds to the test signal level necessary to remove the inserted loss in sending direction in idle mode. The build up time is determined for the test signal burst which is applied with the minimum activation level.

Note: In the following the term minimum activation level refers to the test signal level at the MRP although the test signal may differ at the MRP due to the HATS-HFRP calibration. It is assumed that the tests are carried out with a HATS-HFRP-calibrated set-up to guarantee the appropriate test signal levels at the HATS-HFRP instead of the MRP.

Requirement

The minimum activation level $L_{S,min}$ shall be less or equal to -20 dB_{Pa}, i.e. 17 dB below the nominal level of -3dB_{Pa} (active signal part).

Note: This active signal level corresponds to an average signal level of -4,7 dB_{Pa} at the MRP assuming a pause of 101,38 ms between two CSS bursts.

The build up time shall be less or equal to 50 ms.

Test

The signal structure as given through the Fig. 6 represents signal parts with increasing levels. Periods of the CSS (as a simulation of speech) with increasing levels are suited for this signal.

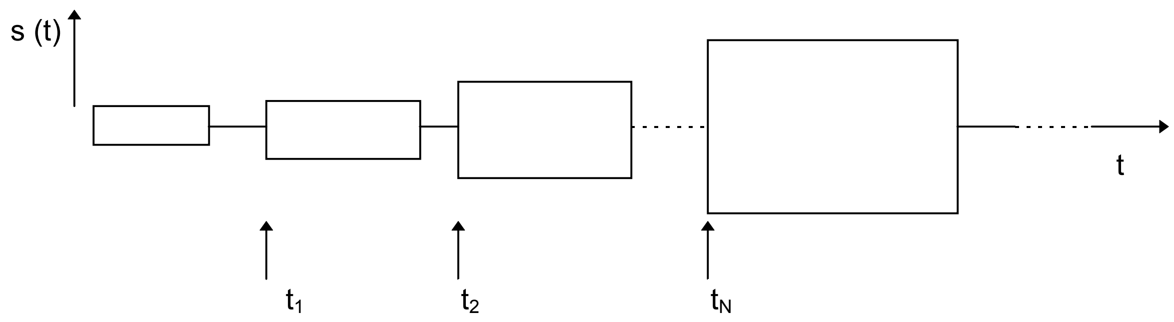


Fig. 6: Structure of test signal to determine the minimum activation level and build up time

Note:

The dotted line indicates the repetition or elongation of the test signal to achieve the suitable length for the measurement

The settings for the test signal in sending direction are as follows:

	Active duration / pause duration	level of the first period (active signal part, at the MRP)	Level difference between two periods
CSS for switching in **Sending direction**	248.62 ms / 451.38 ms	-23 dB_{Pa} *)	1 dB

*) Note: This active signal level corresponds to an average signal level of -24,7 dBPa at the MRP assuming a pause of 101,38 ms between two CSS bursts.

It is assumed that the pause length of 451.38 ms is longer than the hangover time so that the test object mode is the idle mode after each signal burst independent if is was activated or not.

The level of the transmitted signal is measured at the POI and referred to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The minimum activation level is determined from the resulting curve and the build up time is analyzed at the beginning of the corresponding signal burst (t_1, t_2, ..., t_N in Fig. 6).

3.4.7.2 Activation in Receiving Direction

General

The activation in receiving direction is determined by the two parameters

- minimum activation level $L_{R,min}$ and
- build up time $T_{r,R,min}$.

The minimum activation level corresponds to the test signal level necessary to remove the inserted loss in receiving direction in idle mode. The build up time is determined for the test signal burst which is applied with the minimum activation level.

In order to guarantee a higher accuracy recording the transmitted signal in receiving direction a measurement microphone is used for this test and positioned near the loudspeaker of the hands-free telephone.

Requirement

The minimum activation level $L_{R,min}$ shall be less or equal to -38,7 dB_m (active signal part), i.e. approximately 3 dB lower than a minimum expected network speech signal level.

Note: This active signal level corresponds to an average signal level of -40 dB_m assuming a pause of 101.38 ms between two CSS bursts.

The build up time shall be less or equal to 50 ms.

