INTERSTATE COUNCIL
ON STANDARDIZATION, METROLOGY AND CERTIFICATION
(ISC)

**INTERSTATE STANDARD**

**GOST 33468-2015**

## Global navigation satellite system

# ROAD ACCIDENT EMERGENCY RESPONSE SYSTEM

## Test methods for verification of in-vehicle emergency call device/system conformity to requirements for loudspeaker communication in vehicle cabin

**Official Edition**
**English Version Approved by Interstandard**

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## Foreword

The purposes, main principles and basic order of work on interstate standardization are established by GOST 1.0-2015 "Interstate system for standardization. Basic principles" and GOST 1.2-2015 "Interstate System for Standardization. Interstate standards. Rules for development, taking over, renovation and cancellation"

**Details**

1 DEVELOPED by Non-Commercial Partnership "For Promotion of Navigation Technologies Development and Application" and Joint Stock Company "Research and Technical Centre of Advanced Navigation Technologies" "Internavigation" (JSC "Internavigation RTC")

2 INTRODUCED by Federal Agency on Technical Regulating and Metrology

3 ADOPTED by Interstate Council for Standardization, Metrology and cCrtification by means of correspondence (protocol No. 82-П, dated 12.11.2015)

Votes in favour:

| Short name of the country according to IC (ISO 3166) 004—97 | Country code according to IC (ISO 3166) 004—97 | Abbreviated name of national standards body |
|---|---|---|
| Armenia | AM | Ministry of Economics of Republic of Armenia |
| Belarus | BY | Gosstandart of Republic of Belarus |
| Kazakhstan | KZ | Gosstandart of Republic of Kazakhstan |
| Kyrgyzstan | KG | Kyrgyzstandart |
| Russian Federation | RU | Rosstandart |
| Tajikistan | TJ | Tajikstandart |

4 The Interstate Standard GOST 33468-2015 is introduced since 01.01.2017 as a national standard of the Russian Federation by Order No. 2038-ст, dated 15.12.2016, of Federal Agency on Technical Regulating and Metrology.

5 This Standard developed on based GOST R 55530-2013*

6 INTRODUCED FOR THE FIRST TIME

*The information on the amendments to this Standard is published in the annually issued information index "National standards", and the text of the amendments and corrections is published in the monthly issued information indices "National standards". In case of revision (replacement) or cancellation of this Standard the appropriate notice will be published in the monthly issued information index "National standards". The appropriate information, notice and texts are also placed in the general-use information system — on official site of Federal Agency on Technical Regulating and Metrology in the Internet (www.gost.ru)*



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National standard GOST R 55531-2013 is withdrawn from 01.06.2017 by Order No. 2038-ст, dated 15.12.2016, of Federal Agency on Technical Regulating.

## Contents

**I N T E R S T A T E S T A N D A R D**

**Global navigation satellite system**

**ROAD ACCIDENT EMERGENCY RESPONSE SYSTEM**

**Test methods for verification of in-vehicle emergency call device/system conformity to requirements for loudspeaker communication in vehicle cabin**

**Date of Introduction — 2017—01— 01**

## 1 Scope

This Standard applies to in-vehicle emergency call devices and systems intended for installation on wheeled vehicles of Categories M and N in accordance with the requirements of [1].

The Standard sets out the methods that shall be used when in-vehicle emergency call devices/systems are tested for their conformity to the requirements of GOST 33464 in regard to the quality of loudspeaker communications in the vehicle cabin so that two-side duplex voice connections with emergency services may be established in loudspeaker mode over mobile communication networks, as required in the Regulation [1].

## 2 Normative references

The following standards are referred to in this Standard:

GOST 16600-72 Speech transmission over the radiocommunication. Requirements for speech legibility and methods of articulation measurements

GOST 17187-2010 (IEC 61672-1:2002). Sound level meters. Part 1. Technical requirements

GOST 33464-2015 Global navigation satellite system. Road accident emergency response system. In-vehicle emergency call device/system. General technical requirements

N o t e — When using this standard it is expedient to check the validation of the reference standards in the general-use information system — on official site of Federal Agency on Technical regulating and Metrology in Internet or according to the annual information index "National standards" which is published as of January, 1st, of current year, and according to releases of monthly issued information index "National standards" in the current year. If a reference standard which the dated reference is provided to is replaced, it is recommended to use a version of this standard with the above specified year of approval (acceptance). If after the approval of this standard an amendment is inserted in a reference standard which the dated reference is provided to, and this amendment regards the provision referred to, it is recommended to apply this provision without regard to this amendment. If a reference standard is cancelled without a replacement, it is recommended to apply the provision which refers to it to a part which does not engage this reference.

## 3 Terms and definitions

3.1 The following terms with their respective definitions are used for the purposes of this Standard:

3.1.1 **acoustic echo canceller;** AEC: Signal processing device or algorithm included in the IVDS to compensate for acoustic echo signals from a far-end subscriber that enter the transmit channel mixed with the speech of a near-end subscriber, by subtracting the predicted echo signals obtained basing on adaptive filtering of far-end subscriber signals taken from the receive channel.

**Official Edition**

Note — A high-quality AEC enables loudspeaker communication sessions in full duplex mode.

3.1.2 **acoustic echo suppressor;** AES: Signal processing device or algorithm included in the IVDS to suppress acoustic echo signal from a far-end subscriber that enter the transmit channel with the speech of a near-end subscriber, owning to implementation of a send/receive direction switch controlled by voices of near-end and far-end subscribers.

Note — The main difference between an AES and an AEC is that the former attenuates echo signals by introducing losses in receive and transmit channels thereby making full duplex voice communications impossible.

3.1.3 **loudspeaker IVDS:** IVDS has no telephone handset or headset, and makes use of the speakers and microphones at a distance from the subscriber thereby providing for hands-free operation during the communication session.

3.1.4 **artificial ear:** Device modelled as an external human ear consisting of a transition chamber and a calibrated microphone, intended for measurements of sound pressure levels developed by external sources, and characterised by input acoustic impedance close to the average input impedance of an adult person's ear in the range of audible sound frequencies.

3.1.5 **head and torso simulator;** HATS: Human manikin constructed with its head and torso of dimensions close to the average ones of an adult person, and intended for consideration of sound diffraction effects caused by the human head and body when the acoustic measurements are taken in the vehicle cabin.

Note — The manikin's head contains artificial mouth and artificial ear devices.

3.1.6 **artificial voice:** Mathematically pre-defined synthetic test signal of spectral and timing characteristics close to the average ones of male or female speech. Used as a substitute of natural speech in objective measurements in order to ensure reproducibility of their results.

3.1.7 **artificial mouth:** Device modelled as a human mouth, equipped with a loudspeaker in the body with simulated lips, and characterised by near field sound directivity and acoustic radiation pattern similar to the average ones of an adult person's mouth.

3.1.8 **codec:** Device or algorithm used for encoding and decoding of voice or audio signals transmitted in mobile communication systems.

Note — A codec is described by the transmission frequency band; rate of encoded digital stream; signal processing delay; intelligibility and encoding quality of voice and audio signals.

3.1.9 **composite source signal;** CSS: Sound test signal which is a combination of various test signals in succession.

3.1.10 **sending direction:** Direction along the transmit channel from a near-end subscriber in the vehicle to a far-end subscriber in the service centre.

3.1.11 **receiving direction:** Direction along the receive channel from a far-end subscriber in the service centre to a near-end subscriber in the vehicle.

3.1.12 **speech quality assessment as per MOS scale:** Subjective estimation of the speech quality by averaging expert opinion scores expressed in a five-grade scale from 1 (very bad) to 5 (excellent).

3.1.13 **peak factor of a signal:** Ratio of the maximum amplitude of a signal to its r.m.s. value.

Note — For example, the peak factor of a sine-wave signal is 3.01 dB. The peak value of narrowband voice signals rarely exceeds 18 dB.

3.1.14 **sending loudness rating;** SLR: Weighted electro-acoustic attenuation of a loudspeaker IVDS in sending direction; it describes the amount of signal attenuation in the transmit channel from the loudness level of the acoustic signal developed by the near-end subscriber at the mouth reference point to the electric signal level at the reference point of the system simulator.

Note — This value is defined as a frequency-weighted average of ratios between sound pressure levels in Pascals (Pa) and effective signal voltages in Volts (V) measured in one-third-octave frequency bands. Sending loudness ratings take into account the nature of signal loudness perception by humans, are expressed in dB and calculated in accordance with [2].

3.1.15 **receiving loudness rating;** RLR: Weighted electro-acoustic attenuation of a loudspeaker IVDS in receiving direction; it describes the amount of signal attenuation in the receive channel from the electric signal level at the reference point of the system simulator to the loudness level of the acoustic signal perceived by the near-end subscriber at the ear drum reference point.

Note — This value is defined as a frequency-weighted average of ratios between effective signal voltages in Volts (V) and sound pressure levels in Pascals (Pa) measured in one-third-octave frequency bands. The receiving loudness rating takes into account the nature of signal loudness perception by humans, is expressed in dB, and is calculated in accordance with [2].

3.1.16 **full duplex:** Attribute of loudspeaker communication sessions that do not capture attention of the parties, and are possible in simultaneous double-talk mode.

3.1.17 **half duplex:** Attribute of loudspeaker communication sessions that are possible in alternating single-talk mode only.

3.1.18 **double-talk mode;** dt: Operating mode of a loudspeaker IVDS where both far-end and near-end subscribers try talking and listening to each other at the same time, interrupting one another.

3.1.19 **single-talk mode;** st: Operating mode of a loudspeaker IVDS where both far-end and near-end subscribers talk and listen to each other one at a time, not interrupting one another. One subscriber is always silent while the other one is speaking.

3.1.20 **near-end speech:** Speech of the subscriber occupying the cabin (compartment) of the vehicle equipped with a loudspeaker IVDS.

Note — A subscriber may be either a real person, or a test signal fed through an "artificial mouth" of the HATS manikin.

3.1.21 **far-end speech:** Speech of the subscriber located at the remote call service centre.

Note — A subscriber may be either a real person using an ordinary fixed-line phone with a handset, or a test signal fed through electric inputs/outputs of the system simulator.

3.1.22 **in-vehicle emergency call system;** IVS: System supporting the functions of an in-vehicle emergency call device and providing for automatic transmission of vehicle data messages when a road accident or an accident of other kind occurs.

Notes

1 In addition, an in-vehicle emergency call system may be used for manual transmission of vehicle data messages in the case of road accidents or accidents of other type.

2 Categories of vehicles that shall be equipped with in-vehicle emergency call systems are specified in [1].

3.1.23 **system simulator:** Device designed to simulate a mobile communication network, and equipped with a radio interface from the one side and with electric inputs/outputs of transmit and receive channels from the other side.

3.1.24 **narrowband in-vehicle emergency call device/system:** In-vehicle emergency call device/system operating with narrowband voice signals of standard quality (i.e., with an operating bandwidth of 0.3 — 3.4 kHz and a sampling rate not less than 8 kHz).

3.1.25 **in-vehicle emergency call device;** IVD: Device used for measurement and evaluation of vehicle coordinates, speed and direction of movement based on the signals from at least two active Global Navigation Satellite Systems, for manual transmission of vehicle data messages when a road accident or an accident of other kind, and for double-talk voice communication with emergency services over mobile communication networks.

Notes

1 In addition, an in-vehicle emergency call device may be used for automatic transmission of vehicle data messages in the case of road accidents or accidents of other type. The types of road accidents detected automatically and the time frames for implementation of the function for automatic transmission of vehicle data in the device are established in [1].

2 Categories of vehicles that shall be equipped with in-vehicle emergency call devices are specified in [1].

3.1.26 **partial duplex:** Attribute of communication sessions that partially employ double-talk mode if the signal of the other party is audible, but its loudness is bouncing.

3.1.27 **wideband in-vehicle emergency call device/system:** In-vehicle emergency call device/system operating with wideband voice signals of standard quality (i.e., with an operating bandwidth of 0.15 — 7.9 kHz and a sampling rate not less than 16 kHz).

3.1.28 **ear drum reference point;** DRP: Measuring point of sound pressure levels which is located either at the ear tract end near the ear drum of a human ear, or on the diaphragm of the measuring microphone in an "artificial ear" device.

3.1.29 **mouth reference point;** MRP: Measuring point of sound pressure levels which is located at a distance of 25 mm before either the human lips or the signal-emitting ring of an "artificial mouth" device.

3.1.30 **hands-free reference point;** HFRP: Measuring point of sound pressure levels which is located on the "artificial mouth" axis, 50 cm apart from the signal-emitting "lips" ring, where the sound pressure level is calibrated for the "artificial mouth" device in free acoustic field conditions.

3.1.31 **reference point of system simulator;** POI: Point of connection and measurement of electric signal levels in the receive/transmit channels of a mobile system simulator.

3.1.32 **ear reference point;** ERP: Measuring point of sound pressure levels which is located outside the human ear or the "artificial ear" device.

3.1.33 **radio frequency reference point;** RFRP: Hypothetical point of IVDS connection to a mobile communication network or a system simulator through the radio interface. It is used in calculations of the system processing delay in the IVDS.

## 4 Abbreviations

The following symbols and abbreviations are used for the purposes of this Standard:

| | | |
|---|---|---|
| ADPCM | — | Adaptive Differential Pulse Code Modulation; |
| AGC | — | Automatic Gain Control; |
| ADC | — | Analogue-to-Digital Converter (including an analogue LPF at the input); |
| AFR | — | Amplitude-Frequency Response; |
| FFT | — | Fast Fourier Transform; |
| PCM | — | Pulse-Code Modulation; |
| DD | — | Design Documentation; |
| ND | — | Normative Documentation; |
| NTD | — | Normative and Technical Documentation; |
| SNR | — | Signal-to-Noise Ratio; |
| PC | — | Personal Computer; |
| CWP | — | Check Work Place; |
| PSD | — | Power Spectrum Density; |
| VH | — | Vehicle; |
| SPL | — | Sound Pressure Level; |
| PASF | — | Power Amplifier of Sound Frequency; |
| IVDS | — | In-vehicle Emergency Call Device/System; |
| HPF | — | High-Pass Filter; |
| LPF | — | Low-Pass Filter; |
| DAC | — | Digital-to-Analogue Converter (including an analogue LPF at the output); |
| DSP | — | Digital Signal Processing; |
| NRD | — | Noise-Reduction Device; |
| OD | — | Operating Documents; |
| ACR | — | Absolute Category Rating; |
| AMR | — | Adaptive Multi-Rate, (the standard for encoding of sound files with variable bitrate); |
| $A_{H,R}$ | — | Attenuation Range in receiving direction during single talk (in IVDS receive channel); |

$A_{H,R,dt}$ — Attenuation Range in receiving direction during double talk (in IVDS receive channel);
$A_{H,S}$ — Attenuation Range in sending direction during single talk (in IVDS transmit channel);
$A_{H,S,DT}$ — Attenuation Range in sending direction during double talk (in IVDS transmit channel);
CCR — Comparative Category Rating;
DCR — Degradation Category Rating;
DI — Digital Interface;
DTX — Discontinuous Transmission (voice transmission through a communication link, with interruption during the pauses);
ERL — Echo Return Loss (echo signal attenuation);
full rate — Digital standard for speech encoding;
GSM — Global System for Mobile Communications (global standard of mobile communication in digital cellular networks);
$L_{R,min}$ — Minimum channel activation (switch-on) level in receiving direction;
$L_{S,min}$ — Minimum channel activation (switch-on) level in sending direction;
MOS — Mean Opinion Score (speech quality assessment by averaging subjective opinions of several experts);
PN — Pseudorandom Noise;
$S_{diff}$ — Diffuse Field Sensitivity (for diffuse sound field);
$S_{direct}$ — Direct Sound Sensitivity (for direct plane-wave direction);
SPL — Sound Pressure Level;
$TCL_W$ — Weighted Terminal Coupling Loss (weighted crossover attenuation in electro-acoustic signal path);
$T_R$ — Signal processing delay in IVDS for receiving direction;
$T_S$ — Signal processing delay in IVDS for sending direction;
UMTS — Universal Mobile Telecommunications System.

## 5 General

5.1 The test methods described in this Standard are intended for IVDS conformity checks against the requirements of GOST 33464 in regard to quality assurance of loudspeaker communications in order to meet the requirements of [1] for establishing and maintaining IVDS duplex voice communication in loudspeaker mode with the emergency services over mobile communication networks.

5.2 In addition, this Standard specifies the requirements for test procedures and test conditions, testing equipment, and measuring instruments.

5.3 The complete IVDS test cycle includes the following stages:

1) tests of microphones separately from IVDS (such tests are omitted if the microphone of the type specified by the IVDS manufacturer is included in the IVDS delivery package);

2) objective measurements of IVDS specifications;

3) subjective quality assessment of loudspeaker communication, including its assessment in presence of noises for each relevant scenario (see 7.7.6.3 and 7.12.2).

5.4 If an IVDS is of multi-purpose design intended for use in vehicles of various types, the IVDS tests are recommended for at least three types of such vehicles with an identical geometry of their bodies. This is because the IVDS specifications in part of the sound quality are largely dependent on the vehicle body geometry that governs echo signal and background acoustic noise levels, as well as on the types and locations of microphones and speakers in the vehicle compartment.

## 6 Conditions and procedure of tests

### 6.1 Main equipment

6.1.1 Two connection interfaces are used in IVDS quality assurance tests in part of loudspeaker communications: an acoustic interface and a radio interface of the supported mobile communication system.

6.1.2 When an IVDS is connected using the acoustic interface for near-end subscriber simulation, a HATS manikin shall be used; the latter shall include an artificial ear and mouth, and its specifications shall meet the requirements of [3] both in sending and in receiving directions of acoustic fluctuations.

N o t e — As an alternative to HATS, other artificial mouth and ear devices are permitted provided that they do not lead to substantial errors in acoustic measurements compared to those of HATS.

6.1.3 When an IVDS is connected using the radio interface of the supported mobile communication system (at the RFRP point) for far-end subscriber simulation, a system simulator shall be used; the latter shall comply with all requirements of the mobile communication standard used in the IVDS, and shall be equipped with a calibrated (in dBm0) electric input/output for audio signals that is appropriate for connection of testing equipment.

6.1.4 System simulator units shall provide for selection of speech encoding type and rate, as well as for switching-off of additional voice processing, e.g., of DTX which is discontinuous transmission of voice data based on the speech activity detector.

6.1.5 The arrangement of simulator antennas and IVDS antennas, as well as the levels of RF signals, shall be such that no voice packet losses may occur in either direction of the digital communication link.

6.1.6 For the tests based on such measurements where the prevention of additional voice signal distortions in the operator network (system simulator) due to low-rate encoding of voice/audio signals is critical, the best codec from among those available for the simulator and the IVDS shall be selected, and shall be configured for as high data transmission rate as possible.

N o t e — In most tests, as well as in subjective assessments of voice signal transmission quality, notably, where acoustic noises are present, all codec and rate options supported by the mobile communication system and by the IVDS shall also be checked to ensure the quality of IVDS operation in real operating conditions regardless of the encoding type that is selected by a base station automatically when the connection to the IVDS is established.

6.1.7 Electric test signals shall be applied to or picked up from the IVDS through a system simulator, while acoustic ones, through an artificial mouth and an artificial ear located in the HATS manikin's head. The flow diagram of the IVDS testing apparatus is shown in Figure 1.

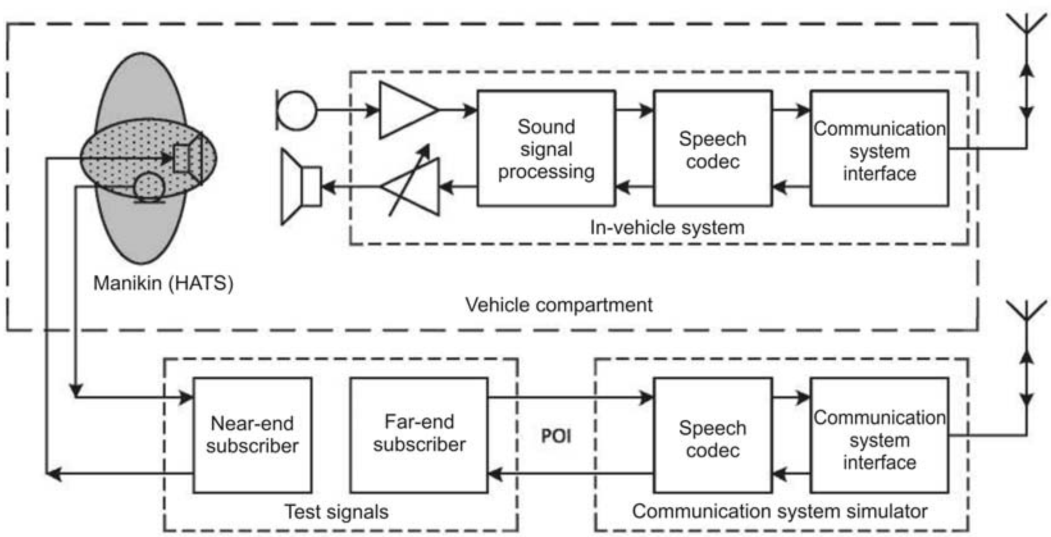



Fig. 1 — Flow diagram of testing apparatus for in-vehicle emergency call devices/systems

6.1.8 IVDS microphones are tested separately from the IVDS in free acoustic field conditions (of an anechoic chamber) using a low-distortion reference loudspeaker, and in the vehicle cabin using an "artificial mouth" device. The flow diagram of the testing apparatus for microphones shall be specified by the manufacturer; one of the conceivable options is shown in Figure 2.

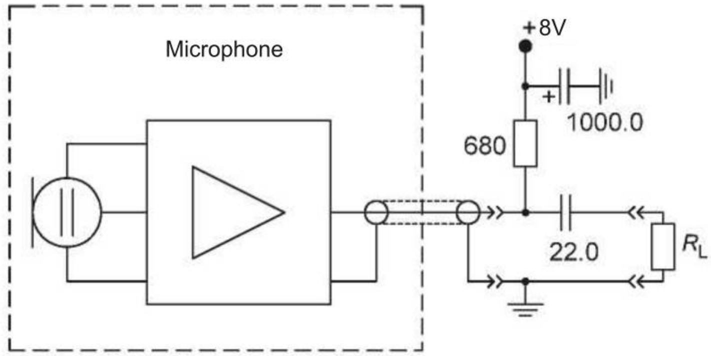



Fig. 2 — Flow diagram of testing apparatus for microphones

6.1.9 The microphone should be powered from a stabilised power supply or a battery with voltage ripples not exceeding 0.5 mV so that any spurious pulses at the microphone output would not exceed 0.5 mV under its loading with the resistance $R_L$ of a value not greater than 10 kOhm.

6.1.10 The measuring systems, testing equipment, measuring instruments and devices that shall be used for the tests are specified in Table A.1 (Appendix A).

**6.2 Simulation of external acoustic noise**

6.2.1 A separate PC-based sound playback unit with a multi-channel sound card, five-channel power amplifier of sound frequency, four broadband loudspeakers and one low-frequency loudspeaker (subwoofer) shall be used for simulation of the acoustic noise surrounding the vehicle driver in real operating conditions. The arrangement of testing equipment in the vehicle is illustrated in Figure 3. The sound playback unit shall be capable of reproducing noise recordings made during the movement of a given vehicle type. Noise signals shall be recorded and reproduced taking into account the requirements of B.2 (Appendix B) and [4].

6.2.2 Different noise scenarios relevant to operation of a given vehicle type shall be used for acoustic noise recording in the vehicle compartment.

A measuring broadband capacitor microphone installed close to the IVDS microphone and a digital recording apparatus of dynamic range not less than 60 dB in the frequency range of at least 20—16000 Hz shall be used for such recordings. Concurrently with recording, the total SPL inside the vehicle shall be monitored and logged using a noise level meter.

6.2.3 Prior to playback of acoustic noise recordings, the sound playback unit shall be calibrated based on the total SPL using the noise level meter and by means of frequency response equalisation, so that the power spectrum densities in the original noise recordings and the ones in any further noise recordings using the measuring microphone and the loudspeakers for simulation of that noise would coincide to a given accuracy in the whole frequency range from 100 Hz to 10 kHz.

6.2.4 After the measurements and frequency-weighting over A-curve, the maximum SPL deviation of the reproduced noise from the original noise shall not exceed ±1 dB. The PSD deviation of the reproduced noise measured in one-third-octave bands in the range frequency from 100 Hz to 10 kHz shall not be greater than ±3 dB with respect to the original noise spectrum.

6.2.5 This noise simulation method can not reproduce the acoustic noise field in the vehicle cabin exactly, but provides a satisfactory approximation for tests of loudspeaker IVDS systems with a single microphone. The frequency response equalisation procedure and the reference set of acoustic noise recordings are presented in [4].

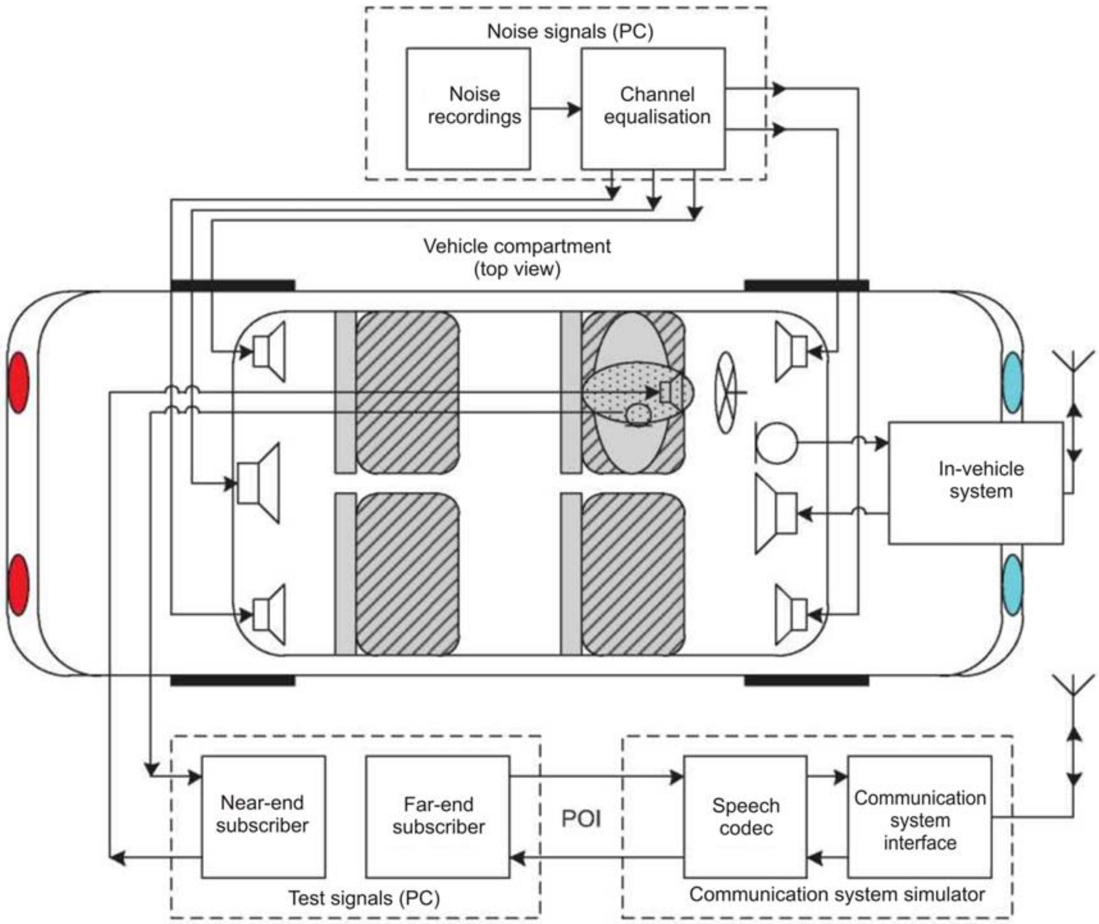



Fig. 3 — Testing equipment for simulation of external acoustic noise

**6.3 Arrangement of loudspeaker IVDS in vehicle compartment**

6.3.1 IVDS installed in standard equipment configurations by vehicle manufacturers shall be tested as originally supplied with IVDS microphones and speakers already installed.

