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TECHNICAL SPECIFICATION

**Speech and multimedia Transmission Quality (STQ);
Test Methods and Performance Requirements for
Active Noise Cancellation Headsets and other Earphones**

Reference

RTS/STQ-295

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Modal verbs terminology

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Introduction

Wired or wireless Active Noise Cancellation (ANC) headsets and earphones are a prevalent acoustic interface for communication systems. ANC aims to cancel ambient noise around the user by producing a sound pressure with opposite phase to the undesired noise. Applications of ANC can be found in noise-cancelling headsets, including over the ear and in ear form factors, reduction of noise in car and aircraft interiors, and control of noise in air conditioning ducts.

ANC in headsets and earphones can be realized with both analog and digital techniques, and ANC performance is dependent not only on the technique chosen, but also on the choice and integration of acoustic components into the device. Therefore, large variations in performance can be expected when comparing different devices, even for devices sharing a common ANC technique.

In addition to ambient noise cancellation, ANC headsets and earphones may provide pass-through functionality, wherein external sounds and self-speech are reproduced, in whole or in part, for the user.

To get a holistic view of an ANC device performance, with a focus on the Quality of Experience (QoE) delivered to the user, a range of test methods and performance metrics are summarized in the present document. The present document provides test methods that achieve repeatable and reproducible results in both inter-lab and intra-lab comparisons.

1 Scope

The present document specifies test methods and performance requirements for wireless and wired Active Noise Cancellation (ANC) devices such as headsets, earphones, headphones, hearables, and other ear worn devices (herein referred to as devices). The test methods and performance requirements cover the various functionalities of ANC devices including e.g. noise cancellation, media playback, voice calls, and pass-through mode.

ANC performance metrics and requirements for handsets are out of scope.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

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- [1] IEC 60268-4:2018: "Sound system equipment -- Part 4: Microphones".
- [2] Recommendation ITU-T P.57: "Artificial Ears".
- [3] Recommendation ITU-T P.58: "Head and torso simulator for telephonometry".
- [4] Recommendation ITU-T P.64: "Determination of sensitivity/frequency characteristics of local telephone systems".
- [5] Recommendation ITU-T P.381: "Technical requirements and test methods for the universal wired headset or headphone interface of digital mobile terminals".
- [6] ETSI TS 103 224: "Speech and multimedia Transmission Quality (STQ); A sound field reproduction method for terminal testing including a background noise database".
- [7] ETSI TS 126 132: "Universal Mobile Telecommunications System (UMTS); LTE; Speech and video telephony terminal acoustic test specification (3GPP TS 26.132)".
- [8] Recommendation ITU-T P.501: "Test signals for use in telephony and other speech-based applications".
- [9] Recommendation ITU-T P.56: "Objective measurement of active speech level".
- [10] Recommendation ITU-T G.100.1: "The use of the decibel and of relative levels in speechband telecommunications".
- [11] ETSI TS 103 737: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".
- [12] ETSI TS 103 739: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".

- [13] ETSI TS 102 924: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for Super-Wideband / Fullband handset and headset terminals from a QoS perspective as perceived by the user".
- [14] ISO 532-1:2017: "Acoustics -- Methods for calculating loudness -- Part 1: Zwicker method".
- [15] IEC 61672-1:2013: "Electroacoustics -- Sound level meters -- Part 1: Specifications".
- [16] IEC 61260-1:2014: "Electroacoustics -- Octave-band and fractional-octave-band filters -- Part 1: Specifications".
- [17] EN 50332-1:2013: "Sound system equipment: Headphones and earphones associated with personal music players - Maximum sound pressure level measurement methodology - Part 1: General method for "one package equipment"" (produced by CENELEC).
- [18] Recommendation ITU-T P.700: "Calculation of loudness for speech communication".
- [19] ETSI TS 103 558: "Speech and multimedia Transmission Quality (STQ); Methods for objective assessment of listening effort".
- [20] Recommendation ITU-T P.863: "Perceptual objective listening quality prediction".
- [21] ANSI/ASA S3.5-1997 (R2017): "American National Standard Methods for Calculation of the Speech Intelligibility Index".
- [22] IEC 60565-1:2020: "Underwater acoustics -- Hydrophones -- Calibration of hydrophones -- Part 1: Procedures for free-field calibration of hydrophones".
- [23] IEC 60268-1:1985: "Sound system equipment. Part 1: General".