Test

The signal structure as given through the Fig. 6 (subclause "Activation in sending direction") is used in the same way for the test in receiving direction with the following signal settings:

	Active duration / pause duration	level of the first period (active signal part)	Level difference between two periods
CSS for switching in **Receiving direction**	248.62 ms / 451.38 ms	-38,7 dB_m *)	1 dB

*) Note: This active signal level corresponds to an average signal level of -40 dB_m assuming a pause of 101,38 ms between two CSS bursts.

The level of the transmitted signal is measured at the acoustical interface and referred to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The minimum activation level is determined from the resulting curve and the build up time is analyzed at the beginning of the corresponding signal burst (t_1, t_2, ..., t_N in Fig. 6).

3.4.7.3 Attenuation Range in Sending Direction

General

The attenuation range is determined by applying a test signal in sending direction after the terminal was activated in opposite direction (receiving direction). Two parameters can be determined

- attenuation range $a_{H,S}$ (complete inserted loss) and
- the build up time $T_{r,S}$ (until the output level reaches 3 dB below its final value)

Requirement

The attenuation range $a_{H,S}$ shall be ≤ 20 dB.

The build up time $T_{r,S}$ shall be ≤ 50 ms (recommended ≤ 15 ms except a residual value of 13 dB below the final value).

Test

The signal structure is given through Fig. 7. Periods of the CSS (as a simulation of speech) and voiced sounds are used for this signal. The signals on both channels are uncorrelated.

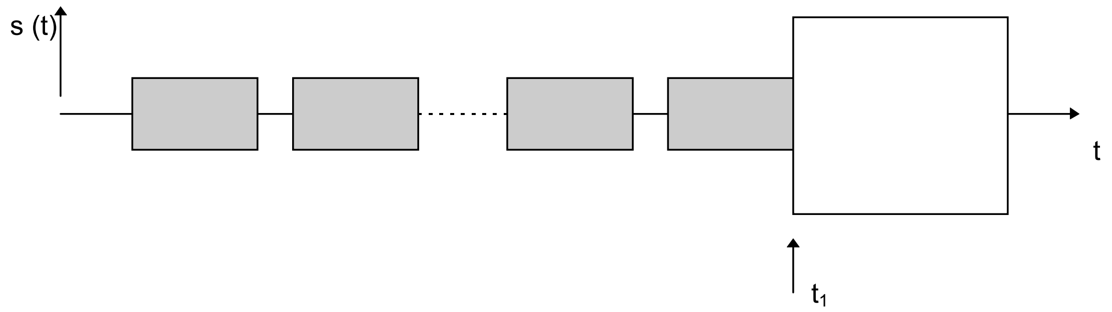


Fig. 7: Structure of test signal for attenuation range measurement

Note: The dotted line indicates the repetition or elongation of the test signal to achieve the suitable length for the measurement

A periodical repetition of CSS bursts as a simulation of speech is used to activate one transmission path (grey colour). In case of the attenuation range measurement in sending direction, the receiving direction is activated first. At the end of the voiced part of the CSS, the measurement signal is applied in the opposite path (white colour). This signal part consists of a periodical repetition of a voiced sound.

The signal settings are as follows:

	receiving direction	sending direction (at the MRP)
average signal level (incl. 101,38 ms pauses)	-16 dB_m	---
active signal parts	-14,7 dB_m	-3 dB_{Pa}

The level of the transmitted signal is measured at the POI and referred to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve and the build up time is analyzed starting from the beginning of the voiced activation signal in sending direction (t_1 in Fig. 7).

3.4.7.4 Attenuation Range in Receiving Direction

General

The attenuation range is determined by applying a test signal in receiving direction after the terminal was activated in opposite direction (sending direction). Two parameters can be determined

- attenuation range $a_{H,R}$ (complete inserted loss) and
- the build up time $T_{r,R}$ (until the output level reaches 3 dB below its final value)

In order to guarantee a higher accuracy recording the transmitted signal in receiving direction a measurement microphone is used for this test and positioned near the loudspeaker of the hands-free telephone.

Requirement

The attenuation range $a_{H,R}$ shall be ≤ 15 dB.

The build up time $T_{r,R}$ shall be ≤ 50 ms (recommended ≤ 15 ms except a residual value of 9 dB below the final value).

