FEDERAL AGENCY
ON TECHNICAL REGULATING AND METROLOGY

NATIONAL STANDARD OF THE RUSSIAN FEDERATION

GOST R 55531-2013

Global navigation satellite system

ROAD ACCIDENT EMERGENCY RESPONSE SYSTEM

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

Official Edition
English Version Approved by Interstandard

Moscow
Standartinform
2014

Foreword

1 DEVELOPED by Non-Profit Partnership "Promotion of Development and Use of Navigation Technologies" (NP "GLONASS")

2 SUBMITTED by Technical Committee for standardization TC 363 "Radio navigation"

3 APPROVED AND INTRODUCED by Decree No. 596-*cm*, dated 28.08.2013, of the Federal agency on technical regulating and metrology

4 INTRODUCED FOR THE FIRST TIME

The rules of this standard application are established in GOST R 1.0-2012 (*section 8). Information on the amendments to this standard is published in the annual (as of January, 1st, of the current year) information index «National standards» and the official text of the amendments and corrections is published in the monthly information index «National standards». In case of revision (replacement) or cancellation of this standard an appropriate notice will be published in the nearest release of the monthly issued information index «National standards». The appropriate information, notice and texts shall also be placed in the general-use information system — on official site of Federal Agency of Technical regulating and metrology in Internet (gost.ru)*

Contents

N A T I O N A L   S T A N D A R D   O F   T H E   R U S S I A N   F E D E R A T I O N

Global navigation satellite system

ROAD ACCIDENT EMERGENCY RESPONSE SYSTEM

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

Date of Introduction — 2014—01—01

1 Scope

This Standard applies to in-vehicle emergency call systems/devices, and sets out the methods that shall be used in tests of such systems/devices in order to verify their conformity to the requirements specified in GOST R 54620 in regard to quality of loudspeaker communication in a vehicle cabin.

2 Normative references

The following standards are referred to in this Standard:

GOST R 8.714-2010 (IEC 61260:1995) State system for ensuring the uniformity of measurements. Octave-band and fractional-octave-band filters. Technical requirements and test methods

GOST R 50840-95 Speech transmission over various communication channels. Techniques for measurements of speech quality, intelligibility and voice identification

GOST R 51061-97 Low bitrate speech transmission systems. Speech quality characteristics and their evaluation

GOST R 53188.1-2008 Sound level meters. Part 1. Technical requirements*

GOST R 53576-2009 (IEC 60268-4:2004) Microphones. Methods of measurement of electroacoustic characteristics

GOST R 54620-2011 Global navigation satellite system. Road accident emergency response system. In-vehicle emergency call system. General technical requirements

GOST R IEC 60942-2009 Sound calibrators. Specifications and test 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.

* Withdrawn; GOST 17187-2010 shall be used starting from 01.11.2012.

Official Edition

3 Terms and definitions

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

3.1.1 **in-vehicle emergency call system/device;** (IVS): System/device installed on a wheeled vehicle of a relevant Category and used to evaluate vehicle location, speed and movement direction based on the signals generated by the GLONASS Global Navigation Satellite System (GNSS) in cooperation with other active GNSS, to transmit messages containing vehicle data in automatic (system) or manual (device) mode when a road accident or an accident of other type occurs, and to ensure duplex voice communication with emergency services over wireless mobile communication networks.

Notes

1 In-vehicle emergency call systems are intended for Category M1 or N1 vehicles of permitted design weight less than 2.5 t.

2 In-vehicle emergency call devices are intended for Category M1 or N1 vehicles of permitted design weight above 2.5 t, as well as for Category M2, M3, N2 and N3 vehicles.

3 In case of road accidents or accidents of other type, in-vehicle emergency call systems are capable of transmitting vehicle data messages in manual mode as well.

4 In case of road accidents or accidents of other type, an in-vehicle emergency call device may also be capable of transmitting vehicle data messages in automatic mode. The types of accidents to be detected automatically and the timing parameters of the automatic message transmission function implemented in the device are established in [1; 2].

3.1.2 **acoustic echo canceller;** (AEC): Signal processing device/algorithm included in the IVS in order 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.

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

3.1.3 **acoustic echo suppressor;** (AES): Signal processing device/algorithm included in the IVS in order to suppress acoustic echo signals from a far-end subscriber that enter the transmit channel mixed with the speech of a near-end subscriber, owing 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 the receive and transmit channels, and thus prohibits from full duplex voice communication.

3.1.4 **loudspeaker IVS:** IVS that 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.

3.1.5 **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 an acoustic impedance close to the average input impedance of an adult person's ear in the range of audible sound frequencies.

3.1.6 **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 a 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.

3.1.7 **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.

3.1.8 **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.

3.1.9 **codec:** Device or algorithm used for encoding and decoding of voice or audio signals that are transmitted in wireless 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.

3.1.10 **composite source signal;** (CS signal, CSS): Sound test signal which is a combination of various test signals in succession.

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

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

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

3.1.14 **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.

3.1.15 **sending loudness rating;** (SLR): Weighted electro-acoustic attenuation of a loudspeaker IVS in sending direction; describes a signal attenuation value in the transmit channel between the loudness level of the acoustic signal developed by the near-end subscriber at the mouth reference point and 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. The sending loudness rating takes into account the nature of signal loudness perception by humans, is expressed in dB, and is calculated in accordance with [13].

3.1.16 **receiving loudness rating;** (RLR): Weighted electro-acoustic attenuation of a loudspeaker IVS in receiving direction; describes a signal attenuation value in the receive channel between the electric signal level at the reference point of the system simulator and 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 [13].

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

3.1.18 **half duplex:** Loudspeaker communication sessions that are possible in alternating single-talk mode only.

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

3.1.20 **single-talk mode;** (st): Operating mode of a loudspeaker IVS where both far-end and near-end subscribers talk and listen to each other one at a time, not interrupting one another.

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

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

3.1.22 **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.

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

3.1.24 **narrowband IVS:** In-vehicle 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 **partial duplex:** Communication sessions that partially employ double-talk mode, i.e., the signal of the other party is audible, but its loudness is bouncing.

3.1.26 **wideband IVS:** In-vehicle system operating with wideband 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).

3.1.27 **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.

3.1.28 **mouth reference point;** (MRP): Measuring point of sound pressure levels which is located 25 mm in front of either the human lips, or the signal-emitting ring of an "artificial mouth" device.

3.1.29 **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.

3.1.30 **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.

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

4 Designations and abbreviations

The following designations and abbreviations are used in 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;
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 (variable bitrate encoding of sound files);
$A_{H,R}$ — Attenuation Range in receiving direction during single talk (in IVS receive channel);
$A_{H,R,DT}$ — Attenuation Range in receiving direction during double talk (in IVS receive channel);
$A_{H,S}$ — Attenuation Range in sending direction during single talk (in IVS transmit channel);
$A_{H,S,DT}$ — Attenuation Range in sending direction during double talk (in IVS transmit channel);
CCR — Comparison 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;
TCLw — weighted Terminal Coupling Loss (i.e., weighted crossover attenuation in electro-acoustic signal path);
T_R — Signal processing delay in IVS for receiving direction;
T_S — Signal processing delay in IVS for sending direction;
UMTS — Universal Mobile Telecommunications System.

5 General

5.1 The test methods described in this Standard are intended for IVS checks against the requirements of GOST R 54620 in regard to quality assurance of loudspeaker communications.

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

5.3 The complete IVS test cycle includes the following stages:

1) tests of microphones separately from IVS;
2) objective measurements of IVS specifications;
3) subjective quality assessment of loudspeaker communication, including its assessment in presence of noises for each relevant scenario from among the ones listed in Table 22 and Table D.1 (Appendix D).

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

6 Conditions and procedure of tests

6.1 Basic equipment

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

6.1.2 When an IVS is connected using the acoustic interface for near-end subscriber simulation, a HATS manikin is used; the latter shall include an artificial ear and mouth, and its specifications shall meet the requirements of [12] 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 IVS is connected using the radio interface for far-end subscriber simulation, a system simulator is used; the latter shall comply with all requirements of the mobile communication standard used in the IVS, 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 location of simulator antennas and IVS antennas, as well as the levels of RF signals, shall be selected so 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 IVS 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 IVS shall also be checked to ensure the quality of IVS operation in real operating conditions regardless of the encoding type that might be selected by a base station when the connection to the IVS is established.

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

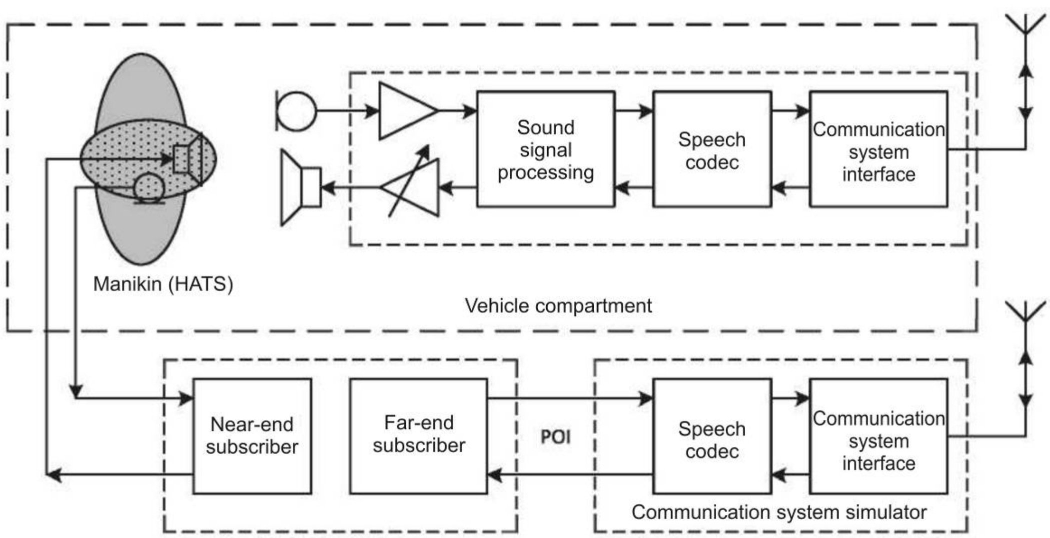


Figure 1 — Flow diagram of testing apparatus for tests of in-vehicle systems

6.1.8 IVS microphones shall be tested separately from the IVS, under free acoustic field conditions (in the 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.

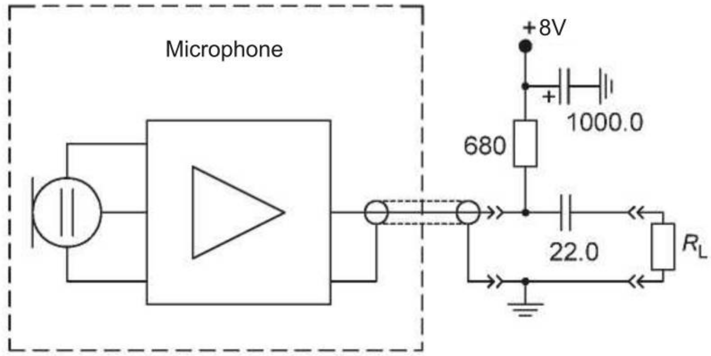


Figure 2 — Flow diagram of testing apparatus for tests of microphones

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 kΩ.

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 [3].

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 (the minimum set of scenarios is presented in Appendix E).

A measuring broadband capacitor microphone installed close to the IVS microphone and a digital recording equipment of a dynamic range not less than 60 dB in the frequency range at least 20—16000 Hz wide 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 Before the playback of acoustic noise recordings, the sound playback unit shall be calibrated in regard to the total SPL using the noise level meter and by 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 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 IVS systems with a single microphone. The frequency response equalisation procedure and the reference set of acoustic noise recordings are presented in [3].

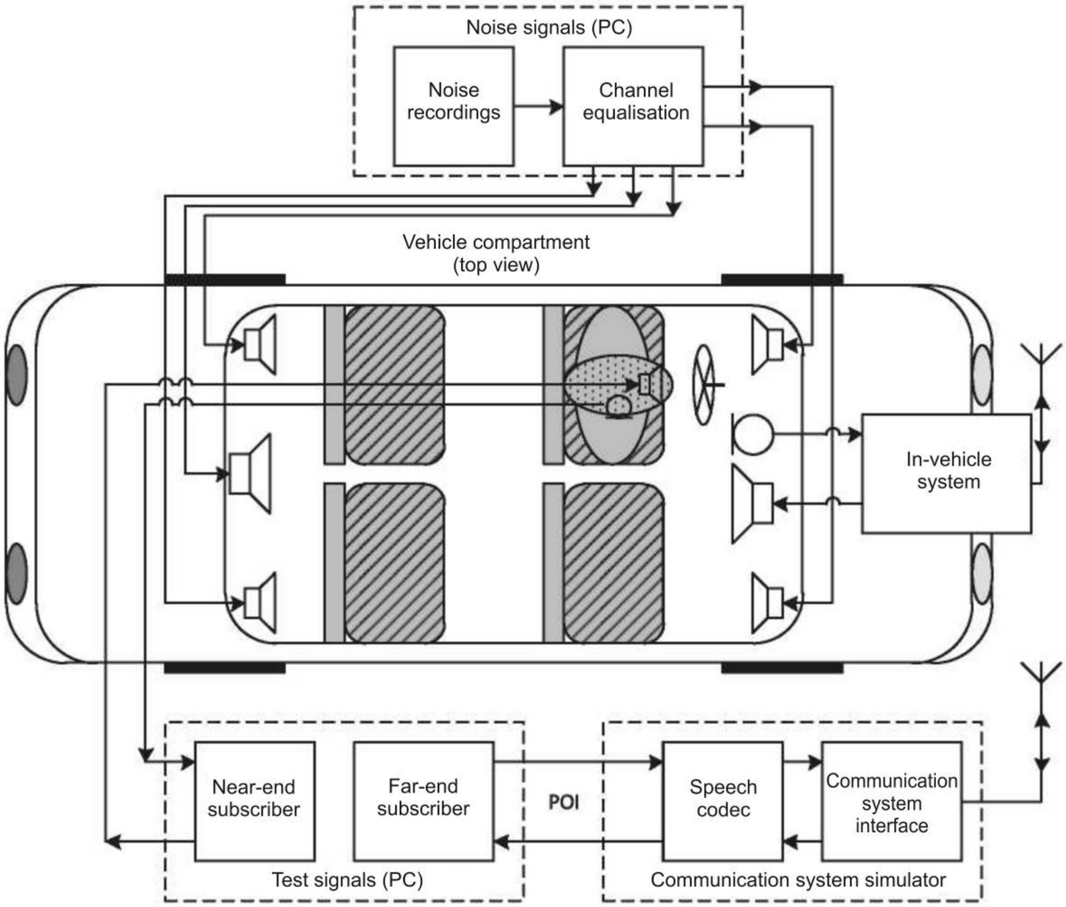


Figure 3 — Testing equipment for simulation of external acoustic noise

6.3 Loudspeaker IVS arrangement in vehicle compartment

6.3.1 In-vehicle systems installed in standard equipment configuration by vehicle manufacturers shall be tested in their originally supplied arrangement with IVS microphones and speakers already installed.