6.3.2 IVDS installed in auxiliary equipment configuration shall be mounted and configured in the vehicle compartment in accordance with the requirements of the IVDS manufacturer. The microphone (or microphone array) and speaker positions shall be clearly indicated by the IVDS manufacturer. If such recommendations are missing, the testing laboratory itself shall determine the position of IVDS component parts. As a rule, the microphone is placed on a rear view mirror or on a sun screen, and the speaker, in a foot well for the side passenger.

6.3.3 The microphone and speaker positions during the tests shall be reflected in the test report and documented as an electronic photographic image.

6.3.4 The main location of a HATS manikin complying with the requirements of [3] shall be the vehicle driver's seat.

The manikin shall be arranged in accordance with the average driver location pattern corresponding to a male person of a 50-percentile representation level.

If the vehicle manufacturer and the IVDS manufacturer do not enforce any clear requirements regarding the distance from the MRP point of the HATS manikin to the IVDS microphone, then the testing laboratory shall determine an MRP to microphone distance on their own accord.

6.3.5 The manikin position and, most importantly, the position of its head shall be measured during the tests with respect to the ceiling, walls, airbags, etc., of the vehicle, shall be detailed in the test report and documented as an electronic photographic image.

Direct application of mark-up both in the vehicle compartment and on the HATS manikin is recommended to ensure the reproducibility of test conditions at any time after the tests are complete.

6.3.6 Different positions of the artificial manikin head may strongly affect the test results, especially the voice transmission quality in acoustic noise conditions. Thus, additional tests are recommended for the largest MRP to microphone distance possible for the driver where the SNR of voice signals for sending direction would be minimal.

**6.4 Requirements for artificial mouth device**

6.4.1 An artificial mouth located in the artificial HATS head shall meet the requirements of [3] and [5], and its transfer function (AFR) in free acoustic field conditions shall be equalised at the MRP for the SPL of voice signals in sending direction equal to minus 4.7 dBPa (89.3 dB SPL) in accordance with the requirements of [6], and then for the SPL increased by 3 dB to minus 1.7 dBPa to allow for loudspeaker communication effects.

6.4.2 For the HATS manikin in free acoustic field conditions, the average SPL for voice signals in sending direction shall be set to minus 25.7 dBPa by means of SPL correction at the MRP. The calibration point of HFRP in the free acoustic field (in an anechoic measuring chamber) is selected at a distance of 50 cm from the "lips" ring in the direction coinciding with the one to the IVDS microphone installed in the vehicle. If the HATS is almost perfect and the HFRP is on the emitting axis of the artificial mouth, the SPL of minus 25.7 dBPa at the HFRP will correspond to the SPL of minus 1.7 dBPa at the MRP. If the HATS is medium quality or the HFRP direction is at an angle to the emitting axis of the artificial mouth, the SPL at the MRP after calibration will be different from minus 1.7 dBPa. This SPL at the MRP corrected in acoustic free field conditions will correspond to a certain level of electric (digital) signals at the input of the "artificial mouth" amplifier. This level is recorded and later used to set a nominal "minus 1.7 dBPa at MRP" level during the tests in the vehicle compartment where direct measurements of real SPL at MRP and HFRP are not possible due to the body geometry and the presence of reflections and standing acoustic waves.

The usage procedures of HATS manikins in tests of loudspeaker devices including their equalisation and calibration procedures are described in [7].

6.4.3 The acoustic level of voice signals in sending direction with an average level equal to minus 25.7 dBPa (62.3 dB SPL) at the HFRP is primary for most tests, and corresponds to "normal" loudness of human voice during the talk at a distance of (0.5 – 1) allowing for a level increase by 3 dB due to loudspeaker communication effects [6].

6.4.4 If the noise level exceeds 50 dB(A) during the tests, the output level of voice signals shall be increased by 3 dB for each 10 dB increment of the noise level averaged for a long period. This accounts for the effect that people start talking more loudly in ambient noise conditions. The output level of voice signals depending on the noise level increments is expressed by the formula:

$$I(N) = \begin{cases} 0 & for \quad N < 50 \quad dB(A) \\ 0.3(N-50) & for \quad 50 \le N \le 77 \; dB(A) \\ 8.1 & for \quad N > 77 \quad dB(A) \end{cases} \tag{1}$$

where $I$ is the increment of the output speech level, dB;

$N$ is the noise level measured near the driver's head, averaged for a long period, dB(A).

For example, if the noise level measured in the vehicle compartment is 70 dB(A), then the output level increment of the artificial mouth will be 6 dB. The maximum increment shall be equal to 8 dB.

This level correction of the near-end subscriber's speech is only used for testing in noise conditions.

A 3 dB level increase of voice signals in sending direction due to loudspeaker communication effects [6] shall also be taken into consideration.

**6.5 Requirements for "artificial ear" devices**

6.5.1 The IVDS tests may employ signals of one or both artificial ears in the HATS manikin's head. A single ear is used by agreement with the IVDS manufacturer and shall be located from the side where the main IVDS loudspeaker is installed.

6.5.2 The transfer function of the artificial ear shall be equalised in free acoustic field conditions in accordance with [7].

**6.6 Eliminating effects of mobile communication system**

6.6.1 The IVDS parameter measurements may be adversely affected by additional processing of test signals as they pass through the communication channels (the use of various speech codecs, speech activity detectors, comfort noise generators, etc.) depending on the mobile communication system and on the settings of the system simulator used in the test.

6.6.2 If the requirements for IVDS specifications can not be met due to distortions introduced by the system simulator, then such behaviour of the simulator shall be checked using a reference IVDS, and shall be included in the relevant report.

**6.7 Calibrating levels and equalising frequency response of acoustic devices**

6.7.1 The following preparatory activities shall be carried out before the tests are started:

1) acoustic calibration of levels for measuring microphones and a microphone in HATS artificial ear;

2) calibration of levels and equalisation of the specified AFR transfer function for an artificial mouth in HATS manikin's head at the MRP and HFRP;

3) equalisation of AFR transfer function for an artificial ear in HATS manikin's head, either in free acoustic field if a single speaker is used in the IVDS, or in diffuse acoustic field if several speakers are used.

6.7.2 The following reference measurements shall be carried out in order to check, and to compensate for, the difference between the power spectrum densities of test signals applied to the IVDS either acoustically or electrically, and their original power spectrum densities in digital form:

1) in sending direction, the reference spectrum of acoustic signals is recorded and analysed at the MRP;

2) in receiving direction, the reference spectrum of electric signals is recorded at the electric input of the system simulator interface, at the POI.

**6.8 Setting-up system simulator**

6.8.1 All settings of the system simulator shall ensure that no additional processing of sound signals (except for encoding/decoding) and no errors occur in the radio channel. DTX mode shall be disabled. The RF signal level shall be set to its maximum value for all supported networks. All settings of the system simulator shall be included in the test report.

6.8.2 For narrowband measurements in GSM networks, the basic "full rate" codec should be used. If a narrowband AMR codec is used instead, its maximum digital bitrate of 12.2 kbit/s shall be configured.

6.8.3 For wideband measurements in GSM and UMTS networks, the AMR-WB codec should be used at a digital bitrate of 12.65 kbit/s.

**6.9 Acoustic conditions of measurements**

6.9.1 The total SPL of extraneous acoustic noise in the testing laboratory shall not be greater than minus 54 dBPa(A), and its spectral density shall not exceed the value specified in Figure 4 (Curve NC 40).

For certain tests, the background acoustic noise level should not exceed minus 64 dBPa(A) whereas its spectral density should be no higher than that of NC 30 profile; in such tests, the maximum level of minus 74 dBPa(A) and the spectral density no higher than NC 20 are recommended.

**6.10 Test conditions and testing equipment for subjective quality assessment of loudspeaker communication**

6.10.1 The test arrangement is shown in Figure 5. The near-end subscriber is inside the vehicle equipped with a loudspeaker IVDS. At the far end, a test team leader using a fixed-line telephone resides in the call centre, controls the progress of tests and makes basic communication quality assessments.

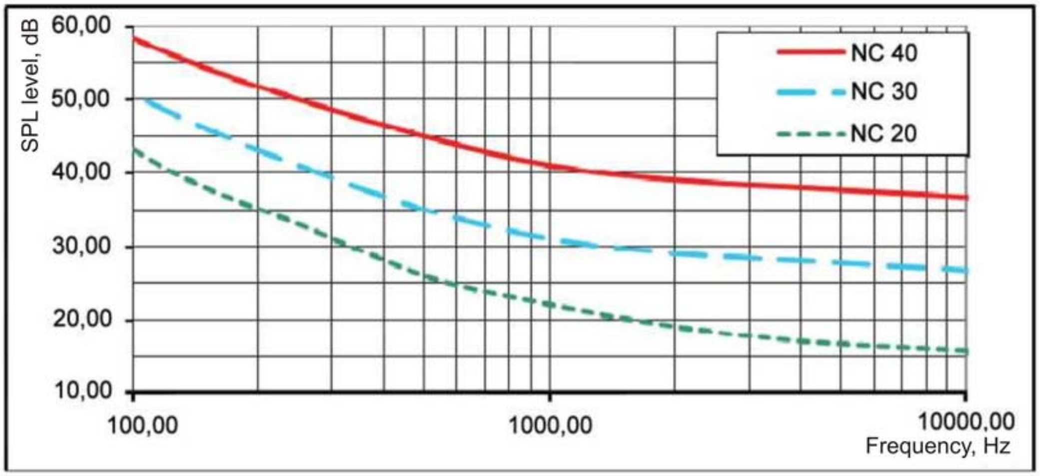



Fig. 4 — Spectral density of ambient noise

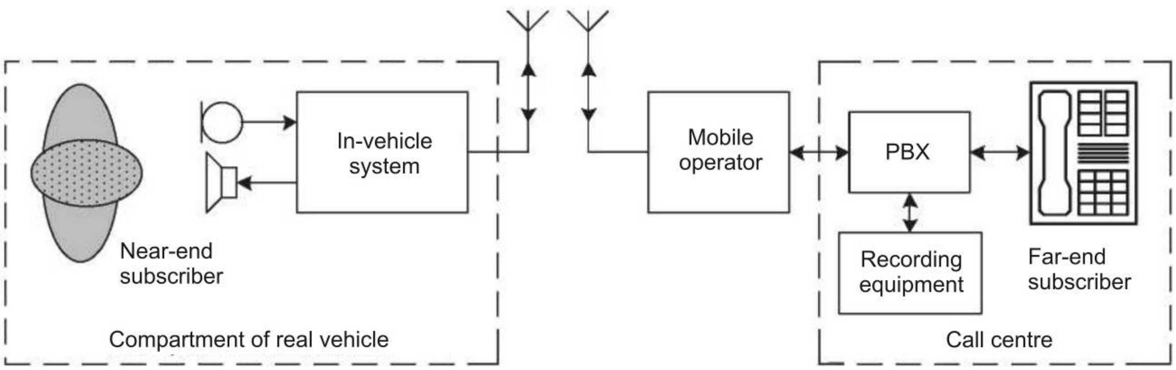



Fig. 5 — Test arrangement for subjective quality assessment of loudspeaker communication

6.10.2 Requirements for equipment:

1) vehicle of given type, mark and model equipped with IVDS under test;

2) equipment for recording signals on operator side (electric connection) and in vehicle cabin (acoustic connection via microphones installed next to near-end subscriber's head);

3) mobile communication network used for tests shall provide for complete coverage of the road area of tests and for an RF level that enables the best quality of uninterrupted services for the whole test duration;

4) if possible, the above mobile communication network should not use any additional signal processing (e.g., AGC, noise suppression [8], echo cancellation, etc.).

6.10.3 Requirements for test team:

1) tests shall be carried out by experts in speech quality assessment who understand what features and subtleties of sounding deserve special attention. Untrained system users and independent experts may be admitted when the test recordings are reproduced;

2) both male and female talkers with no noticeable pronunciation or hearing defects should be in the vehicle, adding up to at least three voices of each type. The low, normal and loud voice (shrill) options shall be checked for normal and accelerated tempos and for different talker's positions relative to the IVDS microphone;

3) both talkers (near-end and far-end) shall know each other voices well so that the checks of natural sounding and of talker's voice recognition would be possible;

4) to prevent from additional disturbances, passengers of the vehicle and people surrounding the far-end subscriber shall keep silent.

6.10.4 Requirements for reporting

A test report shall include the following essential information:

1) vehicle model, year of manufacture, completing options (type of tyres, engine, compartment upholstery, etc.), photograph;

2) IVDS model, hardware and software versions, photograph;

3) IVDS speaker and microphone positions relative to the driver (type, location, orientation, distance, photograph);

4) communication network provider; speech encoding if known;

5) type of subscriber equipment used for tests in the call centre.

The additional information pertaining to management of the subjective testing process is included in clause 7.13.

Note — For laboratory tests, a mobile network simulator (emulator) may be used. For field tests, a public mobile communication network shall be used.

## 7 Test methods

The IVDS test methods described in this section:

- are aimed at conformity assurance as regards the requirements stated in GOST 33464 with respect to the quality of loudspeaker communication in the vehicle cabin;
- apply to both narrowband and wideband IVDS, unless otherwise specified;
- take into account the basic requirements of [9], [10];
- include the minimum requirements for parameters of transmit/receive channels and for performance of digital processing algorithms used for sound signal processing (echo cancellers and other algorithms).

If an IVDS is of wideband type, and it uses the technology of artificial sound bandwidth expansion for narrowband voice signals received via communication networks, this technology shall be disabled in order to obtain correct results of measurements.

If an IVDS uses AGC for sending, it may employ a test mode where the minimum, nominal or maximum transmission amplification can be enforced. The nominal amplification shall correspond to the $SLR_{nom}$ for the vehicle driver.

If an IVDS uses AGC for receiving, it may employ a test mode where the minimum, nominal or maximum transmission amplification can be enforced in accordance with the $RLR_{max}$, $RLR_{nom}$ and $RLR_{min}$ for the vehicle driver.

If an IVDS uses NRD, it may employ a test mode where the noise reduction may be forcibly disabled.

If an IVDS uses AEC, it may employ a test mode where the AEC algorithms can be controlled using the following operations: disabling echo cancellation, disabling echo suppression, disabling (freezing) adaptive filter coefficients, and resetting (clearing) adaptive filter coefficients.

Prior to tests, all acoustic measuring instruments shall be calibrated, and their AFR shall be equalised.

### 7.1 Signal processing delay in IVDS

7.1.1 The delays in signal propagation from one subscriber to another both in receiving direction $T_{RSUM}$ and in sending direction $T_{SSUM}$ are the sums of the delays $T_{RSND}$ and $T_{SSND}$ introduced by sound processing algorithms used in the IVDS (AGC, AEC, noise suppression, etc.), the signal encoding and decoding delays $T_{RDEC}$ and $T_{SC0D}$ in the IVDS telephone part, and the standard delays $T_{RSYS}$ and $T_{SSYS}$ in the mobile communication system which are related to signal encoding and decoding processes on the base station side as well as to signal propagation time in service provider channels:

$$T_{SSUM} = T_{SSND} + T_{SCOD} + T_{SSYS}, \quad (2)$$

$$T_{RSUM} = T_{RSND} + T_{RDEC} + T_{RSYS}. \quad (3)$$

Since the delays $T_{SSND}$, $T_{SCOD}$, $T_{RSND}$ and $T_{RDEC}$ in the IVDS are difficult to measure separately, the total delays in sending ($T_{SSND}$+$T_{SCOD}$) and receiving ($T_{RSND}$+$T_{RDEC}$) direction between the IVDS acoustic interface (sound signal at MRP and ERP) and the IVDS radio interface (RFRP) are specified.

The requirements for signal processing delays in the IVDS are enforced both separately [signal processing delay in sending direction ($T_{SSND}$+$T_{SCOD}$) and signal processing delay in receiving direction ($T_{RSND}$+$T_{RDEC}$)] and cumulatively [overall delay in both directions ($T_{SSND}$+$T_{SCOD}$+$T_{RSND}$+$T_{RDEC}$)].

**7.1.2 Signal processing delay in IVDS in sending direction**

7.1.2.1 Requirements

For any loudspeaker IVDS, the signal processing delay in sending direction ($T_{SSND}$+$T_{SCOD}$) shall not exceed 122 ms for GSM communication systems and 143 ms for UMTS communication systems.

Since the minimum delay $T_{SCOD}$ is governed by the mobile communication system, the processing time of sound signals in sending direction $T_{SSND}$ shall be minimised at the IVDS development stage by reasonable selection of algorithms and of signal buffering methods (a value of at most 50 ms is recommended).

7.1.2.2 Method of measurement

The total signal propagation delay in sending direction $T_{SSUM}$ shall be measured as a signal propagation time from the MRP mouth reference point (acoustic voice signal) to the POI reference point of the communication system simulator (electric voice signal after decoding) as shown in Figure 6.

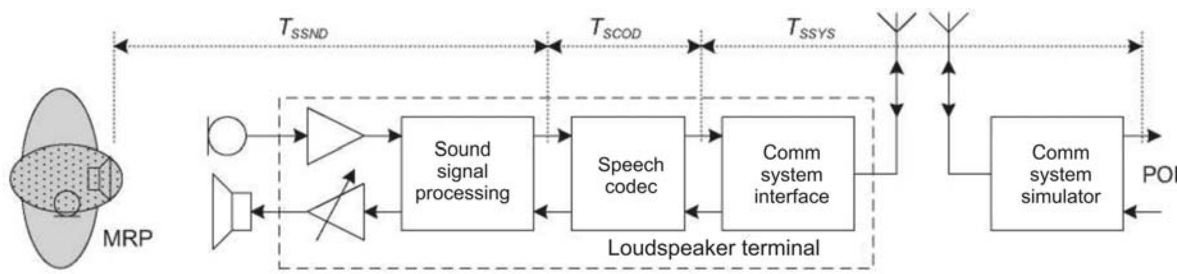



Fig. 6 — Signal propagation delay in sending direction

The signal delay $T_{SSYS}$ depends on the operating mode of the system simulator, and must be known from its passport data.

Composite CSS as per [11] shall be used for measurements. The duration of a single CSS part containing pseudo-random noise shall be greater than the signal propagation time being measured. A PN-sequence consisting of at least 16000 samples at a sampling rate of 48 kHz is recommended. The acoustic SPL of the input test signal applied through the HATS [7] is set equal to minus 25.7 dBPa at the HFRP (see 6.4.2). The frequency response of the artificial mouth is equalised in advance at the MRP (see 6.4.1).

Two electric signals shall be recorded at the same time: an input test signal applied to the artificial mouth, and an output signal picked up at the POI from the detector output of the system simulator.

The signal propagation delay is determined in milliseconds by calculating the cross-correlation function between the two recorded signals and evaluating the primary maximum of the latter function.

The IVDS signal processing delay ($T_{SSND}$+$T_{SCOD}$) in sending direction is calculated by subtraction of the known signal delay in the system simulator $T_{SSYS}$.

**7.1.3 Signal processing delay in IVDS in receiving direction**

7.1.3.1 Requirements

For any loudspeaker IVDS, the signal processing delay in receiving direction ($T_{RSND}$+$T_{REC}$) shall not exceed 122 ms for GSM communication systems and 143 ms for UMTS communication systems.

Since the minimum delay $T_{RDEC}$ is governed by the mobile communication system, the processing time of sound signals in receiving direction $T_{RSND}$ shall be minimised at the IVDS development stage by reasonable selection of algorithms and of signal buffering methods (a value of at most 50 ms is recommended).

7.1.3.2 Method of measurement

The total signal propagation delay in receiving direction $T_{RSUM}$ shall be measured as a signal propagation time from the POI reference point of the communication system simulator (electric voice signal before encoding) to the ERP ear reference point or the DRP ear drum reference point (acoustic voice signal) as shown in Figure 7.

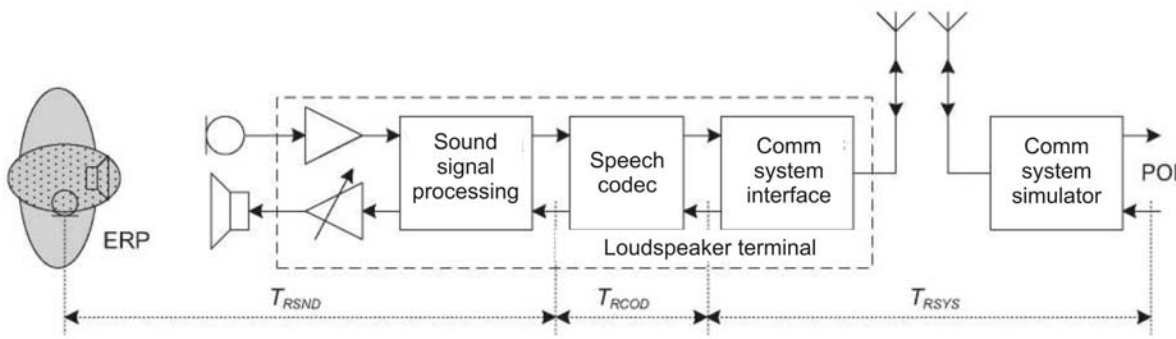



Fig. 7 — Signal propagation delay in receiving direction

The signal delay $T_{RSYS}$ depends on the operating mode of the system simulator, and must be known from its passport data.

Composite CSS as per [11] shall be used for measurements. The duration of a CSS part containing pseudo-random noise shall be greater than the signal propagation time being measured. A PN-sequence consisting of at least 16000 samples at a sampling rate of 48 kHz is recommended. The electric level of the input test signal is set equal to minus 16 dBm0 at the POI reference point of the encoder input of system simulator.

Two electric signals shall be recorded at the same time: an input test signal applied to the encoder input of the system simulator, and an output signal picked up at the DRP reference point of a HATS artificial ear (the ear closest to the IVDS speaker shall be used).

The signal propagation delay is determined in milliseconds by calculating the cross-correlation function between the two recorded signals and evaluating the primary maximum of the latter function.

The IVDS signal processing delay ($T_{RSND}$+$T_{RDEC}$) in receiving direction is calculated by subtraction of the known signal delay in the system simulator $T_{RSYS}$.

**7.1.4 Overall signal processing delay in IVDS in receiving and sending directions**

7.1.4.1 Requirements

For any loudspeaker IVDS, the overall signal processing delay in receiving and sending directions ($T_{SSND}$+$T_{SCOD}$+$T_{RSND}$+$T_{RDEC}$) shall not exceed 214 ms for GSM communication systems and 256 ms for UMTS communication systems.

Since the minimum delays for $T_{SCOD}$ and $T_{RDEC}$ are governed by the mobile communication system, the overall processing time of sound signals in receiving and sending directions ($T_{SSND}$+$T_{RSND}$) shall be minimised at the IVDS development stage by reasonable selection of algorithms and of signal buffering methods (a value of at most 50 ms is recommended).

7.1.4.2 Method of measurement

The values of ($T_{SSND}$+$T_{SCOD}$) and ($T_{RSND}$+$T_{RDEC}$) are measured separately, and then summed up.

**7.2 Loudness ratings**

7.2.1 Loudness ratings of a loudspeaker IVDS in sending and receiving directions provide for standard methods used to describe relative levels of acoustic and electric (digital) signals in the communication network taking into account psychoacoustic properties behind the perception of sound loudness by a human ear.

The loudness ratings of loudspeaker IVDS shall be measured in accordance with the requirements of [2], [6], [12] and [13] taking into account the specifics of their use in vehicles as described in [9] and [10].

**7.2.2 Sending loudness rating (SLR)**

7.2.2.1 The SLR of an IVDS is the frequency-weighted measure of signal attenuation along the path from the IVDS input at the MRP to the electric output of the network simulator decoder at the POI reference point; its physical dimension (dBPa/V) is reverse to that of the sensitivity.

7.2.2.2 Requirements

The SLR measured for an IVDS installed in the vehicle compartment (cabin) shall be equal to (13 ± 4) dB for the driver and closest passengers.

The passenger from the right side of the driver and advisably the passengers in the second row behind the driver's back are considered as the closest passengers.

The microphone shall not be turned off.

Any additional manual gain control of the IVDS in sending direction is not provided for. Whether an automatic gain control (AGC) is necessary for the IVDS in sending direction in order to equalise loudness ratings for the passengers located at different distances from the IVDS microphone shall be decided by either the IVDS manufacturer or the vehicle manufacturer.

N o t e — If an AGC is used in sending direction, the level of transmitted ambient noises or echo signals may increase.

7.2.2.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If an AGC is available for sending direction, it may be switched to a test mode with the amplification fixed at $K_{max}$, $K_{nom}$ and $K_{min}$ levels if the adaptive pattern of AGC amplification changes obstructs the measurements when test signals are used.

2) The acoustic test signal shall be an artificial voice obtained as described in [14]. The artificial mouth of the HATS manikin is calibrated, and its frequency response is equalised at the MRP in accordance with the requirements of [7] (see 6.4.1). The acoustic SPL of the test signal at the HFRP of the IVDS is set to minus 25.7 dBPa (see 6.4.2). The test signal level is determined by averaging along the full signal path.

3) The PSD is assessed at the MRP for the test signal, and this assessment is later used to estimate the IVDS sensitivity in sending direction.

4) The sensitivity of a narrowband IVDS in sending direction is calculated for each of 14 one-third-octave frequency bands listed in [2] (Table 1), bands 4 through 17. The sensitivity of a wideband IVDS in sending direction is calculated for each of 20 one-third-octave frequency bands listed in [2] (Table A.2), bands 1 through 20. To calculate the sensitivity in each frequency band, an r.m.s. level of electric signals at the POI of the decoder output of the system simulator is measured and divided by the SPL of acoustic signals for a given band at the MRP. The sensitivity is evaluated in dBV/Pa.

5) The SLR is calculated in accordance with [2]:

- for narrowband IVDS: using the formula 5-1, in bands 4—17, for $m = 0.175$, with frequency-weighting in accordance with [2] (Table 1);

- for wideband IVDS: in accordance with [2] (Appendix A).

N o t e — As a preliminary rough estimate of the IVDS sensitivity in sending direction, the estimated transfer ratio between the acoustic input of the IVDS at the MRP and the electric output of the system simulator decoder at the POI may be taken in a wide band with frequency-weighting along A-curve.

**7.2.3 Receiving loudness rating (RLR)**

7.2.3.1 The RLR of an IVDS is the frequency-weighted measure of signal attenuation along the path from the electric input of the system simulator encoder at the POI reference point to the acoustic output at the DRP; its physical dimension (dBV/Pa) is reverse to that of the sensitivity.

An additional gain control, if present in receiving direction, may be either manual (using an IVDS volume control), or automatic (AGC) depending on the received signal level as well as on the level of the ambient acoustic noise.