Test

The signal structure is given through Fig. 7 (subclause "Attenuation range in sending direction") The signal is used is used in the same way for the test in receiving direction: A periodical repetition of CSS bursts as a simulation of speech is used to activate the sending direction (grey colour in Fig. 7). At the end of the voiced part of the CSS, the measurement signal is applied in the receiving direction (white colour).

The signal settings are as follows:

	receiving direction	sending direction (at the MRP)
average signal level (incl. 101,38 ms pauses)	---	-4,7 dB_{Pa}
active signal parts	-14,7 dB_m	-3 dB_{Pa}

The level of the transmitted signal in receiving direction is measured at the acoustical interface and referred to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve and the build up time is analyzed starting from the beginning of the voiced activation signal in receiving direction (t_1 in Fig. 7).

3.4.8 Double Talk Performance

General

In duplex conditions the system performance is determined by mostly two parameters: talker echo loudness rating and level variation compared to the single talk condition (attenuation range). In order to achieve a auditory perceived high quality in double talk conditions the double talk echo attenuation should be high and the inserted loss in sending and receiving direction (due to level switching) should by low. For terminals which are not intended to provide double talk capabilities the total echo loss may be achieved by a correspondingly high echo attenuation based on level switching.

The corresponding parameters which determine the double talk performance are

- attenuation range in sending direction during double talk $a_{H,S,dt}$
- attenuation range in receiving direction during double talk $a_{H,R,dt}$
- echo attenuation during double talk

3.4.8.1 Attenuation Range in Sending Direction During Double Talk

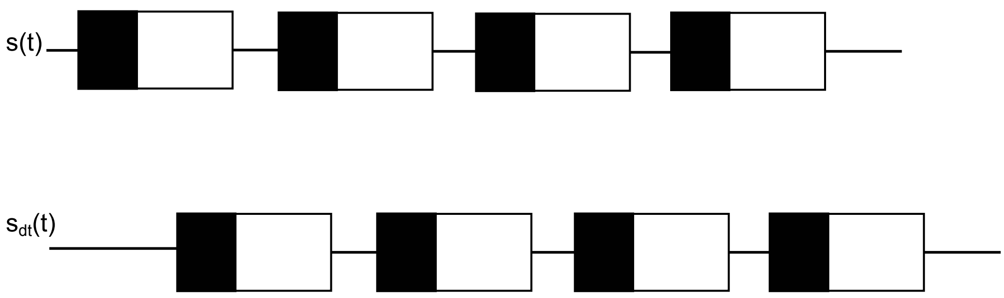
Requirement

Based on the level variation in sending direction during double talk determined by the parameter $a_{H,S,dt}$ (attenuation range in sending direction during double talk) the behaviour can be distinguished according to the following table:

Behaviour	***1***	***2a***	***2b***	***2c***	***3***
a_{HSdt} [dB]	*≤3*	*≤6*	*≤9*	*≤12*	*>12*

Test

The test signal used for the attenuation range test is shown in Fig. 8. This test signal consists of a series of uncorrelated CSS which are fed in sending and receiving direction simultaneously simulating the double talk situation.

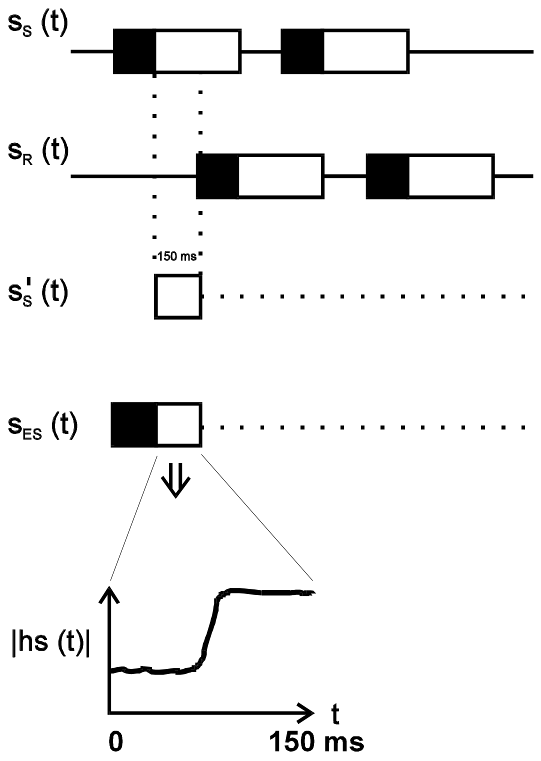


s(t) - signal in one direction

s_{dt}(t) - double talk signal

Fig. 8: Measurement sequence for double talk simulation with overlapped CSS bursts in sending and receiving direction