6.3.2 In-vehicle systems installed in auxiliary equipment configuration shall be mounted and configured in the vehicle compartment in accordance with the requirements of the IVS manufacturer. The microphone (or microphone array) and speaker positions shall be clearly indicated by the IVS manufacturer. If such recommendations are missing, the testing laboratory shall determine, based on their own efforts, the position of IVS 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 [12] 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 IVS manufacturer do not enforce any clear requirements regarding the distance between the MRP point of the HATS manikin and the IVS microphone (HFRP point), then the testing laboratory shall determine an MRP to HFRP 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.

A marking both in the vehicle compartment itself 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 HFRP distance possible for the driver where the SNR of voice signals for sending direction would be minimal.

6.4 Requirements for "artificial mouth" devices

6.4.1 An artificial mouth located in the artificial HATS head shall meet the requirements of [9] and [12], and its transfer function shall be equalised at the MRP, with the SPL for voice signals in sending direction equal to minus 4.7 dBPa (89.3 dB SPL) in accordance with the requirements of [14].

6.4.2 For the IVS at the HFRP, the average SPL for voice signals in sending direction shall be set to minus 28.7 dBPa by means of SPL correction at the MRP. The usage procedures of HATS manikins in tests of loudspeaker devices including their equalisation and calibration procedures are described in [20].

6.4.3 The acoustic level of voice signals in sending direction with an average level equal to minus 28.7 dBPa (65.3 dB SPL) at the HFRP point of the IVS microphone is primary for most tests, and corresponds to "normal" loudness of human voice during the talk at a distance of (0.5 – 1) m.

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:

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

where I is the increment of the output 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.

6.5 Requirements for "artificial ear" devices

6.5.1 The IVS tests may employ signals of one or both artificial ears in the manikin's head. If a single ear is used, it shall be a right-side ear in left-hand steering vehicles, and vice versa.

6.5.2 The transfer function of an artificial ear shall be equalised in free acoustic field conditions in accordance with [20].

6.6 Eliminating effects of mobile communication system

6.6.1 The IVS 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 tests.

6.6.2 If the requirements for IVS 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 IVS, 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 given AFR transfer function for an artificial mouth in HATS manikin's head at the MRP and HFRP points;

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 IVS, 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 IVS 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 System simulator setup

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, a 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 NC40).

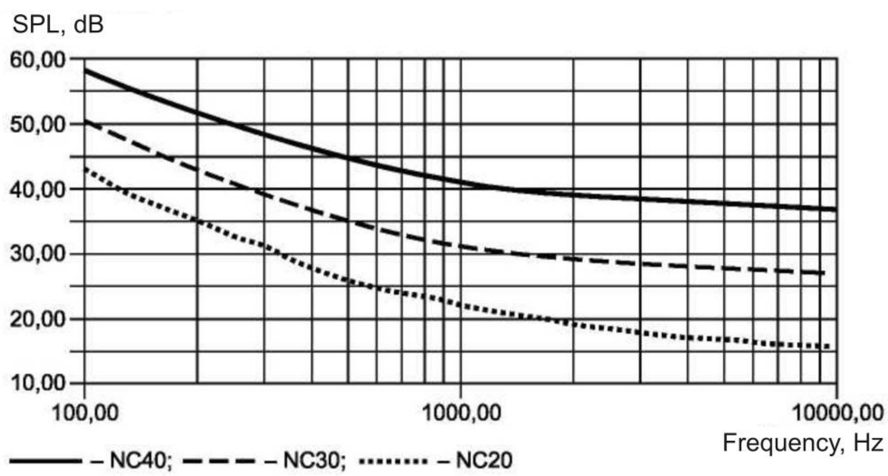


Figure 4 — Spectral density of ambient noise

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 NC30 profile; in such tests, the maximum level of minus 74 dBPa(A) and the spectral density no higher than NC20 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 IVS. 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.

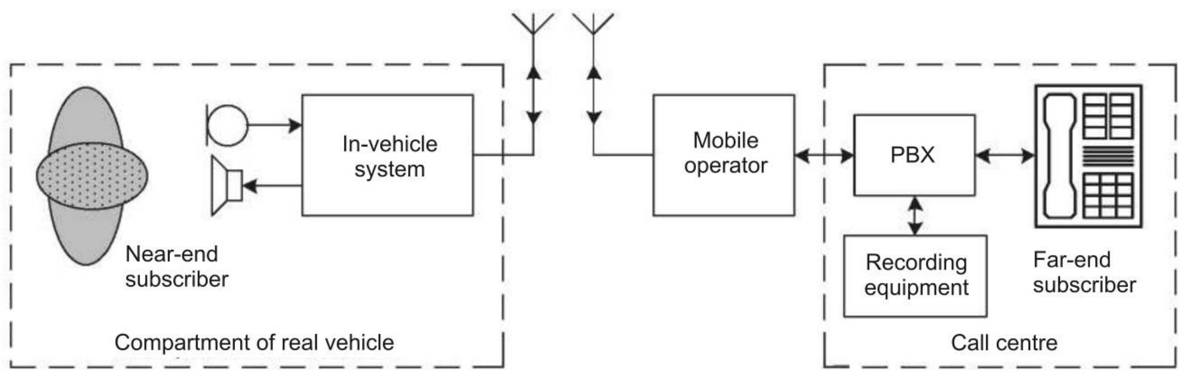


Figure 5 — Test arrangement for subjective quality assessment of loudspeaker communication

6.10.2 Requirements for equipment:

1) vehicle of given type, mark and model equipped with IVS 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 [5], 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 IVS 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

The 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) IVS model, hardware and software versions, photograph;

3) IVS 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.

N o t e — 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 IVS test methods described in this section:

- are aimed at conformity assurance as regards the requirements stated in GOST R 54620 with respect to the quality of loudspeaker communication in the vehicle cabin;
- apply to both narrowband and wideband IVS, unless otherwise specified;
- take into account the basic requirements of [30], [31];
- 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 IVS 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.

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

7.1 Signal processing delay in IVS

7.1.1 The delays in signal propagation from one subscriber to another both in receiving direction $T_{R,SUM}$ and in sending direction $T_{S,\ SUM}$ are the sums of the delays T_R, T_S introduced by sound processing algorithms used in the IVS (AGC, AEC, noise suppression, etc.) and the standard delays of signals in the IVS telephone part t_{system} which are related to processes of signal encoding and decoding as well as to signal propagation time in service provider channels:

$$T_{SSUM} = T_S + t_{system},$$

$$T_{RSUM} = T_R + t_{system}.$$

The requirements pertaining to signal processing delays in the IVS are enforced both separately (signal processing delay for sending T_S, signal processing delay for receiving T_R) and cumulatively (overall delay $T_R + T_S$).

7.1.2 Signal processing delay in IVS in sending direction

7.1.2.1 Requirements

The signal processing delay in sending direction T_S in a loudspeaker IVS shall not exceed 50 ms. The T_S value shall be minimised at the IVS development stage by reasonable selection of algorithms and of signal buffering methods.

7.1.2.2 Method of measurement

The total signal propagation delay in sending direction $T_{S,SUM}$ 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.

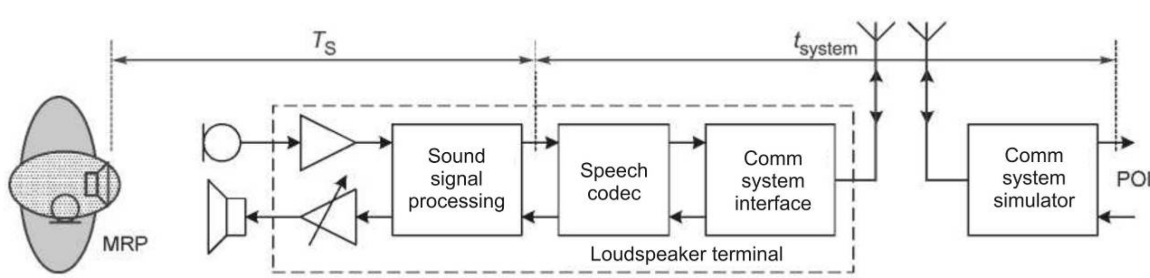


Figure 6 — Signal propagation delay in sending direction

The delay in the communication network t_{system} depends on the system simulator mode, and its value must be known beforehand.

Composite CS signals as per [17] shall be used for measurements. The duration of a CS signal 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 [20] is set equal to minus 28.7 dBPa at the HFRP. The frequency response of the artificial mouth is preliminary equalised at the MRP.

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 IVS signal processing delay T_S is calculated by subtraction of the known signal delay in the communication system t_{system}.

7.1.3 Signal processing delay in IVS in receiving direction

7.1.3.1 Requirements

The signal processing delay in receiving direction T_R in a loudspeaker IVS shall not exceed 50 ms. The T_R value shall be minimised at the IVS development stage owing to reasonable selection of algorithms and of signal buffering methods.

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.

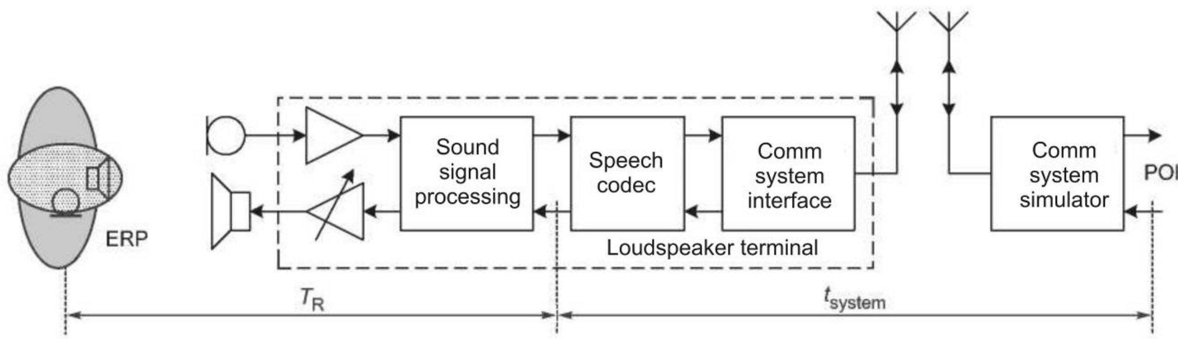


Figure 7 — Signal propagation delay in receiving direction

The delay in the communication network t_{system} depends on the system simulator mode, and its value must be known beforehand.

Composite CS signals as per [17] shall be used for measurements. The duration of a CS signal 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 IVS 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 IVS signal processing delay T_R is calculated by subtraction of the known signal delay in the communication system t_{system}.

7.1.4 Overall signal processing delay in IVS in receiving and sending directions

7.1.4.1 Requirements

The overall signal processing delay in receiving and sending directions ($T_R + T_S$) in a loudspeaker IVS shall not exceed 70 ms.

7.1.4.2 Method of measurement

The T_R and T_S values are measured separately, and then summed up.

7.2 Loudness ratings

7.2.1 Loudness ratings of a loudspeaker terminal in sending and receiving directions provide for standard methods used to describe the levels of acoustic and electric 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 terminals shall be measured in accordance with the requirements of [13]—[16] taking into account the specific features of their use in vehicles as described in [30] and [31].

7.2.2 Sending loudness rating

7.2.2.1 The SLR of an IVS is the frequency-weighted measure of signal attenuation along the path from the IVS 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 IVS installed in the vehicle compartment (cabin) shall be equal to (13 ± 4) dB for the driver and for the passengers in his/her immediate vicinity.

The microphone shall not be turned off.

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

N o t e — An AGC present in an in-vehicle system may increase the transmission level of ambient noises or echo signals.

7.2.2.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If an AGC is present for sending direction, it shall be disabled for the duration of measurements.

2) The acoustic test signal shall be an artificial voice obtained as per [8]. The artificial mouth of the HATS manikin is calibrated, and its frequency response is equalised at the MRP in accordance with the requirements of [20]. The acoustic SPL of the test signal at the HFRP of the IVS microphone is set to minus 28.7 dBPa. 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 IVS sensitivity in sending direction.

4) The sensitivity of a narrowband IVS in sending direction is calculated for each of 14 one-third-octave frequency bands listed in [13] (Table 1), bands 4 to 17. The sensitivity of a wideband IVS in sending direction is calculated for each of 20 one-third-octave frequency bands listed in [13] (Table A.2), bands 1 to 20. To calculate the sensitivity in each frequency band, an r.m.s. level of electric signals at the POI of the output of the system simulator decoder 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 [13]:

- for narrowband IVS: using the formula 5-1/P.79, in bands 4—17, for m = 0.175, with frequency weighting in accordance with Table 1/P.79;

- for wideband IVS: in accordance with Appendix A/P.79.

N o t e — As a preliminary rough estimate of the IVS sensitivity in sending direction, the estimated transfer ratio between the acoustic input of the IVS 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

7.2.3.1 The RLR of an IVS 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 IVS 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 IVS installed in the vehicle compartment (cabin) shall be equal to the value specified by the IVS or vehicle manufacturer, in accordance with the requirements of GOST R 54620 (sub-clause 7.5.3.10).

The RLR_{nom} value shall be selected in the range from (minus 6 ± 4) dB to (2 ± 4) dB. The value of (minus 6 ± 4) dB is recommended.

If a manual gain control in receiving direction is provided for in the IVS, then the selected value of RLR_{nom} corresponding to the nominal IVS 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 IVS volume shall be achieved at the minimum (leftmost) position of the volume control. The required RLR_{max} value shall be specified by the IVS or vehicle manufacturer in accordance with the requirements of GOST R 54620 (sub-clause 7.5.3.11).

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 IVS volume shall be achieved at the maximum (rightmost) position of the volume control. The required RLR_{min} value shall be specified by the IVS or vehicle manufacturer in accordance with the requirements of GOST R 54620, clause И.4.3 (Appendix И).

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 IVS shall be decided by the IVS or vehicle manufacturer.

Note — An AGC present in in-vehicle systems may increase the transmission level of echo signals.

7.2.3.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If an AGC is present for receiving direction, it shall be disabled for the duration of measurements.

2) The test signal shall be an artificial voice obtained as per [8]. 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 IVS 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 [20]. The measurements are taken at the DRP of a single artificial ear, and the results may, if necessary, be reduced to the reference ERP in accordance with [11].