7.2.3.2 Requirements

The nominal receiving loudness rating $RLR_{nom}$ measured for an IVDS installed in the vehicle compartment (cabin) shall be equal to the value specified by the IVDS or vehicle manufacturer in accordance with the requirements of GOST 33464 (sub-clause 7.5.3.10) for the driver and closest passengers.

The passenger from the right side of the driver and advisably the passengers in the second row behind the driver's back are considered as the closest passengers.

The nominal receiving loudness rating (constant for systems with no volume controls and initial for systems with manual or automatic volume controls) shall ensure that duplex voice communications with the system operator in double-talk mode are reliable in all typical conditions of vehicle operation including in the presence of interfering acoustic noises in the vehicle compartment (cabin)

The required nominal value of $RLR_{nom}$ is selected by the IVDS or vehicle manufacturer to ensure that the receiving loudness rating is sufficient for reliable double-talk loudspeaker communications at an acoustic SNR in receiving direction not less than 6 dB in "ordinary noise" conditions [depending on the vehicle category (type) and on the noise scenario]. If the requirements for noise type and level are not specified by the vehicle manufacturer, the minimum SPL of background noises in the vehicle compartment is taken equal to minus 24 dBPa(A) (70 dBa SPL). The $RLR_{nom}$ value selected by the manufacturer shall be the range from (minus 6 ± 4) dB to (2 ± 4) dB. The value of (minus 6 ± 4) dB is recommended.

If manual gain control in receiving direction is provided for in the IVDS, then the selected nominal loudness rating $RLR_{nom}$ corresponding to the nominal IVDS volume shall be achieved at the nominal position of the volume control (mid-range of the gain control marked on the control scale).

The maximum receiving loudness rating $RLR_{max}$ corresponding to the minimum IVDS volume shall be achieved at the minimum (leftmost) position of the volume control. The required $RLR_{max}$ value shall be specified by the IVDS or vehicle manufacturer in accordance with the requirements of GOST 33464 (sub-clause 7.5.3.11).

The IVDS user or the IVDS itself (operating automatically) shall not be able to decrease the receiving loudness level below the minimum value enabling double-talk loudspeaker communication at an acoustic SNR in receiving direction not less than 0 dB in typical situations (depending on the vehicle category/type and on the noise scenario). If the requirements for noise type and level are not specified by the vehicle manufacturer, the minimum SPL of background noises in the vehicle compartment is taken equal to minus 24 dBPa(A) (70 dBa SPL). The $RLR_{max}$ value shall be selected in the range from (0 ± 4) dB to (8 ± 4) dB. The value of (2 ± 4) dB is recommended.

The minimum receiving loudness rating $RLR_{min}$ corresponding to the maximum IVDS volume shall be achieved at the maximum (rightmost) position of the volume control. The required $RLR_{min}$ value shall be specified by the IVDS or vehicle manufacturer in accordance with the requirements of GOST 33464, clause И.4.3 (Appendix И) to provide for reliable double-talk loudspeaker communications at an acoustic SNR in receiving direction not less than 6 dB in "worst case" noise conditions (depending on the vehicle type and on the noise scenario). If the requirements for noise type and level are not specified by the vehicle manufacturer, the SPL of background noises in the vehicle compartment is taken equal to minus 14 dBPa(A) (80 dBa SPL).

The $RLR_{min}$ value shall be selected in the range from minus 10 ± 4 dB to minus 18 ± 4 dB. The value of minus 13 ± 4 dB is recommended.

Whether an automatic gain control in receiving direction is necessary for the IVDS shall be decided by the IVDS or vehicle manufacturer.

N o t e — If an AGC is used in receiving direction, the level of transmitted echo signals may increase.

7.2.3.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If an AGC is available for receiving direction, it may be switched to a test mode with the amplification fixed at $K_{max}$, $K_{nom}$ and $K_{min}$ levels if the adaptive pattern of AGC amplification changes obstructs the measurements when test signals are used.

2) The acoustic test signal shall be an artificial voice obtained as described in [14]. The test signal level at the electric input of the system simulator encoder at the POI reference point is set to minus 16 dBm0. The test signal level is determined by averaging along the full signal path.

3) The PSD is assessed for the test signal in one-third-octave bands at the POI reference point, and this assessment is later used to estimate the IVDS sensitivity in receiving direction.

4) The artificial ear in the HATS manikin's head is calibrated, and its frequency response is equalised in free acoustic field conditions in accordance with the requirements of [7]. The measurements are taken at the DRP, and the results may, if necessary, be reduced to the reference ERP in accordance with [15]. For narrowband IVDS, the RLR is measured using the output of a single artificial air closest to the IVDS loudspeaker. For wideband IVDS, the use of both artificial ears is preferable so that their signal levels are averaged independently in third-octave bands and then summed up.

5) The sensitivity of a narrowband IVDS in receiving direction is calculated for each of 14 one-third-octave frequency bands listed in [2] (Table 1, bands 4 through 17). The sensitivity of a wideband IVDS in receiving direction is calculated for each of 20 one-third-octave frequency bands listed in [2] (Table A.2, bands 1 through 20). To calculate the sensitivity in each frequency band, an r.m.s. level of acoustic signals at the DRP is measured and divided by the r.m.s. level of electric signals at the POI reference point. The sensitivity is evaluated in dBPa/V.

6) The RLR in receiving direction is calculated in accordance with [2]:

- for narrowband IVDS: using the formula 5-1, in bands 4—17, for $m = 0.175$, with frequency-weighting in accordance with Table 1 [13];

- for wideband IVDS: in accordance with Appendix A (with $L_e$ factor ignored).

7) The resulting RLR value shall be corrected down by 14 dB for narrowband IVDS, and by 14 dB or 8 dB for wideband IVDS when one or two artificial ears are used respectively, in accordance with the requirements of [9] and [10].

If an additional volume control in receiving direction is available, the RLR shall be measured for minimum, nominal and maximum positions of the IVDS volume control.

N o t e — As a preliminary rough estimate of the IVDS sensitivity in receiving direction, the estimated transfer ratio between the electric input of the system simulator encoder at the POI and the acoustic output at the DRP may be taken in a wide band with frequency-weighting along A-curve.

The $RLR_{max}$, $RLR_{nom}$ and $RLR_{min}$ values selected by the vehicle or IVDS manufacturer are checked for their fitness to ensure the required SNR in the vehicle compartment. For this purpose, a test signal of artificial voice with the nominal level equal to minus 16 dBm(0) is applied in receiving direction at the POI, and a simulating noise of the specified level corresponding to "typical" and "worst case" noise situations is switched on in the vehicle compartment.

Voice and noise signals at the artificial air output are applied and recorded separately if the IVDS loudness in receiving direction does not depend on the level of external noise in the vehicle compartment; otherwise, the voice signal is applied while the noise signal is simulated. The recorded noise and voice signals are frequency-weighted along A-curve, their power is averaged over time (omitting any pauses between the words in the case of voice), and the ratio $A$ between the power of voice signals and the power of noise in pauses is calculated in decibels.

If the power of noise and the power of voice have been assessed separately, the obtained ratio $A$ is equal to the SNR. If the power of voice has been assessed on the noise background with no correlation of voice and noise signals, then the correction is required: the SNR threshold of 0 dB must correspond to the $A$ threshold of 3 dB, and the SNR threshold of 6 dB, to the $A$ threshold of 7 dB.

**7.2.4 SLR deviations**

7.2.4.1 Requirements

For acoustic signals sent with SPL variations in the range from minus 3 dB to plus 6 dB with respect to the nominal level, the measured loudness rating in sending direction shall not deviate by more than ±1 dB from the SLR value for signals at the nominal SPL (if an AGC in sending direction is not present or is switched off).

This requirement ensures stable sensitivity of the end-to-end path in sending direction.

7.2.4.2 Method of measurement

If the adaptive pattern of NDR amplification changes obstructs the measurements for test signals, the NRD may be disabled (or switched to its test mode).

The SLR is measured as specified above for two additional SPL values of input acoustic test signals equal to minus 28.7 dBPa and minus 19.7 dBPa at the HFRP (in accordance with 6.4.2).

For both measurements, the obtained values are compared with the SLR value for the nominal loudness level equal to minus 25.7 dBPa at the HFRP.

N o t e — The fact that the latter requirement holds is also an evidence that the IVDS microphone input is not overloaded by signals of a level higher than the nominal one.

7.2.4.3 IVDS with AGC for sending

If an AGS for sending is implemented in the IVDS and a test mode where the AGS may be disabled is available, then the measurements may be carried out in such mode. In this case, the AGS is turned off and switched to the test mode with a fixed amplification ensuring the SLR specified for a given location of the speaker.

If an AGS for sending is implemented in the IVDS but it can not be disabled in the test mode, then the SLR deviations in sending direction are not measured.

**7.2.5 RLR deviations**

7.2.5.1 Requirements

For electric signals received with variations in the range ±5 dB with respect to the nominal electric level, the measured loudness rating in receiving direction shall not deviate by more than ±1 dB from the RLR value for signals at the nominal level [with the volume control in its nominal position and with the AGC in receiving direction present not present or switched off].

This requirement ensures stable sensitivity of the end-to-end path in receiving direction.

7.2.5.2 Method of measurement

If the adaptive pattern of NDR amplification changes obstructs the measurements for test signals, the NRD may be disabled (or switched to its test mode).

The RLR is measured as specified above for two additional levels of input electric test signals equal to minus 11 dBm0 and minus 21 dBm0 at the POI.

For both measurements, the obtained values are compared with the RLR value for the nominal level of electric signals equal to minus 16 dBm0.

N o t e — The fact that the latter requirement holds is also an evidence that the IVDS acoustic output is not overloaded by signals of a level higher than the nominal one. Combining such signals with the maximum IVDS volume level may lead to overloading and distortions, and, as a consequence, may degrade the IVDS duplex capability.

7.2.5.3 IVDS with AGC for receiving

If an AGS for receiving is implemented in the IVDS and a test mode where the AGS may be disabled is available, then the measurements may be carried out in such mode. In this case, the AGS is turned off and switched to the test mode with a fixed amplification ensuring the SLR specified for a given location of the speaker.

If an AGS for receiving is implemented in the IVDS but it can not be disabled in the test mode, then the RLR deviations in receiving direction are not measured.

**7.3 Frequency sensitivity response**

7.3.1 Frequency sensitivity response of sending IVDS part

7.3.1.1 The frequency response of the IVDS sensitivity in sending direction shall be measured with the IVDS installed in the vehicle compartment (cabin), along the path from the IVDS acoustic input at the MRP to the electric output of the speech codec on the operator side at the POI of the system simulator.

7.3.1.2 Requirements

The requirements for relative tolerances in regard to the frequency response in sending direction are specified in Table 1 for narrowband IVDS, and in Table 2 for wideband IVDS. Linear interpolation on log-log scale may be used for intermediate frequencies.

An ideal frequency response in sending direction should be flat in the range from 200 Hz to 4 kHz for narrowband IVDS and from 100 Hz to 7 kHz for wideband IVDS. However (and especially when disturbing acoustic noises are present), a frequency response making use of additional frequency-weighting may be more preferable, for example, in the case where the frequency response has an LF drop and a slight HF rise within the specified tolerances.

T a b l e 1 — Frequency sensitivity response in sending direction for narrowband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 200 | 0 | – ∞ |
| 250 | 0 | – ∞ |
| 315 | 0 | –14 |
| 400 | 0 | –13 |
| 500 | 0 | –12 |
| 630 | 0 | –11 |
| 800 | 0 | –10 |
| 1000 | 0 | –8 |
| 1300 | 2 | –8 |
| 1600 | 3 | –8 |
| 2000 | 4 | –8 |
| 2500 | 4 | –8 |
| 3100 | 4 | –8 |
| 4000 | 0 | – ∞ |

T a b l e 2 — Frequency sensitivity response in sending direction for wideband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 100 | 4 | – ∞ |
| 125 | 4 | –10 |
| 200 | 4 | –4 |
| 1000 | 4 | –4 |
| 5000 | 8.5 | –4 |
| 6300 | 9 | –7 |
| 8000 | 9 | – ∞ |

Digital correction of the frequency response in sending direction is permitted (using a built-in IVDS equaliser).

7.3.1.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

2) The acoustic test signal shall be an artificial voice as per [14]. The acoustic mouth shall be calibrated by level and by frequency response at the MRP in accordance with 6.4.1. The SPL of the input test signal as per 6.4.2 at the HFRP is set to its nominal value corresponding to minus 25.7 dBPa (test signal level averaged along the full signal path).

3) The frequency sensitivity response of the IVDS is determined in one-third-octave bands in accordance with IEC 61260-1, in the frequency range from 100 Hz to 4 kHz inclusive for narrowband IVDS and from 100 Hz to 8 kHz for wideband IVDS. To calculate the signal level in each frequency band, the signal is averaged along its full path.

4) At the input, the SPL is measured in one-third-octave bands at the MRP.
5) At the output, the electric signal level is measured in one-third-octave bands at the POI.
6) The IVDS sensitivity in sending direction in each frequency band is expressed in dBV/Pa.

N o t e — As the frequency response in sending direction is measured in the closed vehicle compartment rather than in an anechoic chamber, the results will be affected by standing acoustic waves inside the vehicle compartment; thus, the frequency response may vary for different HATS manikin positions.

**7.3.2 Frequency sensitivity response of receiving IVDS part**

7.3.2.1 The frequency response of the IVDS sensitivity in receiving direction shall be measured with the IVDS installed in the vehicle compartment (cabin), along the path from the electric input of the speech codec on the operator side at the POI of the system simulator to the IVDS acoustic output at the DRP.

7.3.2.2 Requirements

The requirements for relative tolerances in regard to the frequency response in receiving direction are specified in Table 3 for narrowband IVDS, and in Table 4 for wideband IVDS. Linear interpolation in log-log scale shall be used for intermediate frequencies.

T a b l e 3 — Frequency sensitivity response in receiving direction for narrowband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 200 | 0 | – ∞ |
| 250 | 0 | – ∞ |
| 315 | 0 | – ∞ |
| 400 | 0 | –15 |
| 630 | 0 | –12 |
| 3100 | 0 | –12 |
| 4000 | 0 | – ∞ |

T a b l e 4 — Frequency sensitivity response in receiving direction for wideband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 125 | 8 | – ∞ |
| 200 | 8 | –12 |
| 250 | 8 | –9 |
| 315 | 7 | –6 |
| 400 | 6 | –6 |
| 5000 | 6 | –6 |
| 6300 | 6 | –9 |
| 8000 | 6 | – ∞ |

Digital correction of the frequency response in receiving direction is permitted (using an IVDS equaliser).

7.3.2.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

2) The electric test signal shall be an artificial voice as per [14]. The test signal level at the electric input of the system simulator at the POI is set to minus 16 dBm0 (calculated as a test signal level averaged along the full signal path).

3) The frequency sensitivity response of the IVDS is determined in one-third-octave bands as per [10] in the frequency range from 100 Hz to 4 kHz inclusive for narrowband IVDS and from 100 Hz to 8 kHz for wideband IVDS. To calculate the signal level in each frequency band, the signal is averaged along its full path.

4) At the input, the electric signal level is measured in one-third-octave bands at the POI.

5) At the output, the SPL is measured in one-third-octave bands at the DRP. The received acoustic signal is picked up from the microphone in the artificial ear of the HATS manikin's head; in advance, the latter shall be calibrated and its frequency response equalised in accordance with the requirements of [7].

6) The IVDS sensitivity in receiving direction is expressed in dBPa/V.

N o t e — As the frequency response in sending direction is measured in the closed vehicle compartment rather than in an anechoic chamber, the results will be affected by standing acoustic waves inside the vehicle compartment; thus, the frequency response may vary for different HATS manikin positions.

### 7.4 IVDS self-noise level

7.4.1 The term "IVDS self-noise" shall be understood as a total noise level in the active end-to-end voice communication channel in receiving or sending direction at the time moments when no subscriber's speech is present therein.

Such noise is composed of microphone self-noise, noise in analogue electric circuits and ADC/DAC of the IVDS, digital noise of speech encoding and DSP algorithms in the IVDS, and cumulative noise of the system simulator.

The maximum level of ambient acoustic noises inside the vehicle shall not exceed minus 64 dBPa(A) during the measurements; the levels not greater than minus 74 dBPa(A) are recommended.

The maximum level of electric self-noise at the encoder input and decoder output of the communication system simulator shall not exceed minus 74 dBm0(A) at the POI reference point.

As the noise from radio-transmitting devices that are used for wireless mobile communication and are included in the IVDS also contributes to total noise and interference levels in the acoustic communication channel, the testing shall employ different transmitter power and receiver sensitivity settings of the system simulator.

#### 7.4.2 Noise level in transmit channel

7.4.2.1 Requirements

In silence conditions when the near-end subscriber is not speaking, the maximum permitted level of the IVDS self-noise in the transmit channel measured at the electric output of the speech codec on the operator side shall not exceed minus 64 dBm0(A) for narrowband systems or for wideband systems with noise reduction in the transmit channel switched on, and minus 58 dBm0(A) for narrowband systems or for wideband systems with noise reduction in the transmit channel switched off. Individual spectral peaks in the frequency region shall not overrun the mean spectral envelope of the self-noise by more than 10 dB.

7.4.2.2 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

2) Test signals are not used. However, some IVDS systems may require a signal exceeding a certain channel activation level in order to switch to their operating transmit mode. To ensure reliable channel activation in sending direction, a provisional activation signal that includes four instances of the CSS sequence as per [11] is applied prior to measurements. The spectrum of the acoustic activation signal is first calibrated at the MRP in free-field conditions (see 6.4.1). The SPL of the activation signal at the HFRP shall be equal to minus 25.7 dBPa (see 6.4.2).

3) The idle channel noise is measured at the decoder output of the system simulator at the POI reference point in the frequency range from 100 Hz to 4 kHz for narrowband IVDS, or from 100 Hz to 8 kHz for wideband IVDS. The analysis window size for averaging shall be 1 s. The measurement starts right after the activation and its related transient processes (including reverberation of the activation signal in the vehicle compartment) are complete. The checks shall be made to ensure that the channel always remains active during the measurements. The power spectrum density of the channel noise is determined using the FFT and a running window for Hann analysis containing 8192 samples at a sampling rate of 48 kHz.

4) The noise level is calculated by weighting along A-curve. Individual spectral peaks are analysed in the frequency region with respect to the arithmetic average of a spectral envelope expressed in dBm0(A) and obtained by frequency averaging in one-third octave bands.

**7.4.3 Noise level in receive channel**

7.4.3.1 Requirements

When measured in silence conditions at the IVDS acoustic output under the nominal loudness rating $RLR_{nom}$, the maximum permitted level of the IVDS self-noise in the receive channel with no operator speech present shall not exceed minus 51 dBPa(A) with the $RLR_{nom}$ value subtracted (e.g., minus 53 dBPa(A) for $RLR_{nom}$ equal to 2 dB). Individual spectral peaks in the frequency region shall not overrun the mean spectral envelope of the self-noise by more than 10 dB.

7.4.3.2 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

2) Test signals are not used. Some IVDS systems may require signals exceeding a certain channel activation level in order to switch to their operating receive mode. To ensure reliable channel activation in receiving direction, a provisional activation signal that includes a sequence of four CSS as per [11] is applied prior to measurements. The level of the electric activation signal at the input of the system simulator shall be minus 16 dBm0 at the POI reference point. The level is measured and averaged along the full path of the activation signal.

3) The received acoustic signal is picked up from the microphone in the artificial ear of the HATS manikin's head; the latter shall preliminary be calibrated, and its frequency response equalised in accordance with the requirements of [7].

The idle channel noise is measured at the DRP in the frequency range from 50 Hz to 7 kHz for narrowband IVDS and from 50 Hz to 10 kHz for wideband IVDS. The analysis window size for averaging shall be 1 s. The measurement starts right after the activation and its related transient processes (including reverberation of the activation signal in the vehicle compartment) are complete. The checks shall be made to ensure that the channel always remains active during the measurements. The power spectrum density of the channel noise is determined using the FFT and a running window for Hann analysis containing 8192 samples at a sampling rate of 48 kHz.

4) The noise level is calculated by weighting along A-curve. Individual spectral peaks are analysed in the frequency region with respect to the arithmetic average of a spectral envelope expressed in dBPa(A) and obtained by frequency averaging in one-third-octave bands.

N o t e — The band used for noise assessment should be wider than the IVDS frequency band in order to allow for any out-of-band noise or induced interference generated by the IVDS and perceived by the closest subscriber.

**7.5 Suppression of out-of-band signals**

7.5.1 Signals with spectral components outside the IVDS operating frequency range are called out-of-band signals. They may be either external, or generated by the IVDS itself. If inadequately suppressed, they may impair the quality of communication and echo cancellation, and lead to out-of-band interferences.

The maximum level of ambient acoustic noise inside the vehicle shall not exceed minus 64 dBPa(A) during the measurements; the levels not greater than minus 74 dBPa(A) are recommended.

The maximum level of electric self-noise at the encoder input and decoder output of the communication system simulator shall not exceed minus 74 dBm0(A) at the POI reference point.

**7.5.2 Suppression of out-of-band signals in transmit channel**

7.5.2.1 If out-of-band sound signals (of a frequency above 4 kHz for narrowband IVDS or above 8 kHz for wideband IVDS) are present at the acoustic input of the IVDS and are poorly suppressed by analogue filters before the ADC, they can create out-of-band interferences in the communication channel due to superposition of frequencies in analogue-to-digital conversion.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the IVDS acoustic input at the MRP to the electric output of the speech codec on the operator side at the POI reference point.

7.5.2.2 Requirements

For input out-of-band acoustic signal of the nominal level represented by white Gaussian noise that is limited in the frequency range from 4.6 to 8 kHz for narrowband IVDS and from 9 kHz to 16 kHz for wideband IVDS, the electric level of noise at the output of the system simulator codec measured in the base frequency band from 300 Hz to 3.4 kHz for narrowband IVDS and from 100 Hz to 7 kHz for wideband IVDS shall be either below the noise level in the transmit channel, or at least 35 dB below the output level of the in-band test signal of the nominal level.

7.5.2.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If the frequency-shifting technology is used in the IVDS, it shall be disabled for the duration of these measurements.

2) To ensure reliable channel activation in sending direction, a provisional activation signal that includes a sequence of four CSS as per [11] is applied prior to measurements. The sound pressure level of the activation signal at the MRP shall be minus 4.7 dBPa (allowing for calibration as per 6.4.2). The r.m.s. level of the activation signal is calculated by averaging along the full signal path.

3) The out-of-band test signal shall be turned on right after the activation and transient processes including reverberation in the vehicle compartment are terminated, and shall exactly follow the vocalised CSS part rather than its pseudo-noise part. The duration of the out-of-band test signal shall be at least 200 ms.

4) The input out-of-band acoustic test signal is a white Gaussian noise limited in the band specified above, with its SPL at the MRP equal to minus 4.7 dBPa (allowing for calibration as per 6.4.2). Band-pass filters with suppression in the rejection band not less than 60 dB shall be used to generate the test signal.

5) The signal is analysed at the electric output of the system simulator decoder at the POI reference point. The output signal corresponding to the test range is extracted using a rectangular window of at least 200 ms in duration, and the level is evaluated in the frequency band specified above.

6) In addition, the reference output level is determined for the input in-band test signal in the form of artificial voice as per [14] limited in the band specified above, with its SPL at the MRP also equal to minus 4.7 dBPa (allowing for calibration as per 6.4.2). The level of this signal is calculated by r.m.s. averaging along its full path.

7) The measured levels are compared with each other and with the noise level that is observed in the communication channel during voice pauses.

**7.5.3 Suppression of out-of-band signals in receive channel**

7.5.3.1 Out-of-band noises in receiving direction may, for example, arise due to insufficient suppression of image frequencies in an analogue filter at the DAC output, or in digital filters if the sampling rate is changed.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the electric input of the speech codec on the operator side at the POI reference point to the IVDS acoustic output (at the point in the immediate vicinity of the IVDS speaker).

7.5.3.2 Requirements

For input electric signal in the form of artificial voice limited in the operating frequency range from 300 Hz to 3.4 kHz for narrowband IVDS and from 100 Hz to 7 kHz for wideband IVDS and applied at the level of minus 12 dBm0, the acoustic level of out-of-band noise at the IVDS output measured in the frequency band from 4.6 KHz to 8 kHz for narrowband IVDS and from 8.6 kHz to 16 kHz for wideband IVDS shall be either below the noise level in the receive channel in the said frequency band, or at least 45 dB below the output level of the main signal measured in the operating frequency band from 300 Hz to 3.4 kHz for narrowband IVDS or from 100 Hz to 7 kHz for wideband IVDS.

7.5.3.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If the frequency-shifting technology is used in the IVDS, it shall be disabled for the duration of these measurements.

2) The test signal of artificial voice is generated as per [14] at a sampling rate of 48 kHz and digitally filtered in the specified frequency band. The signal is applied to the encoder input of the system simulator at the POI reference point, with its electric level set to minus 12 dBm0.

3) The output signal is picked up from a measuring microphone located as close as possible to the IVDS speaker. This is necessary for two reasons: first, it increases the SNR of measurements compared to the case when the DRP of the artificial ear of the HATS manikin is used, and second, it enables the measurements at frequencies above 10 kHz.

4) The signal from the measuring microphone is recorded at a sampling rate of 48 kHz and filtered using digital filters of a coupling loss not less than 60 dB over the two analytic frequency bands specified above; then, the ratio between the power of spurious out-of-band signals and the power of voice signals is calculated in the primary frequency band.

**7.6 Signal distortions in IVDS**

7.6.1 The term "IVDS distortion" is understood as an overall level of harmonic distortions in the active end-to-end voice communication channel in receiving or sending direction for sine-wave test signals.

Such distortions are made up of distortions in the microphone, speaker, analogue/digital IVDS circuits, of distortions introduced by speech encoding and DSP in the IVDS, and of system simulator distortions.

The distortions introduced by speech encoding are unrecoverable, and depend on the selected speech codec as well as on its bitrate setting. Not all speech codecs (and not all IVDS noise suppression systems) are suitable for measurements because some of them are unable to pass pure tone signals. A codec of the highest bitrate and processing quality of sine-wave signals shall be selected.

The maximum level of ambient acoustic noises in the testing laboratory that penetrate inside the vehicle shall not exceed minus 64 dBPa(A) during the tests; the levels not greater than minus 74 dBPa(A) are recommended.

The maximum level of electric self-noise of the system simulator at the POI of the encoder input and at the decoder output shall not exceed minus 74 dBm0(A).

**7.6.2 Signal distortions in sending direction**

7.6.2.1 The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the IVDS acoustic input at the MRP to the electric output of the speech codec of the communication system simulator at the POI reference point on the operator side, in the frequency band up to 4 kHz for narrowband IVDS and up to 8 kHz for wideband IVDS.

7.6.2.2 Requirements

Total harmonic distortion of sine-wave signals of the nominal level in sending direction shall not exceed 3 % for each of the following test frequencies:

- 300, 500, and 1000 Hz — for narrowband IVDS;
- 300, 500, 1000, 2000, and 3000 Hz — for wideband IVDS.

7.6.2.3 Method of measurement

1) Noise suppression [8], AGC, frequency shifting, spectrum spreading and other non-linear DSP algorithms affecting propagation of pure tone signals in sending direction shall be disabled, including such algorithms in the system simulator. They will degrade measurement results otherwise.