From Fig. 8 it can be seen that the overlap of the sequences is only partially. Always the voiced sound (black) overlaps the end of the noise sequence (white) of the opposite channel. The sequence is constructed in that way, that during the pauses in receiving direction, the sending direction can be measured; during the pauses in sending direction, the receiving direction can be evaluated. The principle of signal construction and the analysis procedure is given in Fig. 9.



$s_S(t)$: Measured signal sending direction

$s_R(t)$: Measured signal receiving direction

$s'_S(t)$: Extracted signal sending direction

$s_{ES}(t)$: Excitation signal for sending direction

$|h_S(t)|$: Level vs. time

$$|h_s(t)| = \left|\int s_s{}'(t) e^{t/10ms}\right| - \left|\int s_{ES}\ (t) e^{t/10ms}\right|$$

Note: The time constant of 10 ms in this equation is shown as one example

Fig. 9: Principle of signal extraction and analysis during double talk.

The signal settings are as follows:

	receiving direction	sending direction
pause length between two signal bursts	151,38 ms	151,38 ms
average signal level (assuming a original pause length of 101,38 ms)	-16 dB_{m0}	-4,7 dB_{Pa}
active signal parts	-14,7 dB_{m0}	-3 dB_{Pa}

The level of the transmitted signal is measured at the POI and referenced to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve.

3.4.8.2 Attenuation Range in Receiving Direction During Double Talk

General

In order to guarantee a higher accuracy recording the transmitted signal in receiving direction under double talk conditions a measurement microphone is used for this test and positioned near the loudspeaker of the hands-free telephone.

Requirement

Based on the level variation in receiving direction during double talk determined by the parameter $a_{H,R,dt}$ (attenuation range in receiving direction during double talk) the behaviour can be distinguished according to the following table:

Behaviour	***1***	***2a***	***2b***	***2c***	***3***
a_{HRdt} [dB]	*≤3*	*≤5*	*≤8*	*≤10*	*>12*

Test

The test signal given in Fig. 8 (subclause “Level variation in sending direction during double talk”) is also used for the test in receiving direction. The signal settings are identical as for the measurement in sending direction.

The level of the transmitted signal is measured at the acoustical interface and referenced to the original test signal level. The levels are calculated from the time domain using a time constant of 5 ms. The attenuation range is determined from the resulting curve.

3.4.8.3 Detection of Echo Components During Double Talk

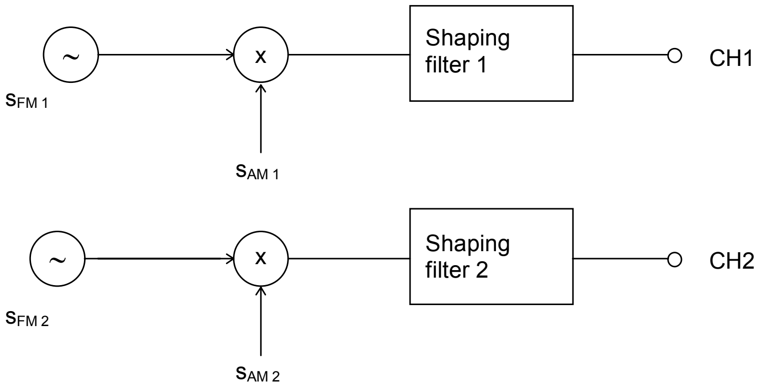
Requirement

The behaviour of a terminal in a conversation based on the echo attenuation during double talk is determined by the parameter $TELR_{dt}$. Values resulting from subjective tests are given in Annex A of ITU Recommendation P.340 [3] for a complete end to end scenario. Assuming a nominal SLR + RLR of 10 dB these values can be reduced by 10 dB to set up requirements for terminals under test. The resulting values (echo loss during double talk EL_{dt}) are given in the following table:

Behaviour	***1***	***2a***	***2b***	***2c***	***3***
EL_{dt} [dB]	*≥ 27*	*≥ 23*	*≥ 17*	*≥ 11*	*< 11*

Test

The double talk test signal consists of orthogonal sequences generated by a set of voice like modulated sine waves, spectrally shaped. The general construction principle which in detail can be found in ITU-T Recommendation P.501 is shown below.