5) The sensitivity of a narrowband IVS in receiving direction is calculated for each of 14 one-third-octave frequency bands listed in [13] (Table 1, bands 4 to 17). The sensitivity of a wideband IVS in receiving direction is calculated for each of 20 one-third-octave frequency bands listed in [13] (Table A.2, bands 1 to 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 is calculated in accordance with [13]:

- for narrowband IVS: using the formula 5—1, in bands 4—17, for m = 0.175, with frequency weighting in accordance with Table 1 [13];
- for wideband IVS: in accordance with Appendix A (Le factor ignored), as per [13].

7) The resulting RLR value shall be corrected downwards by 14 dB for narrowband IVS, or by 8 dB for wideband IVS, in accordance with the requirements of [30] and [31].

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

N o t e — As a preliminary rough estimate of the IVS 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.

7.2.4 SLR deviations

7.2.4.1 Requirements

For acoustic signals sent under 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 ±0.5 dB from the SLR value for signals at the nominal SPL (if no AGC in sending direction is present).

7.2.4.2 Method of measurement

The SLR is measured as specified above for two additional SPL values of acoustic test signals equal to minus 31.7 dBPa and minus 22.7 dBPa at the HFRP.

For both measurements, the obtained values are compared with the SLR value for the nominal loudness level equal to minus 28.7 dBPa.

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

7.2.5 RLR deviations

7.2.5.1 Requirements

For electric signals received under 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 ±0.5 dB from the RLR value for signals at the nominal level (with the volume control at its nominal position and with no AGC in receiving direction present).

7.2.5.2 Method of measurement

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

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 IVS output is not overloaded by signals of a level higher than the nominal one. Combining such signals with the maximum IVS volume level may lead to overloading and distortions, and, as a consequence, may degrade the IVS duplex capability.

7.3 Frequency sensitivity response

7.3.1 Frequency sensitivity response of sending IVS part

7.3.1.1 The frequency response of the IVS sensitivity in sending direction shall be measured with the IVS installed in the vehicle compartment (cabin), along the path from the IVS 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 IVS, and in Table 2 for wideband IVS. Linear interpolation on log-log scale may be used for intermediate frequencies.

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

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

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

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

An ideal frequency response in sending direction should be flat in the range from 200 Hz to 4 kHz for narrowband IVS and from 100 Hz to 7 kHz for wideband IVS. 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.

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

7.3.1.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. If an AGC is present for sending direction, it shall be disabled for the duration of measurements.

2) The acoustic test signal shall be an artificial voice as per [8]. The acoustic mouth shall be calibrated by level and by frequency response at the MRP. The SPL of the input test signal at the HFRP is set to minus 28.7 dBPa (test signal level averaged along the full signal path).

3) The frequency sensitivity response of the terminal is determined in one-third-octave bands as per GOST R 8.714 in the frequency range from 100 Hz to 4 kHz inclusive for narrowband terminals and from 100 Hz to 8 kHz for wideband terminals. 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 terminal 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 IVS part

7.3.2.1 The frequency response of the IVS sensitivity in receiving direction shall be measured with the IVS 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 of the system simulator to the IVS acoustic output at the DRP reference point.

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 IVS, and in Table 4 for wideband IVS. Linear interpolation in log-log scale shall be used for intermediate frequencies.

Table 3 — Frequency sensitivity response in receiving direction for narrowband IVS

Frequency [Hz]	Upper limit [dB]	Lower limit [dB]	Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
200	0	–∞	400	0	-12
250	0	–∞	3100	0	-12
315	0	–15	4000	0	–∞

Table 4 — Frequency sensitivity response in receiving direction for wideband IVS

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

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

7.3.2.3 Method of measurement

1) 1) The test conditions shall meet the requirements of Section 6. If an AGC is present for receiving direction, it shall be disabled for the duration of measurements.

2) The electric test signal shall be an artificial voice as per [8]. 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) 3) The frequency sensitivity response of the terminal is determined in one-third-octave bands as per GOST R 8.714 in the frequency range from 100 Hz to 4 kHz inclusive for narrowband terminals and from 100 Hz to 8 kHz for wideband terminals. 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, it shall be calibrated and its frequency response equalised in accordance with the requirements of [20].

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

Note — As the frequency response in receiving 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 IVS self-noise level

7.4.1 The term "IVS 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 electric circuits and ADC/DAC of the IVS, digital noise of speech encoding and DSP algorithms in the IVS, and 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 IVS 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 free of near-end subscriber's speech, the maximum permitted level of the IVS self-noise in the transmit channel shall not exceed minus 64 dBm0(A) when measured at the electric output of the speech codec on the operator side. 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. If an AGC is present for sending direction, it shall be disabled for the duration of measurements.

2) Test signals are not used. However, some IVS 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 CS signal sequence as per [17] is applied prior to measurements. The spectrum of the acoustic activation signal is first calibrated at the MRP in free-field conditions. The SPL of the activation signal at the HFRP shall be equal to minus 28.7 dBPa.

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 (from 50 Hz to 4 kHz) for narrowband terminals, or from 100 Hz to 8 kHz (from 50 Hz to 8 kHz) for wideband terminals. The analysis window size for averaging shall be 1 s. 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 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).

7.4.3 Noise level in receive channel

7.4.3.1 Requirements

When measured in silence conditions at the IVS acoustic output under the nominal loudness rating RLR_{nom}, the maximum permitted level of the IVS self-noise in the receive channel with no operator speech present shall not exceed (minus 51 minus RLR_{nom}) dBPa(A). 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. If an AGC is present for receiving direction, it shall be disabled for the duration of measurements.

2) Test signals are not used. Some IVS 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 CS signals as per [17] 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; it shall preliminary be calibrated, and its frequency response equalised in accordance with the requirements of [20].

The idle channel noise at the DRP is measured in the frequency range from 50 Hz to 7 kHz for narrowband IVS, or from 50 Hz to 10 kHz for wideband IVS. The analysis window size for averaging shall be 1 s. 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 during the measurements. The power spectrum density 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).

7.5 Suppression of out-of-band signals

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

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 terminals or above 8 kHz for wideband terminals) are present at the acoustic input of the terminal and are poorly suppressed by analogue filters before the ADC, they can create out-of-band noise in the communication channel due to superposition of frequencies in analogue-to-digital conversion.

The measurement is carried out for an IVS installed in the vehicle compartment (cabin), along the path from the IVS 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 IVS and from 9 kHz to 16 kHz for wideband IVS, 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 IVS and from 100 Hz to 7 kHz for wideband IVS 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 a frequency-shifting technology is used in the IVS, 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 CS signals as per [17] is applied prior to measurements. The sound pressure level of the activation signal at the MRP shall be minus 4.7 dBPa. 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 signal is terminated, and shall exactly follow the vocalised CS signal 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. 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 [8] limited in the band specified above, with its SPL at the MRP also equal to minus 4.7 dBPa. 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 IVS 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 IVS acoustic output (at the point in the immediate vicinity of the IVS 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 IVS and from 100 Hz to 7 kHz for wideband IVS and applied at the level of minus 12 dBm0, the acoustic level of out-of-band noise at the IVS output measured in the frequency band from 4.6 KHz to 8 kHz for narrowband IVS and from 8.6 kHz to 16 kHz for wideband IVS 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 IVS or from 100 Hz to 7 kHz for wideband IVS.

7.5.3.3 Method of measurement

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

2) The test signal of artificial voice is generated as per [8] 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 IVS 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 IVS

7.6.1 The term "IVS distortion" shall be 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 IVS circuits, of distortions introduced by speech encoding and DSP in the IVS, 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 IVS noise suppression systems) are suitable for measurements because some of them are unable to pass pure tone signals. A codec of the highest quality and bitrate shall be selected.

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

7.6.2 Signal distortions in sending direction

7.6.2.1 The measurement is carried out for an IVS installed in the vehicle compartment (cabin), along the path from the IVS 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 IVS and up to 8 kHz for wideband IVS.

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

7.6.2.3 Method of measurement

1) Noise suppression, 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.

3) To ensure reliable channel activation in sending direction, a provisional activation signal that includes a sequence of four CS signals as per [17] is applied prior to measurements. The sound pressure level of the activation signal at the MRP shall be minus 4.7 dBPa. 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 CS signal 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 signal 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 terminals and up to 8 kHz for wideband terminals, for each test signal separately.

N o t e — If a digital debug interface is available for the IVS, the level of distortions introduced by the IVS and unaffected by the system simulator may be evaluated.

7.6.3 Signal distortions in receiving direction

7.6.3.1 The measurement is carried out for an IVS 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 reference point on the operator side to the DRP, in the frequency band of up to 8 kHz for narrowband terminals and up to 15 kHz for wideband terminals.

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 IVS volume control, for each of the following test frequencies:

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

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 reference point.

3) To ensure reliable channel activation in receiving direction, a provisional activation signal that includes a sequence of four CS signals as per [17] is applied prior to measurements. The activation signal level at the POI reference point 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 CS signal part rather than its pseudo-noise part. The duration of the test signal shall be at least 200 ms.

5) When wideband terminals are tested, the artificial ear of the HATS manikin is replaced by a measuring microphone installed where the centre of the manikin's head would be located in other tests. This is because the frequency response of the manikin's head is specified for up to 10 kHz only in [12] while 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 IVS speaker, in the same way as in out-of-band noise measurements. This allows 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 terminals and up to 15 kHz for wideband terminals, for each test signal separately.

Note — If a digital debug interface is available for the IVS, the level of distortions introduced by the IVS and unaffected by the system simulator may be evaluated.

7.7 IVS 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 In regard to the electro-acoustic path in the IVS, the terminal coupling loss shall be 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.

As 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 IVS.

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

The echo canceller or acoustic echo suppressor implemented in an IVS 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 IVS 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 IVS 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 IVS 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 50 dB at the nominal position of the volume control ($RLR = RLR_{nom}$) or at least 40 dB for the maximum volume ($RLR = RLR_{min}$) after the time period required to complete configuration of all factors for the acoustic echo canceller (AEC). These TCL_W values shall be achieved in a wide range of possible acoustic conditions inside the vehicle (different number of passengers, open or closed windows, etc.).

7.7.2.3 Method of measurement

1) The test conditions shall meet the requirements of Section 6. The noise level in the receive channel measured at the decoder output of the system simulator shall not exceed minus 63 dBm0 at the POI reference point.

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 voice generated in accordance with [8]. 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 [17] 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. The 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) The TCL_W is measured as per [4] (clause B.4, trapezoidal rule). The mean energy of the test signal and of echo signals is calculated per 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).

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 IVS 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 CS signal and the test artificial voice signal of the nominal level are applied to the IVS receive channel, the echo signal attenuation in the IVS 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 [17]. 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 8 CSS periods excluding the pauses. Then, the test is repeated for an artificial voice signal as per [8]. 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.

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 IVS 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.

7.7.4.2 Requirements

When the CS signal of the nominal level is applied to the IVS receive channel, the frequency dependence of the echo signal attenuation in the transmit channel shall be below the limits specified in Table 5 for narrowband IVS and in Table 6 for wideband IVS.

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

Frequency [Hz]	Upper limit [dB], minus	Frequency [Hz]	Upper limit [dB], minus
100	20	1500	33
200	30	2600	24
300	38	4000	24
800	34		

T a b l e 6 — Frequency dependence of echo signal attenuation in wideband IVS

Frequency [Hz]	Upper limit [dB], minus	Frequency [Hz]	Upper limit [dB], minus
100	41	5200	46
1300	41	7500	37
3450	46	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 satisfied at any time; 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 an echo signal but neither a "comfort noise" injected in the transmit channel in order to mask echo signal attenuation, and nor a received background noise, e.g., the one induced from the terminal speakers.

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 [8]. 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 CS signal 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 of adaptive filter factors in the AEC, and is described by the dependence of the echo signal loss on the time passed after the AEC start-up.

The measurement is carried out for an IVS 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 on the operator side.

7.7.5.2 Requirements

After the CS signal and the test artificial voice signal of the nominal level are applied to the IVS receive channel, the ERL value for echo signals in the IVS transmit channel depending on the time passed after the initial start-up of the AEC with a volume control set at its maximum level shall not exceed the limits shown in Figure 8.

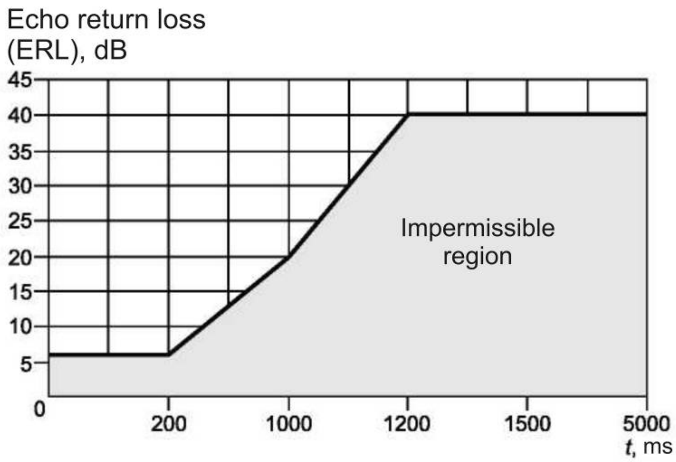


Figure 8 — Time dependence of echo return loss (ERL)

Special attention should be given to the IVS 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 the TCL on the electro-acoustic path not less than 6 dB in all operating frequency range at any time moment, while the transient process shall not be accompanied with abrupt loudness jumps, noise bursts, or excitation of tone signals.

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 maximum level.

3) The first test signal is a periodically repeated CSS as per [17]. The average signal level shall be equal to minus 16 dBm0. The echo signal shall be analysed in regard to the active test signal level (excluding the pauses) for at least 5 s after the connection is established.

4) Then, the test is repeated for the second test signal of artificial voice as per [8]. 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 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 in assessments of the original and echo signal levels shall be 35 ms. After the envelopes of direct and return 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.

7.7.6 Initial convergence of AEC in presence of noise

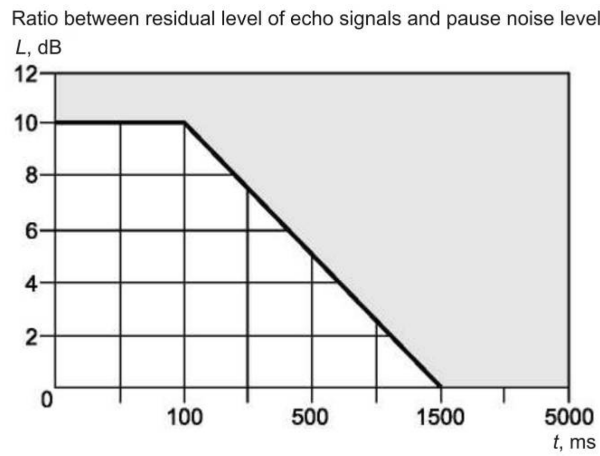
7.7.6.1 As a rule, an initial AEC set-up for yet unknown parameters of the echo path during single talk with acoustic noise present depends not only on the convergence of adaptive filter factors in the AEC, but also on the character and level of acoustic noise. This being the case, the adaptive filter 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 after the AEC start-up.