2.2 Informative references

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The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

- [i.1] Recommendation ITU-T P.1150: "In-car communication audio specification".
- [i.2] ANSI/ASA S12.2-2019: "Criteria For Evaluating Room Noise".
- [i.3] Gordon, C.C., et al. (2014): "2012 Anthropometric Survey of U.S. Army Personnel: Methods and Summary Statistics". U.S. Army Natick Soldier Research Development & Engineering Center.
- [i.4] Recommendation ITU-T P.76: "Determination of Loudness Ratings; Fundamental Principles".
- [i.5] ANSI/CTA 2051-2017: "Personal Sound Amplification Performance Criteria".
- [i.6] ANSI/ASA S3.22-2014: "Specification of Hearing Aid Characteristics".

3 Definition of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the following terms apply:

pass-through: operational mode of an ANC device where external sound and self-speech are reproduced, in whole or in part, for the user

3.2 Symbols

For the purposes of the present document, the following symbols apply:

dB	decibel
dB _{Pa}	dB referenced to 1 Pascal
dB _V	dB referenced to 1 Volt
dB _{m0}	dB _V + 2,2 dB
dB _{SPL}	dB referenced to sound pressure level (2e-5 Pascal)
dB _{SPL(A)}	dB referenced to sound pressure level (2e-5 Pascal), A-weighted according to IEC 61672-1 [15]
ms	milliseconds
N _A	Noise level in dB _{SPL(A)} at the talker's ears

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ANC	Active Noise Cancellation
ASL	Active Speech Level
BGN	BackGround Noise
DF	Diffuse-Field
DRP	Drum Reference Point
DUT	Device Under Test
EEP	Ear canal Entry Point
ES	External Speech
FB	FullBand
FFT	Fast Fourier Transform
HATS	Head And Torso Simulator
KPI	Key Performance Indicator
MOS	Mean Opinion Score
MOS-LEO	MOS - Listening Effort Objective
MOS-LQO	MOS - Listening Quality Objective
MRP	Mouth Reference Point
NB	NarrowBand
NC	Noise Criteria
PCM	Pulse Code Modulated
POI	Point Of Interconnection
PT	Pass-Through (mode)
QoE	Quality of Experience
RF	Radio Frequency
RLR	Receive Loudness Rating
SFR	Send Frequency Response
SLR	Send Loudness Rating
SNR	Signal to Noise Ratio
SPL	Sound Pressure Level
SS	Self-Speech
STMR	SideTone Masking Rating
SWB	Super-WideBand
TCL	Terminal Coupling Loss

TCL _w	Terminal Coupling Loss (weighted)
THD	Total Harmonic Distortion
THD+N	Total Harmonic Distortion Plus Noise
WB	WideBand
WN	Wind Noise

4 Setup requirements

4.1 Test room requirements

4.1.1 Test room size

Tests for ANC devices require an acoustically isolated test room. The test room size should meet the recommendations of ETSI TS 103 224 [6]. In addition to the recommended size range for test rooms in ETSI TS 103 224 [6], smaller sound isolating booths or boxes are also allowed. For laboratories using a sound isolating booth or box, the following additional requirements apply:

- 1) The distance from any background noise generation loudspeaker to the acoustic test equipment shall be > 0,5 m.
- 2) The diameter of any background noise generation loudspeaker diaphragm shall be such that the far-field approximation for sound wave propagation holds true for the frequency range of interest. The following two criteria (from IEC 60565-1 [22]) shall be met simultaneously:
 - a) The distance from the DUT to the speaker shall be at least 10 times larger than the radius of the speaker (frequency independent).
 - b) The distance from the DUT to the speaker shall be greater than $4a^2/k$, where a is the radius of the speaker and k is the acoustic wavenumber (frequency dependent).