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

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

Fig. 10: Two channel test signal generation for double talk evaluations based on AM - FM signals

The settings are given in the following table:

Sending direction			**Receiving direction**		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]	f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	± 5	3	270	± 5	3
500	± 10	3	540	± 10	3
750	± 15	3	810	± 15	3
1000	± 20	3	1080	± 20	3
1250	± 25	3	1350	± 25	3
1500	± 30	3	1620	± 30	3
1750	± 35	3	1890	± 35	3
2000	± 40	3	2160	± 35	3
2250	± 40	3	2400	± 35	3
2500	± 40	3	2650	± 35	3
2750	± 40	3	2900	± 35	3
3000	± 40	3	3150	± 35	3
3250	± 40	3	3400	± 35	3
3500	± 40	3	3650	± 35	3
3750	± 40	3	3900	± 35	3

Parameters of the shaping filter: LP, 5 dB/oct.

Both signals are applied simultaneously in sending direction via the artificial mouth and in receiving direction of the terminal with nominal level. The recorded signal in sending direction consists of the measured echo signal and the double talk signal from the artificial mouth. The echo signal is separated by appropriate filtering (comb filter). This filter suppresses the frequency components of the double talk signal (near end).

The echo loss calculation is carried out according to ITU-T Recommendation G.122. This calculation method is originally based on a wide band excitation signal. This test here uses an amplitude and frequency modulated test signal with a comb filter structure which has to be considered using an appropriate correction formula.

3.4.9 Background Noise Transmission

General

Mobile terminals are often used in noisy environments. Consequently the transmission quality of the ambient background noise in sending direction is an important factor. The background noise is regarded as the test signal for the following tests (and not a disturbing factor).

3.4.9.1 Level of Transmitted Background Noise

Requirement

The maximum noise level measured in sending direction at the POI shall not exceed -37 dBm0p.

Moreover a noise level < -45 dB_{m0}(P), but not lower than –50 dBm0(P) is recommended.

Test

a) The background noise is used as test signal. The signal level is determined through the characteristic of the background noise as described in table 1.

b) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”.

c) The test are carried out under steady state conditions, the noise must have been applied for at least 5 s in order to adapt implemented noise reduction. The noise spectrum is calculated by FFT (8k/48 kHz sampling rate) using Hanning windowing and psophometric weighting.

3.4.9.2 Background Noise Transmission After Call Set-up

Requirement

The analysis based on the Relative Approach [15] should not indicate remarkable characteristics exceeding 6 cp/cPa. The first transmitted signal peak in sending direction should not cause higher excitation than 15 cp/cPa .

Test

a) Background noise is played back. The signal level is determined through the characteristic of the background noise as described in table 1.

b) A measurement microphone is positioned near the HFT microphone to record the interior vehicle background noise signal. This recorded signal is not necessarily needed for the analysis but should be stored for information.

c) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. The terminal is switched off and on again (to provide a reset) and a call is send by the SS. The incoming call is answered at the terminal. Special care should be taken not to produce any disturbances or unwanted noise by touching the terminals housing while answering the incoming call.

d) The transmitted signal in sending direction is recorded at the POI starting at least 1 s before the call is answered and for at least 7 s after the call is established. The analysis range is chosen to 8 s including an initial pause of 1 s before the call was established. The recorded signal is analysed using the Relative Approach [14] and represented with scaling between 0 cp/cPa and 10 cp/cPa.

3.4.9.3 Quality of Background Noise Transmission (with far end speech)

Requirement

1. After the end of the last composite source signal burst (representing the end of far end speech simulation) the signal level in sending direction should not vary more than 10 dB (during transition to transmission of background noise without far end speech).
2. In addition this measurement can be used to detect comfort noise implementation (for information).