2) Acoustic test signals shall be sine-wave signals of the SPL equal to minus 4.7 dBPa at the MRP, (allowing for calibration as per 6.4.2). The intrinsic harmonic distortion of the "artificial month" should be monitored; they shall not exceed 1 %.

3) To ensure reliable channel activation in sending direction, a provisional activation signal that includes a sequence of four CSS as per [11] is applied prior to measurements. The sound pressure level of the activation signal at the MRP shall be minus 4.7 dBPa (allowing for calibration as per 6.4.2). The r.m.s. level of the activation signal is calculated by averaging along the full signal path.

4) The sine-wave test signal is turned on right after the activation signal is terminated, and shall exactly follow the vocalised CSS part rather than its pseudo-noise (PN) part. The duration of the test signal shall be at least 200 ms.

5) The measurement is taken at the electric output of the simulator decoder at the POI reference point once the activation and its related transient processes are completed. The checks shall be made to ensure that the channel always remains active during the measurements. The power spectrum density of the output signal is determined using the FFT and a window for Hann analysis containing 8192 samples at a sampling rate of 48 kHz.

6) The total harmonic distortion of sine-wave signals is calculated in the frequency band up to 4 kHz for narrowband IVDS and up to 8 kHz for wideband IVDS, for each test signal separately.

N o t e — If a digital debug interface described in Appendix C is available for the IVDS, the level of distortions introduced by the IVDS and unaffected by the system simulator may be evaluated. Live speech is characterised by the peak factor of 12—18 dB whereas a sine-wave signal, of 3 dB; thus, an additional check at an increased level of the sine-wave test signal is required to detect disturbances of voice signals of the nominal level.

**7.6.3 Signal distortions in receiving direction**

7.6.3.1 The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the electric output of the speech codec of the communication system simulator at the POI on the operator side to the DRP, in the frequency band of up to 8 kHz for narrowband IVDS and up to 15 kHz for wideband IVDS.

7.6.3.2 Requirements

Total harmonic distortion of sine-wave signals of the nominal level in receiving direction shall not exceed 3 % at the nominal, minimum and maximum positions of the IVDS volume control, for each of the following test frequencies:

- 500, and 1000 Hz — for narrowband IVDS;
- 300, 500, 1000, 2000, and 3000 Hz — for wideband IVDS.

7.6.3.3 Method of measurement

1) Noise suppression, AGC, frequency shifting, spectrum spreading and other non-linear signal processing algorithms affecting propagation of pure tone signals in receiving direction shall be disabled, including such algorithms in the system simulator. They will degrade measurement results otherwise.

2) Test signals shall be sine-wave signals of the electric level equal to 16 dBm0 at the POI.

3) To ensure reliable channel activation in receiving direction, a provisional activation signal that includes a sequence of four CSS as per [11] is applied prior to measurements. The activation signal level at the POI is set to 16 dBm0. The r.m.s. level of the activation signal is calculated by averaging along the full signal path.

4) The sine-wave test signal is turned on right after the activation signal is terminated, and shall exactly follow the vocalised CSS part rather than its pseudo-noise part. The duration of the test signal shall be at least 200 ms.

5) The artificial ear of the HATS manikin may be replaced by a measuring microphone installed where the centre of the manikin's head would be located in other tests. This change is, in particular, justified because the frequency response of the manikin's head is stated for up to 10 kHz only in [3], but the distortions in these tests are evaluated for the frequency band of up to 15 kHz.

An alternative test method would employ a measuring microphone located as close as possible to the IVDS loudspeaker, in the same way as in out-of-band noise measurements. This enables both to increase the SNR of measurements in comparison with those performed at the DRP of the artificial ear of the HATS manikin, and to take measurements at frequencies above 10 kHz.

6) The measurement starts right after the activation signal and its related transient processes are completed. The checks shall be made to ensure that the channel always remains active. The power spectrum density is determined using the FFT and a window for Hann analysis containing 8192 samples at a sampling rate of 48 kHz.

7) The total harmonic distortion of sine-wave signals is calculated in the frequency band of up to 8 kHz for narrowband IVDS and up to 15 kHz for wideband IVDS, for each test signal separately.

N o t e — If a digital debug interface described in Appendix C is available for the IVDS, the level of distortions introduced by the IVDS and unaffected by the system simulator may be evaluated. Live speech is characterised by the peak factor of 12—18 dB whereas a sine-wave signal, of 3 dB; thus, an additional check at an increased level of the sine-wave test signal is required to detect disturbances of voice signals of the nominal level.

### 7.7 IVDS performance in single-talk mode

7.7.1 In single-talk mode where the subscribers speak one at a time and do not try to interrupt each other while speaking or hearing, echo signals and other sounding artefacts should not be noticeable.

**7.7.2 Weighted terminal coupling loss ($TCL_W$)**

7.7.2.1 As regards the electro-acoustic path in the IVDS, the terminal coupling loss is understood as a ratio between the induced electric signal level in the transmit channel at the decoder output of the communication system simulator at the POI and the initial electric signal level applied to the receive channel at the encoder input.

Since the TCL value generally depends on the signal frequency, an averaged frequency-weighted value $TCL_W$ is used for assessment of the terminal coupling loss in the IVDS.

The reasons why signals can penetrate from receive to transmit channels and lead to decreased $TCL_W$ values may be: acoustic echo signals, electric pickups, mechanical pickups, incorrect operation of DSP algorithms (in particular, of AEC).

Any echo canceller or acoustic echo suppressor implemented in the IVDS shall ensure that all echo signals are damped efficiently for all typical operating set-ups in a wide range of pulse response durations on the echo path in the vehicle compartment.

If an AES is used to damp echo signals by introducing an extra attenuation in the transmit channel when an active speech is present in the receive channel, then each time the AES suppresses input signals in the transmit channel it shall replace them by generating a "comfort noise" signal that is close to the background pause noise in regard to its energy and tone so as to mask any switchover effects and improve the perception of the transmitted speech by the far-end subscriber.

Due to relatively long signal delays that occur in the IVDS and in communication networks, echo signals become largely retarded and thus more apparent. Therefore, the requirements in regard to $TCL_W$ are more stringent for the IVDS than for local wired communications, and are much harder to meet because of high levels of acoustic echo signals in loudspeaker communication, varying acoustic features of vehicle compartments, changing parameters of echo paths, and for other reasons.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), from the electric input to the electric output of the speech codec of the system simulator on the operator side.

7.7.2.2 Requirements

When the PN test signal of the maximum level is routed to the receive channel and no external acoustic noise is present in the vehicle compartment, the weighted terminal coupling loss $TCL_W$ along the electro-acoustic path for echo signals in the transmit channel shall be at least 46 dB (at least 50 dB is recommended) at the nominal position of the volume control ($RLR_{nom}$) or at least 40 dB for the maximum volume ($RLR_{min}$) after the time period required to complete configuration of all factors for the acoustic echo canceller (AEC).

7.7.2.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

The maximum level of ambient acoustic noises in the testing laboratory that penetrate inside the vehicle shall not exceed minus 64 dBPa(A) during the tests; the levels not greater than minus 74 dBPa(A) are recommended.

The level of self-noise in the transmit channel measured at the decoder output of the system simulator at the POI of shall not exceed minus 58 dBm0(A).

2) Prior to $TCL_W$ measurement, the AEC shall be configured for the largest echo signal attenuation by applying a training test signal in the form of a sequence containing 10 s of male and 10 s of female artificial voices generated in accordance with [14]. The signal level of the training sequence shall be minus 16 dBm0.

3) The $TCL_W$ electric signal attenuation between the encoder input and decoder output of the system simulator at the POI reference point is measured using a speech-like test signal which is a pseudo noise (PN) sequence as per [11] containing 4096 samples (at a sampling rate of 48 kHz) with its peak factor equal to 6 dB. The duration of the test signal shall be 250 ms, and its level shall be minus 3 dBm0. Low peak factor of the test signal including numerous sine-wave signals is achieved by random switchover of their phases from minus 180° to plus 180°.

4) Signals at the encoder input and decoder output of the communication simulator are recorded at the POI. The $TCL_W$ is measured as per [17] (clause B.4, trapezoidal rule). The mean energy of the test signal and of echo signals is calculated for each frequency band using a time window of 250 ms (with the respective time-shift relative to the test signal, to make provision for the delay of echo signals).

When the $TCL_W$ is measured, it should be checked that the signal analysed at the decoder output is indeed the echo signal rather than self-noise of the IVDS transmit channel, "comfort noise" injected in this channel in order to mask suppression of echo signals, transmitted background noise signal, or direct electric disturbance (e.g., induced by IVDS loudspeakers).

**7.7.3 Temporal stability of echo signal attenuation**

7.7.3.1 This test is intended for verification of stable echo signal attenuation in single-talk mode of the far-end subscriber when no interfering acoustic noises are present in the vehicle compartment.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), from the electric input to the electric output of the speech codec of the system simulator on the operator side.

7.7.3.2 Requirements

After the CSS and the test artificial voice signal of the nominal level are applied to the IVDS receive channel, the echo signal attenuation in the IVDS transmit channel shall not decrease by more than 6 dB from its maximum value for a long time period provided that the echo path inside the vehicle is stable, and the measurement is carried out at least 5 s after the signal start time and only for those signal sections that include active speech.

7.7.3.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

2) The test signal shall be a periodically repeated CSS corresponding to [11]. Two medium signal levels are checked: minus 5 dBm0 and minus 25 dBm0. Signal duration of at least 2.8 s is analysed; this corresponds to eight CSS periods excluding the pauses. Then, the test is repeated for an artificial voice signal as per [14]. One sequence of "male" voice and one sequence of "female" voice with an average level of 16 dBm0 are used. The analysis is carried out for the complete signal duration. The controlled parameter in the measurements using CSS is the echo signal attenuation. The controlled parameter in the measurements using an artificial voice is the echo signal level.

3) When the original and echo signal levels are evaluated, the integration time constant shall be 35 ms. After the envelopes are calculated, their ratio shall be determined providing that the signals are accurately synchronised. The curve of echo signal suppression versus time is plotted.

**7.7.4 Frequency dependence of echo signal attenuation**

7.7.4.1 The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), from the electric input to the electric output of the speech codec of the system simulator at the POI.

7.7.4.2 Requirements

When the test CSS of the nominal level is routed to the IVDS receive channel, the frequency curve of echo signal attenuation in the transmit channel shall be below the limits specified in Table 5 for narrowband IVDS and in Table 6 for wideband IVDS.

T a b l e 5 — Frequency dependence of echo signal attenuation in narrowband IVDS

| Frequency, Hz | Upper limit, dB, negated |
|---|---|
| 100 | 20 |
| 200 | 30 |
| 300 | 38 |
| 800 | 34 |
| 1500 | 33 |
| 2600 | 24 |
| 4000 | 24 |

Table 6 — Frequency dependence of echo signal attenuation in wideband IVDS

| Frequency, Hz | Upper limit, dB, negated |
|---|---|
| 100 | 41 |
| 1300 | 41 |
| 3450 | 46 |
| 5200 | 46 |
| 7500 | 37 |
| 8000 | 37 |

The loss values for intermediate frequencies may be linearly interpolated when a log scale for the frequency and a linear scale for the attenuation in dB are used.

These requirements shall be met at all times; therefore, they shall be checked for different time slices of the test signal.

The check should be made during the test that the signal measured in the transmit channel is indeed the echo signal rather than self-noise of the IVDS transmit channel, "comfort noise" injected in this channel in order to mask suppression of echo signals, transmitted background noise signal, or direct electric disturbance (e.g., induced by IVDS loudspeakers).

7.7.4.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6.

2) Prior to measurements, the AEC shall be set up by applying a training test signal in the form of a sequence containing 10 s of male and 10 s of female artificial voice generated in accordance with [14]. The signal level of the training sequence shall be minus 16 dBm0.

3) The test signal shall be a periodically repeated CSS. The measurement is carried out in steady state conditions. The test signal level shall be equal to minus 16 dBm0. A sequence of four CSS instances of total duration 1.4 s including the pauses is used for measurements.

4) The ratio between the power spectrum density of the measured echo signals and the one of the original test signals is used to evaluate the frequency dependence of echo signal attenuation. The analysis is carried out using the FFT for 8192 samples at a sampling rate of 48 kHz, and a rectangular window.

5) The curve of echo signal attenuation versus frequency is plotted.

**7.7.5 Initial convergence of AEC in silence**

7.7.5.1 As a rule, an initial AEC set-up for yet unknown parameters of the echo path during single talk with no acoustic noise present depends on the convergence rate of adaptive filter factors in the AEC, and is described by the dependence of the echo signal loss on the time passed since the AEC start-up.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin). The echo return loss (ERL) is understood as an attenuation of an electric signal along the path from the encoder input to the decoder output of the system simulator at the POI reference point.

7.7.5.2 Requirements

After the CSS of the nominal level is applied to the IVDS receive channel, the ERL curve for echo signals in the IVDS transmit channel versus the time passed after the initial start-up of the AEC with the volume control set to its nominal level shall be below the limits shown in Figure 8a).

After the test artificial voice signal of the nominal level is applied to the IVDS receive channel, the ERL curve for echo signals in the IVDS transmit channel versus time passed after the initial start-up of the AEC with the volume control set to its nominal level shall be below the limits shown in Figure 8b).

Special attention should be given to the IVDS behaviour at the moment when the AEC is switched on (when the connection with the operator is established). The system shall remain stable for any position of the volume control, i.e., shall ensure that the TCL along the electro-acoustic path is not less than 6 dB in all operating frequency range at any time moment, and transient processes are not accompanied with abrupt loudness jumps, noise bursts, or excitation of tone signals.

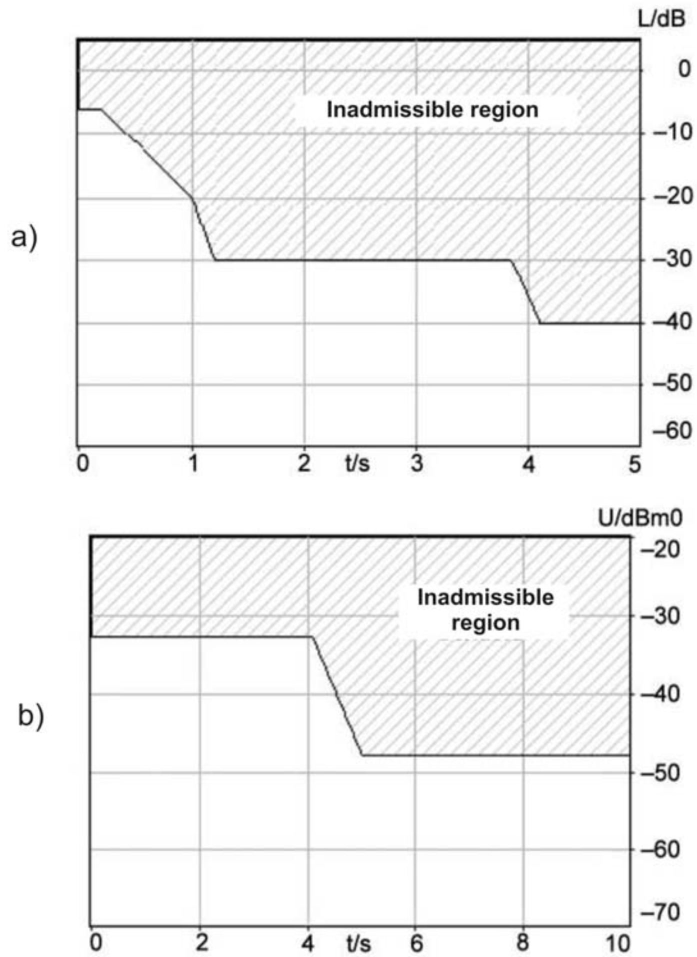



Fig. 8 — Time dependence of echo return loss *L* and echo signal level *U*

7.7.5.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6. The noise level in the receive channel measured at the decoder output of the system simulator at the POI shall not exceed minus 63 dBm0.

2) The test signal is applied immediately after the connection is established and the volume control is set to its nominal level.

3) The first test signal is a periodically repeated CSS as per [11]. The average signal level shall be minus 16 dBm0. The echo signal shall be analysed for at least a 5 s interval after the connection is established at the active test signal level (excluding the pauses).

4) Then, the test is repeated for the second test signal of artificial voice as per [14]. One sequence for "male" voice and one sequence for "female" voice with an average level of minus 16 dBm0 are used. The analysis covers all voice signal duration (excluding the pauses). The echo signal shall be analysed for at least 5 s after the connection is established. Since the convergence rate of the adaptive filter factors in the AEC depends on the signal type, different initial signal points are selected as starting points when the convergence process is checked.

5) The integration time constant used to assess the original and echo signal levels shall be 35 ms. After the envelopes of direct and return signals are calculated, their ratio (for the CSS) and level (for the artificial voice) shall be determined providing that the signals are accurately synchronised. The time curves of the echo signal suppression for the CSS and of the echo signal level for the artificial voice are plotted.

**7.7.6 Initial convergence of AEC in presence of noise**

7.7.6.1 As a rule, an initial AEC set-up for yet unknown parameters of the echo path during single talk with acoustic noise present depends not only on the convergence of adaptive filter factors in the AEC, but also on the character and level of acoustic noise. This being the case, the adaptive filter operation stops as soon as the residual echo signal level becomes equivalent to the pause noise. So the AEC set-up process is described by the dependence of the ratio between the residual echo signal level and the pause noise level on the time passed since the AEC start-up.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the electric input to the electric output of the speech codec of the communication simulator at the POI, with simultaneous simulation of acoustic noises of different in the vehicle cabin.

7.7.6.2 Requirements

After the CSS and the test artificial voice signal of the nominal level are applied to the IVDS receive channel, the curves of the ratio *L* between the residual echo signal level and the pause noise level versus time passed after the initial start-up of the AEC with the volume control at its maximum position shall lie below the limits shown in Figure 9.

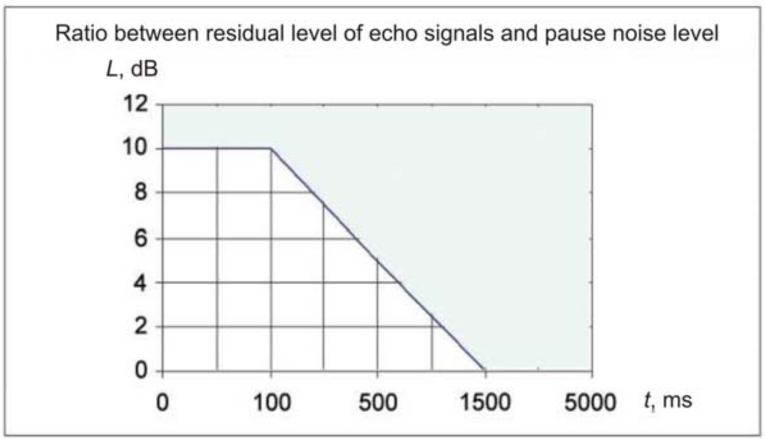



Fig. 9 — Time dependence of ratio between residual level of echo signals and pause noise level

7.7.6.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. Noise shall be turned on at least 5 s prior to measurements. This will allow adaptive algorithms such as AGC or NRD included in the IVDS to configure their parameters, and to achieve a steady state.

3) The test signal is applied immediately after the connection is established and the volume control is set to its maximum level.

4) The first test signal is a periodically repeated CSS as per [11]. The average signal level shall be minus 16 dBm0. The echo signal shall be analysed for a period of at least 5 s, excluding the pauses. Then, the test is repeated for the second test signal of artificial voice as per [14]. One sequence for "male" voice and one sequence for "female" voice with an average level of minus 16 dBm0 are used. The echo signal shall be analysed for a period of at least 5 s, excluding the pauses. Since the convergence rate depends on the signal type, different initial signal points are selected as starting points when the convergence process is checked.

5) The integration time constant used to assess echo signal levels shall be 35 ms. After the envelopes of echo signals and pause noise are calculated, the time curve of the echo return loss shall be plotted.

**7.7.7 Echo signal loss depending on echo path changes**

7.7.7.1 After the EAC is configured, it shall be capable of adapting and maintaining the required echo signal suppression under continual echo path changes inside the vehicle (e.g., while the passengers are moving in it).

The acoustic echo path with time-changing parameters is modelled inside the vehicle by rotating a reflective rectangular screen 30 cm wide and 40 cm high (for example, a piece of cardboard, veneer or plastic) located symmetrically on the passenger seat next to the driver and co-centred with the HATS manikin. The initial position of the reflecting surface (zero angle position) corresponds to the one of a plane perpendicular to the windscreen of the vehicle. It is rotated clockwise (as seen from top) to an angle of 90° where the surface plane is parallel to the windscreen, then rotated back. The surface is rotated continuously between 0° and 90° positions at a rate of 90°/s. This way an additional reflected signal beam from the speaker to the microphone is modelled with time dependent properties. To obtain reproducible results, the rotation of the reflecting surface shall be synchronised with the start of test signals using the control channel.

7.7.7.2 Requirements

The degradation of echo signal suppression after echo path changes in the vehicle compartment shall not exceed 6 dB for a signal level of minus 25 dBm0 or 15 dB for a signal level of minus 16 dBm0 with respect to the maximum value observed during the test of the echo path with constant parameters.

7.7.7.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) The reflecting surface shall not move before the measurements, and the AEC shall be allowed to configure itself completely.

3) The time when the reflective surface starts rotating from the position 0° shall be synchronised with the time when the playback of test signals starts.

4) The test signal is a periodically repeated CSS as per [11], of an average level equal to minus 16 dBm0 or to minus 25 dBm0. The signal is analysed for at least 2.8 s; this corresponds to eight CSS periods, excluding the pauses. Then, the test is repeated for the second test signal of artificial voice as per [14]. One sequence for "male" voice and one sequence for "female" voice with an average level of minus 16 dBm0 are used. The levels shall be analysed for all duration of test signals.

5) The integration time constant used to assess the original and echo signal levels shall be 35 ms. After the envelopes of direct and echo signals are calculated, their ratio shall be determined providing that the signals are accurately synchronised. The curve of echo signal suppression versus time is plotted. The degradation of suppression due to continuously varying echo path parameters is evaluated.

### 7.8 Performance of voice direction switching

7.8.1 If AES algorithms, combined AES/AEC, speech activity detectors or other DSP algorithms are used in the IVDS, the IVDS characteristics that pertain to activation of receive and transmit channels as well as to switching of voice direction in half-duplex mode shall be checked.

**7.8.2 Channel activation in sending direction**

7.8.2.1 The process of channel activation (turning-on) in sending direction is described using two parameters: the minimum turn-on time $T_{r,S,min}$ and the minimum acoustic level of activation $L_{S,min}$.

The minimum activation level is defined as a minimum level of the transmitted signal that is required to turn on the transmit channel, i.e., to completely remove signal attenuation taking place in inactive state. The time of channel activation is a time required to start up the channel when the signal of a level higher than the activation threshold is applied.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the acoustic input of the IVDS at the MRP to the electric output of the speech codec of the system simulator at the POI.

7.8.2.2 Requirements

The activation level $L_{S,min}$ measured for active areas of voice signals shall not exceed minus 20 dBPa. The activation time $T_{r,S,min}$ for input signals of the minimum activation level shall not exceed 50 ms.

7.8.2.3 Method of measurement

The waveform of a test signal is shown in Figure 10. The test signal is a CSS sequence as per [11] of gradually increased levels with pauses in-between. The test signal parameters are given in Table 7.

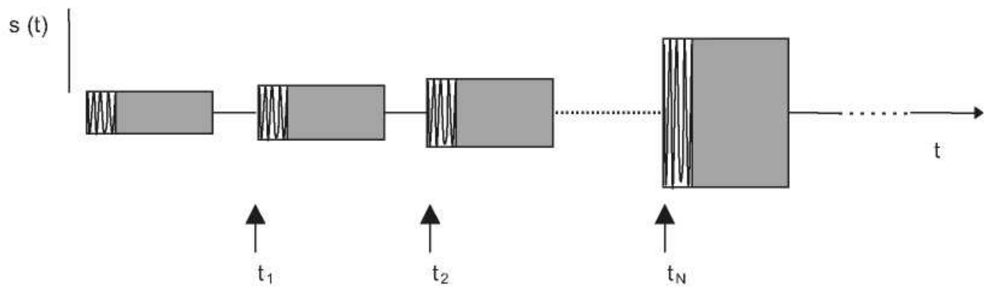



Fig. 10 — Test signal for evaluation of minimum activation level and channel activation time [10]

T a b l e 7 — Parameters of test signal in sending direction

| Designation of test signal | CSS/pause duration | Active level of first CSS (at MRP) | Level increment between two periods |
|---|---|---|---|
| Signal for evaluation of channel activation performance in sending direction | 248.62 ms/451.38 ms | minus 23 dBPa | 1 dB |

It is assumed that the pause duration of 451.38 ms is always longer than the time required to return to inactive state, and the channel is able to return to it after each CSS period.

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) An input acoustic test signal is applied through an "artificial mouth" of the HATS manikin, and its level is monitored at the MRP.

3) The electric signal at the decoder output of the communication system simulator is recorded at the POI. The recorded signal is time-synchronised with the original test signal, and its level is evaluated over time using an integration time constant of 5 ms.

4) The minimum activation level is determined as a CSS level which resulted in full activation of the transmit channel. The activation time is determined as a time between the CSS start and the full channel activation.

The level is only measured for active regions of the test CSS, and thus it is slightly higher than the average CSS level [11] that includes the pause 101.38 ms in duration. For example, an active part level of minus 23 dBPa corresponds to the average signal level equal to minus 24.7 dBPa.

If the measurements using CSS can not provide for exact evaluation of the minimum activation level, they may be repeated using the recording of a particular single-syllable word, e.g., "test." The technique used for generation of test signal levels and pause durations shall remain the same.

**7.8.3 Channel activation in receiving direction**

7.8.3.1 The process of channel activation in receiving direction is described using two parameters: the minimum turn-on $T_{r,R,min}$ and the minimum electric level of activation $T_{R,min}$.

The minimum activation level is defined as a minimum level of the received signal that is required to turn on the receive channel, i.e., to completely remove signal attenuation taking place in inactive state. The time of channel activation is a time required to turn on the channel when the signal of a level higher than the activation threshold is applied.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the electric input of the speech codec of the system simulator at the POI to the acoustic output of the IVDS. Instead of the artificial ear of the HATS manikin, a measuring microphone installed close to the IVDS speaker is used in the test in order to ensure high accuracy of signal recordings in receiving direction.

7.8.3.2 Requirements

The level $L_{R,min}$ measured for active regions of test signals shall not exceed 35.7 dBm0. The activation time $T_{r,R,min}$ for an input signal of the minimum activation level shall not be longer than 50 ms.

7.8.3.3 Method of measurement

The waveform of test signals is shown in Figure 10. The test signal is a CSS sequence as per [11] of gradually increased levels with pauses in-between. The test signal parameters are listed in Table 8.

T a b l e 8 — Parameters of test signal in receiving direction

| Designation of test signal | CSS/pause duration | Active level of first CSS (at MRP) | Level increment between two periods |
|---|---|---|---|
| Signal for evaluation of channel activation performance in receiving direction | 248.62 ms/451.38 ms | -38.7 dBm0 | 1 dB |

It is assumed that the pause duration of 451.38 ms is always longer than the time required to return to inactive state, and the channel is able to return to it after each CSS period.

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) An input electric test signal is applied to the speech codec input of the communication system simulator at the POI reference point.