The measurement is carried out for an IVS installed in the vehicle compartment (cabin), along the path from the electric input to the electric output of the speech codec of the system simulator on the operator side, under different levels of simulated acoustic noises in the vehicle cabin.

7.7.6.2 Requirements

After the CS signal and the test artificial voice signal of the nominal level are applied to the IVS receive channel, the values of the ratio L between the residual echo signal level and the pause noise level depending on the time passed after the initial start-up of the AEC with a volume control set at its maximum position shall not exceed the limits shown in Figure 9.

Figure 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 terminal 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 [17]. The average signal level shall be equal to 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 [8]. 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 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 in assessments of echo signal levels shall be 35 ms. After the envelopes of return signals and pause noise are calculated, the curve of echo return loss versus time 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 using a rotating 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 above) to an angle of 90° where the surface plane is parallel to the windscreen, then rotated back. The surface rotates cyclically 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. In order to obtain reproducible results, the rotation of the reflecting surface shall be synchronised using the control channel with the start of test signals.

7.7.7.2 Requirements

The degradation of echo signal suppression with echo path changes in the vehicle compartment shall not exceed 6 dB 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 [17], of an average level equal to minus 5 dBm0, and to minus 25 dBm0. The signal is analysed for at least 2.8 s; this corresponds to 8 periods of CS signals, excluding the pauses. Then, the test is repeated for the second test signal of artificial voice as per [8]. 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 in assessments of the original and echo signal levels shall be 35 ms. After the envelopes of direct and return 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 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 IVS, the IVS 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 turn on the channel when the signal of a level higher than the activation threshold is applied.

The measurement is carried out for an IVS installed in the vehicle compartment (cabin), along the path from the acoustic input of the IVS 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 $T_{r,S,min}$ with an input signal of the minimum activation level shall not exceed 50 ms.

7.8.2.3 Method of measurement

The waveform of test signals is shown in Figure 10. The test signal is a sequence of CS signals as per [17] of gradually increased levels with pauses in-between.

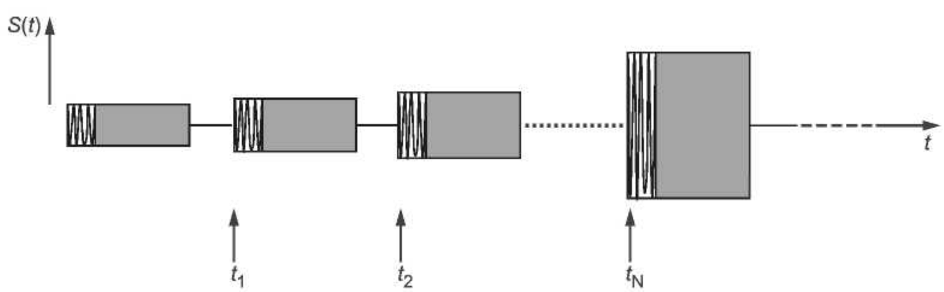


Figure 10 — Test signal for evaluation of minimum activation level and channel activation time [31]

The test signal parameters are listed in Table 7.

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

Designation of test signal	CS signal/pause duration	Active level of first CS signal (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 CS signal 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.

4) 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.

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

The level is only measured for active regions of a CS test signal, and thus it is slightly higher than the average CS signal level [17] 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 CS signals 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 time $T_{r,R,min}$ and the minimum electric level of activation $L_{R,min}$.

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 IVS 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 IVS. Instead of the artificial ear of the HATS manikin, a measuring microphone installed close to the IVS 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 sequence of CS signals as per [17] 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	CS signal/pause duration	Active level of first CS signal (at MRP)	Level increment between two periods
Signal for evaluation of channel activation performance in receiving direction	248.62 ms/451.38 ms	minus 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 CS signal 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 IVS 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 CS signal level which resulted in full activation of the receive channel. The activation time is determined as a time between the CS signal start and the full channel activation.

The level is only measured for active regions of a CS test signal, and thus it is slightly higher than the average CS signal level [17] 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 CS signals 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 in half-duplex mode

7.8.4.1 When the subscribers are talking one at a time (in half-duplex mode), an IVS 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 IVS 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 sequence of CS signals as per [17] 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 IVS. The waveform diagram of signals is shown in Figure 11.

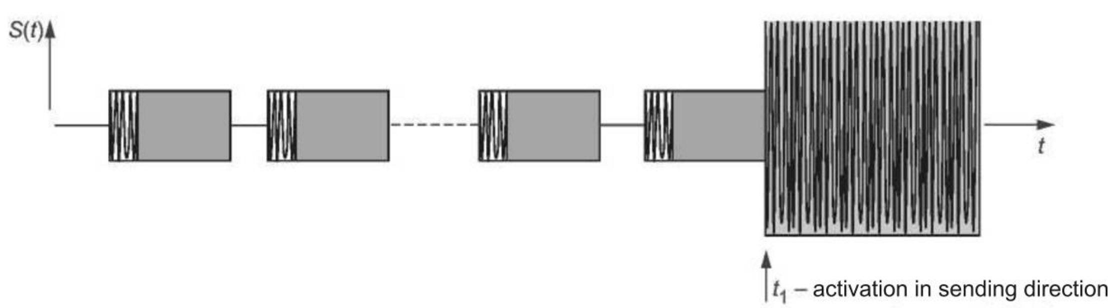


Figure 11 — Waveform diagram of signals for measurement of attenuation in transmit channel [31].

Example test signals are shown in Table 9.

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	minus 16 dBm0 (including pause of 101.38 ms)	minus 3 dBPa
Active signal level	minus 14.7 dBm0	minus 3 dBPa

Test signals shall be time-synchronised at the acoustic interface of the terminal 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.

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

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

5) 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 in half-duplex mode

7.8.5.1 When the subscribers are talking one at a time (in half-duplex mode), an IVS 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 IVS 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 sequence of CS signals as per [17] of a nominal level sufficient for transmit channel activation is applied to the acoustic input of the terminal. 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.

Table 10 — Signal levels for attenuation measurements in receive channel

Measured value	Receiving direction (voice at POI)	Sending direction (CSS at MRP)
Average signal level	minus 14.7 dBm0	minus 4.7 dBPa (including pause of 101.38 ms)
Active signal level	minus 14.7 dBm0	minus 3 dBPa

Test signals shall be time-synchronised at the acoustic interface of the terminal 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 CS signals of a nominal average level equal to minus 4.7 dBPa is applied to the acoustic input of the terminal.

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 terminal 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 IVS performance in double-talk mode

7.9.1 This section deals with the measurements of those IVS 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 IVS 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.

IVS that do not support full duplex mode shall provide for good suppression of echo signals in half duplex mode (high value of TCL_W).

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

The value $A_{H,S,dt}$ defines how noticeable the loudness jumps are in the transmit channel when the switching between single-talk and double-talk occurs, and the value $A_{H,R,dt}$ 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, a terminal may introduce an 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 in the transmit channels 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 IVS in the transmit channel during double-talk depends on the IVS performance (quality class) as regards its duplex communication capability, and shall correspond to the values specified in Table 11.

Table 11 — IVS performance parameters for duplex communication

Parameter	Quality class				
	1	2a	2b	2c	3
	Full duplex	Partial duplex			Half duplex only
$A_{H,S,dt}$ [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

The requirements shall hold both for the nominal and for the maximum position the IVS volume control. They shall also be satisfied both for nominal signal levels in sending/receiving directions and for a ± 6 dB disbalance of the levels, e.g., when the receiving level rises by 6 dB while the sending level drops by 6 dB, and vice versa. Thus, six combinations of signal levels and volume control positions shall be checked.

It should be taken into account that if the signal level is increased in receiving direction when the volume control is set to its maximum position, non-linear effects may occur in the echo path which lead to echo signal attenuation in the AEC and impair the double-talk quality.

Table 11 includes the requirements on the $A_{H\ Sdt}$ parameter that are necessary to attribute an IVS 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, where s(t) is the signal in sending direction, $s_{dt}(t)$ is the signal in receiving direction. Two sequences of non-correlated CS signals are used; they are applied to the transmit channel and receive channel simultaneously with partial overlapping in time to model a double-talk effect. The length of test sequences shall be at least 25 periods 400 ms each. 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.

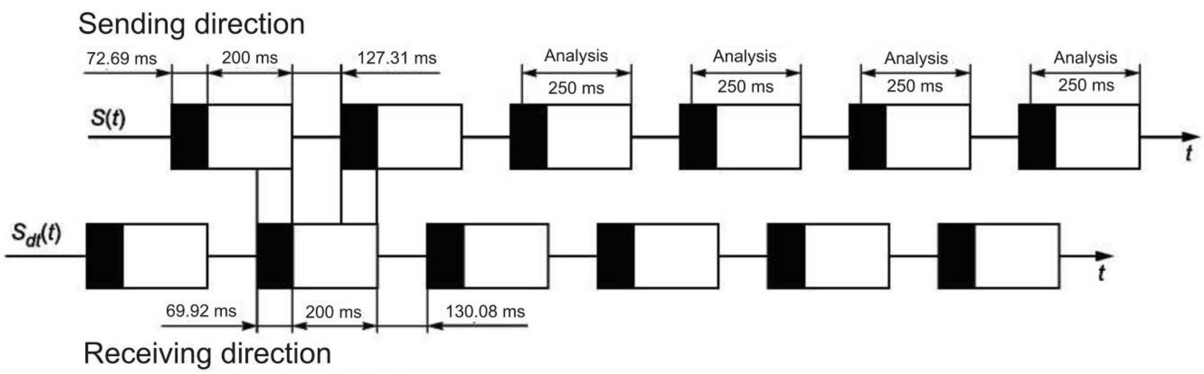


Figure 12 — Test signal for evaluation of attenuation range in sending direction during double-talk

The initial portion of each CS signal period (vocalised sound painted black) in one direction is overlapped by the final portion of each CS signal 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. 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 pause 101.38 ms long)	minus 16 dBm0	minus 4.7 dBPa
Active signal level	minus 14.7 dBm0	minus 3 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 [8] 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 direction is applied at the acoustic input of the terminal at the MRP. The test signal for receiving direction 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 period of the CS signal is analysed.

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

7.9.3 Attenuation in receive channel in double-talk mode

7.9.3.1 During double-talk, a terminal may apply an 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 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 IVS in the receive channel during double-talk depends on the IVS 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 — IVS performance parameters for duplex communication

Parameter	Quality class				
	1	2a	2b	2c	3
	Full duplex	Partial duplex			Half duplex only
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

The requirements shall hold both for the nominal and for the maximum position of the IVS volume control. They shall also be satisfied both for nominal signal levels in sending/receiving directions and for a ± 6 dB disbalance of the levels, e.g., when the receiving level rises by 6 dB while the sending level drops by 6 dB, and vice versa. Thus, six combinations of signal levels and volume control positions shall be checked.

It should be taken into account that if the signal level is increased in receiving direction when the volume control is set to its maximum position, non-linear effects may occur in the echo path which lead to echo signal attenuation in the AEC and impair the double-talk quality.

Table 13 includes the requirements on the $A_{H,R,dt}$ parameter that are necessary to attribute an IVS 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 IVS 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 the previous subsection, and are shown in Figure 13, where s(t) is the signal in receiving direction, and $s_{dt}(t)$ the signal in sending direction.

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 [8] with a level of 16 dBm0, applied at the electric input of the system simulator encoder at the POI reference point.

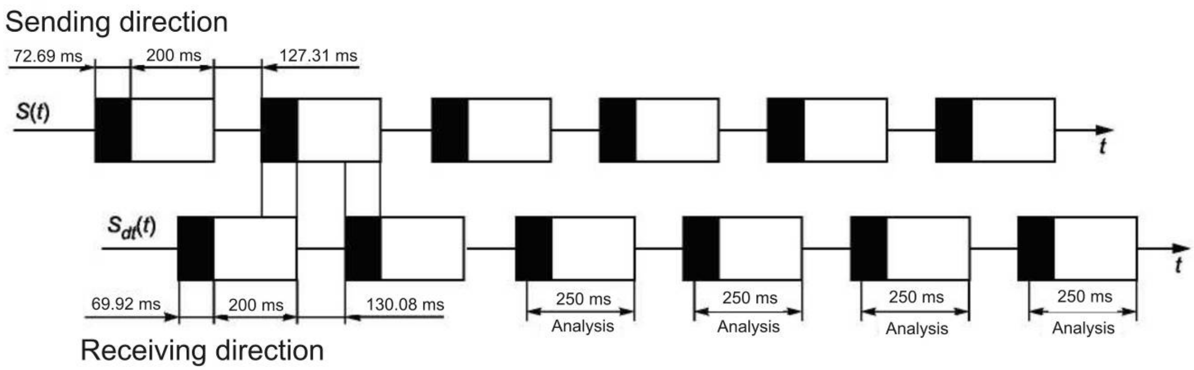


Figure 13 — Test signal for evaluation of attenuation range in receiving direction during double-talk

3) The test signal for sending direction is applied at the acoustic input of the terminal at the MRP. The test signal for receiving direction 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 terminal using a measuring microphone located close to the terminal 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 period of the CS signal is analysed.

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

N o t e — The electric output of the IVS 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 IVS 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 IVS performance type in regard to duplex communication, and shall comply with the values specified in Table 14.

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

Parameter	Quality class				
	1	2a	2b	2c	3
	Full duplex	Partial duplex			Half duplex only
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.

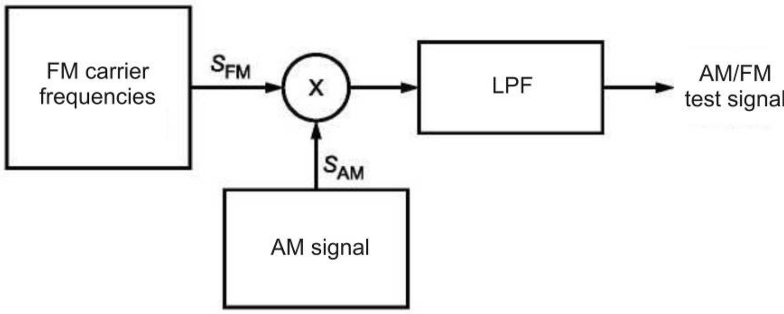


Figure 14 — Flow diagram of test signal generation based on AM/FM modulated sine sets

Initially, a set of N sine-wave signals is formed with FM carrier frequencies f_n, of the frequency deviation Δf_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 the same. 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 φ_n affect the signal waveform only.

The signal generation parameters for receiving and sending directions are listed in Table 15 for narrowband IVS and in Table 16 for wideband IVS. 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 [17]— [19].