Test

a) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. Background noise is played back.

b) A test signal is applied in receive direction with nominal level consisting of an initial pause of ≥ 5 s and a periodical repetition of the composite source signal (duration ≥ 2 signal bursts).

c) The sending signal is recorded at the POI. The test signal level versus time is calculated using a time constant of 35 ms.

3.4.9.4 Quality of Background Noise Transmission (with near end speech)

Requirement

After the end of the last composite source signal burst (representing the end of near end speech simulation) the signal level in sending direction should not vary more than 10 dB (during transition to transmission of background noise without near end speech).

Test

a) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. Background noise is played back.

b) A test signal is applied in sending direction via the artificial with nominal level consisting of an initial pause of ≥ 5 s and a periodical repetition of the composite source signal (duration ≥ 2 signal bursts).

c) The sending signal is recorded at the POI. The test signal level versus time is calculated using a time constant of 35 ms.

3.4.9.5 Adjustment of Comfort Noise Injection

Requirement

1. The level of comfort noise is adjusted in a range of +2 and -5 dB to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

2. The spectral difference between comfort noise and original (transmitted) background noise shall be in between below the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in the following table.

Table 10: Spectral adjustment of comfort noise (mask)

Frequency (Hz)	Upper limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2000	10	-10
2000	6	-6
4000	6	-6
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.		

Test

a) The hands-free terminal is set-up as described in subclause “Set-up for Vehicle Mounted Hands-free Terminals”. Background noise is played back.

b) A test signal is applied in receive direction consisting of an initial pause of 10 s and a periodical repetition of the composite source signals in receiving direction (duration ≥ 10 s) with nominal level to enable comfort noise injection.

c) The transmitted signal is recorded in sending direction at the POI.

d) The power density spectra measured in sending direction during the initial pause of the test signal (8k FFT / 48kHz sampling rate, averaged over ≥ 5 s) is referred to the power density spectrum determined during the period with the periodical repetition of the composite source signals in receiving direction (8k FFT / 48kHz sampling rate, averaged over ≥ 5 s).

3.5 Recordings Using Real Speech

General

In addition to the objective tests described in this specification, recordings in sending and receiving direction via the simulated GSM or UMTS network are made using real speech samples. These recordings are not used for objective analyses but provide

- an impression about the transmission quality for different terminals under test,
- the possibility to verify that the transmission quality is not significantly influenced by parameters which where not covered by the objective tests and
- additional material that can be used to demonstrate certain transmission quality aspects to someone, who was not involved in the tests.

The test speech sequences are identical for all terminals under test and are therefore not designed to be used for subjective Third Party Listening Tests (e.g. according to ITU-T Recommendation P.800 [15]).

Special care should be taken during playback to guarantee a realistic sound reproduction. Recordings in receiving direction are binaural recordings provided by the HATS. The equalisation is described in subclause “Calibration and Equalisation of the Test Set-up”. The headphones should be free-field equalised. Both single and double talk situations are considered under quiet test conditions and with background noise playback.

The recordings in sending direction are mono-channel and are carried out at the POI. In order to reproduce the listening situation in sending direction these recordings at the POI are filtered using IRS send characteristic as described in ITU-T Recommendation P.48 [16]. Again single and double talk situations under quiet test conditions and with background noise playback are recorded.

For the recordings in sending direction under double talk conditions and under single talk conditions in receiving direction, the receive signal for the terminal is fed via the system simulator. The recorded signal in sending direction consists of the send signal played back by the artificial mouth at the near end and possible echo components. It should be considered, that echo signals – if present – appear in the recordings without the original test signal.

Requirement

- none -

3.5.1 Recordings in Sending Direction

The signal is recorded mono-channel at the POI. In sending direction the following situations are recorded:

- single talk
- double talk
- single talk echo

The recordings are carried under quite test conditions and with background noise.

Under quiet test conditions the test sequences in both directions are applied with nominal level. For the recordings in the presence of background noise the test signal level in sending direction is adjusted according to the kind of background noise as follows:

Recordings for handset terminals in the presence of background noise (university cafeteria noise according to table 1) are carried out with an increased test signal level in sending direction set to –1.7 dB_{Pa} at the MRP.