3) The acoustic signal at the output of the receive channel is recorded using a microphone installed next to the IVDS speaker. The recorded signal is time-synchronised with the original test signal, and its level is evaluated over time using an integration time constant of 5 ms.

4) The minimum activation level is determined as a CSS level which resulted in full activation of the receive channel. The activation time is determined as a time between the CSS start and the full channel activation.

The level is only measured for active regions of the test CSS, and thus it is slightly higher than the average CSS level [11] that includes the pause 101.38 ms in duration. For example, an active part level of minus 38.7 dBm0 corresponds to the average signal level equal to minus 40 dBm0.

If the measurements using CSS can not provide for exact evaluation of the minimum activation level, they may be repeated using the recording of a particular single-syllable word, e.g., "test." The technique used for generation of test signal levels and pause durations shall remain the same.

**7.8.4 Attenuation in transmit channel during reception**

7.8.4.1 When the subscribers are talking one at a time (in half-duplex mode), an IVDS may induce attenuation in the transmit channel to mitigate echo signals if the receive channel is currently active.

The attenuation in sending direction is described using two parameters: attenuation value $A_{H,S}$ and attenuation turn-off time (switching from receiving to sending direction) $T_{r,S}$.

7.8.4.2 Requirements

The value $A_{H,S}$ of attenuation induced by the IVDS in the transmit channel when the receive channel is active shall not exceed 20 dB, and the attenuation turn-off time (switching from receiving to sending direction) $T_{r,S}$ for signals of a nominal level shall not exceed 50 ms. The recommended approach is to achieve an attenuation less than 13 dB for a time interval no longer than 15 ms.

7.8.4.3 Method of measurement

A pair of time-synchronised test signals is used. In receiving direction, a CSS sequence as per [11] of a nominal level sufficient for receive channel activation is applied to the electric input of the system simulator encoder. In sending direction, a vocalised sound of a level higher than the nominal one is then applied to the acoustic input of the IVDS. The waveform diagram of signals is shown in Figure 11. The test signal parameters are listed in Table 9.

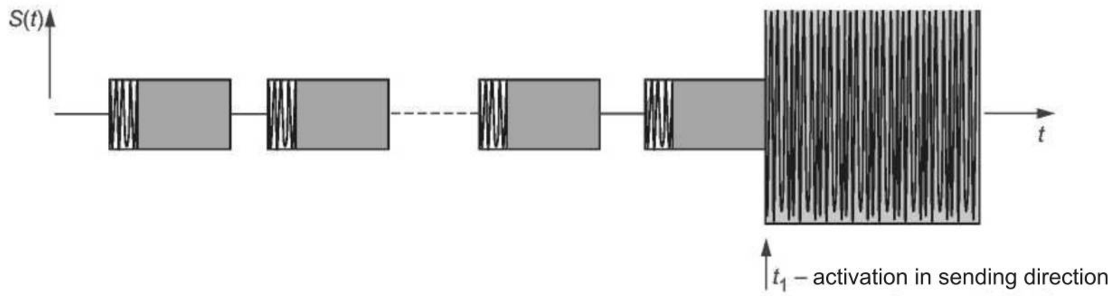



Fig. 11 — Waveform diagram of signals for measurement of attenuation in transmit channel [10].

Table 9 — Signal levels for attenuation measurements in transmit channel

| Measured value | Receiving direction (CSS at POI) | Sending direction (voice at MRP) |
|---|---|---|
| Average signal level | –16 dBm0 (including 101.38 ms pause) | 0 dBPa |
| Active signal level | –14.7 dBm0 | 0 dBPa |

Test signals shall be time-synchronised at the acoustic interface of the IVDS taking into account the total signal propagation delay in the receive channel (this delay must be constant).

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) First, an activation signal in receiving direction in the form of CSS of a nominal average level equal to minus 16 dBm0 is applied to the electric input of the system simulator encoder.

3) Once the activation signal is over, a test signal (vocalised sound) in sending direction is applied to the acoustic input of the IVDS.

4) The signal at the electric output of the system simulator decoder is recorded, and the changes of its level calculated with an integration time constant of 5 ms are analysed over time. The attenuation value $A_{H,S}$ is determined as a difference between the measured signal levels at the signal start (point $t_1$ in Figure 11) and at the time of full channel activation channel in sending direction, and the activation time $T_{r,S}$ is determined as a difference between the latter moments.

**7.8.5 Attenuation in receive channel during transmission**

7.8.5.1 When the subscribers are talking one at a time (in half-duplex mode), an IVDS may induce attenuation in the receive channel to mitigate echo signals if the transmit channel is currently active.

The attenuation in receiving direction is described using two parameters: attenuation value $A_{H\ R}$ and attenuation turn-off time (switching from sending to receiving direction) $T_{r,R}$.

7.8.5.2 Requirements

The value $A_{H,R}$ of attenuation induced by the IVDS in the receive channel when the transmit channel is active shall not exceed 15 dB, and the attenuation turn-off time (switching from sending to receiving direction) $T_{r,R}$ for signals of a nominal level shall not exceed 50 ms. The recommended approach is to achieve an attenuation less than 9 dB for a time interval no longer than 15 ms.

7.8.5.3 Method of measurement

A pair of time-synchronised test signals is used. In sending direction, a CSS sequence as per [11] of a nominal level sufficient for transmit channel activation is applied to the acoustic input of the IVDS. In receiving direction, a vocalised sound of a level higher than the nominal one is then applied to the electric input of system simulator encoder. The waveform diagram of signals is shown in Figure 11. The parameters of test signals are listed in Table 10.

T a b l e 10 — Signal levels for attenuation measurements in receive channel

| Measured value | Receiving direction (voice at POI) | Sending direction (CSS at MRP) |
|---|---|---|
| Average signal level | –14.7 dBm0 | –1.7 dBPa (including 101.38 ms pause) |
| Active signal level | –14.7 dBm0 | 0 dBPa |

Test signals shall be time-synchronised at the acoustic interface of the IVDS taking into account the total signal propagation delay in the receive channel (this delay must be constant).

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) First, an activation signal in sending direction in the form of CSS of a nominal average level equal to minus 1.7 dBPa is applied to the acoustic input of the IVDS.

3) Once the activation signal is over, a test signal (vocalised sound) in receiving direction is applied to the electric input of the system simulator encoder.

4) The signal at the acoustic output of the IVDS is recorded, and the changes of its level calculated with an integration time constant of 5 ms are analysed over time. The attenuation value $A_{H,R}$ is determined as a difference between the measured signal levels at the signal start (point $t_1$ in Figure 11) and at the time of full channel activation channel in sending direction, and the activation time $T_{r,R}$ is determined as a difference between the latter moments.

### 7.9 IVDS performance in double-talk mode

7.9.1 This section deals with the measurements of those IVDS characteristics that pertain to activation of receive and transmit channels as well as to switching of voice direction in double-talk mode where both subscribers try talking and listening to each other at the same time, interrupting one another. Depending on the quality class, an IVDS may provide for full duplex, partial duplex or only half duplex mode (where the communication link is captured by one of the subscribers).

The speech quality in double-talk mode is mainly dependent on distortions and spurious side-tones caused by echo signals and by AEC operation, as well as on loudness changes (jumps) taking place during the switches from single-talk to double-talk mode and vice versa due to activation/deactivation of additional signal attenuation in the receive and transmit channels (AES functions).

In order to ensure the required quality of communication in full duplex mode, the primary attenuation of noise signals in the AEC should be as high as possible, whereas all additional attenuation introduced in the receive and transmit channels should be as low as possible.

The most important IVDS parameters that govern the speech quality in duplex mode are: attenuation in the transmit channel during double-talk $A_{H,S,dt}$, attenuation in the receive channel during double-talk $A_{H,R,dt}$, and attenuation of echo signals in the AEC during double-talk $EL_{W,dt}$.

The value of $A_{H,S,dt}$ defines how noticeable the loudness jumps are in the transmit channel when the switching between single-talk and double-talk occurs, and the value of $A_{H,R,dt}$ defines how noticeable they are in the receive channel for such switching.

#### 7.9.2 Attenuation in transmit channel in double-talk mode

7.9.2.1 During double-talk, an IVDS may introduce extra attenuation $A_{H,S,dt}$ in the transmit channel in order to damp acoustic echo signals sneaking from the receive channel. In practice, this leads to loudness jumps of the near-end subscriber's speech in the transmit channel during the switches from single-talk to double-talk mode and back.

7.9.2.2 Requirements

The maximum attenuation $A_{H,S,dt}$ introduced by an IVDS in the transmit channel during double-talk depends on the IVDS performance (quality class) as regards its duplex communication capability, and shall correspond to the values specified in Table 11.

T a b l e 11 — IVDS performance parameters for duplex communication

| Parameter | Value of $A_{H,S,dt}$ vs. quality class | | | | |
|---|---|---|---|---|---|
| | 1 | 2a | 2b | 2c | 3 |
| | Full duplex | Partial duplex | | | Half duplex |
| $A_{H,S,dt}$. dB | ≤ 3 | ≤ 6 | ≤ 9 | ≤ 12 | > 12 |

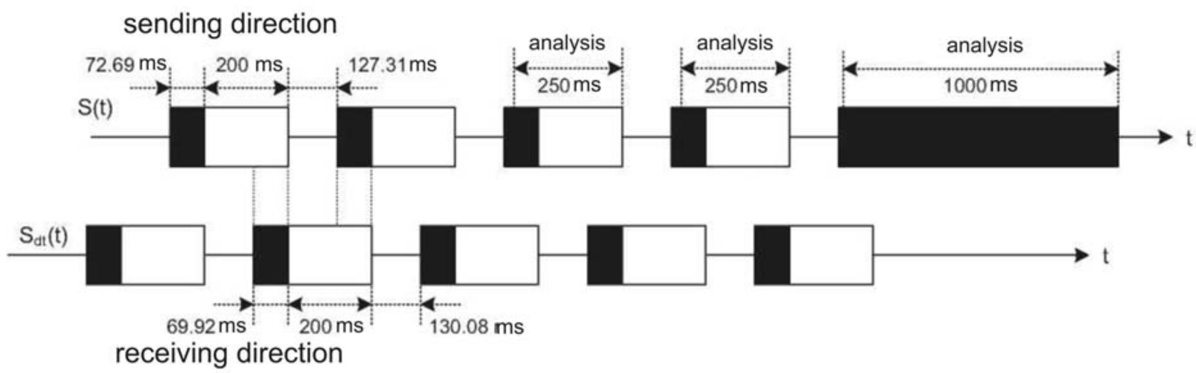
The requirements shall hold both for nominal signal levels in sending/receiving directions and for disbalances of these levels as listed below. The following two combinations of signal levels shall be checked:

- nominal signal levels in sending/receiving direction;
- signal level in sending direction is 6 dB higher, signal level in receiving direction is 6 dB lower.

Table 11 includes the requirements for $A_{H,S,dt}$ that are necessary to attribute an IVDS to a particular quality class.

7.9.2.3 Method of measurement

The test signals used to determine the range of attenuation jumps $A_{H,S,dt}$ during double-talk are shown in Figure 12. Two sequences of non-correlated CSS are used; they are applied to the transmit channel and receive channel simultaneously with partial overlapping in time to model the double-talk effect. The length and shape of test sequences are shown in Figure 12. The signals shall be synchronised at the acoustic interface point as shown in Figure 12, and the receive signal delay in network transfers shall be fixed.




$s(t)$ — signal in sending direction, $s_{dt}(f)$ — signal in receiving direction

Fig. 12 — Test signal for evaluation of attenuation range in sending direction during double-talk

The initial portion of each CSS period (vocalised sound painted black) in one direction is overlapped by the final portion of each CSS period (pseudo-noise painted white) in opposite direction. The analysis is performed over the active signal intervals in sending direction that are shown in Figure 12. The test signal parameters are listed in Table 12.

T a b l e 12 — Parameters of test signals for double-talk simulation

| Parameter | Receiving direction (*POI* point) | Sending direction (*MRP* point) |
|---|---|---|
| Vocalised part | 69.92 ms | 72.69 ms |
| Pseudo-noise part | 200 ms | 200 ms |
| Pause between signals | 130.08 ms | 127.31 ms |
| Average signal level (including 101.38 ms pause) | –16 dBm0 | –1.7 dBPa |
| Active signal level | –14.7 dBm0 | 0 dBPa |

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) Before testing, the AEC shall be configured for the highest echo suppression using a training sequence in the receive channel consisting of 10 s of male and 10 s of female voice as per [14] with a level of 16 dBm0, applied at the electric input of the system simulator encoder at the POI reference point.

3) The test signal for sending is applied at the acoustic input of the IVDS at the MRP. The test signal for receiving is applied at the electric input of the system simulator encoder at the POI. The measurements are carried out at the electric output of the system simulator decoder at the POI.

4) The signal level in the transmit channel is evaluated in the time region with an integration time constant equal to 5 ms. The curve of signal level versus time is plotted. The signal attenuation in the transmit channel is evaluated by comparison of the signal level during double-talk and the one during single-talk (in pauses of the signal in receiving direction) when the transmit channel is fully activated. All the test sequence starting from the second CSS period is analysed.

5) The test is repeated for all combinations of signal levels.

**7.9.3 Attenuation in receive channel in double-talk mode**

7.9.3.1 During double-talk, an ECSD may introduce extra attenuation $A_{H,R,dt}$ to the receive channel in order to damp acoustic echo signals sneaking from the transmit channel. In practice, this leads to loudness jumps of the far subscriber's speech in the receive channels during the switches from single-talk to double-talk mode and back.

7.9.3.2 Requirements

The maximum attenuation $A_{H,R,dt}$ introduced by an IVDS in the receive channel during double-talk depends on the IVDS performance (quality class) as regards its duplex communication capability, and shall correspond to the values specified in Table 13.

T a b l e 13 — IVDS performance parameters for duplex communication

| Parameter | Value of $A_{H,R,dt}$ vs. quality class | | | | |
|---|---|---|---|---|---|
| | 1 | 2a | 2b | 2c | 3 |
| | Full duplex | Partial duplex | | | Half-duplex |
| $A_{H,R,dt}$, dB | ≤ 3 | ≤ 5 | ≤ 8 | ≤ 10 | > 10 |

The requirements shall hold both for nominal signal levels in sending/receiving directions and for disbalances of these levels as listed below. The following two combinations of signal levels shall be checked:

- nominal signal levels in sending/receiving direction;
- signal level in sending direction is 6 dB higher, signal level in receiving direction is 6 dB lower.

Table 13 includes the requirements for $A_{H,R,dt}$ that are necessary to attribute an IVDS to a particular quality class.

7.9.3.3 Method of measurement

In order to ensure high accuracy of signal recordings in receiving direction, a measuring microphone installed as close as possible to the IVDS speaker is used instead of the artificial ear of the HATS manikin.

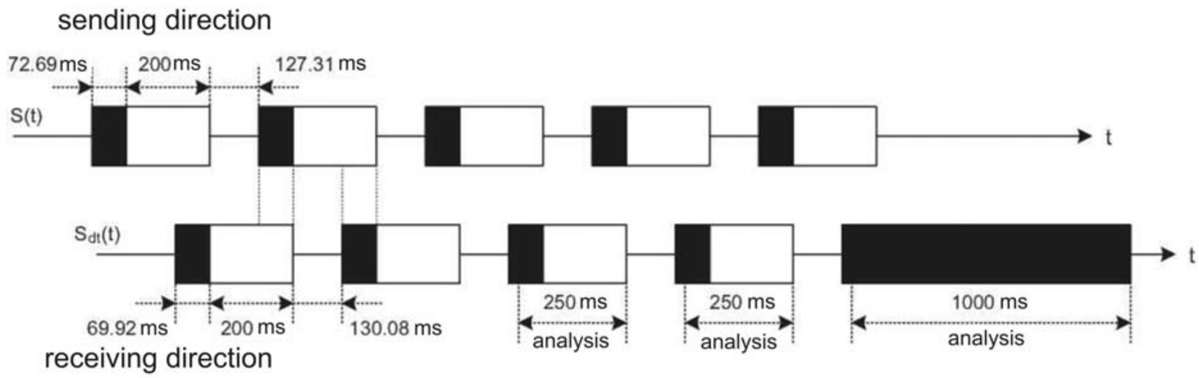
However, even such placement of the microphone does not protect the analysed signal in receiving direction from direct overlapping with echo signals and acoustic signals in sending direction, thereby making evaluation of its level more complex. So, great care shall be exercised during the measurements.

Test signals used to determine the range of attenuation jumps $A_{H,R,dt}$ during double-talk are analogous to those used for evaluation of $A_{H,S,dt}$ in 7.9.3.2. They are shown in Figure 13.

The analysis is performed over the active signal intervals in receiving direction that are shown in Figure. The test signal parameters are listed in Table 12.

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) Before testing, the AEC shall be configured for the highest echo suppression using a training sequence in the receive channel consisting of 10 s of male and 10 s of female voice as per [14] with a level of 16 dBm0, applied at the electric input of the system simulator encoder at the POI reference point.




s(t) — signal in receiving direction, $s_{dt}(t)$ — signal in sending direction

Fig. 13 — Test signals for evaluation of attenuation range in receiving direction during double-talk

3) The test signal for sending is applied at the acoustic input of the IVDS at the MRP. The test signal for receiving is applied at the electric input of the system simulator encoder at the POI. The measurements are carried out at the acoustic output of the IVDS using a measuring microphone located close to the speaker.

4) The signal level in the receive channel is evaluated in the time region with an integration time constant equal to 5 ms. The curve of signal level versus time is plotted. The signal attenuation in the receive channel is evaluated by comparison of the signal level during double-talk and the one during single-talk (in pauses of the signal in sending direction) when the transmit channel is fully activated. All the test sequence starting from the second CSS period is analysed.

5) The test is repeated for all combinations of signal levels.

N o t e — The electric output of the IVDS may be used for evaluation of signal levels. This signal pickup method is free of acoustic superposition of a signal from the near-end subscriber.

**7.9.4 Attenuation of echo signals in double-talk mode**

7.9.4.1 The attenuation of echo signals in double-talk mode $EL_{dt}$ is one of the key quality criteria, and the basis the very possibility of full-fledged duplex communication in loudspeaker mode rests upon.

A precise measurement of this parameter is only conceivable when the residual echo signal is separated from the speech of the near-end subscriber at the transmit channel output. The latter is not feasible in voice signal tests due to overlaying of spectra. Therefore, special test signals that include two orthogonal sets of sine-wave signals are used.

The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), from the electric input to the electric output of the speech codec of the system simulator at the POI reference point.

7.9.4.2 Requirements

The minimum permitted attenuation of echo signals $EL_{dt}$ during double-talk of subscribers (in duplex mode) depends on the IVDS performance type in regard to duplex communication, and shall comply with the values specified in Table 14.

T a b l e 14 — IVDS performance parameters for duplex communication

| Parameter | Value of $EL_{dt}$ vs. quality class | | | | |
|---|---|---|---|---|---|
| | 1 | 2a | 2b | 2c | 3 |
| | Full duplex | Partial duplex | | | Half-duplex |
| $EL_{dt}$, dB | ≥ 27 | ≥ 23 | ≥ 17 | ≥ 11 | < 11 |

7.9.4.3 Test signals

Test signals (synthetic vocalised sound) consist of two orthogonal sets of frequency and phase modulated sine-wave signals with a spectral envelope similar to vocalised speech sounds. The flow diagram describing generation of test signals is shown in Figure 14.

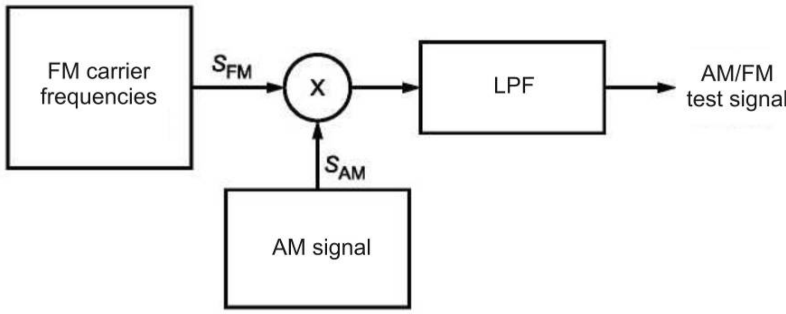



Figure 14 — Flow diagram of test signal generation based on AM/FM modulated sine sets

Initially, a set of $N$ sine-wave signals is formed with FM carrier frequencies $f_n$, of the frequency deviation $\Delta f_n$ and modulation frequency $F_{FM}$ equal to 5 Hz:

$$S_{FM}(t) = \sum_{n=1}^{N} A_n \sin\left(2\pi f_n t + \frac{\Delta f_n}{F_{FM}} \sin(2\pi F_{FM} t) + \varphi_n\right).$$

Then, the amplitude modulation with the modulation index $M = 0.7$ and frequency $F_{AM}$ equal to 3 Hz is applied:

$$S_{AM}(t) = 1 + M \cdot \sin(2\pi F_{AM} t).$$

The initial amplitudes of sine-wave signals $A_n$ are selected equal. Then, the spectral envelope is formed using an LPF of an attenuation equal 5 dB/octave for signals above 250 Hz, and a HPF for signals below 250 Hz. The initial phases $\varphi_n$ affect the signal waveform only.

The signal generation parameters for receiving and sending directions are listed in Table 15 for narrowband IVDS and in Table 16 for wideband IVDS. Comb filters are used to separate receive and transmit signals in the case of their superposition. The detailed description of test signals is presented in [11], [18] and [19].

T a b l e 15 — Parameters of test signals based on AM/FM modulated sine-wave set for narrowband IVDS

| | Receiving direction | | Sending direction | |
|---|---|---|---|---|
| | $f_n$, Hz | $\Delta f_n$, Hz | $f_n$, Hz | $\Delta f_n$, Hz |
| 1 | 250 | ± 5 | 270 | ± 5 |
| 2 | 500 | ± 10 | 540 | ± 10 |
| 3 | 750 | ± 15 | 810 | ± 15 |
| 4 | 1000 | ± 20 | 1080 | ± 20 |
| 5 | 1250 | ± 25 | 1350 | ± 25 |
| 6 | 1500 | ± 30 | 1620 | ± 30 |
| 7 | 1750 | ± 35 | 1890 | ± 35 |
| 8 | 2000 | ± 40 | 2160 | ± 35 |
| 9 | 2250 | ± 40 | 2400 | ± 35 |
| 10 | 2500 | ± 40 | 2650 | ± 35 |

*Table 15 (continued)*

| | Receiving direction | | Sending direction | |
|---|---|---|---|---|
| | $f_n$, Hz | $\Delta f_n$, Hz | $f_n$, Hz | $\Delta f_n$, Hz |
| 11 | 2750 | + 40 | 2900 | + 35 |
| 12 | 3000 | + 40 | 3150 | + 35 |
| 13 | 3250 | + 40 | 3400 | + 35 |
| 14 | 3500 | + 40 | 3650 | + 35 |
| 15 | 3750 | + 40 | 3900 | + 35 |

T a b l e 16 — Parameters of test signals based on AM/FM modulated sine-wave set for wideband IVDS

| | Receiving direction | | Sending direction | |
|---|---|---|---|---|
| | $f_n$, Hz | $\Delta f_n$, Hz | $f_n$, Hz | $\Delta f_n$, Hz |
| 1 | 125 | ± 2.5 | 150 | ± 2.5 |
| 2 | 250 | ± 5 | 270 | ± 5 |
| 3 | 500 | ± 10 | 540 | ± 10 |
| 4 | 750 | ± 15 | 810 | ± 15 |
| 5 | 1000 | ± 20 | 1080 | ± 20 |
| 6 | 1250 | ± 25 | 1350 | ± 25 |
| 7 | 1500 | ± 30 | 1620 | ± 30 |
| 8 | 1750 | ± 35 | 1890 | ± 35 |
| 9 | 2000 | ± 40 | 2160 | ± 35 |
| 10 | 2250 | ± 40 | 2400 | ± 35 |
| 11 | 2500 | ± 40 | 2650 | ± 35 |
| 12 | 2750 | ± 40 | 2900 | ± 35 |
| 13 | 3000 | ± 40 | 3150 | ± 35 |
| 14 | 3250 | ± 40 | 3400 | ± 35 |
| 15 | 3500 | ± 40 | 3650 | ± 35 |
| 16 | 3750 | ± 40 | 3900 | ± 35 |
| 17 | 4000 | ± 40 | 4150 | ± 35 |
| 18 | 4250 | ± 40 | 4400 | ± 35 |
| 19 | 4500 | ± 40 | 4650 | ± 35 |
| 20 | 4750 | ± 40 | 4900 | ± 35 |
| 21 | 5000 | ± 40 | 5150 | ± 35 |
| 22 | 5250 | ± 40 | 5400 | ± 35 |
| 23 | 5500 | ± 40 | 5650 | ± 35 |
| 24 | 5750 | ± 40 | 5900 | ± 35 |
| 25 | 6000 | ± 40 | 6150 | ± 35 |
| 26 | 6250 | ± 40 | 6400 | ± 35 |
| 27 | 6500 | ± 40 | 6650 | ± 35 |
| 28 | 6750 | ± 40 | 6900 | ± 35 |
| 29 | 7000 | ± 40 | | |

7.9.4.4 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) The test signals are applied at the same time in sending and receiving directions. The acoustic test signal in sending direction is applied at the MRP, and has a nominal level of minus 1.7 dBPa (allowing for calibration as per 6.4.2). The electric test signal in receiving direction is applied at the POI to the input of the system simulator encoder, and has a nominal level of minus 16 dBm0.

3) The analysed signal is picked up from the electric output of the system simulator decoder at the POI. This signal contains the transmitted test signal of the near-end subscriber and the partially attenuated echo signal of the far-end subscriber. The echo signal is extracted using a comb filter where the only frequency components of the test signal that are permitted to pass are those of the far-end subscriber, while those of the near-end subscriber are rejected according to [11].

4) The value of echo signal attenuation is measured separately in each frequency band of the test signal transferred in receiving direction. The requirements of Quality Class 1 are deemed satisfied if the echo signal in each frequency band is either below the required limit as specified in Table 14, or below the noise level in the channel. If the echo signal level is above the Class 1 limits, then the IVDS is classified as per Table 14. The check is carried out for all frequencies in the range from 200 to 3450 Hz for narrowband IVDS, or from 200 to 6950 Hz for wideband IVDS.

During the IVDS tests, it is also necessary to check the leakage degree of the near-end subscriber signal components through the comb filter in use, and to make sure that they do not distort the filtered echo signal.

**7.9.5 Attenuation in transmit channel in double-talk mode (additional test)**

7.9.5.1 During double-talk, an IVDS may introduce extra attenuation $A_{H,S,dt}$ in the transmit channel so as to damp acoustic echo signals sneaking from the receive channel; this leads to loudness jumps in the transmit channel during the switches from single-talk to double-talk mode and back.

The main test for verification $A_{H,S,dt}$ is described in 7.9.1. However, it can not be used to separate near-end subscriber signals from residual echo signals, and to perform accurate level measurements of signals transmitted during double-talk.

An additional test is required to make sure that an AES with a short switching time will never be erroneously classified as a full duplex or partial duplex system.

7.9.5.2 Requirements

The requirements for $A_{H,S,dt}$ are analogous to those specified in 7.9.1.

7.9.5.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) Test signals similar to the ones described in 7.9.3 consist of two orthogonal sets of frequency and phase modulated sine-wave signals with a spectral envelope close to that of vocalised speech sounds.