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

Receiving direction			Sending direction		Receiving direction			Sending direction	
n	f_n [Hz]	Δf_n [Hz]	f_n [Hz]	Δf_n [Hz]	n	f_n [Hz]	Δf_n [Hz]	f_n [Hz]	Δf_n [Hz]
1	250	± 5	270	± 5	9	2250	± 40	2400	± 35
2	500	± 10	540	± 10	10	2500	± 40	2650	± 35
3	750	± 15	810	± 15	11	2750	± 40	2900	± 35
4	1000	± 20	1080	± 20	12	3000	± 40	3150	± 35
5	1250	± 25	1350	± 25	13	3250	± 40	3400	± 35
6	1500	± 30	1620	± 30	14	3500	± 40	3650	± 35
7	1750	± 35	1890	± 35	15	3750	± 40	3900	± 35
8	2000	± 40	2160	± 35					

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

Receiving direction			Sending direction		Receiving direction			Sending direction	
n	f_n [Hz]	Δf_n [Hz]	f_n [Hz]	Δf_n [Hz]	n	f_n [Hz]	Δf_n [Hz]	f_n [Hz]	Δf_n [Hz]
1	125	± 2,5	150	± 2,5	16	3 750	± 40	3 900	± 35
2	250	± 5	270	± 5	17	4 000	± 40	4 150	± 35
3	500	± 10	540	± 10	18	4 250	± 40	4 400	± 35
4	750	± 15	810	± 15	19	4 500	± 40	4 650	± 35
5	1 000	± 20	1 080	± 20	20	4 750	± 40	4 900	± 35
6	1 250	± 25	1 350	± 25	21	5 000	± 40	5 150	± 35
7	1 500	± 30	1 620	± 30	22	5 250	± 40	5 400	± 35
8	1 750	± 35	1 890	± 35	23	5 500	± 40	5 650	± 35
9	2 000	± 40	2 160	± 35	24	5 750	± 40	5 900	± 35
10	2 250	± 40	2 400	± 35	25	6 000	± 40	6 150	± 35
11	2 500	± 40	2 650	± 35	26	6 250	± 40	6 400	± 35
12	2 750	± 40	2 900	± 35	27	6 500	± 40	6 650	± 35
13	3 000	± 40	3 150	± 35	28	6 750	± 40	6 900	± 35
14	3 250	± 40	3 400	± 35	29	7 000	± 40		
15	3 500	± 40	3 650	± 35					

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 4.7 dBPa. 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 [17].

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 IVS is classified as per Table 14. The check is carried out for all frequencies in the range from 200 to 3450 Hz for narrowband IVS, or from 200 to 6950 Hz for wideband IVS.

During the IVS 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, a terminal may introduce an 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 of the $A_{H,S,dt}$ value 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 4.7 dBPa. 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 [17].

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 IVS is classified as per Table 11. The check is carried out for all frequencies in the range from 200 to 3550 Hz for narrowband IVS, or from 200 to 6900 Hz for wideband IVS.

6) The test is repeated for all combinations of signal levels and volume control positions. During the IVS 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 Speech quality in receive and transmit channels

7.10.1 The IVS checks in regard to speech intelligibility and quality shall be carried out by expert persons using the technique described in GOST R 51061 and GOST R 50840 that provides for objective and well repeatable results.

7.10.2 Requirements

The expert evaluation of the loudspeaker communication quality of an IVS installed in the vehicle compartment is performed in transmit and receive channels. For single-talk in silence conditions, the speech quality of the IVS loudspeaker communication at a five-grade rating scale of speech quality and intelligibility classes specified in GOST R 51061 (Table 1) and GOST R 50840 shall correspond to Class 1 or higher, or to Class 2 in the case of disturbing acoustic noise.

Additional subjective appraisals of speech quality are carried out in accordance with GOST R 51061 and GOST R 50840 during double-talk between the driver and the system operator when they talk one at a time and when they talk simultaneously, in normal as well as in accelerated speech tempo, both in silence and in conditions of background acoustic noises in the vehicle cabin, at the levels specified for "ordinary" and "worst case" noise environments defined by the vehicle manufacturer.

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

The key properties subject to assessment are:

- good word articulation of speech;
- ability to identify talker's voice;
- natural voice sounding;
- no sounding artefacts;

- no need to exert extra attention;
- understanding transmitted speech without difficulties; no need to ask and listen again.

The average rating at five-grade rating scales for the above properties shall be at least 3.0 for narrowband IVS and at least 3.6 for wideband IVS when the IVS is operated either in silence or in "ordinary" noise environment (depending on the vehicle type and noise scenario).

7.11 IVS performance in acoustic noise conditions

7.11.1 Operation of transmit channel in acoustic noise conditions

7.11.1.1 The measurement is carried out for an IVS installed in the vehicle compartment (cabin), along the path from the acoustic input of the IVS 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 IVS microphone position and its optimum directional properties as well as the use of additional algorithms in the IVS (AGC in sending direction, and nose suppression).

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.

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 terminal to configure their parameters, and to achieve a steady state.

3) The input acoustic test signal in the form of artificial voice as per [8] is applied using an "artificial mouth" with an average SPL equal to minus 4.7 dBPa at the MRP. 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 4.7 dBPa for active speech regions as per [10]. 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 output of the simulator.

7.11.2 Operation of receive channel in acoustic noise conditions

7.11.2.1 The measurement is carried out for an IVS 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 IVS at the DRP.

7.11.2.2 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 IVS microphone position and directional properties, as well as for the use of additional algorithms in the IVS (AGC in receiving direction).

7.11.2.3 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 terminal to configure their parameters, and to achieve a steady state.

3) The input electric test signal in the form of artificial voice as per [8] 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 IVS 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 [10]. 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 IVS at the DRP.

7.12 Background noise quality in transmit channel

7.12.1 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 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.

7.12.2 Background noise after connection

7.12.2.1 The background noise in the transmit channel right after the connection is usually louder than the one observed several seconds later. This is related to transient processes in AEC, noise suppression, AGC and speech encoding algorithms. The initial noise level increase in the IVS shall not cause any discomfort for the far-end subscriber.

7.12.2.2 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 IVS or from 150 Hz to 7.0 kHz for wideband IVS.

7.12.2.3 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 IVS is turned off then on again (for initial reset of adaptive algorithms included in the IVS, such as AGC, NRD and AEC). The connection is initiated using the system simulator, and the IVS answers an incoming call. Care should be taken in this process in order 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 terminal 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.

7.12.3 Quality of background noise transmission in case of near-end subscriber speech

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

7.12.3.2 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).

7.12.3.3 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 terminal to configure their parameters, and to achieve a steady state.

3) To ensure that the AEC of the terminal will configure itself in the receive channel, a training sequence in the form of artificial voice as per [8] 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 speech present. The noise signal level is averaged using a time constant of 35 ms.

5) Then, the test CS signal as per [17] at a level from minus 4.7 dBPa to 1.3 dBPa is periodically applied in the transmit channel to the acoustic input of the terminal at the MRP for at least two CS signal 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 35 ms.

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

7.12.4 Quality of background noise transmission in case of far-end subscriber speech

7.12.4.1 The test is carried out using a CS signal 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.

7.12.4.2 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).

7.12.4.3 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 terminal to configure their parameters, and to achieve a steady state.

3) To ensure that the AEC of the terminal will configure itself in the receive channel, a training sequence in the form of artificial voice as per [8] 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 speech present. The noise signal level is averaged using a time constant of 35 ms.

5) Then, the test CS signal as per [17] 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 CS signal 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 speech is switched on and off.

7.12.5 Quality of background noise transmission using "comfort noise" for pauses

7.12.5.1 This test is only carried out if the IVS 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.

7.12.5.2 Requirements

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

1) "comfort noise" level in pauses shall not differ from the original transmitted background noise by more than +2 dB and minus 5 dB. The noise level is assessed by frequency averaging 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 IVS 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.

Table 17 — Tolerances for "comfort noise" spectrum

Frequency [Hz]	Upper limit	Lower limit	Frequency [Hz]	Upper limit	Lower limit
200	12	–12	2001	6	–6
800	12	–12	4000	6	–6
801	10	–10	8000	6	–6
2000	10	–10			

7.12.5.3 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 terminal to configure their parameters, and to achieve a steady state.

3) To ensure that the AEC of the terminal will configure itself in the receive channel, a training sequence in the form of artificial voice as per [8] 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 from a pause of at least 10 s and a periodically repeated CS signal as per [17] 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.

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

7.13 Properties of electro-acoustic components

7.13.1 This section specifies the parameters and requirements for microphones used in loudspeaker IVS if such microphones may be tested separately from an IVS, and the IVS itself has an input for external microphone connection.

The tests described herein pertain to the checks of single external microphones (directional or omni-directional, passive or active) and do not apply to microphone arrays using additional signal processing (direction pattern beam generation, noise removal, etc.).

The requirements for acoustic measurements during the tests are established in GOST R 53576.

The microphone tests are first completed in standard conditions of the anechoic chamber, and then at the location selected for installation of the microphone in the vehicle compartment.

7.13.2 Measurements in anechoic chamber

The measurements in the anechoic camera are aimed at the evaluation of original microphone parameters in standard free-field acoustic environment with no regard to any effects on their characteristics that may be caused by acoustic properties of the vehicle compartment or by the microphone location and orientation. The tests are carried out using a reference speaker with low level of its intrinsic electro-mechanic distortions.

7.13.3 Microphone sensitivity

7.13.3.1 The IVS microphone sensitivity measurement is mandatory. The sensitivity is measured in the anechoic chamber and then checked after the microphone is placed in the vehicle compartment. The requirements for IVS sensitivity unification of GOST R 54620 are treated as recommendations.

7.13.3.2 Requirements

The microphone sensitivity at a frequency of 1 kHz measured in direction where its direction pattern (DP) has a maximum shall be equal to the value specified by the IVS manufacturer. Unification of sensitivity at a level of 300 mV/Pa ± 3 dB is recommended for IVS microphones to ensure their interchangeability.

7.13.3.3 Method of measurement

1) The measurements shall be carried out in free sound field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed at a distance of 1 m from the measuring speaker, on a line through the centre of that speaker.

3) The acoustic test signal is a sine-wave signal of a frequency equal to 1 kHz and an SPL equal to 0 dBPa at the microphone location for undisturbed sound field with no microphone present.

4) The microphone under test is directed to the speaker, as judged by the DP (output voltage) maximum.

5) The microphone sensitivity is evaluated in mV/Pa.

N o t e — A narrowband (one-third-octave) noise signal of an average frequency equal to 1 kHz and an SPL equal to 0 dBPa is permitted as a test signal.

7.13.4 Microphone frequency response

7.13.4.1 Requirements

The frequency response of an IVS microphone measured in free sound field conditions shall be within the tolerance limits listed in Table 18 for narrowband and in Table 19 for wideband IVS.

T a b l e 18 — Frequency response of microphones for narrowband IVS

Frequency [Hz]	Upper limit [dB]	Lower limit [dB]	Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
200	0	–∞	500	0	–12
250	0	–∞	630	0	–11
315	0	–14	800	0	–10
400	0	–13	1000	0	–8

Table 18 (continued)

Frequency [Hz]	Upper limit [dB]	Lower limit [dB]	Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
1300	+2	–8	2500	+4	–8
1600	+3	–8	3100	+4	–8
2000	+4	–8	4000	+4	–∞

T a b l e 19 — Frequency response of microphones for wideband IVS

Frequency [Hz]	Upper limit [dB]	Lower limit [dB]	Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
100	0	–∞	1000	+0	–8
125	0	–∞	1300	+2	–8
200	0	–14	1600	+3	–8
315	0	–13	2000	+4	–8
400	0	–12	3100	+4	–8
500	0	–11	4000	+4	–8
630	0	–10	8000	+4	–∞

The microphone frequency response shall be flat in the range of 200 Hz — 4 kHz for narrowband IVS and in the range of 100 Hz — 7 kHz for wideband IVS.

7.13.4.2 Method of measurement

1) The measurements shall be carried out in free sound field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed at a distance of 1 m from the measuring speaker, on a line through the centre of the speaker.

3) The acoustic test signals are sine-wave signals of the specified frequencies and an SPL equal to 0 dBPa at the microphone location for undisturbed sound field with no microphone present.

4) The microphone under test is directed to the speaker, as judged by the maximum output voltage at a frequency of 1 kHz.

5) The microphone sensitivity for all frequencies is evaluated in mV/Pa.

7.13.5 Microphone harmonic distortion level

7.13.5.1 Requirements

The total harmonic distortion factor of a microphone shall not exceed 1 % for test sine-wave signals of an SPL equal to 0 dBPa.

7.13.5.2 Method of measurement

1) The measurements shall be carried out in free sound field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed on a line through the centre of the measuring speaker.

3) The acoustic test signals are sine-wave signals of frequencies equal to 300 Hz, 500 Hz and 1 kHz for narrowband IVS or to 300 Hz, 500 Hz, 1 kHz and 2 kHz for wideband IVS and an SPL equal to 0 dBPa at the microphone location for undisturbed sound field with no microphone present.

4) The microphone under test is directed to the speaker, as judged by the maximum DP at a frequency of 1 kHz.

5) The total harmonic distortion factor of the microphone is determined in % for each frequency. The measurements are taken in the IVS operating frequency range.

N o t e s

1 Before the measurements, the measuring microphone shall be used to check that the reference speaker itself exhibits smaller distortions at a given SPL than those of the microphone under test.

2 An additional check is necessary to ensure that the IVS microphone distortions do not exceed the specified value at SPL below 0 dBPa as well.

7.13.6 Maximum sound pressure level

7.13.6.1 The maximum SPL of an IVS microphone is defined as a level limited by harmonic distortions of that microphone.

7.13.6.2 Requirements

For the test signals of 1 kHz frequency, the maximum SPL limited by microphone distortions equal to 3 % shall be less than 12 dBPa (106 dB SPL) for microphones of a standard sensitivity 300 mV/Pa.

7.13.6.3 Method of measurement

1) The measurements shall be carried out in free sound field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed on a line through the centre of the measuring speaker.

3) The acoustic test signal is a 1 kHz sine-wave signal of an SPL that is gradually increased until the microphone distortion level reaches 3 %.

4) The microphone under test is directed to the speaker, as judged by the maximum DP.

5) The SPL at the microphone location for an undisturbed sound field with no microphone present is expressed in dB SPL or dBPa.

N o t e s

1 Before the measurements, the measuring microphone shall be used to check that the reference speaker itself exhibits smaller distortions at a given SPL than those of the microphone under test.

2 The maximum SPL of a properly designed microphone is limited by electric distortions due to limitations of its measuring circuit rather than its mechanical structure. For microphones of sensitivity equal to 300 mV/Pa at an SPL equal to 106 dB SPL, the peak-to-peak value (V_{pp}) of the sine-wave signal at the output will be up to 3.3 V.