NOTE: The extra flexibility in using smaller sound isolating booths or boxes is cognizant of the substantial costs in producing a large test environment compliant to the end-to-end idle noise requirements in clause 4.2.3.

4.1.2 Test room acoustic requirements

The test room or sound isolating booth shall have a clarity index (C80) consistent with the requirements of ETSI TS 103 224 [6], i.e. suitable rooms shall have $C80 > 20$ dB. Furthermore, the test room should have a noise floor less than 15 dB SPL(A) and a Noise Criteria (NC), as defined in ANSI/ASA S12.2 2019 [i.2], of NC15 in order to meet the end-to-end test equipment idle noise requirements presented in clause 4.2.3.

NOTE: Frequency dependent test room recommendations, such as *Noise Criteria* (NC) defined in ANSI/ASA S12.2 2019 [i.2] provide helpful guidelines when designing an acoustic space for testing. However, ANSI/ASA S12.2-2019 [i.2] does not specify NC values below NC15. A truly acceptable NC recommendation to meet the end-to-end test equipment idle noise requirements presented in clause 4.2.3 falls within this undefined region. Therefore, NC15 is recommended as an absolute upper limit.

4.2 Test equipment requirements

4.2.1 Acoustic test equipment

4.2.1.1 Overview

The acoustic test equipment setup includes two artificial ear simulators for DUT mounting and signal recording and an artificial mouth simulator for speech reproduction.

An artificial mouth simulator is required for tests which incorporate speech reproduction. For such tests, a Recommendation ITU-T P.58 [3] HATS shall be used. However, an artificial mouth simulator is not required for those tests which exclude speech reproduction. Therefore, the acoustic equipment setup for such tests may consist of only a pair of ear simulators. The requirements for standalone ear simulators are provided in clause 4.2.1.2. For all tests, the end-to-end test equipment setup shall comply with the idle noise requirements described in clause 4.2.3.

4.2.1.2 Artificial ear simulators

Eardrum recordings shall be made using a Type 4.3, Type 4.4, or Type 3.3 artificial ear defined in Recommendation ITU-T P.57 [2]. Test labs shall report which ear type is used for testing. If not specified otherwise, Diffuse-Field (DF) equalization according to Recommendation ITU-T P.58 [3], clause 8.1.3 shall be applied for all recordings made during the testing procedure.

NOTE 1: In the analysis of some tests (e.g. in clauses 5.1 and 5.2), measured spectra are referenced to another measured one and the DF-equalization cancels out. In such cases, DRP-measurements may be used directly for the calculation, without applying the DF-equalization first.

Low-noise artificial ear simulators with an inherent noise floor of < 7 dB_{SPL(A)} should be used in order to meet the end-to-end test equipment idle noise requirements presented in clause 4.2.3.

In the case that standalone artificial ear simulators are used without a mouth simulator, the ear simulators shall be mounted parallel to one-another such that both EEPs lie along the intersection of the test equipment's horizontal (reference) and transverse planes (as defined in Recommendation ITU-T P.58 [3]). Furthermore, the distance between the EEPs of the two ear simulators shall be between 130 mm and 150 mm. Lastly, the mounted ear simulators shall meet the sound leakage requirement (≥ 35 dB of attenuation) presented in clause 6.1.5 of Recommendation ITU-T P.58 [3].