3) The test signals are applied at the same time in sending and receiving directions. The acoustic test signal in sending direction is applied at the MRP, and has a nominal level of minus 1.7 dBPa (allowing for calibration as per 6.4.2). The electric test signal in receiving direction is applied at the POI to the input of the system simulator encoder, and has a nominal level of minus 16 dBm0.

4) The analysed signal is picked up from the electric output of the system simulator decoder at the POI. This signal contains the transmitted test signal of the near-end subscriber and the partially attenuated echo signal of the far-end subscriber. The near-end subscriber signal in sending direction is extracted using a comb filter where the only frequency components that are permitted to pass are those of the near-end subscriber, while those of the far-end subscriber echo noise are rejected according to [11].

5) The value of echo signal attenuation $A_{H,S,dt}$ is measured separately in each frequency band of the test signal in sending direction. The requirements of Quality Class 1 are deemed satisfied if the $A_{H,S,dt}$ value in each frequency band is below the limit specified in Table 11. If the attenuation value is above the Class 1 limits, then the IVDS is classified as per Table 11. The check is carried out for all frequencies in the range from 200 to 3550 Hz for narrowband IVDS, or from 200 to 6900 Hz for wideband IVDS.

6) The test is repeated for all combinations of signal levels and volume control positions.

During the IVDS tests, it is also necessary to check the leakage degree of the far-end subscriber signal components through the comb filter in use, and to make sure that they do not distort the filtered signal of the near-end subscriber.

### 7.10 IVDS performance in acoustic noise conditions

#### 7.10.1 Operation of transmit channel in acoustic noise conditions

7.11.1.1 The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the acoustic input of the IVDS at the MRP to the electric output of the speech codec of the system simulator at the POI.

7.11.1.2 Requirements

For voice signals of the nominal level in conditions of background acoustic noises in the vehicle, the signal-to-noise ratio (SNR) at the output of the transmit channel shall not be less than 6 dB as measured at the seats of the driver and its closest passengers both in "ordinary" and in "worst case" noise environments. The SNR values not less than 12 dB are recommended.

This requirement may imply the selection of the optimum IVDS microphone position and its optimum directional properties in accordance with Appendix E as well as the use of additional algorithms in the IVDS (AGC in sending direction, and nose reduction).

If the requirements for noise wave-form and level are not specified by the manufacturer, the minimum sound pressure level of background noise in the vehicle compartment is taken equal to minus 24 dBPa(A) for "ordinary" and to minus 14 dBPa(A) for "worst case" noise environments.

7.11.1.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. The noise shall be switched on at least 5 s prior to measurements. This will allow adaptive algorithms such as AGC or NRD included in the IVDS to configure their parameters, and to achieve a steady state.

3) The input acoustic test signal in the form of artificial voice as per [14] is applied using an "artificial mouth" with an average SPL equal to minus 1.7 dBPa at the MRP. The test signal level shall be increased in accordance with formula (1) depending on the noise level. One sequence for "male" voice and one sequence for "female" voice with pauses are used.

4) The analysed signal is picked up from the electric output of the system simulator encoder at the POI. The integration time constant for evaluation of signal and noise levels shall be 35 ms. After the signal level and pause noise envelope is calculated, the SNR is assessed in the transmit channel.

N o t e — The SNR assessment is also possible using the recordings of natural male and female speech with an average SPL of minus 1.7 dBPa for active speech regions as per [20]. The test is carried out in silence and in noise conditions, thus providing for assessments of both SNR and intelligibility/quality reduction in noise conditions at the simulator output.

#### 7.10.2 Operation of receive channel in acoustic noise conditions

7.10.2.1 The measurement is carried out for an IVDS installed in the vehicle compartment (cabin), along the path from the electric input of the speech codec of the system simulator at the POI to the acoustic output of the IVDS at the DRP.

7.10.2.2 Requirements

For voice signals of the nominal level in the receive channel, the SNR in the vehicle compartment shall not be less than 0 dB at the minimum volume level and 6 dB at the nominal volume level at the seats of the driver and its closest passengers for the specified level of background acoustic noise in the vehicle compartment in the case of "ordinary" noise environment, and also not less than 6 dB at the maximum volume level in "worst case" noise environment.

If the noise wave-form and level requirements are not specified by the manufacturer, the minimum sound pressure level of background noise in the vehicle compartment is taken equal to minus 24 dBPa(A) for "ordinary" and to minus 14 dBPa(A) for "worst case" noise environments.

This requirement may imply the selection of the optimum values for $RLR_{min}$, $RLR_{norm}$ and $RLR_{max}$ parameters, optimum IVDS microphone position and directional properties, as well as for the use of additional algorithms in the IVDS (AGC in receiving direction).

7.10.2.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. The interfering noise shall be switched on at least 5 s prior to measurements. This will allow adaptive algorithms such as AGC or NRD included in the IVDS to configure their parameters, and to achieve a steady state.

3) The input electric test signal in the form of artificial voice as per [14] is applied at the input of the system simulator encoder at the POI with its level equal to 16 dBm0. One sequence for "male" voice and one sequence for "female" voice with pauses are used.

4) The analysed signal is picked up from the acoustic output of the IVDS at the DRP. The integration time constant for evaluation of signal and noise levels shall be 35 ms. After the signal level and pause noise envelope is calculated, the SNR is assessed in the transmit channel.

Note — The SNR assessment is also possible using the recordings of natural male and female speech with an average SPL of minus 16 dBm0 applied at the input of the system simulator encoder at the POI reference point for active speech regions as per [20]. The test is carried out in silence and in noise conditions, thus providing for assessments of both SNR and speech intelligibility/quality reduction in noise conditions at the acoustic output of IVDS at the DRP.

**7.11 Background noise quality in transmit channel**

7.11.1 The measurements are carried out in conditions of background acoustic noise of a given level in the vehicle compartment, for "ordinary" and "worst case" noise environments. If the requirements for noise wave-form and level are not specified by the manufacturer, the minimum sound pressure level of background noise in the vehicle compartment is taken equal to minus 24 dBPa(A) for "ordinary" and to minus 14 dBPa(A) for "worst case" noise environments.

**7.11.2 Background noise after connection**

7.11.2.1 Right after the connection is established, the background noise in the transmit channel is usually louder than the one observed several seconds later. This is related to transient processes in AEC, noise reduction (NRD), AGC and speech encoding algorithms. The initial noise level increase in the IVDS shall not cause any discomfort for the far-end subscriber.

7.11.2.2 Requirements

An initial pulse of background noise that occurs in the transmit channel after the connection is established shall not exceed the average noise level by more than 12 dB during the frequency measurements in the range from 300 Hz to 3.4 kHz for narrowband IVDS or from 150 Hz to 7.0 kHz for wideband IVDS.

7.11.2.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) The test noise signal is reproduced in the vehicle compartment using a sound amplifier unit as stated in 6.2. The noise level shall be equal to the one observed when the sound signal is recorded for the vehicle of a given type. All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. The acoustic noise shall be switched on at least 5 s prior to measurements.

3) The IVDS is turned off then on again (for initial reset of adaptive algorithms included in the IVDS, such as AGC, NRD and AEC). The connection is initiated using the system simulator, and the IVDS answers the incoming call. Care should be taken in this process to prevent additional acoustic noise from occurring in the vehicle compartment as the result of operator's actions.

4) The signal recording in the transmit channel is performed at the electric output of the system simulator decoder, and is started at least one second before the call is answered by the IVDS and stopped at least 15 s after the connection is established. The period 8 s in length including 1 s of pause before the connection is analysed.

**7.11.3 Quality of background noise transmission in presence of near-end subscriber's speech**

7.11.3.1 The test in the transmit channel proceeds using a CSS simulating the speech of the near-end subscriber, and a noise signal simulating the ambient acoustic noise.

7.11.3.2 Requirements

The background noise level in the transmit channel before, during and after the speech activity in this channel shall not change by more than 10 dB (while the speech of the near-end subscriber is turned on and off in the transmit channel).

7.11.3.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) The test noise signal is reproduced in the vehicle compartment using a sound amplifier unit as stated in 6.2. The noise level shall be equal to the one observed when the sound signal is recorded for the vehicle of a given type. All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. The acoustic noise shall be switched on at least 5 s prior to measurements and to transmission of a training sequence. This will allow adaptive algorithms such as AGC or NRD included in the IVDS to configure their parameters, and to achieve a steady state.

3) To ensure that the AEC of the IVDS will configure itself in the receive channel, a training sequence in the form of artificial voice as per [14] consisting of 10 s of "male" and 10 s of "female" speech at a level of minus 16 dBm0 is applied to the electric input of the system simulator encoder at the POI.

4) The first measurement is taken with no voice signal in the transmit channel. The noise signal of at least 10 s in length is recorded at the electric output of the system simulator decoder at the POI. The curve of noise signal level versus time is plotted with no far-end subscriber's speech present. The noise signal level is averaged using a time constant of 5 ms.

5) Then, the test CSS as per [11] at a level from minus 1.7 dBPa to 4.3 dBPa is periodically applied in the transmit channel to the acoustic input of the IVDS at the MRP for at least two CSS periods. The noise signal is recorded at the electric output of the system simulator decoder at the POI. The curve of noise signal level versus time is plotted. The noise signal level is averaged using a time constant of 5 ms.

6) The changes of noise signal levels are evaluated at the moments when the near-end subscriber's speech is switched on and off.

**7.11.4 Quality of background noise transmission in presence of far-end subscriber's speech**

7.11.4.1 The test is carried out using a CSS that simulates the speech of the far-end subscriber and is applied in receiving direction, and a noise signal that simulates the ambient acoustic noise and is applied in sending direction.

7.11.4.2 Requirements

The background noise level in the transmit channel before, during and after the speech activity in the receive channel shall not change by more than 10 dB (while the speech of the far-end subscriber is turned on and off in the receive channel).

7.11.4.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) The test noise signal is reproduced in the vehicle compartment using a sound amplifier unit as stated in 6.2. The noise level shall be equal to the one observed when the sound signal is recorded for the vehicle of a given type. All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. The acoustic noise shall be switched on at least 5 s prior to measurements and to transmission of a training sequence. This will allow adaptive algorithms such as AGC or NRD included in the IVDS to configure their parameters, and to achieve a steady state.

3) To ensure that the AEC of the IVDS will configure itself in the receive channel, a training sequence in the form of artificial voice as per [14] consisting of 10 s of "male" and 10 s of "female" speech at a level of minus 16 dBm0 is applied to the electric input of the system simulator encoder at the POI.

4) The first measurement is taken with no voice signal in the receive channel. The noise signal of at least 10 s in length is recorded at the electric output of the system simulator decoder at the POI. The curve of noise signal level versus time is plotted with no far-end subscriber's speech present. The noise signal level is averaged using a time constant of 35 ms.

5) Then, the test CSS as per [11] at a level of minus 16 dBm0 is periodically applied in the receive channel to the electric input of the system simulator encoder at the POI for at least two CSS periods. The noise signal is recorded at the electric output of the system simulator decoder at the POI. The curve of noise signal level versus time is plotted. The noise signal level is averaged a time constant of 35 ms.

6) The changes of noise signal levels are evaluated for the moments when the far-end subscriber's speech is switched on and off.

**7.11.5 Quality of background noise transmission using "comfort noise" for pauses**

7.11.5.1 This test is only carried out if the IVDS generates an artificial "comfort noise" in pauses instead of transmitting real noise when the far-end subscriber is active. Such "comfort noise" generator is used in AEC algorithms to mask residual echo suppression effects.

7.11.5.2 Requirements

If the IVDS generates an artificial "comfort noise" in pauses instead of transmitting real background noise while the subscribers keep silent in the vehicle compartment, then:

1) "comfort noise" level in pauses shall not differ from the original transmitted background noise by more than plus 2 dB and minus 5 dB. The noise level is assessed by frequency-weighting along A-curve;

2) difference between the spectrum of the "comfort noise" in pauses and the one of the original transmitted noise shall be within the tolerance specified in Table 17. Intermediate frequency values may be obtained by linear interpolation using a log scale for frequencies, and a linear scale for levels in dB. The last line of the Table applies to wideband IVDS only;

3) "comfort noise" on/off switching effects shall not cut off the speech at word starts or ends in the transmit channel, and shall not impair speech intelligibility.

7.11.5.3 Method of measurement

1) Environment conditions in the vehicle cabin shall comply with the requirements of Section 6.

2) The test noise signal is reproduced in the vehicle compartment using a sound amplifier unit as stated in 6.2. The signal level shall be equal to the one when the acoustic noise signal is recorded for the vehicle of a given type. All scenarios of acoustic noise recording specified in Table D.1 (Appendix D) and related to vehicle movement at a constant speed shall be checked. The acoustic noise shall be switched on at least 5 s prior to measurements and to transmission of a training sequence. This will allow adaptive algorithms such as AGC or NRD included in the IVDS to configure their parameters, and to achieve a steady state.

3) To ensure that the AEC of the IVDS will configure itself in the receive channel, a training sequence in the form of artificial voice as per [14] consisting of 10 s of "male" and 10 s of "female" speech at a level of minus 16 dBm0 is applied to the electric input of the system simulator encoder at the POI.

4) Then, the test signal containing a pause of at least 10 s and a periodically repeated CSS as per [11] of a level of minus 16 dBm0 and duration of at least 10 s is applied in the receive channel to the electric input of the system simulator encoder at the POI in order to activate the "comfort noise" generator at the start of transmission. The noise signal at the transmit channel output is recorded at the electric output of the system simulator decoder at the POI. A pair of regions is selected in the recording: the one with an ordinary pause noise while the far-end subscriber is silent, and the one with the "comfort noise" in pauses while the far-end subscriber is active.

5) The power spectrum density is analysed using the FFT for 8192 samples (at a sampling rate of 48 kHz) for the two mentioned regions not less than 5 s in duration each. The spectrum difference is checked against the requirements of Table 17.

T a b l e 17 — Tolerances for "comfort noise" spectrum

| Frequency, Hz | Upper limit | Lower limit |
|---|---|---|
| 200 | 12 | –12 |
| 800 | 12 | –12 |
| 801 | 10 | –10 |
| 2000 | 10 | –10 |
| 2001 | 6 | –6 |
| 4000 | 6 | –6 |
| 8000 | 6 | –6 |

6) The noise signal level in pauses is calculated using frequency-weighting along A-curve for the said signal regions not less than 5 s in duration each. The changes of noise signal levels are evaluated for the moments when the far-end subscriber speech's is switched on and off.

**7.12 Subjective quality assessment of IVDS loudspeaker communication**

In addition to objective measurements of the loudspeaker IVDS parameters, a subjective expert quality assessment shall also be performed for different external operation conditions.

The performance quality of a loudspeaker IVDS depends on:

- IVDS specifications and configuration parameters;
- acoustic properties of vehicle compartment;
- selection of installation locations for IVDS microphone and loudspeakers;
- current (noise-related) vehicle operation mode;
- talkers' speech and hearing capabilities;
- additional external conditions (road, weather, communication network).

The first three conditions are constant, and are determined when the IVDS is installed and configured in the vehicle compartment. The latter three are variable, and depend on specific circumstances taking place when the vehicle is operated.

The tests described in this section are carried out in real operating conditions with a particular vehicle type and with a particular IVDS installed and configured in it.

As an alternative to these tests, those described in 7.13 are permitted.

**7.12.1 Organisation of tests**

In subjective assessments of the loudspeaker IVDS quality, a driver in the vehicle with an installed IVDS is considered as a near-end subscriber using loudspeaker communication. An operator of the service centre who is using a fixed telephone handset (preventing from acoustic echo signals on the operator side) is considered as a far-end subscriber.

The loudspeaker communication quality shall be assessed for various noise scenarios in the vehicle cabin that are governed by the speed of movement, quality of pavement, operating mode of the engine and conditioning or heating systems, the presence of open windows, etc.

The Russian language and the national languages of the target operating region of the system shall be used during the tests.

The activities of subscribers shall be strictly regulated and coordinated. A clear test plan shall be in place, and alternative duplex communication shall be maintained. The test results shall be documented in a test report. On each subscriber side, talks shall be recorded (using standard call recording equipment in service centres and binaural recording tools in the vehicle compartment).

**7.12.2 Assessed parameters**

The voice tests shall be carried out both in alternating single-talk (half duplex) mode, and in simultaneous double-talk mode (duplex if possible).

The communication quality shall be assessed on both near-end and far-end subscriber sides.

An overall quality assessment is based on the combined assessments of the following parameters:

- AEC performance quality of IVDS during single-talk (echo signal perception degree and intensity, AEC convergence rate, call direction switching time, etc.);
- AEC performance quality of IVDS during double-talk (echo signal perception degree and intensity, loudness jumps of voice signals, etc.);
- quality of speech and of background noise in pauses in sending direction (loudness rating, loudness jumps, intelligibility, speech natural sounding and quality, SNR, voice signal distortions, etc.);
- quality of speech in receiving direction (loudness rating, loudness jumps, intelligibility, speech natural sounding and quality, SNR, voice signal distortions, etc.);

The methods described in [21] — [26] shall be used for assessment of individual parameters.

The decision on the quality of each indicator shall be based on its rating scale. Such scales, as well as the MOS assessment, generally include five grades (with Grade 1 denoting the worst quality and Grade 5 denoting the best quality).

Specific indicators shall be evaluated by specialists experienced in quality assessment of loudspeaker systems, and the expert opinions shall then be averaged. The audition of test results and the overall IVDS quality assessment may be participated by ordinary unqualified IVDS users. During the tests, the acoustic signals in the vehicle and the electric signals on the operator side shall be recorded and included in reports for later audition and comparison where individual experts may also be engaged.

The set of typical conditions for subjective testing is listed in Table 18. The set of tests is detailed in Table 19.

T a b l e 18 — Set of typical conditions for subjective testing

| | |
|---|---|
| Vehicle movement | Standing with engine switched off;<br>Standing with engine switched on;<br>Moving along urban road at 70 km/h;<br>Moving along highway at 120 km/h. |
| Vehicle location | Quiet street;<br>Heavy-traffic route. |
| Vehicle windows | Closed<br>Open |
| Ventilation, conditioning, heating | On<br>Off |
| IVDS volume control position | Minimum<br>Nominal<br>Maximum |
| Talker speech | Male voice (three talkers)<br>Female voice (three talkers) |
| Talker location | Driver seat;<br>Passenger seat next to driver;<br>Passenger seat back from driver. |
| Talker speech tempo | Normal;<br>Accelerated; |
| Talker speech level | Normal;<br>Weak;<br>Loud (shrill). |

T a b l e 19 — Set of tests for subjective quality assessment

| Echo canceller | | Parameters to be assessed |
|---|---|---|
| Alternating single-talk<br><br>Simultaneous double-talk | Parameters to be assessed are those describing inconvenience caused by AEC operation in IVDS during the talk.<br>Combinations of test conditions correspond to Table 18. | Echo signal occurrence rate;<br>Echo signal perception and level (talker's ability to speak when his/her own echo signals are present);<br>Duration of AEC setup for echo signal suppression when the connection is established or the echo path is changed;<br>Perception degree of effects related to call direction switching processes (beginning or ending of words, phrases or sentences dropped at the other party side);<br>Audible loudness jumps of talker voice and background noise |

*Table 19 (continued)*

| Speech quality in noise conditions in sending direction | | Parameters to be assessed |
|---|---|---|
| Stationary acoustic noise inside vehicle | Parameters to be assessed are those describing speech intelligibility and quality in sending direction from vehicle to operator.<br>Combinations of test conditions correspond to Table 18. | Overall speech quality (distortions, artefacts, etc.);<br>Overall intelligibility of driver speech, efforts required to understand its meaning |
| **Speech quality in noise conditions in receiving direction** | | **Parameters to be assessed** |
| Transient acoustic noise in vehicle cabin | Change of movement mode<br>On/off switching of ventilation, conditioning, heating<br>Opening/closing of windows<br>Noise from vehicle units (beeps, clicks), wind noise from windows,<br>Noise from traffic passing by | Quality of background noise transmission;<br>No sounding artefacts;<br>Adaptation to changes of noise conditions |
| Acoustic noise inside vehicle | Parameters to be assessed are those describing speech intelligibility and quality in receiving direction from operator to vehicle.<br>Combinations of test conditions correspond to Table 18. | Overall speech quality (distortions, artefacts, etc.);<br>Overall intelligibility of driver speech, efforts required to understand its meaning |

The mean grade based on five-grade rating scales shall be at least 3.0 for narrowband IVDS and at least 3.6 for wideband IVDS when they operate in ordinary noise conditions (depending on the vehicle type and noise scenario).

**7.12.3 Driving scenarios and acoustic noise cases**

The basic driving scenarios and cases where acoustic noises can arise, to be considered in subjective testing, are listed in Table 18. They shall correspond to typical operating conditions of a particular vehicle and, as a consequence, this list may be corrected for a given vehicle type and its operating environment. The assessments shall be carried out for conditions that are both "ordinary" and "worst case" as regards the disturbing noises.

Noises in the vehicle compartment may be caused by either inside factors (engine operation, movement along the route, operation of internal climate-control devices, air flows in the compartment), or outside factors (street noise, traffic passing by). According to their type, noises may be classified as stationary (of a steady level, waveform and spectral content) and transient ones.

**7.12.4 Procedure of subjective testing**

At the beginning of each test, one of its participants pronounces the test number aloud so that the sound recordings may be numbered. The sound recording proceeds as follows:

- on the far-end subscriber side: two-channel separate recording of received and transmitted signals;
- in the vehicle compartment: two-channel binaural recording of acoustic environment.

After each test, the experts deliver their quality judgements regarding an indicator, using the scale that corresponds to that indicator. In doing so, they shall take into account all potential quality limitations caused by signal transmission and processing in the communication network.

GOST 16600 shall be used to assess the speech quality and intelligibility. The calculation of syllabic intelligibility in the vehicle is complicated; therefore, the phrase intelligibility is assessed basing on phonetically balanced test phrases. The quality and identification ability of speech are assessed using the five-grade scale with Grade 1 corresponding to the worst quality and Grade 5, to the best quality.

When the performance or configuration parameters of different IVDS are compared basing on signal recordings made in the same acoustic environment, the pair-wise comparison method employing a relative rating scale shall be used for quality assessment because such scale is more accurate than the absolute one.

**7.12.5 Speech and background noise quality in sending direction**

The voice signal level is assessed from the side of the far-end subscriber in the service call centre during a single-talk with the near-end subscriber in the vehicle. Three scenarios are used for background noise in the vehicle, i.e., silence, moderate noise and loud noise, and various heating/ventilation/conditioning system modes as well as open/closed window positions are considered.

7.12.5.1 Speech intelligibility and efforts required for its understanding

The speech intelligibility (by words and by phrases) and the efforts required for understanding the meaning of words and sentences shall be assessed. The phonetically balanced phrases shall be taken from GOST 16600 for their use as test phrases.

The assessment is carried out using the rating scale detailed in Table 20.

T a b l e 20 — Rating scale for assessment of speech intelligibility and efforts to understand its meaning

| Description of indicator | Grade |
|---|---|
| Each word sounds clearly and is understood with no efforts | 5 |
| Speech is understood without noticeable efforts | 4 |
| Some words are hard to understand, moderate efforts are required for that | 3 |
| Many words are hard to understand, significant efforts are required for that | 2 |
| Nothing can be understood whatever efforts are taken | 1 |

7.12.5.2 Transmission quality of transient background noise

Transient background noises may be generated by windscreen wipers, turn indicators and other sources of acoustic background noise, and their intensity may change in time abruptly. The natural sounding in transmission of such sounds is described by the following indicators:

- noise similarity to natural background noise of vehicle;
- synthetic sounding of noise;
- presence of noise distortions.

The assessment is carried out using the ACR rating scale as per [21] described in Table 31.

T a b l e 21 — Transmission quality of transient background noise

| Description of indicator | Grade |
|---|---|
| Comfortable natural sounding | 5 |
| Almost natural sounding<br>Slightly distorted/synthetic sounding | 4 |
| Moderately natural sounding<br>Moderately distorted/synthetic sounding | 3 |
| Clearly unnatural/distorted/synthetic sounding | 2 |
| Completely unnatural/distorted/synthetic sounding | 1 |

**7.12.6 Speech quality in receiving direction**

The voice signal level shall be assessed from the side of the near-end subscriber in the vehicle during a single-talk with the far-end subscriber in the service call centre. The assessment proceeds in silence, at a nominal volume control position of the IVDS.

The phonetically balanced phrases shall be taken from GOST 16600 for their use as test phrases. The assessment shall be based on the five-grade rating scale described in Table 20.

**7.12.7 Speech quality during double-talk**

The speech quality during double-talk (intelligibility, loudness changes) depends on the AEC operation quality, and is assessed for both near-end and far-end subscribers' speech.

7.12.7.1 Speech level variations during double-talk

The speech level variations during double-talk shall be assessed by both subscribers (near-end and far-end) simultaneously at different noise levels in the vehicle cabin. The level variations are assessed basing on the following indicators:

- speech level drop effects related to current direction switching;
- gradual changes of levels;
- sharp level rises or falls (dropped beginning/end of words);
- intermittent voice.

The rating scale detailed in Table 22 is used in this assessment.

Table 22 — Speech level variations during double-talk

| Description of indicator | Speech level range | Grade |
|---|---|---|
| During the speech of one party and in pauses, no level variations in speech of other party are audible | Full duplex | 5 |
| Slight level variations in speech of other party, barely audible or rarely occurring | | 4 |
| Moderate and fairly frequent level variations in other party's speech, with individual syllables and words sometimes excessively attenuated, or dropped. Or, moderate attenuation of other party's speech due to switching of call direction | Partial duplex | 3 |
| Significant level variations in speech of other party. Many drops, lost syllables or words, intermittent voice. Or, strong attenuation due to switching of call direction | | 2 |
| Other party completely inaudible during double-talk. Receive or transmit channel is blocked | Half-duplex only | 1 |

7.12.7.2 Speech intelligibility and efforts required to understand it during double-talk

The speech intelligibility is assessed by both subscribers (near-end and far-end) simultaneously at different noise levels in the vehicle cabin. The phonetically balanced phrases shall be taken from GOST 16600 for their use as test phrases. The assessment shall be based on the rating scale detailed in Table 23.

Table 23 — Speech intelligibility and efforts to understand its meaning during double-talk

| Description of indicator | Grade |
|---|---|
| Each word of other party sounds clearly and understood with no efforts | 5 |
| Speech of other party is understood without noticeable efforts | 4 |
| Some words of other party are hard to understand, moderate efforts are required for catch meaning of phrases | 3 |
| Many words of other party are hard to understand, essential efforts are required for catch meaning of phrases | 2 |
| Speech of other party is cannot be understood during double-talk whatever efforts are taken | 1 |

**7.12.8 Assessment of acoustic echo canceller performance**

The key parameter of acoustic echo canceller (AEC) performance is the perception level of the residual echo signal and its disturbing influence during conversation. The AEC perception quality shall be assessed using the following indicators:

- nature and magnitude of echo signals during single-talk;
- nature and magnitude of echo signals during double-talk;

- AEC convergence in case of echo path changes, e.g., volume changes, driver position shifts, front seat passenger movements, etc.;
- stability of AEC operation in connection process, at maximum volume control position, and upon loudspeaker communication switch-on in call centre.

The tests shall be carried out both when no noise is present (silence in the vehicle compartment) and in the presence of background noises that correspond to different vehicle operation scenarios, in order to evaluate the AEC noise immunity.