7.13.7 Microphone self-noise

7.13.7.1 Requirements

The microphone self-noise shall not exceed minus 72 dBV(A) for sensitivity of 300 mV/Pa (the values of up to minus 66 dBV(A) are permitted provided that the microphone noise does not impair the IVS noise level performance in sending direction).

7.13.7.2 Method of measurement

1) No test signals are used.

2) The microphone shall be powered from a source with low self-noise level.

3) The microphone self-noise is measured at the output of the circuit shown in Figure 2, in the frequency range from 100 Hz to 4 kHz using psophometric frequency-weighting along A-curve.

4) The self-noise is expressed in dBV (A).

N o t e — The check shall be made to ensure that the acoustic noise is lower than the equivalent microphone self-noise expressed in dB SPL.

7.13.8 Spatial selectivity

7.13.8.1 The spatial selectivity of a microphone is described by the direction pattern (DP) of that microphone which describes the dependence of its sensitivity on the incidence angle of a flat sound wave.

The front-to-back ratio is the ratio between the sensitivity in direction of the microphone DP maximum to the one in direction of its minimum measured at a frequency of 1 kHz and expressed in dB.

7.13.8.2 Requirements

In order to achieve the required suppression of background noise in the vehicle compartment, the recommended front-to-back ratio shall be at least 10 dB.

N o t e — The final benefit as regards the SNR value depends on the microphone location and orientation in the vehicle compartment. If placed inefficiently, a super-directional microphone may lead to worse results than a near omni-directional one.

7.13.8.3 Method of measurement

1) The measurements shall be carried out in free sound field conditions using the arrangement shown in Figure 2.

2) The microphone under test is placed at a distance of 1 m from the measuring speaker, on a line through the centre of the speaker.

3) The acoustic test signal is a 1 kHz sine-wave signal of an SPL equal to 0 dBPa at the microphone location for an undisturbed sound field with no microphone present.

4) In the first test, the microphone is directed to the speaker, as judged by the maximum DP. In the second test, it is to the speaker, as judged by the minimum DP. If the exact minimum position is unknown, it must be determined by rotation of the microphone.

5) The front-to-back ratio is determined in dB.

7.13.9 Measurements in vehicle compartment

7.13.9.1 The measurements in the vehicle compartment are aimed at the selection of the best IVS microphone position and orientation, and at the assessment how much the acoustic properties of the vehicle compartment affect the microphone parameters. The tests are carried out using an "artificial mouth" device located in the HATS manikin's head.

7.13.9.2 Selection of microphone location in vehicle

The optimum microphone location is specific to a vehicle type, and shall be selected experimentally based on the recommendations listed below.

1) The microphone shall be placed as close as possible to the talker so as to fall within his near-field acoustic zone (normally not farther than 50—100 cm) where the energy of a direct sound beam is higher than the total energy of reflected beams, and the reverberation is small. It has a considerable impact on speech intelligibility and on its spectral content.

2) The direct beam energy of the talker's voice is inversely proportional to the squared distance from the microphone while the acoustic noise energy does not depend on the distance to the talker; thus, the SNR of the voice signal falls with the distance. The SNR of a single microphone located close to the talker may be higher than the one of a microphone array placed more distantly.

3) No obstructions shall exist between the microphone and the talker's mouth. Such obstructions reduce the SNR and increase the reverberation.

4) On the average, the DP maximum direction for the microphone shall coincide with the direction to the talker's mouth.

5) The microphone of the terminal shall be protected from direct air flows in the vehicle compartment from half-open windows, the conditioning system, etc.

6) The IVS microphone shall be protected from limitation (saturation) caused by closely located speakers, especially low frequency ones, by air vibration in the vehicle compartment due to engine operation, or by pressure differentials.

7) The microphone of the terminal shall be properly suspended to prevent entry of noise caused by vibration of the vehicle body.

The vehicle or IVS power system is recommended for power supply of the microphone during the measurements of its parameters.

7.13.10 Microphone sensitivity in vehicle compartment

7.13.10.1 The microphone sensitivity is measured in the anechoic chamber, followed by the check in the IVS that the microphone and its position in the vehicle compartment are suitable for creating the required voltages at the IVS input. The requirements for the nominal IVS input level are specified by the IVS manufacturer. The unification requirements stated in GOST R 54620 in regard to the input level should be treated as recommendations.

7.13.10.2 Requirements

For an acoustic signal with SPL at the MRP equal to 0 dBPa, the recommended signal level at the microphone output shall be 19 mV ±3 dB; this is equivalent to the microphone sensitivity of 300 mV/Pa and the measurement in the anechoic chamber with the microphone at a distance of 50 cm from the MRP.

Making allowance for potential variations of microphone sensitivity, acoustic design and position, this requirement may be corrected by the IVS manufacturer in regard to the vehicle compartment.

7.13.10.3 Method of measurement

1) The measurements in the vehicle compartment shall be carried out using the arrangement shown in Figure 2.

2) The acoustic test signal is a narrowband (one-third-octave) noise signal of an average frequency equal to 1 kHz and an SPL equal to 0 dBPa at the MRP.

3) The output voltage of the microphone is measured in mV.

7.13.11 Frequency response of microphone in vehicle compartment

7.13.11.1 The frequency response of the microphone is measured in the anechoic chamber, followed by the check in the vehicle compartment that the microphone and its position are capable to ensure the required frequency response tolerance in the IVS sending direction taking into account the acoustic properties of the vehicle compartment.

7.13.11.2 Requirements

The frequency response of the microphone is checked along the path from the MRP to the electric output of the measuring circuit shown in Figure 2.

The relative tolerance requirements for the frequency response of microphones in the IVS compartment in sending direction are specified in Table 20 for narrowband IVS and in Table 21 for wideband IVS. Linear interpolation on log-log scale may be used for intermediate frequencies.

T a b l e 20 — Frequency response of microphones for narrowband IVS

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

T a b l e 21 — Frequency response of microphones for wideband IVS

Frequency [Hz]	Upper limit [dB]	Lower limit [dB]	Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
100	4	–∞	5000	8.5	–4
300	4	–4	6300	9	–7
1000	4	–4	8000	9	–∞

The frequency response of the microphone for a vehicle terminal shall be flat in the range of 200 Hz — 4 kHz for narrowband terminals and 100 Hz — 7 kHz for wideband terminals. 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.

7.13.11.3 Method of measurement

1) The measurements in the vehicle compartment shall be carried out using the arrangement shown in Figure 2.

2) The acoustic test signal is an artificial voice as per [8], periodic wideband noise signal, or CS signal as per [17]. The acoustic mouth shall be calibrated and equalised at the MRP. The SPL of the test signal at the HFRP of the HATS is set to minus 28.7 dBPa (level averaged along the full signal path).

3) The power spectrum density of the test signal measured at the MRP is used as an initial density when the frequency response of the terminal is calculated in sending direction.

4) The frequency sensitivity response is measured in one-third-octave frequency bands according to GOST R 8.714 in the frequency range from 100 Hz to 4 kHz for narrowband IVS and from 100 Hz to 8 kHz for wideband IVS. Frequency averaging along the full path of the test signal is used in calculations of the signal level in each frequency band.

5) The microphone sensitivity is expressed in dBV/Pa.

7.13.12 Directional properties of microphone in vehicle compartment

7.13.12.1 Requirements

The recommended SNR enhancement for the driver speech in noise conditions which is ensured by the directional properties of the IVS microphone shall be at least 3 dB compared to an omni-directional wideband microphone installed at the same place, after all distinctions in frequency-weighting of signals are taken into account.

7.13.12.2 Method of measurement

1) The environment conditions of testing shall comply with the requirements of Section 6. The measurements in the vehicle compartment shall be carried out using the arrangement shown in Figure 2.

2) All types of external acoustic noise corresponding to the scenarios of Table D.1 (Appendix D) shall be tested. The noises shall be turned on at least 5 s prior to measurements in order to provide for adaptation of noise suppression algorithms [5] if any.

3) First, only a disturbing noise signal of a given SPL is turned on and recorded at the output of the measuring circuit shown in Figure 2. To calculate the noise level (denoted as LN_hft_mic), frequency-weighting along A-curve is used along with averaging in the frequency band from 200 Hz to 4 kHz for narrowband IVS and from 100 Hz to 8 kHz for wideband IVS. The noise level is expressed in dBV/Pa(A).

4) Then, only a useful signal (without noise) is applied in sending direction. This signal is a CS signal as per [17] that contains more than two elementary sequences. To calculate the speech level (denoted as LS_hft_mic), frequency-weighting along A-curve is used along with averaging in the frequency band from 200 Hz to 4 kHz for narrowband IVS and from 100 Hz to 8 kHz for wideband IVS. The speech level is expressed in dBV/Pa(A).

5) The real frequency response of the IVS microphone is evaluated for the useful signal (without noise) as per 7.13.2.2; it is recorded, and used for later normalising.

6) The SNR for the IVS microphone is calculated using the formula: SNR_hft_mic=LS_hft_mic minus LN_hft_mic.

7) An omni-directional measuring microphone with a flat frequency response is placed as close as possible to the IVS microphone. Using the CS signal for both microphones, the real frequency response of the omni-directional microphone is measured, and then weighted with the one obtained at step 5 for the microphone of the terminal. This is necessary to ensure that the differences in frequency-weighting for the two microphones may not affect the comparison of their directional properties.

8) Steps 3) — 4) are repeated using the omni-directional microphone and taking into account the additional weighting of its frequency response. The measured noise level is denoted as LN_omni_mic, and speech level as LS_omni_mic.

9) The SNR for the omin-directional microphone is calculated using the formula: SNR_omni_mic = LS_omni_mic minus LN_omni_mic.

10) The SNR enhancement for the directional microphone with respect to the omni-directional one is assessed as SNR_hft_mic minus SNR_omni_mic.

7.14 Subjective quality assessment of IVS loudspeaker communication

Objective measurements of the loudspeaker IVS parameters shall be supplemented by a subjective expert assessment of the IVS performance for different external operation conditions.

The loudspeaker IVS performance depends on:

- IVS specifications and configuration parameters;
- acoustic properties of vehicle compartment;
- selection of installation locations for IVS microphone and speakers;
- 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 IVS 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 IVS installed and configured.

7.14.1 Test management

In subjective assessments of the loudspeaker IVS quality, a driver in the vehicle with an installed IVS is considered as a near-end subscriber of 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, presence of open windows, etc.

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

The subscriber activities 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).

7.14.2 Measured 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 IVS during single-talk (echo signal perception degree and intensity, AEC convergence, call direction switching time, etc.);
- AEC performance quality of IVS 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 and of background noise in pauses in receiving direction (loudness rating, loudness jumps, intelligibility, speech natural sounding and quality, SNR, voice signal distortions, etc.);
- stability of AEC and of the whole system during communication sessions (no echo signal bursts, acoustic back-coupling, oscillation transients, sounding artefacts, etc.).

The methods described in GOST R 50840 and GOST R 51061 as well as in [21] — [26] shall be used for assessment of individual indicators.

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 one.

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 IVS quality assessment may be participated by ordinary unqualified IVS 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 situations for subjective testing is listed in Table 22. The set of tests is detailed in Table 23.

T a b l e 22 — Set of typical conditions for subjective testing

Test condition	Status
Vehicle movement	1) Standing, engine switched off; 2) Standing, engine switched on; 3) Movement along urban road at 70 km/h; 4) Movement along highway at 120 km/h.
Vehicle environment	1) Quiet street; 2) Heavy-traffic route.
Vehicle windows	- Closed - Open
Ventilation, conditioning, heating	- On - Off
IVS volume control position	- Minimum - Nominal - Maximum
Talker speech	- Male voice (3 talkers) - Female voice (3 talkers)
Talker location	1) Driver seat; 2) Passenger seat next to driver; 3) Passenger seat back from driver.
Talker speech tempo	1) Normal; 2) Accelerated;
Talker speech level	1) Normal; 2) Weak; 3) Loud (shrill).

T a b l e 23 — Set of tests for subjective quality assessment

Echo canceller		Parameters to be assessed
Alternating single-talk	Parameters to be assessed are those describing inconvenience caused by AEC operation during the talk. Combinations of test conditions according to Table 5.	1) Echo signal occurrence rate; 2) Echo signal perception and level (talker's ability to speak when his/her own echo signals are present); 3) Duration of AEC setup for echo signal suppression when the connection is established or the echo path is changed; 4) Perception degree of effects related to call direction switching processes (beginning or ending of words, phrases, sentences dropped at the other party side); 5) Distinctive loudness jumps of talker voice and of background noise
Simultaneous double-talk		

Table 23 (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. Combinations of test conditions according to Table 5.	1) Speech loudness level; 2) Jumps of speech loudness level; 3) SNR of voice signals; 4) Overall speech quality (distortions, artefacts, etc.); 5) Overall intelligibility of driver speech, efforts required to understand its meaning; 6) Natural sounding and talker identification ability; 7) Quality of background noise transmission

Table 23 (continued)

Speech quality in noise conditions in sending direction		Parameters to be assessed
Transient acoustic noise in vehicle cabin	Change of movement mode On/off switches of ventilation, conditioning, heating Opening/closing of windows Noise from vehicle instruments (beeps, clicks), wind noise from windows, Noise from traffic passing by	1) Quality of background noise transmission; 2) No sounding artefacts; 3) Adaptation to changes of noise conditions

Table 23 (continued)

Speech quality in noise conditions in receiving direction		Parameters to be assessed
Acoustic noise inside vehicle	Parameters to be assessed are those describing speech intelligibility and quality in receiving direction from operator to vehicle. Combinations of test conditions according to Table 5.	1) Speech loudness level; 2) Jumps of speech loudness level; 3) Rate and degree of IVS loudness adaptation to changes of ambient noise level; 4) Acoustic SNR for voice signals; 5) Overall speech quality (distortions, artefacts, etc.); 6) Overall intelligibility of operator speech, efforts required to understand its meaning; 7) Natural sounding and talker identification ability
System stability		Parameters to be assessed
	Parameters to be assessed are those describing stable IVS operation during communication sessions. Checks to be made at nominal and maximum volume control positions	1) Echo signal suppression stability; 2) AEC adaptation rate; 3) No acoustic back-coupling (ring, whistle or howl at maximum IVS volume); 4) No sounding artefacts due to transient noise

7.14.3 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 22. 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 inner reasons (engine operation, movement along the route, operation of internal climate-control devices, air flows in the compartment), or outer reasons (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.

7.14.4 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 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 R 50840 and GOST R 51061 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 included in these standards. The speech quality and identification ability are assessed using the indicators and scales also detailed in these standards.

A five-grade absolute assessment scale is most common, with Grade 1 corresponding to the worst quality, and Grade 5 to the best one.

When the performance or configuration parameters of different IVS 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.

7.14.5 Speech and background noise quality in sending direction

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

7.14.5.1 Voice signal level

The voice signal loudness is assessed using the absolute ACR scale as per [21] listed in Table 24.