Recordings for vehicle mounted hands-free terminals in the presence of background noise (130 km/h according to table 1) are carried out with an increased test signal level in sending direction set to +1.3 dB_{Pa} at the MRP.

3.5.2 Recordings in receiving direction

Recordings in receiving direction are binaural recordings provided by the HATS with mounted handset terminal or at the driver's seat position for vehicle-mounted hands-free terminals.

In receiving direction the following situations as recorded:

- single talk
- double talk

The recordings are carried under quite test conditions and in single talk as well with background noise.

Under quiet test conditions the test sequences in both directions are applied with nominal level. For the recordings in the presence of background noise the test signal level in sending direction is adjusted according to the kind of background noise as follows:

Recordings for handset terminals in the presence of background noise (university cafeteria noise according to table 1) are carried out with an increased test signal level in sending direction set to -1.7 dB_{Pa} at the MRP.

Recordings for vehicle mounted hands-free terminals in the presence of background noise (130 km/h according to table 1) are carried out with an increased test signal level in sending direction set to +1.3 dB_{Pa} at the MRP.

4 Appendix

4.1 Test Result Table

Test Result Table should be taken to report the battery measurement test results. This test result tables will be presented soon.

5 Reference

[1] ITU Recommendation P.58, Head and Torso Simulators for Telephonometry

[2] ITU Recommendation P.57, Artificial Ears

[3] ITU Recommendation P.340, Transmission Characteristics and Speech Quality Parameters of Hands-free Telephones

[4] ITU Recommendation P.501, Test Signals for Use in Telephonometry

[5] ITU-T Recommendation P.50, Artificial voices

[6] ITU Recommendation P.502, Objective analysis methods for speech communication systems, using complex test signals

[7] ITU Recommendation P.64, Telephone transmission quality objective measuring apparatus, Determination of sensitivity/frequency characteristics of local telephone systems

[8] ITU Recommendation P.79, Calculation of Loudness Ratings for Telephone Sets

[9] ISO 3: “Preferred Numbers – Series of preferred numbers”.

[10] ITU-T Recommendation B.12, Use of the decibel and the neper in telecommunications

[11] ITU Recommendation G.122, Influence of national systems on stability and talker echo in international connections

[12] IEC 225, Third octave filters for electro-acoustical measurements

[13] ITU Recommendation P.581, Use of Head and Torso Simulators (HATS) for Hands-free Terminal Testing

[14] Manual ArtemiS, HEAD acoustics GmbH

[15] ITU Recommendation P.800, Methods for subjective determination of transmission quality

[16] ITU Recommendation P.48, Specification for an intermediate reference system

[17] ITU-T Rec. P.862, Perceptual evaluation of speech quality (PESQ), an objective method for end-to-end speech quality assessment of narrowband telephone networks and speech codecs

6 Abbreviations and Definitions

For the purposes of this specification, the following abbreviations and definitions apply:

ADC	Analogue to Digital Converter
AEC	Acoustic Echo Canceller
BGN	Background Noise
CC	Centre Clipper
DAC	Digital to Analogue Converter
DAI	Digital Audio Interface
DRP	Drum Reference Point
DTX	Discontinuous Transmission
EC	Echo Canceller
EEC	Electrical Echo Control
EL	Echo Loss
ERL	Echo return Loss
ERP	Ear Reference Point
HATS	Head and Torso Simulator
HFRP	Hands-free Reference Point
HATS-HFRP	HATS Hands-free Reference Point
HFT	Hands-free Telephone
LSTR	Listener Sidetone Rating
MRP	Mouth Reference Point
NLP	Non-linear Processor
OLR	Overall Loudness Rating
PCM	Pulse Code Modulation
POI	Point of Interconnection (with PSTN)
PSTN	Public Switched Telephone Network
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
SPL	Sound Pressure Level
STMR	Sidetone Masking Rating
SS	System Simulator
TX	Transmission
UE	User Equipment
UPCMI	13-bit Uniform PCM Interface

For the purposes of this specification, the following terms: dB, dBr, dB_{m0}, dBm0p and dBA, shall be interpreted as defined in ITU-T Recommendation B.12 [10]; the term dB_{Pa} shall be interpreted as the sound pressure level relative to 1 pascal expressed in dB (0 dB_{Pa} is equivalent to 94 dB SPL).