The tests shall be carried out for single-talk and double-talk of the far-end subscriber, in low to intensive noise scenario conditions inside the vehicle cabin, for different window positions and different operating modes of ventilation, heating and conditioning systems. The IVDS volume control shall be set to its nominal and to its maximum positions. The near-end subscriber and his passenger may move (e.g., may turn about).

The degrees of perception and disturbing effects of residual echo signals shall be assessed by the far-end subscriber who hears his own reflected voice in the transmit channel so that his ability to speak and to understand the other party is reduced.

The rating scales for the indicators used in this section apply to steady state operation of the AEC, except for those in the AEC convergence test.

7.12.8.1 Perception degree of residual echo signal

The assessment shall be carried out using the scales of talk quality degradation as per [21]. The following indicators shall be considered: intensity, duration, rate and intelligibility of echo signals (of either speech or noise ones).

The assessment result shall be presented using the rating scale detailed in Table 24.

T a b l e 24 — Degree of perception and disturbing effects of echo signals

| Description of indicator | Grade |
|---|---|
| Echo signals are not perceivable | 5 |
| Echo signals are audible, but not disturbing | 4 |
| Echo signals are slightly disturbing | 3 |
| Echo signals are disturbing and possibly repeated | 2 |
| Echo signals are very disturbing and cause stuttering or are repeated multiple times; other party's speech can not be understood during double-talk | 1 |

Additional subjective assessments of speech intelligibility in transmit and receive channels in accordance with Appendix F are recommended.

### 7.13 Subjective quality assessment of IVDS loudspeaker communication based on reference recordings

The test methods described in this section are based on the use of a previously prepared collection of recordings with noise disturbances that occur in real operating conditions of the vehicle, including external noises penetrating the vehicle compartment or noises from operating in-vehicle systems, and degrade the communication quality in conversations between the driver and the operator. In such tests, various recordings of noise and driver's speech are reproduced in the vehicle compartment, completed with the operator's speech played back from the operator's side to simulate his/her standard responses and queries in emergency calls.

After that, the recorded conversation is handed to experts for listening and assessing based on the criteria specified for the modelled call situation.

This approach ensures that all vehicle and IVDS types are tested in the same conditions regardless of any external and human factors.

#### 7.13.1 Organisation of tests

In subjective assessments of the loudspeaker IVDS quality that employ reference recordings, the HATS manikin in the vehicle with an installed IVDS is considered as a near-end subscriber using loudspeaker communication. The operator of the service centre is considered as a far-end subscriber.

The loudspeaker communication quality shall be assessed for various noise scenarios in the vehicle cabin that are governed by the speed of movement, quality of pavement, operating mode of the engine and conditioning or heating systems, the presence of open windows, etc.

The Russian language and the national languages of the target operating region of the system shall be used during the tests.

Units accepted for acoustic measurements in these tests are listed in Appendix G.

The results shall be included in a test report. Recording of conversations at the side of the far-end subscriber and binaural recording in the vehicle compartment shall be made.

All IVDS samples are tested using identical recordings of conversations. Each recording is done using a single HATS located in the vehicle with IVDS equipment, accompanied by playback of recordings for noise scenarios indicated in Table D.1 (Appendix D) for a given vehicle. In this process, both the IVDS parameters and the influence of a particular vehicle are taken into account.

The IVDS is connected to the system simulator for signal transmission in receiving direction and signal reception on the side of the far-end subscriber. Playback and recording of voice signals in the vehicle compartment are completed using the HATS.

During the tests, the recording samples made both in the vehicle and on the side of the far-end subscriber are appraised by the experts.

**7.13.2 Assessed parameters**

The voice tests shall be carried out both in alternating single-talk (half duplex) mode, and in simultaneous double-talk (duplex) mode.

The communication quality shall be assessed on both near-end and far-end subscriber sides. An overall quality assessment is based on the combined assessments of the following parameters:

- AEC performance quality of IVDS during single-talk (echo signal perception degree and intensity, AEC convergence rate, call direction switching time, etc.);
- AEC performance quality of IVDS during double-talk (echo signal perception degree and intensity, loudness jumps of voice signals, etc.);
- quality of speech and of background noise in pauses in sending direction (loudness rating, loudness jumps, intelligibility, speech natural sounding and quality, SNR, voice signal distortions, etc.);
- quality of speech in receiving direction (loudness rating, loudness jumps, intelligibility, speech natural sounding and quality, SNR, voice signal distortions, etc.);

The mean grade based on five-grade rating scales shall be at least 3.0 for narrowband IVDS and at least 3.6 for wideband IVDS when they operate in ordinary noise conditions (depending on the vehicle type and noise scenario).

**7.13.3 Set of conditions used in tests**

The basic driving scenarios and cases where acoustic noises can arise, to be considered in subjective testing, are listed in Table D.1 (Appendix D). They shall correspond to typical operating conditions of a particular vehicle and, as a consequence, this list may be corrected for a given vehicle type and its operating environment. The assessments shall be carried out for conditions that are both "ordinary" and "worst case" as regards the disturbing noises.

The noises in the vehicle compartment may be caused by either inside factors (engine operation, movement along the route, operation of internal climate-control devices, air flows in the compartment), or outside factors (street noise, traffic passing by). According to their type, noises may be classified to stationary (of a steady level, waveform and spectral content) and transient ones.

The tests using the following scenarios in addition to the ones listed in Table D.1 (Appendix D) shall be carried out:

- noise scenario of emergency call in parking conditions at a highway verge, with engine switched off, all four windows closed, HATS manikin placed on driver's seat, and noise of passing-by vehicles audible. The scenario presented in [27] (scenario A1) is recommended;
- noise scenario of emergency call in parking conditions at a non-busy road (noise of vehicles passing by from time to time), with engine switched off, all four windows closed and HATS manikin placed on driver's seat. The scenario presented in [27] (scenario A4) is recommended. This scenario is used to assess the transmission quality of transient noises (in accordance with Table 21).

The recordings in the noise scenarios of Table D.1 (Appendix D) are made in real driving conditions of the vehicle, and reproduced through the loudspeaker installed as shown in Figure 3. The vehicle shall be equipped with a noise reproduction system in accordance with Figure 3.

**7.13.4 Playback of recorded conversations**

During the tests, the recorded conversations shall be reproduced in the sequence shown in Figure 15.

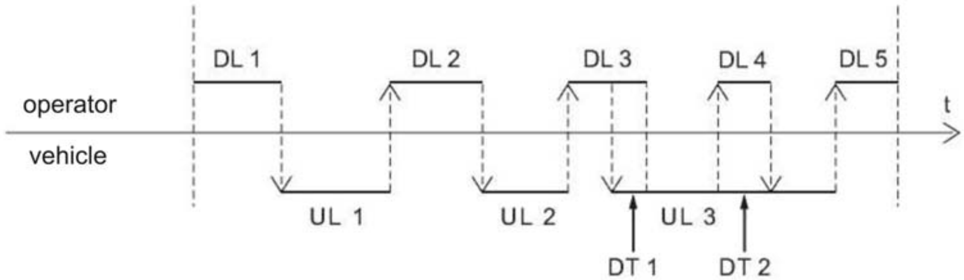



Fig. 15 — Playback sequence of recordings

T a b l e 25 — Parameters of reproduced recordings

| Test signal name | Description | Signal duration |
|---|---|---|
| DL1 | Signal in receiving direction, single-talk; assessment of initial echo at far-end subscriber's side; operator answering call | From 2 to 4 s |
| UL1 | Signal in sending direction, single-talk; driver describing call reason | From 2 to 6 s |
| DL2 | Signal in receiving direction, single-talk; echo assessment at operator side; operator requesting additional information from driver | From 2 to 4 s |
| UL2 | Signal in sending direction, single-talk; driver providing additional information | From 2 to 4 s |
| DT1 | Double-talk conversation; driver interrupting operator; operator requesting additional information | From 2 to 4 s |
| DT2 | Double-talk conversation; operator interrupting driver | From 2 to 4 s |
| DL3 | Signal in receiving direction; operator requesting additional information | From 2 to 4 s |
| UL3 | Signal in sending direction; driver answering and interrupting operator | From 8 to 10 s |
| DL4 | Signal in receiving direction; operator confirming acceptance of information | From 2 to 3 s |
| DL5 | Signal in receiving direction, single-talk; echo assessment at operator side; operator reporting driver on departure of emergency services | From 2 to 4 s |

**7.13.5 Recording of speech samples**

The operator's speech is recorded in silence using an omni-directional microphone located next to the month of the talker.

One male voice is used to model the operator's speech. The voice signal is filtered into the range from 200 Hz to 3.4 kHz [20] using appropriate settings of the system simulator (cellular network simulator). The level is set to minus 16 dBm0.

The driver voices (one male and one female) shall be recorded taking into account the Lombard effect (formula 1). These recordings shall be made in the vehicle as required for the noise scenarios being modelled (e.g., a noise level of (65 ± 3) dB(*A*), (70 ± 3) dB(*A*), etc.), with different characteristics of the Lombard effect. The voice material that will be used for recording of conversations shall be selected taking into account the Lombard method, as appropriated for the actual noise in the vehicle.

For example, if the actual noise in the vehicle is closer to 65 dB(*A*) than to 70 dB(*A*), the recording made in the conditions with the noise level of 65 dB(*A*) shall be used. Then, the signal level shall be corrected to the one taking into account the Lombard effect in accordance with formula 1 (see 6.4.4).

**7.13.6 Recording of conversations**

The IVDS shall be connected to the cellular network simulator during the recording. The HATS manikins shall be in the respective seats of the driver and its nearest passenger.

The second HATS manikin simulating the operator shall use a handset pressed with a force of 8 N to its artificial ear of type 3.3 [15]. The headphones shall be connected to the cellular network simulator.

Alternatively, the headphones and microphone (terminal) of the operator (far-end subscriber) may be modelled by suitable filtering as appropriate for the basic parameters: RLR, receiving AFR, side-tone masking rating (STMR), side-tone AFR.

The filtering shall be made taking into account the requirements for telephone apparatus with handsets to achieve the following simulation parameters:

- signal level at POI of system simulator: minus 16 dBm0 [20] (this simulates the operator's terminal with SLR equal to (8 ± 3) dB);
- RLR: (2 ± 3) dB;
- flat AFR in the range from 200 Hz to 3.4 kHz;
- STMR: (16 ± 4) dB.

The following test signals are used in accordance with Figure 15:

- driver's voice reproduced in the vehicle using the artificial mouth of the HAT manikin;
- operator's voice filtered into the range from 200 Hz to 3.4 kHz using appropriate settings of the system simulator, with the level of minus 16 dBm0.

**7.13.7 Accounting for time delays on IVDS side**

The signal delays in the IVDS for both directions shall be taken into account for an accurate reproduction sequence of recordings.

The delay shall be taken into account because the conversation tests have the same duration for all IVDS; in particular, this is important for time selection of call direction switching from transmission to reception and back, and for duration accounting of double-talk conversations.

The IVDS delay shall be taken into account when the speech is reproduced using the HATS manikin in the vehicle.

The IVDS delay should be taken into account when each speech fragment is reproduced during an outgoing call.

**7.13.8 Language of speech material**

The Russian language or the national languages of the target operating region of the system shall be used during the tests.

**7.13.9 Quality assessment of loudspeaker communication**

The experts provide their assessment after listening to the recorded conversations as described in 7.13.6. Both the recording in the vehicle and the one on the side of the far-end subscriber are assessed.

7.13.9.1 Quality of speech transmission during single-talk

The voice transmission quality during a single-talk shall be assessed considering:

1) efforts required for understanding speech at near-end and far-end side, in rating scale of Table 20;

2) echo levels at far-end side, in rating scale of 24.

7.13.9.2 Quality of background noise transmission

This quality is assessed basing on the rating scale of Table 21.

7.13.9.3 Quality of speech transmission in sending direction during double-talk

This quality is assessed on the side of the far-end subscriber by considering:

1) speech level variations, in rating scale of Table 22;

2) echo level, in rating scale of Table 24.

7.13.9.4 Quality of speech transmission in receiving direction during double-talk

This quality shall be assessed on the vehicle side in accordance with the rating scale of Table 20 by considering the efforts required to understand the meaning of speech.

# Appendix A
# (obligatory)

## List of measuring instruments, testing equipment and devices used in tests

T a b l e A.1 — List of measuring instruments, testing equipment and devices used in tests

| Designation of measuring instruments, testing equipment and devices | Basic requirements on functional properties, engineering (metrological) characteristics |
|---|---|
| Simulator of in-vehicle emergency call system | Emulator type: selected according to the wireless mobile communication network type (GSM, UMTS) used in the IVDS and taking into account the requirements of 6.6 –6.8.<br>Maximum level of electric self-noise at encoder input and at decoder output: not greater than minus 74 dBm0(A).<br>Harmonic distortion in receiving and in sending direction: not greater than 1 % (for a codec supported by the IVDS, at its highest bitrate) |
| HATS manikin in form of artificial head and torso | Basic requirements for manikin: as per [3] and [7].<br>Additional requirements for artificial mouth and ear of manikin: in accordance with 6.4, 6.5 and 6.7 |
| Artificial mouth | Basic error in generated sound pressure level: not greater than ±0.5 dB.<br>Variation of frequency response of sound pressure in frequency range from 100 to 10000 Hz: not greater than ±3 dB.<br>Harmonic distortion at sound pressure of 3 Pa: not greater than 3 % in a frequency range from 100 to 300 Hz, and not greater than 2 % at frequencies above 300 Hz.<br>Additional requirements: in accordance with 6.4, 6.7 and [3], [5], [7] |
| Artificial ear | Basic error in sound pressure measurements: not greater than ±0.5 dB.<br>Variation of frequency response in frequency band range 100 to 8000 Hz: not greater than ±2 dB.<br>Harmonic distortion at sound pressure of 10 Pa: not greater than 1 %.<br>Additional requirements: in accordance with 6.5, 6.7 and [3], [7], [15] |
| Noise meter | In accordance with GOST 17187.<br>Accuracy class: not greater than 2 |
| Additional measuring microphone | Type: condenser 1/2", pressure-based measurements.<br>Basic error: not greater than ±0.5 dB.<br>Variation of frequency response in frequency range from 0.1 to 16 kHz: not greater than 2 dB.<br>Harmonic distortion at sound pressure of 10 Pa: not greater than 1 %. |
| Microphone amplifier | Controlled gain for adjustment between output signal level of the measuring microphone and input level of the I/O card of the PC.<br>Harmonic distortion: not greater than 0.1 % |
| PC with I/O card for test signals (ADC/DAC) and a set of special purpose software for measurement of IVDS characteristics | ADC/DAC sampling rates: 8, 16, 32, 48 KHz.<br>Sample width: not less than 16 bits.<br>Number of channels: not less than two.<br>Dynamic range of ADC/DAC: not less than 80 dB.<br>The software used for each type of measurements shall comply with the requirements specified in Section 7. |

*Table A.1 (continued)*

| Designation of measuring instruments, testing equipment and devices | Basic requirements on functional properties, engineering (metrological) characteristics |
|---|---|
| PC with card for output of noise signal PC (DAC) and set of special purpose software for noise simulation in vehicle cabin | DAC sampling rates: 8, 16, 32, 48 kHz.<br>Sample width: not less than 16 bits<br>Number of channels: not less than five<br>Dynamic range of DAC: not less than 80 dB.<br>Additional hardware and software requirements: in accordance with 6.2 and 6.3 |
| Set of active acoustic systems for noise simulation in vehicle cabin | Number of channels: not less than five (four broadband and one subwoofer).<br>Rated power sufficient for noise generation in vehicle compartment: not less than 90 dBA.<br>Additional requirements: in accordance with 6.2 and 6.3 |
| Digital microphone for binaural recording of acoustic signals in vehicle cabin | Hardware or software (PC-based) DAT.<br>Sampling rate: not less than 32 kHz.<br>Sample width: not less than 16 bits.<br>Number of channels: not less than two.<br>Dynamic range: not less than 80 dB. |
| Digital microphone for recording of electric signals in receive and transmit paths on operator side | Hardware or software (PC-based) DAT.<br>Sampling rate: not less than 32 kHz.<br>Sample width: not less than 16 bits.<br>Number of channels: not less than two.<br>Dynamic range: not less than 80 dB. |
| Electronic voltmeter for measurements of sine-wave signals | Accuracy class: 1.5.<br>Frequency range: from 20 to 20000 Hz.<br>Measurement range: from 1 mV to 10 V.<br>Input resistance: not less than 1 MOhm. |
| Acoustic calibrator of sound pressure level | As per [28] for $^1/_2$" pressure microphone. |
| N o t e — The microphone parameters are measured in an anechoic chamber using the equipment specified in [29] taking into account the requirements of 7.12.1. | |

# Appendix B
# (obligatory)

## Test signals and their levels

**B.1 Speech and speech-like signals**

Both narrowband and wideband artificial speech-like signals used in measurements are generated in accordance with [11] and [14]. For a test CSS in the case of wideband IVDS, an additional spectrum spreading from 4 to 8 kHz with a 5 dB fall per octave in high frequency direction is used with the properties specified in Figure 6 of [11].

The detailed information regarding the levels and durations of the respective test signals is directly included in the description of each test.

All test signals used in receiving direction (applied to the system simulator) shall be frequency-limited. For narrowband IVDS, this is achieved using a band-pass filter with the low cut at 200 Hz, high cut at 4 kHz and a frequency response slope of at most 24 dB per octave. For wideband IVDS, a band-pass filter with the low cut at 50 Hz, high cut at 8 kHz and a frequency response slope of at most 24 dB per octave is used.

In sending direction, no frequency limitations are specified for any test signal of artificial voice.

Unless otherwise specified, all test signal levels used in this Standard are r.m.s. values derived by averaging along the full signal path including pauses. The active signal level (excluding pauses) is calculated in accordance with [20].

The following test signal levels are considered nominal:

1) for electric signals in receiving direction: minus 16 dBm0 (typical signal level in a communication network);

2) for acoustic signals in sending direction: minus 1.7 dBPa at the MRP (typical average level of speech increased taking into account the loudspeaker communication effect [6]) or minus 25.7 dBPa at the microphone HFRP (allowing for correction as per 6.4.2) except for the tests carried out in acoustic noise conditions where a person may inadvertently increase speech loudness (allowing for correction as per 6.4.4).

Certain tests require accurate time synchronisation for signals applied in receiving and sending directions. Such tests shall make provisions for signal delays that occur in the IVDS, in speech codecs and in communication networks.

**B.2 Noise signals**

Noise signals are used in some measurements to simulate external acoustic noise in the vehicle cabin. They are specific to each vehicle model; therefore, they shall be recorded separately during real movements of each vehicle, based on a number of noise situation scenarios described in Table D.1 (Appendix D) and in Table 18.

As regards the loudspeaker communication in acoustic noise conditions, an IVDS shall be tested for all noise scenarios listed herein. If any additional essential vehicle features may affect the noise level in the cabin, they shall be taken into account as well, and the list of noise scenarios for the tests shall be extended.

In general, it is recommended to keep the noise scenario unchanged for the duration of the test so that its parameters (SNR, vehicle speed, noise spectrum content, etc.) remain approximately the same. These conditions enable reproducibility of measurements.

If the noise file description or the documents of the vehicle manufacturer do not specify an exact noise signal level, the latter is assumed to be minus 24 dBPa(A) (70 dBPa SPL) for the "ordinary" noise environment, and minus 14 dBPa(A) (80 dBPa SPL) for the "worst case" one. The noise SPL is measured in the right ear of the HATS manikin's artificial head (with the driver to the left from it in the vehicle cabin).

**B.2.1 Noise signal recording**

Noise signals are recorded in the actual vehicle under test. The measuring microphone is installed as close as possible to the IVDS microphone.

If the IVDS is equipped with a digital debug interface described in Appendix C, noise signals may be recorded from the IVDS microphone directly and then mixed into the transmit channel either digitally or electrically. This is especially advantageous if a microphone array is used as an input converter of the IVDS, because the use of four speakers for external noise modelling in the IVDS cabin can not provide for accurate reproduction of spatial acoustic field parameters that describe real noise signals, wind noise or other disturbing effects.

Acoustic noises are recorded for each vehicle model under test. Table D.1 (Appendix D) and Table 22 contain the recommended list of noise scenarios to be used for recording and IVDS performance checks.

If the test is intended for performance quality comparison of different IVDS or algorithms, then such comparison shall be carried out in the identical conditions, i.e., using the same vehicles, noise scenarios and sound signal recordings.

**B.2.2 Playback of noise signals**

Three methods are recommended for playback of noise signals depending on the purpose of the test:

1) Acoustic method

The noise in the vehicle cabin is reproduced using four speakers (see 6.2.1). Two speakers are installed at the front (from the left and from the right) and the other two, at the rear (from the left and from the right). The exact locations shall be selected so that the direct visibility between the IVDS microphone and the artificial head of the HATS manikin would not be compromised. The amplification in the playback channels shall be calibrated with respect to the sound pressure level, and the frequency response, including the one of the speakers, shall be equalised. The detailed information is given in [4].

2) Electric method

Noise signals may be electrically injected into a signal from the IVDS microphone or microphones. For this purpose, noise recordings made using the IVDS microphone of a given type installed at a given point of the vehicle compartment shall be used, and an appropriate electronic circuit that allows mixing the signals from different sources shall be inserted at a microphone circuit break, for example, in a way similar to the one illustrated in Figure B.1. In addition, the signal levels shall be calibrated, and the check shall be made to make sure that the adder circuit does not introduce additional noises into microphone signals.

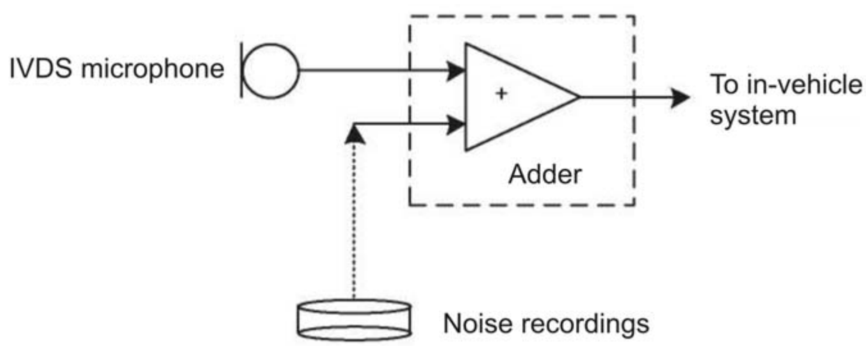



Fig. B.1 — Flow diagram of electric method used to inject previously recorded noises into test signals

3) Digital method

Noise signals may be recorded and later mixed with the IVDS microphone signal digitally, using a DI-S2 interface shown in Figure B.1.

The last two methods enable modelling acoustic noises of complex types, e.g., wind noise, and taking into account spatial properties of noise fields which is very important when directional microphones or microphone arrays are used in the IVDS.

## Appendix C
## (recommended)

## Digital interface used in tests

C.1 The recommended application of this Standard by loudspeaker IVDS developers for the purpose of IVDS performance assessment at the prototyping stage includes implementation of an additional digital debug IVDS interface for signal I/O.

On the one hand, this will avoid labour-consuming acoustic measurements, and on the other hand this will obviate the use of the system simulator for intermediate measurements.

The use of such digital interface is also recommended when an IVDS used for a particular vehicle model is adapted and configured for loudspeaker communication. The final IVDS tests are always completed using the standard (acoustic or electric) method.

**C.2 Recommended digital interface**

The digital interface is intended for tests of speech processing algorithms included in the IVDS, and considers the IVDS as a black box with two signal processing directions: receive channel and transmit channel. Sound signals in digital form at the channel inputs and outputs may be read and transmitted from the IVDS to the PC for their recording in files, or read from PC files and transmitted to the IVDS in real time. It provides for simulation of input acoustic signals in the transmit channel or of electric input signals in the receive channel.

Figure C.1 illustrates the digital debug interface and its possible read/write access points for signals.

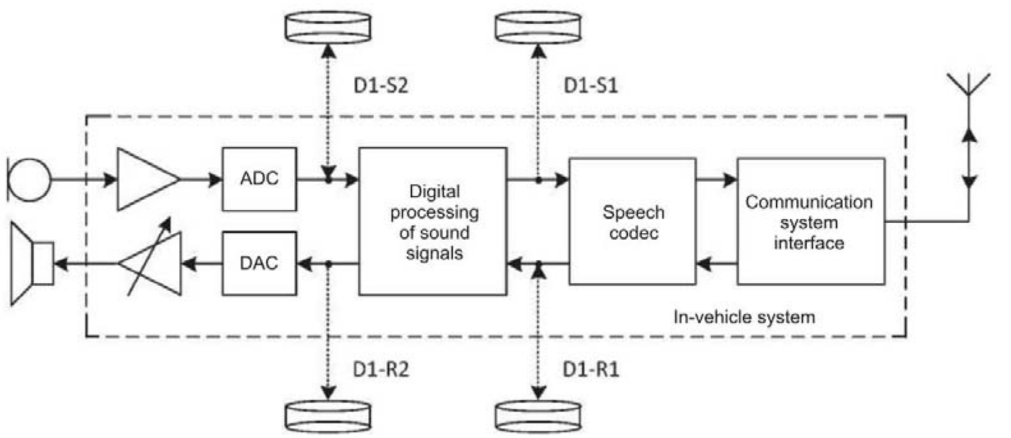



Fig. C.1 — Digital debug interface

Points DI-Rx are related to the receive channel, and DI-Sx, to the transmit channel, where x is the reference point number.

Point DI-R1 (Rin) may be used either for output of signals received from the far-end subscriber and their recording in a file, or for local input of such signals from a PC file to the IVDS without the use of the system simulator.

Point DI-R2 (Rout) may be used for output of signals processed in the receive channel, e.g., of signals processed by an AGC that responds to the noise level variations in the vehicle cabin.

Point DI-S2 (Sin) may be used either for output of echo signals, acoustic signals and signals transmitted from the near-end subscriber as well as for their recording in a file, or for local input of such signals form a PC file to the IVDS without the use of the acoustic input. The latter ensures good reproducibility of test results, enables the use of artificial test signals at a given SNR without the use of the acoustic noise simulator, and also does not require HATS manikin calibration or fixing its position.

Point DI-S1 (Sout) may be used for output of signals processed in the transmit channel, e.g., of signals processed by an echo canceller, noise suppressor, AGC, etc., and for their recording in a PC file.

The analysis of the transmitted signals at point DI-S1 does not include their encoding/decoding in speech codecs of mobile communication systems.

The latter must be kept in mind when the end-to-end transmission quality of voice signals from the vehicle to the operator is assessed, because many speech codecs are extremely sensitive to the noise level, and exhibit low intelligibility for the speech with an SNR below a certain threshold.

If a narrowband IVDS implements this digital interface, it shall support at least one of the following I/O formats:

1) linear PCM 16 bit with a sampling rate of 8 kHz;

2) non-linear PCM 8 bit a sampling rate of 8 kHz, A-Law or μ-Low encoding as per [30].

If a wideband IVDS implements this digital interface, it shall support at least one of the following I/O formats:

1) linear PCM 16 bit with a sampling rate of 16 kHz;

2) two-band ADPCM with a bitrate of 64 kbit/s as per [31].

If different sampling rates are used for signal processing in the IVDS, they also may be used for I/O of test signals provided that the required resampling is performed on the PC.

Acoustic and electric signal levels are converted to digital levels in the implemented digital interface. For digital signals, the nominal signal level at the reference points is selected by the IVDS manufacturer basing on the required dynamic range of signals and on the overload margin. The recommended nominal values of digital signals for narrowband and wideband IVDS are given in Appendix B.