T a b l e 24 — Rating scale for assessment of voice signal loudness

Description of indicator	Grade
Much louder than necessary	1
Louder than necessary	3
Normal loudness	5
Quieter than necessary	3
Much quieter than necessary	1

7.14.5.2 Speech level fluctuations

The speech level fluctuations assessed in this test are described by the following indicators:

- slow changes of signal level;
- abrupt signal drops (e.g., due to lost data packets);
- switch-on effects cutting off weak voice regions in the beginning and in the end of words;
- clipped or intermittent voice.

The loudness fluctuations are assessed using the scale described in Table 25.

T a b l e 25 — Rating scale for assessment of speech level loudness fluctuations

Description of indicator	Grade
No audible fluctuation of voice signal loudness	5
Small level fluctuations barely audible or rarely occurring	4
Frequent moderate level fluctuations	3
Words or syllables are much suppressed or lost sometimes	2
Many losses, heavily clipped or intermittent voice	1

7.14.5.3 Speech quality and natural sounding

The speech quality and natural sounding assessed in this test are described by the following indicators:

- synthetic or robot-like voice sounding;
- voice signal distortions in the form of additional rattling, crackling or hissing;
- limitation of speech frequency range, excessive rise of low or high frequencies;
- additional timbre colouring, such as shriek or whistle, or sharp, thin, metallic or muffled sounding.

The speech quality and natural sounding shall be assessed using the rating scale DCR as per [21] listed in Table 26. The best quality is considered be the one comparable to that of a common fixed telephone.

T a b l e 26 — Rating scale for assessment of speech quality and natural sounding

Description of indicator	Quality range	Grade
Speech sounding is comparable to one of common telephone. Speech sounds clearly and transparently. Voice is fully natural.	Natural	5
Minor degradation compared to common telephone; natural sounding persists. Slight distortions or slight timbre colouring possible.		4
Voice may sometimes sound slightly synthetic. Slight distortions or moderate timbre colouring possible.		3
Very noticeable synthetic sounding. Heavy distortions and apparent timbre colouring.		2
Signal hardly recognised as speech.	Unnatural	1

7.14.5.4 Speech intelligibility and efforts required to understand it

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 as per GOST R 50840 and GOST R 51061 shall be used as test phrases.

The assessment is carried out using the rating scale listed in Table 27.

T a b l e 27 — Rating scale for assessment of speech intelligibility and efforts to understand its meaning

Description of indicator	Grade
Each word sounds clearly and understood without any 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

7.14.5.5 Speech signal-to-noise ratio

The subjective assessment of the speech SNR shall be carried out for voice signals of the near-end subscriber in conditions of loud disturbing acoustic noise. The assessment is made at the output of the transmit channel, and heavily depends on the SNR on the IVS acoustic input as well as on the directional properties of the microphone and the use of additional sound suppression algorithms. This complicates the assessment of the absolute rating, and the assessment shall be based on comparison with operation of another IVS under the same test conditions.

The assessment is carried out using the CCR rating scale as per [21] listed in Table 28.

T a b l e 28 — Rating scale for assessment of speech SNR

Description of indicator	Grade
Noise very low and barely audible	5
Noise audible but its level much lower than speech level, and is not disturbing	4
Medium noise level lower than speech level, noise is slightly disturbing	3
High noise level almost equal to speech level, noise is moderately disturbing but conversation is still possible	2
Noise level higher than speech level, noise is intolerable, conversation is not possible	1

7.14.5.6 Transmission quality of stationary background noise

The transmission quality of stationary background noise in sending direction shall be assessed basing on the following indicators:

- time changes in sounding and level of transmitted noise;
- sounding artefacts (clicks, snaps, chatter) that may be associated by a listener with natural sounds taking place during the vehicle movement;
- natural sounding of noise: noise either similar to natural background noise of the vehicle or distinctive by its synthetic sounding (musical sound resembling purl of water), any audible distortions and rattling.

All sounding artefacts, temporal level and timbre fluctuations, unnatural sounding of the background noise shall reduce its subjective rating.

The assessment is carried out using the ACR rating scale as per [21] listed in Table 29.

Table 29 — Transmission quality of stationary background noise

Description of indicator	Perception range of background noise	Grade
Comfortable natural sounding with no artefacts, constant level and timbre	Natural	5
Slight distortions/synthetic sounding with almost no artefacts, almost constant level and timbre		4
Moderate distortions/synthetic sounding, or artefacts like clicks/gurgle, or moderate level and timbre fluctuations		3
Evident distortions/synthetic sounding, or many artefacts like clicks/gurgle, or frequent level and timbre fluctuations		2
Completely unnatural/distorted/synthetic sounding, or permanent artefacts, or continual variations in level and timbre, very uncomfortable sounding	Unnatural	1

7.14.5.7 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 the 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] listed in Table 30.

Table 30 — Transmission quality of transient background noise

Description of indicator	Perception range of background noise	Grade
Comfortable natural sounding	Natural	5
Almost natural sounding Slight distortions/synthetic sounding		4
Moderately natural sounding Moderate distortions/synthetic sounding		3
Clearly unnatural/distorted/synthetic sounding		2
Completely unnatural/distorted/synthetic sounding	Unnatural	1

7.14.5.8 Adaptation rate in regard to background noise changes

The rate of transmit channel adaptation to background noise changes is assessed if an AGC or NRD algorithm or device is used in the terminal.

The assessment is carried out at the transmit channel output with respect to abrupt changes of noise level at its input, for example, when heating, ventilation or conditioning devices are switched on/off, when the widows are opened or closed, etc.

The assessment is carried out using the rating scale detailed in Table 31.

Table 31 — Adaptation rate in regard to background noise changes

Description of indicator	Adaptation rate	Grade
Immediate adaptation	Very fast	5
Adaptation time ≤ 1 second		4
Adaptation time 2 ... 3 seconds		3
Adaptation time 3 ... 10 seconds		2
Adaptation time > 10 seconds	Very slow	1

7.14.6 Speech quality and background noise quality in receiving direction

The voice signal level shall be evaluated on the side of the near-end subscriber located in the vehicle, with respect to single-talk speech of the far-end subscriber in the call centre. The assessment is made in silence, with the IVS volume control at its nominal position.

7.14.6.1 Speech quality and natural sounding

The speech quality and natural sounding assessed in this test are described by the following indicators:

- synthetic or robot-like voice sounding;
- voice signal distortions in the form of additional rattling, crackling or hissing;
- limitation of speech frequency range, or excessive rise of low or high frequencies;
- additional timbre colouring, such as shriek or whistle, or sharp, thin, metallic or muffled sounding.

The speech quality and natural sounding shall be assessed using the scale described in Table 26. The best quality is considered be the one comparable to that of a common fixed telephone.

7.14.6.2 Speech intelligibility and efforts required to understand it

The phonetically balanced phrases taken from test tables of GOST R 50840 and GOST R 51061 are used as test phrases. The assessment shall be carried out using the rating scale listed in Table 27.

7.14.6.3 Voice signal level in noise conditions (at maximum volume level)

The signal loudness of the voice heard from the IVS speaker is assessed in conditions of loud background noise in the vehicle compartment, at the maximum position of the IVS volume control. The assessment result shall be presented using the rating scale detailed in Table 24.

7.14.6.4 Speech level for new connection (at nominal volume level)

The signal loudness of the voice heard from the IVS speaker for a new call is assessed in conditions of loud background noise in the vehicle compartment, at the nominal position of the IVS volume control. The assessment result shall be presented using the rating scale detailed in Table 24.

7.14.7 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 subscriber speech.

7.14.7.1 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 using the following indicators:

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

The assessment result shall be presented using the rating scale detailed in Table 32.

T a b l e 32 — Speech level variations during double-talk

Description of indicator	Speech level range	Grade
During subscriber's own speech, no level variations in speech of other party and in pauses 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, with syllables and words sometimes heavily attenuated or dropped. Or, moderate attenuation due to switching of call direction	—	3
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	Half duplex only	1

7.14.7.2 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 as per GOST R 50840 and GOST R 51061 are used as test phrases. The assessment result shall be presented using the rating scale detailed in Table 33.

T a b l e 33 — Speech intelligibility and efforts to understand its meaning during double-talk

Description of indicator	Grade
Each word of other party sounds clearly and understood without any 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

7.14.8 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.;
- stable 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 IVS 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.

7.14.8.1 Perception degree of residual echo signal

The assessment shall be carried out using the scales of talk quality degradation as per [21; 24]. 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 34.

Table 34 — Degree of perception and disturbing effects of echo signals

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

7.14.8.2 Echo signal intensity

The test is only carried out if echo signals are present. The assessment result shall be presented using the rating scale detailed in Table 35.

Table 35 — Echo signal intensity

Description of indicator	Grade
Weak	4
Moderate	3
Loud	2
Very loud	1

7.14.8.3 Echo signal duration

The test is only carried out if echo signals are present. The assessment result shall be presented using the rating scale detailed in Table 36.

Table 36 — Echo signal duration

Description of indicator	Grade
Very short	5
Short	4
Medium	3
Long	2
Very long, permanent	1

7.14.8.4 Echo signal rate

The test is only carried out if echo signals are present. The assessment result shall be presented using the rating scale detailed in 37.

T a b l e 37 — Echo signal rate

Description of indicator	Grade
Only once during whole test	5
Only twice during whole test	4
Several times during whole test	3
Occur rather frequently	2
Always present	1

7.14.8.5 Residual intelligibility of echo signals

The test is only carried out if echo signals are present. The assessment result shall be presented using the rating scale detailed in 38.

T a b l e 38 — Intelligibility of echo signals

Description of indicator	Grade
Pure artefacts	5
Hardly identified as speech	4
Severely distorted voice	3
Slightly distorted voice	2
Clear voice	1

7.14.8.6 "Comfort noise" quality at AEC output

The test is carried out if the AEC creates "comfort noise" to mask echo signals. The following characteristic sounding indicators shall be considered:

- intensity and timber changes of background noise in time, when the transmission in sending channel is switched from natural pause noise to artificial comfort noise and vice versa;
- noticeable artefacts (clicks, claps, cracks, chatter, etc.);
- natural sounding, distortions or synthetic character (musical noise, simulated purl of water) of background noise.

The artificial "comfort noise" shall be audibly indistinguishable from the natural pause noise. If the above characteristic side tones are present, the grade as per Table 39 shall be decreased.

T a b l e 39 — "Comfort noise" quality at AEC output

Description of indicator	Perception range	Grade
Artificial "comfort noise" indistinguishable from natural pause noise. Noise is comfortable, natural, level and timbre is unchanging, no switching artefacts are present	Natural	5
Slight difference between "comfort noise" and natural pause noise. Slightly distorted/synthetic sounding; almost unchanging level and timbre; almost no switching artefacts		4
Moderate difference between "comfort noise" and natural pause noise. Moderately distorted/synthetic sounding; moderate variations of level and timbre; audible switching artefacts		3
Clearly audible distinction between "comfort noise" and natural pause noise. Clearly distorted/synthetic sounding; frequent variations of level and timbre; many artefacts		2
No similarity between "comfort noise" and natural pause noise. Very uncomfortable, unnatural, distorted/synthetic sounding; continual variations of level and timbre	Unnatural	1

7.14.9 IVS stability in regard to loudspeaker communication

The assessment of stable IVS behaviour in regard to loudspeaker communication shall include the check that no acoustic back-coupling (whistle, howl) occur when the connection is established, the IVS volume control is set to its maximum position, and loudspeaker communication mode is used at the far-end subscriber side.

In this test, the IVS is first turned off for a short time then turned on again in order to reset all adaptive filter factors in the AEC to their initial settings. Then, the connection is initiated at the maximum volume, and the IVS behaviour as well as the convergence rate of AEC factors (settings) are evaluated.

The communication path stability is assessed both for the speech of the far-end subscriber and for the pulse noise (when the microphone is clicked) of the near-end subscriber. In both cases, the AEC is not yet configured at the start of the test, so the echo signal suppression is lowest, whereas the loop amplification possibly leading to acoustic back-coupling is highest.

The assessment result shall be presented using the rating scale detailed in 40.

Table 40 — IVS stability in regard to loudspeaker communication

Description of indicator	Grade
Echo signal is not audible	5
Echo signal is audible, but disappears very quickly	4
Echo disappears slowly, repetitions are audible for several seconds	3
Echo disappears very slowly, repetitions are audible for more than 10 seconds	2
Echo does not disappear, persistent sustained repetitions occur, possibly with whistle or howl. The system is utterly unstable.	1

Appendix A
(normative)

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
ERA-GLONASS System emulator	Emulator type: selected according to the wireless mobile communication network type (GSM, UMTS) used in the IVS and taking into account the requirements of 6.6, 6.7 and 6.8. Maximum level of electric self-noise at encoder input and at decoder output: not greater than minus 74 dBm0(A). Harmonic distortion in receiving and in sending direction: not greater than 1 % (for a codec supported by the IVS, at its highest bitrate)
HATS manikin in form of artificial head and torso	Basic requirements for manikin: as per [12], [20]. 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. Variation of frequency response of sound pressure in frequency range from 100 to 10000 Hz: not greater than ±3 dB. 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. Additional requirements: in accordance with 6.4, 6.7 and [9], [12], [20]
Artificial ear	Basic error in sound pressure measurements: not greater than ±0.5 dB. Variation of frequency response in frequency band range 100 to 8000 Hz: not greater than ±2 dB. Harmonic distortion at sound pressure of 10 Pa: not greater than 1 %. Additional requirements: in accordance with 6.5, 6.7 and [11], [12], [20]
Noise meter	In accordance with GOST R 53188.1. Accuracy class: not greater than 2
Additional measuring microphone	Type: condenser 1/2", pressure-based measurements. Basic error: not greater than ±0.5 dB. Variation of frequency response in frequency range from 0.1 to 16 kHz: not greater than 2 dB. 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. 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 IVS characteristics	ADC/DAC sampling rates: 8, 16, 32, 48 KHz. Sample width: not less than 16 bits. Number of channels: not less than two. Dynamic range of ADC/DAC: not less than 80 dB. 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 a card for output of noise signal PC (DAC) and a set of special purpose software for noise simulation in vehicle cabin	DAC sampling rates: 8, 16, 32, 48 kHz. Sample width: not less than 16 bits Number of channels: not less than five Dynamic range of DAC: not less than 80 dB. 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). Rated power sufficient for noise generation in vehicle compartment: not less than 90 dBA. 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. Sampling rate: not less than 32 kHz. Sample width: not less than 16 bits. Number of channels: not less than two. 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. Sampling rate: not less than 32 kHz. Sample width: not less than 16 bits. Number of channels: not less than two. Dynamic range: not less than 80 dB.
Electronic voltmeter for measurements of sine-wave signals	Accuracy class: 1.5. Frequency range: from 20 to 20000 Hz. Measurement range: from 1 mV to 10 V. Input resistance: not less than 1 MΩ.
Acoustic calibrator	In accordance with GOST R IEC 60942 for microphone rated for pressure 1/2"
N o t e — The microphone parameters are measured in an anechoic chamber using the equipment specified in GOST R 53576 and taking into account the requirements of 7.12.1.	