NOTE 2: Recommendation ITU-T P.58 [3] requires an EEP-to-EEP distance of 130 mm to 133 mm. Although quantifiable differences in measuring over- and on-ear devices using a head breadth outside of this range are unknown, the present document allows for EEP-to-EEP distances of up to 150 mm. This is motivated by the data in Gordon et al. (2014) [i.3], which demonstrate that the Recommendation ITU-T P.58 requirement represents the 45th percentile and 5th percentile of Bizygomatic Breadths for female and male subjects, respectively [i.3]. Although this anatomical measure may slightly underestimate EEP-to-EEP distance, it provides a reasonable estimate. Thus, 130 mm to 133 mm demonstrates a short EEP-to-EEP distance compared to an average user. The extended requirement of 130 mm to 150 mm is adopted in the present document as a more representative head breadth, which may provide for more realistic device mounting.

4.2.1.3 Artificial mouth simulator

Speech playback shall be accomplished using a mouth simulator which meets the requirements established in Recommendation ITU-T P.58 [3], clause 6.2 and Recommendation ITU-T P.64 [4], clause 6. Furthermore, the mouth simulator shall meet the playback accuracy requirements in Table 4.2.1.3-1 (adapted from ETSI TS 126 132 [7], clause 5.3).

Table 4.2.1.3-1: Mouth simulator accuracy requirements

Item	Accuracy
Sound pressure	$\pm 0,7$ dB
Sound pressure level at MRP in 1/3 rd octave bands	± 3 dB for 100 Hz to 200 Hz
	± 1 dB for 200 Hz to 8 kHz
	± 3 dB for 8 kHz to 20 kHz (see note)
Mouth simulator equalization	The flatness of the mouth simulator transfer characteristics after equalization, measured in 1/3 rd octave bands with the signal used for equalization, shall be within ± 1 dB from 100 Hz to 200 Hz and shall be within $\pm 0,5$ dB above 200 Hz (see note).
NOTE: Not all mouth simulators can be successfully equalized up to 20 kHz.	

The artificial mouth is equalized at the Mouth Reference Point (MRP) according to Recommendation ITU-T P.58 [3] and the upper equalization frequency shall be reported. The validity of the equalization, especially with respect to super-wideband and fullband, shall be checked.

4.2.2 Audio interface

Unless specified otherwise, the audio interface shall meet the requirements presented in Table 4.2.2-1 (adapted from ETSI TS 126 132 [7], clause 5.3).

Table 4.2.2-1: Audio interface requirements

Item	Requirement
Signal to noise ratio	> 110 dB(A)
Channel separation	> 110 dB
Measured maximum frequency	20 kHz (see note 1)
Sampling rate	48 kHz
Electrical Signal Power	$\pm 0,2$ dB for levels ≥ -50 dBm
	$\pm 0,4$ dB for levels < -50 dBm
Time	± 5 %
Frequency	$\pm 0,2$ %
Electrical excitation levels	$\pm 0,4$ dB (see note 2)
Frequency generation	± 2 % (see note 3)
NOTE 1: The measured maximum frequency is due to limitations given by Recommendation ITU-T P.58 [3].	
NOTE 2: Across the whole frequency range.	
NOTE 3: When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ± 2 % on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2 % tolerance may be used.	

4.2.3 Idle noise

The end-to-end test system shall provide a minimal noise floor. This requirement is due to the need of accurately measuring low noise levels, which are present after ANC reduction or as part of the ANC device self-noise in quiet.

Since the idle noise captured at the artificial ears may depend on different sources (e.g. room, audio interface or ear simulator), the loudness level in phon of the whole test setup shall be evaluated in advance:

- The acoustic test equipment is placed into the test room and set into operational mode (in the absence of any DUT).
- An idle noise measurement of 10 s is conducted with the artificial ears. DF-equalization shall be applied.
- The specific loudness vs. time (in unit sone/Bark) and loudness vs. time (in unit *sone*) are calculated according to the time-varying Zwicker loudness model in ISO 532-1 [14] for the left and right channels using the diffuse sound field calculation method.
- The overall specific loudness and overall loudness are calculated using the N5 statistical method, which returns the *sone* threshold (per Bark critical band in the case of specific loudness) surpassed by 5 % of analysis frames.
- The overall specific loudness and loudness are converted to unit *phon* according to the transformation described in clause 5.3 of ISO 532-1 [14] resulting in the overall specific loudness level and overall loudness level, respectively.