A hardware implementation of the interface in regard to real-time signal exchange between the IVDS and the PC is not standardised, and is specified by the IVDS manufacturer. For real-time tests, an implementation of simultaneous signal I/O through several channels with a fixed delay between those channels is necessary.

If such digital interface is available, the PC software for signal exchange and for recording of signals received from the IVDS shall be provided.

An alternative method used to test the processing algorithms for signals entering the IVDS is their modelling on a PC with the file I/O of the signals when such algorithms are developed.

**C.3 Test using digital interface**

The digital interface may be used to perform most tests described in Section 7. If the digital interface is implemented in an IVDS, the following otherwise unavailable recordings and tests are recommended.

**C.3.1 Recording of acoustic noise and generation of test signals with specified SNR**

Many tests require acoustic noise recordings in the IVDS cabin with the noise waveform coinciding with the one at the IVDS input connected to the microphone. For this purpose, the DI-S2 interface may be used for digital recording of noise. Test signals of a given SNR may then be generated on the PC for the near-end subscriber's speech and sent to the IVDS input through the same DI-S2 interface.

**C.3.2 Recording of near-end subscriber's speech**

Many tests require speech recordings of the near-end subscriber (either real person or manikin). For this purpose, the DI-S2 interface may be used for digital recording of speech. Test signals of a given SNR may then be generated on the PC for the near-end subscriber's speech and sent to the IVDS input through the same DI-S2 interface.

Two male and two female talkers are engaged for test speech recording where each talker must pronounce several phonetically balanced phrases as per GOST 16600.

**C.3.3 Objective speech quality evaluation in sending direction during single-talk**

This evaluation is based on the PESQ-MOS criterion as per [32] and [33] for narrowband IVDS and [34] for wideband IVDS, and may be carried out using the digital interface. The tests are performed with test signals prepared before, applied to the DI-S2 interface and picked up from the DI-S1 interface as well as from the electric output of the system simulator (at the POI).

The speech quality assessed objectively using the PESQ-MOS criterion and denoted as MOS-LQO (S1) shall be higher than the speech quality at the system simulator output MOS-LQO (POI) since no network transfer or low bitrate encoding of speech is involved now.

For a narrowband IVDS, the following requirements shall be met:

$$\text{MOS-LQON(SI)} \geq \text{MOS-LQON(POI)} \geq 3{,}0.$$

For a wideband IVDS, the following requirements shall be met:

$$\text{MOS-LQOW(S1)} \geq \text{MOS-LQOW(POI)} \geq 3{,}6.$$

The values of the difference

$$\text{DELTAN} = \text{MOS-LQON(SI) minus MOS-LQON(POI) and}$$
$$\text{DELTAW} = \text{MOS-LQOW(SI) minus MOS-LQOW(POI)}$$

may be considered as quality degradation values due to encoding and to transfer through the mobile communication systems.

**C.3.4 Objective speech quality evaluation in sending direction during double-talk**

The digital interface enables distortion measurements for voice signals transmitted in double-talk mode. This test makes use of the objective speech quality assessment in order to optimise speech processing parameters in the IVDS for double-talk mode.

The test signal in sending direction with the near-end subscriber's speech and echo signals of the far-end subscriber are recorded using the DI-S2 interface. The near-end subscriber's speech is used as a reference signal for evaluation of distortion level in sending direction during double-talk.

The test signal in receiving direction with the far-end subscriber's speech shall be uncorrelated with the signals of the near-end subscriber.

The following test procedure is used.

1) Before the test, make sure that the acoustic echo canceller of the IVDS has completed its setup for the current echo path, and is set to maximum echo suppression. This may be achieved by applying a training signal sequence to the DI-R1 input and its reflected echo signal picked up from the microphone directly, to the DI-S2 input.

2) In real time, apply the test speech signals to the IVDS from the two sides: the signal in receiving direction to the DI-R1 input, and the reflected acoustic echo signal mixed with the recorded near-end subscriber's speech in sending direction, to the DI-S2 input. Always use test recordings of two different talkers in receiving and in sending direction in order to avoid false AEC convergence, namely, use two different female voices in 25 % of cases, two different male voices in 25 % of cases, and a male voice combined with a female voice in other cases.

3) Save the reflected echo signal through the DI-S2 interface, and the processed speech signal with suppressed echo signal, through the DI-S1 interface.

4) Calculate the objective speech quality indicators PESQ-MOS as per [32]—[34] using the original signal of the near-end subscriber's speech in sending direction as a reference, and the processed speech signals with distortions the picked up at the DI-S1 and at the output of the system simulator in receiving direction (at the POI).

For a narrowband IVDS, the following requirements shall be met:

$$\text{MOS-LQON(SI)} \geq \text{MOS-LQON(POI)} \geq 2{,}5$$

For a wideband IVDS, the following requirements shall be met:

$$\text{MOS-LQOW(S1)} \geq \text{MOS-LQOW(POI)} \geq 2{,}5.$$

The values of the difference

$$\text{DELTAN} = \text{MOS-LQON(SI) minus MOS-LQON(POI) и}$$
$$\text{DELTAW} = \text{MOS-LQOW(SI) minus MOS-LQOW(POI)}$$

may be considered as quality degradation values due to encoding and to transfer through the mobile communication systems.

## Appendix D
## (obligatory)

## Minimum standard set of noise scenarios

Table D.1 — Minimum standard set of scenarios for recording of noise signals

| | Description | Speed, km/h | Operating mode of HVAC system fan | Windows | Windscreen wipers | Turn signal | Background talk | Road pavement |
|---|---|---|---|---|---|---|---|---|
| 1 | Stopped with engine on. Low fan noise. | 0 | Set to minimum level | | | | | — |
| 2 | Moving in town. Medium fan noise. | 60 | Set to medium level. Air flows in direction from IVDS microphone | Closed | Off | Off | Not present | Dry and rough road |
| 3 | Moving along route. Low fan noise. | 120 | Set to minimum level | | | | | |

N o t e s
1 Weather conditions: temperature above minus 20 °C and below plus 40 °C, wind speed not exceeding 5 m/s, no precipitation. The road shall be dry and fairly rough, but with no dents and bumps. A concrete pavement is preferable because it generates the loudest noise in the vehicle compartment. It shall be checked that the air flow from the HVAC system does not directly strike the microphone which is recording acoustic noises in the vehicle compartment.
2 Instead of 120 km/h, noise signals may be recorded at a speed typical to movement of this vehicle along the route.

# Appendix E
# (recommended)

## Characteristics of electro-acoustic elements and assessment methods of these characteristics

E.1 This Appendix specifies the basic parameters and requirements for microphones used in a loudspeaker IVDS if such microphones may be tested separately from the IVDS and the IVDS itself has an input for connection of an external microphone.

The tests involve the checks of individual external microphones (directional or non-directional, passive or active), and shall not apply to microphone arrays that use any additional signal processing (beam forming of directivity pattern, clearing of noise, etc.).

If a microphone is included in the IVDS package submitted for certification, then the measurements described in this section are not mandatory.

The requirements for acoustic measurements carried out during the tests are established in [29].

Microphones are first tested in standard conditions of an anechoic chamber, and then at their selected locations in the vehicle compartment.

**E.2 Measurements in anechoic chamber**

Measurements in standard conditions of acoustic free fields in an anechoic chamber ensure that the original parameters of microphones are unaffected by the acoustic properties of the vehicle compartment and by the microphone location or orientation. A reference speaker with a low level of intrinsic electro-mechanical distortions is used in these measurements.

**E.3 Microphone sensitivity**

E.3.1 The measurement of the IVDS microphone sensitivity is mandatory. This parameter is first measured in an anechoic chamber and then checked after the microphone is installed in the vehicle compartment. The requirements for the microphone sensitivity are specified by the IVDS manufacturer. The requirements for unification of IVDS sensitivity as per GOST 33464 are treated as recommendations.

E.3.2 Requirements

The microphone sensitivity at 1 kHz measured along the maximum of its directivity pattern (DP) shall be as specified by the IVDS manufacturer. To ensure the interchangeability of microphones, unifying their sensitivity for IVDS at a level of 300 mV/Pa ± 3 dB is recommended.

**E.3.3 Method of measurement**

1) The measurements shall be performed in free acoustic field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed at a distance of 1 m on a line passing through the centre of the measuring loudspeaker.

3) The acoustic test signal shall be a 1 kHz sine-wave signal of the SPL equal to 0 dBPa at the microphone installation point as measured for an unperturbed acoustic field with no microphone present.

4) The microphone under test is oriented towards the speaker to achieve the maximum of DP (output voltage).

5) The microphone sensitivity is determined in mV/Pa.

Note — A narrowband (one-third octave) noise signal with a centre frequency of 1 kHz and the SPL equal to 0 dBPa may be used as the test signal.

**E.4 Frequency response of microphone**

E.4.1 Requirements

The frequency response of the IVDS microphone measured in free acoustic field conditions shall be within the tolerances listed in Table E.1 for narrowband IVDS or in Table E.2 for wideband IVDS.

Table E.1 — Frequency response of microphones for narrowband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 200 | 0 | – ∞ |
| 250 | 0 | – ∞ |
| 315 | 0 | –14 |

*Table E.1 (continued)*

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 400 | 0 | –13 |
| 500 | 0 | –12 |
| 630 | 0 | –11 |
| 800 | 0 | –10 |
| 1000 | 0 | –8 |
| 1300 | +2 | –8 |
| 1600 | +3 | –8 |
| 2000 | +4 | –8 |
| 2500 | +4 | –8 |
| 3100 | +4 | –8 |
| 4000 | +4 | – ∞ |

T a b l e E.2 — Frequency response of microphones for wideband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 100 | 0 | – ∞ |
| 125 | 0 | – ∞ |
| 200 | 0 | –14 |
| 315 | 0 | –13 |
| 400 | 0 | –12 |
| 500 | 0 | –11 |
| 630 | 0 | –10 |
| 1000 | +0 | –8 |
| 1300 | +2 | –8 |
| 1600 | +3 | –8 |
| 2000 | +4 | –8 |
| 3100 | +4 | –8 |
| 4000 | +4 | –8 |
| 8000 | +4 | – ∞ |

The frequency response shall be flat in the range of 200 Hz — 4 kHz for narrowband IVDS or in the range of 100 Hz — 7 kHz for wideband IVDS.

**E.4.2 Method of measurement**

1) The measurements shall be performed in free acoustic field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed at a distance of 1 m on a line passing through the centre of the measuring loudspeaker.

3) The acoustic test signal shall be a 1 kHz sine-wave signal of the SPL equal to 0 dBPa at the microphone installation point as measured for unperturbed acoustic fields with no microphone present.

4) The microphone under test is oriented towards the speaker to achieve the maximum output voltage at 1 kHz.

5) The microphone sensitivity is determined in mV/Pa for all frequencies.

**E.5 Harmonic distortion level of microphone**

**E.5.1 Requirements**

The total harmonic distortion of the microphone for test sine-wave signals with the SPL equal to 0 dBPa shall not exceed 1 %.

**E.5.2 Method of measurement**

1) The measurements shall be performed in free acoustic field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed on a line passing through the centre of the measuring loudspeaker.

3) The acoustic test signals shall be sine-wave signals of frequencies 300 Hz, 500 Hz and 1 kHz for narrowband IVDS, or 300 Hz, 500 Hz, 1 kHz and 2 kHz for wideband IVDS, with the SPL equal to 0 dBPa at the microphone installation point as measured for unperturbed acoustic fields with no microphone present.

4) The microphone under test is oriented towards the speaker to achieve the DP maximum at 1 kHz.

5) The total harmonic distortion is determined in percentage for each frequency. The measurement is carried out in the operating frequency band of the IVDS.

N o t e s

1 The measuring microphone shall be used in advance to check that the reference speaker itself has lower distortions than the microphone under test at a given SPL and given frequencies.

2 The IVDS microphone distortions at an SPL below 0 dBPa shall be also checked not to exceed a given value.

**E.6 Maximum sound pressure level**

E.6.1 The maximum SPL of the IVDS microphone is determined as a level limited by harmonic distortions of the microphone.

**E.6.2 Requirements**

The maximum SPL limited by microphone distortions equal to 3 % for a 1 kHz test signal shall be at least 12 dBPa (106 dB SPL) for microphones with a typical sensitivity of 300 mV/Pa.

**E.6.3 Method of measurement**

1) The measurements shall be performed in free acoustic field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed on a line passing through the centre of the measuring loudspeaker.

3) The acoustic test signal shall be a 1 kHz sine-wave signal of the SPL gradually increased until the distortion level of the microphone reaches 3 %.

4) The microphone under test is oriented towards the speaker to achieve the DP maximum.

5) The SPL at the microphone installation point as measured for unperturbed acoustic fields with no microphone present is expressed in dB SPL or dBPa (see Appendix G).

N o t e s

1 The measuring microphone shall be used in advance to check that the reference speaker itself has lower distortions than the microphone under test at a given SPL and given frequencies.

2 For a properly designed microphone, its maximum SPL is limited by electric distortions due to limitations of the measuring circuit rather than by its mechanical structure. For the microphone sensitivity of 300 mV/Pa at the SPL equal to 106 dB SPL, the peak-to-peak voltage ($V_{pp}$) of the output sine-wave signal may reach 3.3 V.

**E.7 Microphone self-noise**

**E.7.1 Requirements**

The microphone self-noise at a 300 mV/Pa sensitivity shall not exceed minus 72 dBV(*A*) [at most minus 66 dBV(*A*) accepted if this noise does not impair the IVDS noise performance in sending direction].

**E.7.2 Method of measurement**

1) No test signal is used.

2) The microphone is powered from a source with a low level of self-noise.

3) The microphone self-noise is measured at the output of the circuit shown in Figure 2 in the frequency range between 100 Hz and 4 kHz, using psophometric frequency-weighting along curve *A*.

4) The self-noise is expressed in dBV(*A*).

N o t e — It should be checked that the ambient acoustic noise is lower than the equivalent self-noise of the microphone expressed in dB SPL (see Appendix G).

**E.8 Spatial selectivity**

E.8.1 The special selectivity of the microphone is described by its directivity pattern (DP) that defines the microphone sensitivity versus the incidence angle of a flat acoustic wave.

The front-to-back ratio is a ratio of sensitivity in the direction of the DP maximum to sensitivity in the direction of the DP minimum, as measured at a 1 kHz frequency and expressed in decibels.

E.8.2 Requirements

To achieve the required suppression of the background noise in the vehicle compartment, the front-to-back ratio of least 10 dB is recommended.

Note — The final SNR improvement depends on the microphone orientation and its place in the vehicle compartment. If located improperly, a less isotropic microphone may lead to results worse than a more isotropic one.

**E.8.3 Method of measurement**

1) The measurements shall be performed in free acoustic field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed at a distance of 1 m on a line passing through the centre of the measuring loudspeaker.

3) The acoustic test signal shall be a 1 kHz sine-wave signal of the SPL equal to 0 dBPa at the microphone installation point as measured for unperturbed acoustic fields with no microphone present.

4) The microphone under test is oriented towards the speaker to achieve the DP maximum for the first measurement, and to achieve the DP minimum for the second measurement. If the exact minimum position is unknown, it shall be determined by rotating the microphone.

5) The front-to-back ratio is determined in decibels.

**E.9 Measurements in vehicle compartment**

E.9.1 These measurements are aimed at finding the best location/orientation of the IVDS microphone and at assessing the acoustic effects of the vehicle compartment on the microphone parameters. They are carried out using an artificial mouth device inside the HATS manikin head.

**E.9.2 Selection of microphone location in vehicle compartment**

The optimum microphone location is different for various vehicle types, and shall be selected experimentally based on the general guidelines given below:

1) the microphone shall be located as close as possible to the talker to fall into the closest area of his acoustic field (usually, at most 50—100 cm away) where the energy of a direct sound beam is greater than the total energy of reflected beams, and the reverberation level is low. This condition largely affects the illegibility and spectral contents of speech;

2) the energy of the direct talker's voice beam is inversely proportional to the squared distance from the microphone, and the acoustic noise energy does not depend on the distance to the talker; thus, increasing this distance decreases the SNR. A single microphone located next to the talker may provide a better SNR than a microphone array located at an increased distance;

3) no obstacles shall be present between the microphone and the talker's mouth: they would decrease the SNR and increase reverberation;

4) the direction of DP maximum shall on an average coincide with the one to the talker's mouth;

5) the IVDS microphone shall be protected from direct air flows from partially opened windows, conditioning systems, etc., to the vehicle compartment;

6) the IVDS microphone shall be protected from limitation (saturation) caused by nearby speakers, especially, low-frequency ones, air vibration in the compartment due to engine operation or pressure drops;

7) the IVDS microphone shall be properly suspended so that any noises from vehicle body vibration can not enter the transmit channel.

Powering the microphone from the vehicle power network or from the IVDS is recommended when the microphone parameters are measured.

**E.10 Microphone sensitivity in vehicle compartment**

E.10.1 This parameter is measured in an anechoic chamber and checked in the vehicle compartment to ensure that the microphone and its location are suitable for achieving the required voltage at the IVDS input. The requirements for the nominal input level of the IVDS are specified by the IVDS manufacturer. The requirements of GOST 33464 for unification of the nominal input level are treated as recommendations.

E.10.2 Requirements

The recommended signal level at the microphone output for acoustic signals with the SPL of 0 dBPa at the MRP is 19 mV ± 3 dB, which is equivalent to the microphone sensitivity of 300 mV/Pa and anechoic chamber measurement at a distance of 50 cm between the microphone and the MRP.

This requirement may be corrected by the IVDS manufacturer allowing for possible variations in the microphone sensitivity or microphone location in the compartment.

**E.10.3 Method of measurement**

1) The measurements in the compartment shall be done using the arrangement shown in Figure 2.

2) The acoustic test signal shall be a narrowband (one-third octave) noise signal with a centre frequency of 1 kHz and the SPL equal to 0 dBPa at the MRP.

3) The output voltage of the microphone is measured in millivolts.

**E.11 Frequency response of microphone in vehicle compartment**

E.11.1 The frequency response is measured in an anechoic chamber and checked in the vehicle compartment to ensure that the microphone and its location are suitable for observing the required AFR tolerances in sending direction allowing for acoustic effects of the vehicle compartment.

**E.11.2 Requirements**

The microphone frequency response in the vehicle compartment is checked along the path from the MRP to the electric output of the measuring circuit shown in Figure 2.

The relative tolerances for this frequency response in sending direction are given in Table E.3 for narrowband IVDS and in Table D.4 for wideband IVDS. For intermediate frequencies, linear interpolation in log-log scale may be used.

Table E.3 — Frequency response of microphones for narrowband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 200 | 0 | – ∞ |
| 250 | 0 | – ∞ |
| 315 | 0 | –14 |
| 400 | 0 | –13 |
| 500 | 0 | –12 |
| 630 | 0 | –11 |
| 800 | 0 | –10 |
| 1000 | 0 | –8 |
| 1300 | 2 | –8 |
| 1600 | 3 | –8 |
| 2000 | 4 | –8 |
| 2500 | 4 | –8 |
| 3100 | 4 | –8 |
| 4000 | 4 | – ∞ |

Table E.4 — Frequency response of microphones for wideband IVDS

| Frequency, Hz | Upper limit, dB | Lower limit, dB |
|---|---|---|
| 100 | 4 | – ∞ |
| 300 | 4 | –4 |
| 1000 | 4 | –4 |
| 5000 | 8.5 | –4 |
| 6300 | 9 | –7 |
| 8000 | 9 | – ∞ |

The frequency response of microphones shall be flat in the range of 200 Hz — 4 kHz for narrowband IVDS or in the range of 100 Hz — 7 kHz for wideband IVDS. A frequency response with additional frequency-weighting, e.g., with a roll-off to the LF part or slight rise in the HF part within the specified tolerances may be more preferable.

**E.11.3 Method of measurement**

1) The measurements in the vehicle compartment shall be performed using the arrangement shown in Figure 2.

2) The acoustic test signal shall be an artificial voice as per [8], periodic broadband noise or CSS as per [17]. The artificial mouth shall be calibrated and equalized at the MRP. The SPL of the test signal at the HFRP of the HATS is set to minus 28.7 dBPa (level averaged over the whole test signal path).

3) The power spectrum density of the test signal measured at the MRP is used as an initial value when the frequency response of the IVDS in sending direction is calculated.

4) The frequency response of the microphone sensitivity is determined in one-third octave bands as per [16], in the frequency range from 100 Hz to 4 kHz for narrowband IVDS, and from 100 Hz to 8 kHz for wideband IVDS. To calculate the signal level in each frequency band, the signal is averaged along its full path.

5) The microphone sensitivity is expressed in dBV/Pa (see Appendix G).

**E.12 Directional properties of microphone in vehicle compartment**

**E.12.1 Requirements**

The recommended SNR improvement ensured by directional properties of the IVDS microphone for the driver's speech in noises should be at least 3 dB as compared with a non-directional broadband microphone located at the same place, allowing for any effects of differences in frequency-weighting of the signals.

**E.12.2 Method of measurement**

1) The ambient conditions during the tests shall meet the requirements of Section 6. The measurements in the vehicle compartment shall be performed using the arrangement shown in Figure 2.

2) All types of ambient acoustic noises shall be tested in accordance with the scenarios listed in Table D.1. (Appendix D). Noises shall be switched on at least 5 s prior to measurements to let noise reduction algorithms, if any, adapt to such noises [8].

3) Only a disturbing noise signal with a given SPL is first switched on in sending direction and recorded at the output of the measuring circuit shown in Figure 2. The noise level (denoted as LN_hft_mic) is calculated using frequency-weighting along curve *A* and averaging in the frequency band from 200 Hz and 4 kHz for narrowband IVDS, or from 100 Hz and 8 kHz for wideband IVDS. The noise level is expressed in dBV/Pa (*A*) (see Appendix G).

4) Then, a useful signal only (with no noise) is applied in sending direction. This signal shall be a CSS as per [11] of more than two elementary sequences in its duration. The speech level (denoted as LS_hft_mic) is calculated using frequency-weighting along curve *A* and averaging in the frequency band from 200 Hz and 4 kHz for narrowband IVDS, or from 100 Hz and 8 kHz for wideband IVDS. The speech level is expressed in dBV/Pa (*A*) (see Appendix G).

5) An actual frequency response of the IVDS microphone in sending direction is determined for the useful signal (with no noise) in accordance with E.2.2 and recorded for its further use in normalization.

6) The SNR for the IVDS microphone is calculated as SNR_hft_mic=LS_hft_mic minus LN_hft_mic.

7) A non-directional measuring microphone with a flat frequency response is placed as close as possible to the IVDS microphone. Using the CSS signal, both the frequency response of the IVDS microphone and the actual frequency response of the non-directional microphone are measured, and then weighted by the frequency response of the IVDS microphone obtained at step 5. This is required to ensure that the differences in frequency-weighting for the two microphones will not affect comparison of their directional properties.

8) Steps 3) through 4) are repeated with the non-directional microphone taking into account the additional weighting of its frequency response. The measured levels are denoted as LN_omni_mic for the noise and LS_omni_mic for the speech.

9) The SNR defined as SNR_omni_mic = LS_omni_mic minus LN_omni_mic is calculated for the non-directional microphone.

10) The SNR improvement for the directional microphone as compared to the non-directional one is assessed as SNR_hft_mic minus SNR_omni_mic.

## Appendix F
**(recommended)**

## Speech intelligibility in transmit and receive channels

F.1 The speech intelligibility is assessed by the articulatory test method that is detailed in GOST 16600, and gives objective and steadily reproducible results. This method is a standard method of assessing intelligibility for radio communication and wired telephony equipment.

**F.2 Requirements**

F.2.1 The speech intelligibility for an IVDS installed in the vehicle compartment and switched to loudspeaker communication mode is assessed in transmit and receive channels separately, both in silence and in acoustic noise conditions inside the vehicle.

F.2.2 The speech illegibility of the IVDS loudspeaker communication shall conform to at least Class 1 GOST 16600 for single-talk in silence conditions, and at least Class 2 GOST 16600 in the presence of disturbing acoustic noise (Table 1).

F.2.3 If the requirements for noise type and level are not specified by the vehicle manufacturer, the minimum sound pressure level of background noises in the vehicle compartment is taken equal to minus 24 dBPa(A) (70 dBa SPL) for the ordinary noise conditions, and minus 14 dBPa(*A*) (80 dB*A* SPL) for the worst case noise conditions.

F.2.4 The mean grade based on five-grade rating scales shall be at least 3.0 for narrowband IVDS and at least 3.6 for wideband IVDS when they operate in silence or in normal ordinary conditions (depending on the vehicle type and noise scenario).

# Appendix G
## (reference)

## Units of acoustic measurements used in tests

dBm — Power level of electric signals that is relative to 1 mW and expressed in decibels.

dBm0 — Power level of electric signals that is measured in dBm at the reference point of the communication path with zero relative level. At a load resistance of 600 Ohm, the level of 0 dBm corresponds to the effective (r.m.s.) voltage level of 0.775 V, or minus 2.2 dBV.

dBov — Effective (r.m.s.) level of a digital signal in decibels relative to the highest digital signal amplitude (limitation start) possible for a given bit grid. Thus, the maximum possible undistorted level of digital signals is always lower than the 0 dBov level by the value of their peak factor expressed in dB. For example, the maximum undistorted level of a digital sine-wave signal is equal to minus 3.01 dBov.

The relation between digital and electric levels is defined in the ADC/DAC. In telephony applications for narrowband speech at a 8 kHz sampling rate, levels equal to +3.15 dBm0 for A-Law and to +3.18 dBm0 μ-Law as per [30] are taken for an overflow (limitation start) point of analogue sine-wave signals. It follows that the ratio between digital and electric levels will be:

$Y$ [dBov] = $X$ [dBm0] minus 6.15 (for A-Law);

$Y$ [dBov] = $X$ [dBm0] minus 6.18 (for μ-Law).

Thus, the electric signal of a nominal level equal to minus 16 dBm0 will correspond to the digital signal of a level equal to minus 22 dBov.

For wideband speech at a sampling rate of 16 kHz as per [31], a level of +9 dBm0 is taken for an overflow (limitation start) point of analogue sine-wave signals. Therefore, the ratio between digital and electric levels will be:

$Y$ [dBov] = $X$ [dBm0] minus 12.

Thus, the electric signal of a nominal level equal to minus 16 dBm0 will correspond to the digital signal of a level equal to minus 28 dBov.

kbit/s — Transmission rate of a digital stream.

dBPa — Sound pressure level of acoustic signals relative to 1 Pa, expressed in dB.

dBPa(A) — Sound pressure level of acoustic signals relative to 1 Pa, frequency-weighted along A-curve, and expressed in dB.

dB (SPL) — Sound pressure level of acoustic signals measured with respect to 20 μPa and expressed in dB. The SPL scale is useful for evaluation of loudness level of sound signals. The loudness level of 0 dB SPL roughly corresponds to the audibility threshold, and the level of 120 dB SPL, to the human pain audibility threshold. The SPL of 0 dBPa corresponds to 94 dB SPL.

dBV — Voltage level of electric signals with respect to 1 V expressed in dB.

dBV(A) — Voltage level of electric signals with respect to 1 V frequency-weighted along A-curve and expressed in dB.

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| | | |
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* GOST R IEC 60942-2009 Sound calibrators. Specifications and test requirements is valid in the territory of the Russian Federation.

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