Appendix B
(normative)

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 [8] and [17]. If a composite CS test signal is used for a wideband IVS, an additional spectrum spreading from 4 to 8 kHz with a 5 dB fall by an octave in high frequency direction is used with the properties presented in Figure 6 of [17].

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 IVS, this is achieved using a band-pass filter with the low cut at 200 Hz, high cut at 4 kHz and frequency response slope at most 24 dB per octave. For wideband IVS, a band-pass filter with the low cut at 50 Hz, high cut at 8 kHz and frequency response slope at most 24 dB per octave is used.

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

Unless otherwise specified, 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 [10].

The following test signal levels are considered as nominal ones:

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 4.7 dBPa at the MRP (typical average level of speech) or minus 28.7 dBPa at the microphone HFRP (except for the tests carried out in acoustic noise conditions where a person may inadvertently increase speech loudness).

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

B.2 Noise signals

Noise signals are used in certain measurements in order 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 22 (subsection 7.13).

As regards the loudspeaker communication in acoustic noise conditions, an IVS 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 vehicle manufacturer documents do not specify an exact level of the noise signal, it is assumed to be equal to minus 24 dBPa(A) (70 dBPa SPL) for an "ordinary" noise environment, and minus 14 dBPa(A) (80 dBPa SPL) for a "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 IVS microphone.

If the IVS is equipped with a digital debug interface, noise signals may be recorded from the IVS 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 IVS, because the use of four speakers for external noise modelling in the IVS 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 and Table 22 contain the recommended list of noise scenarios that shall be used for noise recording and IVS performance checks.

If the test is intended for performance quality comparison of different IVS systems 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 possible methods are recommended for playback of noise signals depending on the scope of the test:

1) Acoustic method

The noise in the vehicle cabin is reproduced using four speakers as illustrated in Figure 3. 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 installation locations shall be selected so that the direct visibility between the IVS microphone and the artificial head of the HATS manikin would not be compromised. In addition, the recordings made using a broadband measuring microphone shall be used, the amplification shall be calibrated and the frequency response equalised for all playback channels including the speakers. The detailed information is given in [3].

2) Electric method

Noise signals may be electrically injected into a signal from the IVS microphone or microphones. For this purpose, noise recordings made using the IVS 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 the 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.

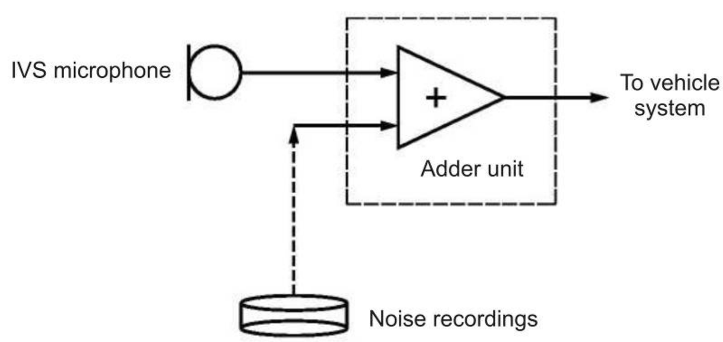


Figure B.1 — Flow diagram of electric method used to inject previously recorded noises into test signals

3) Digital method

Noise signals may be recorded and later mixed with the IVS microphone signal digitally, using the DI-S2 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 directed microphones or microphone arrays are used in the IVS.

Appendix C
(recommended)

Digital interface used in tests

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

On the one hand, this will allow avoiding labour-consuming acoustic measurements, and on the other hand this will obviate the use of a system simulator during intermediate measurements.

The use of the digital interface is also recommended during IVS adaptation and configuration in part of loudspeaker communication for a particular vehicle model. The final IVS 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 IVS, and considers the IVS as a black box with two signal processing directions: receive channel and transmit channel. The sound signals in digital form at the channel inputs and outputs may be read and transmitted from the terminal to the PC for their recording in files, or read from the files on the PC and transmitted to the IVS 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.

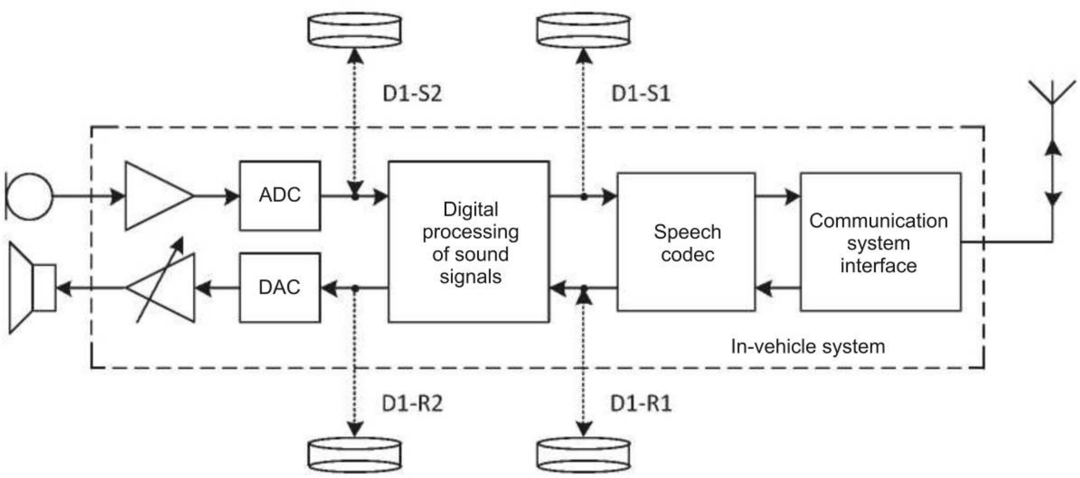


Figure C.1 — Digital debug interface

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 the file on the PC to the IVS 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., those 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 recording of such signals in a file, or for local input of similar signals form the file on the PC to the IVS 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., using an echo canceller, noise suppressor, AGC, etc., and for their recording in a file on the PC.

The analysis of the transmitted signals at point DI-S1 does not include their encoding/decoding in mobile communication systems; this 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 IVS 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-Las or μ-Low encoding as per [6].

If a wideband IVS 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 [7].

If different sampling rates are used for signal processing in the IVS, 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 IVS manufacturer basing on the required dynamic range of signal and on the overload margin. The recommended nominal values of digital signals for narrowband and wideband IVS are given in Appendix B.

A hardware implementation of the interface in regard to real-time signal exchange between the IVS and the PC is not standardised, and is specified by the IVS 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 IVS shall be implemented.

An alternative method that may be used to test processing algorithms for the signals included in the IVS when such algorithms are developed is their modelling on the PC with the file I/O of the signals.

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 the IVS, the following otherwise unavailable recordings and tests are recommended.

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

Many tests require acoustic noise recordings in the IVS cabin with the noise waveform coinciding with the one at the IVS 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 speech and sent to the IVS input through the same DI-S2 interface.

C.3.2 Recording of near-end subscriber 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 speech and sent to the IVS 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 R 50840.

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

This evaluation is based on the PESQ-MOS criterion as per [27] and [28] for narrowband IVS and [29] for wideband IVS, 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 reference point).

The speech quality assessed 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 IVS, the following requirements shall be met:

$$\text{MOS-LQON(S1)} \geq \text{MOS-LQON(POI)} \geq 3.0.$$

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

$$\text{MOS-LQOW(S1)} \geq \text{MOS} - \text{LQOW(POI)} \geq 3.6.$$

The values of the difference

DELTAN = MOS-LQON(S1) minus MOS-LQON(POI), and
DELTAW = MOS-LQOW(S1) minus MOS-LQOW(POI)

may be considered as a quality degradation value due to encoding and to transfer through the mobile communication system.

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 IVS for double-talk mode.

The test signal in sending direction with near-end subscriber speech and echo signals is recorded using the DI-S2 interface. The near-end subscriber 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 far-end subscriber speech shall be uncorrelated with near-end subscriber signals.

The following test procedure is used.

1) Before the test, make sure that the acoustic echo canceller of the IVS has completed its setup for the current echo path, and is in maximum echo suppression mode. 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 IVS from 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 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 using the DI-S2 interface, and the processed speech signal with suppressed echo signal, using the DI-S1 interface.

4) Calculate the objective speech quality indicators PESQ-MOS as per [27]—[29] using the original signal of the near-end subscriber 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 IVS, the following requirements shall be met:

$$\text{MOS-LQON(S1)} \geq \text{MOS - LQON(POI)} \geq 2.5.$$

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

$$\text{MOS-LQOW(S1)} \geq \text{MOS - LQOW(POI)} \geq 2.5.$$

The values of the difference

DELTAN = MOS-LQON(S1) минус MOS-LQON(POI) и
DELTAW = MOS-LQOW(S1) минус MOS-LQOW(POI)

may be considered as a quality degradation value due to encoding and to transfer through the mobile communication system.

Appendix D
(normative)

Minimum standard set of noise scenarios

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

No. of scenario	Description	Speed, km/h	Conditioning/heating settings	Windows	Windscreen wipers	Turn signal	Background talk	Road pavement
1	Stop. Engine on. Low conditioner noise	0	Turned on for minimum level	Closed	Off	Off	No	—
2	Street movement. Loud conditioner noise.	60	Turned on so that conditioner noise is 6 dBA higher than vehicle movement noise. Air flows in IVS microphone direction.	Closed	Off	Off	No	Dry rough road
3	Highway movement. Low conditioner noise.	120	Turned on for minimum level	Closed	Off	Off	No	Dry rough road
4	Highway movement. Loud conditioner noise.	120	Turned on as in No. 2	Closed	Off	Off	No	Dry rough road

Note — 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 road that is too smooth does not fit as it creates insufficient noise from its coupling to vehicle tyres. A road with dents or bumps creates noise bursts that should also be avoided when the signals are recorded. A concrete pavement is preferable because it usually generates the loudest noise in the vehicle compartment. For scenarios No. 2 and No. 4, it shall be checked that the air flow from the conditioning/heating/ventilation system does not directly strike the microphone which is recording acoustic noises in the vehicle compartment.

Appendix E
(reference)

Units of 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 Ω, 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, levels equal to +3.15 dBm0 for A-Law and to +3.18 dBm0 μ-Law as per [6] are taken for an overflow (limitation start) point of analogous 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 [7], a level of +9 dBm0 is taken for an overflow (limitation start) point of analogous 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.**

Bibliography

[1] Technical Regulation on "On Safety of Wheeled Vehicles" (approved by Order of the RF Government No. 720 dated September 10, 2009)
[2] Technical Regulation of the Customs Union "On Safety of Wheeled Vehicles" TP TC (018/2011), approved by Order No. 877 dated December 9, 2011, of the Customs Union Commission (in edition of the European Economic Council, Order No. 6 dated 30.01.2013)

[3] ETSI EG 202 396-1:2011	Speech and multimedia Transmission Quality (STQ); Speech quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database
[4] ITU-T G. 122	Recommendation ITU-T G.122. Influence of national systems on stability and talker echo in international connections
[5] ITU-T G. 160	Recommendation ITU-T G.160. Voice enhancement devices
[6] ITU-T G.711	Recommendation ITU-T G.711. Pulse code modulation (PCM) of voice frequencies
[7] ITU-T G.722	Recommendation ITU-T G.722. 7 kHz audio-coding within 64 kbit/s
[8] ITU-T P.50	Recommendation ITU-T P.50. Artificial voices
[9] ITU-T P.51	Recommendation ITU-T P.51. Artificial mouth
[10] ITU-T P.56	Recommendation ITU-T P.56. Objective measurement of active speech level
[11] ITU-T P. 57	Recommendation ITU-T P.57. Artificial ears
[12] ITU-T P.58	Recommendation ITU-T P.58. Head and torso simulator for telephonometry
[13] ITU-T P.79	Recommendation ITU-T P.79. Calculation of loudness ratings for telephone sets
[14] ITU-T P. 340	Recommendation ITU-T P.340. Transmission characteristics and speech quality parameters of hands-free terminals
[15] ITU-T P. 341	Recommendation ITU-T P.341. Transmission characteristics for wideband (150— 7000 Hz) digital hands-free telephony terminals
[16] ITU-T P.342	Recommendation ITU-T P.342. Transmission characteristics for narrow-band digital loudspeaking and hands-free telephony terminals
[17] ITU-T P.501	Recommendation ITU-T P.501. Test signals for use in telephonometry
[18] ITU-T P.502	Recommendation ITU-T P.502. Objective test methods for speech communication systems using complex test signals
[19] ITU-T P.502 A1	Recommendation ITU-T P.502 Amendment 1. Objective test methods for speech communication systems using complex test signals. Amendment 1: New Appendix III —Automated double talk analysis procedure
[20] ITU-T P.581	Recommendation ITU-T P.581. Use of head and torso simulator (HATS) for hands-free terminal testing
[21] ITU-T P.800	Recommendation ITU-T P.800. Methods for subjective determination of transmission quality
[22] ITU-T P.800.1	Recommendation ITU-T P.800.1. Mean Opinion Score (MOS) terminology
[23] ITU-T P.830	Recommendation ITU-T P.830. Subjective performance assessment of telephone-band and wideband digital codecs
[24] ITU-T P.831	Recommendation ITU-T P.831. Subjective performance evaluation of network echo cancellers
[25] ITU-T P.832	Recommendation ITU-T P.832. Subjective performance evaluation of hands-free terminals
[26] ITU-T P.835	Recommendation ITU-T P.835. Subjective test methodology for evaluating speech communication systems that include noise suppression algorithm
[27] ITU-T P.862	Recommendation ITU-T P.862. Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs
[28] ITU-T P.862.1	Recommendation ITU-T P.862.1. Mapping function for transforming P.862 raw result scores to MOS-LQO

[29] ITU-T P.862.2 Recommendation ITU-T P.862.2. Wideband extension to Recommendation P.862 for the assessment of wideband telephone networks and speech codecs

[30] ITU-T P. 1100 Recommendation ITU-T P. 1100. Narrowband hands-free communications in motor vehicles

[31] ITU-T P. 1110 Recommendation ITU-T P. 1110. Wideband hands-free communications in motor vehicles

UDC 621.396.931:006.354 OKS 17.140
33.070.040

Keywords: in-vehicle emergency call system/device, acoustic echo canceller, near-end subscriber, far-end subscriber, artificial mouth, artificial ear, tests, quality of loudspeaker communication, receiving (sending) loudness rating, narrowband voice signal, wideband voice signal

Editor *E.S. Kotlyarova*
Technical Editor *V.N. Prusakova*
Proof-Reader *E.D. Dul'neva*
Computer Imposition *A.V. Bestuzheva*