Test labs shall report their end-to-end test equipment idle noise overall loudness level and overall specific loudness level for both artificial ears. The idle noise overall loudness level at each ear shall be less than 5 phon and should be as close to 0 phon as possible. Furthermore, the overall specific loudness level should be as close to 0 sone as possible for every Bark critical band.

4.2.4 Diffuse background noise generation system

A setup for simulating realistic background noises in a lab-type environment is described in ETSI TS 103 224 [6].

ETSI TS 103 224 [6] contains a description of the recording arrangement for realistic background noises, a description of the setup for a loudspeaker arrangement suitable to simulate a background noise field in a lab-type environment and a database of realistic background noises, which can be used for testing the terminal performance with a variety of different background noises.

The principal loudspeaker setup for the simulation arrangement is shown in Figure 4.2.4-1.

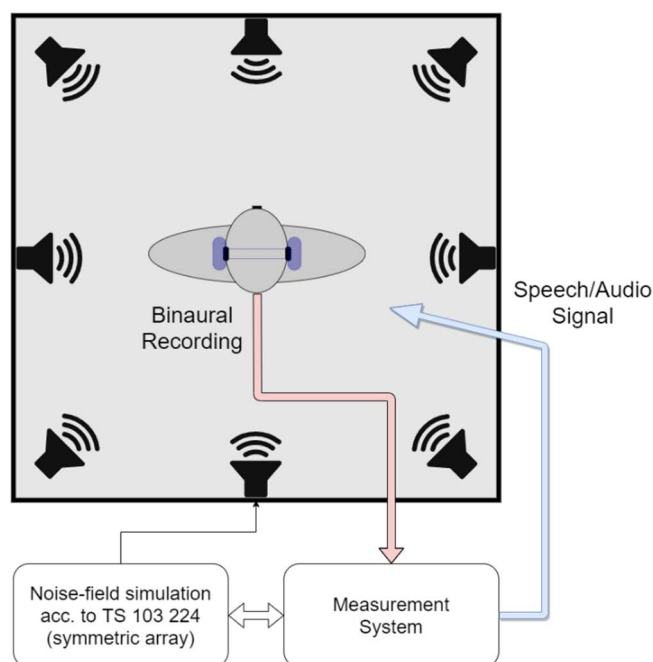


Figure 4.2.4-1: Loudspeaker arrangement for background noise simulation

The flexible equalization and calibration procedure described in clause 7 of ETSI TS 103 224 [6] shall be used in conjunction with the eight-channel symmetric microphone array for binaural applications (clause 5.4 of ETSI TS 103 224 [6]). Figure 4.2.4-2 illustrates the microphone positions used as equalization points around the acoustic test equipment.

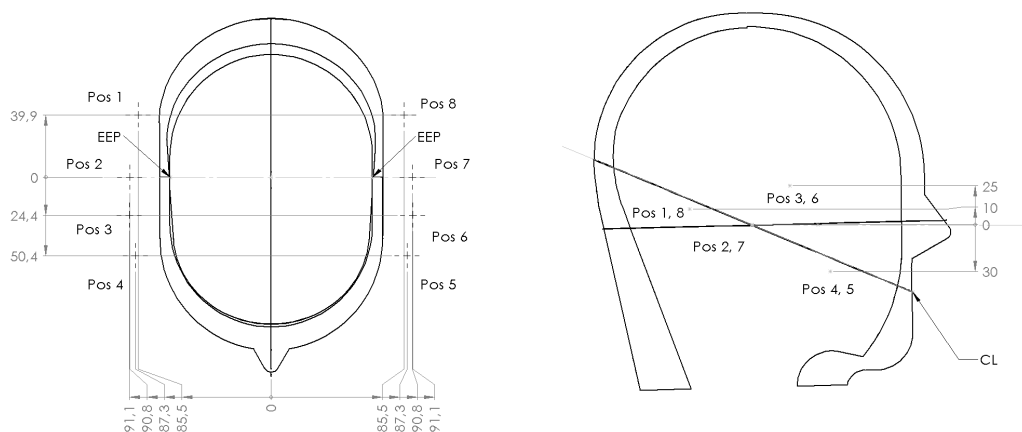


Figure 4.2.4-2: Positions of the recording microphones for binaural applications (vertical positions are related to the vertical position of the EEP)

If not stated otherwise this setup is used in all measurements where background noise playback is required. The noise types as specified in Table 4.2.4-1 shall be used for testing. The corresponding eight-channel sound files for the symmetric microphone array are provided in clause 8.3 of ETSI TS 103 224 [6]. The filename (in parentheses and italic font), length and levels per channel of the electronic attachments are provided in Table 4.2.4-1 for reference.

In addition, a time range for analysis is provided. In advance to each analysis range, a convergence time of at least 5 s shall be used, which allows the signal processing of the ANC device to adapt. If not stated otherwise, all measurements where background noise playback is required shall analyse the specified time range. In case the analysis time range exceeds the overall length of a noise, playback shall be repeated with a cross-fade of 20 ms.

Table 4.2.4-1: Noise types for ANC testing

Index	Noise Type	Description	Length (s)	Levels at microphones (dB _{SPL(A)})	Analysis Start (s)	Analysis Duration (s)
1	Diffuse-Field Pink Noise (<i>PinkNoise</i>)	HATS and microphone array at centre of a sound sphere with 25 uncorrelated pink noise sources	20	1: 80,0 dB 2: 80,0 dB 3: 80,0 dB 4: 80,0 dB 5: 80,0 dB 6: 80,0 dB 7: 80,0 dB 8: 80,0 dB	5	10
2	Railway Platform (<i>RailwayPlatform</i>)	HATS and microphone array on a railway platform	35	1: 76,6 dB 2: 76,8 dB 3: 76,7 dB 4: 76,3 dB 5: 76,3 dB 6: 76,2 dB 7: 76,4 dB 8: 76,2 dB	5	10
3	Inside Bus (<i>Inside_Bus</i>)	HATS and microphone array in passenger cabin of a bus	35	1: 66,5 dB 2: 66,6 dB 3: 66,6 dB 4: 66,3 dB 5: 66,6 dB 6: 66,2 dB 7: 66,5 dB 8: 66,6 dB	5	10
4	Crossroadnoise (<i>Crossroadnoise</i>)	HATS and microphone array standing outside near a crossroad	35	1: 68,3 dB 2: 68,6 dB 3: 68,3 dB 4: 67,7 dB 5: 66,8 dB 6: 66,4 dB 7: 66,7 dB 8: 66,7 dB	5	10

NOTE 1: Convergence time may highly variate across different types of devices. For this reason, a common minimum duration of 5 s is used.

NOTE 2: The analysis start and range values include the required minimum convergence time.

4.2.5 Directional sound source generation systems

4.2.5.1 Directional speech generation system

For the directional speech generation system, a secondary mouth simulator (standalone or included in a HATS) is placed in the vicinity of the primary test equipment. The HATS/mouth simulator shall fulfil the requirements as described in clause 4.2.1.3. For the generation of external speech signals, the secondary HATS/mouth simulator shall be positioned in front of the primary test equipment, with the mouth simulator directed towards the front of the primary test equipment. The secondary mouth simulator MRP shall be at the same height as the primary test equipment reference plane. The distance between the secondary mouth simulator MRP and the transverse plane of the primary test equipment shall be 75 cm.

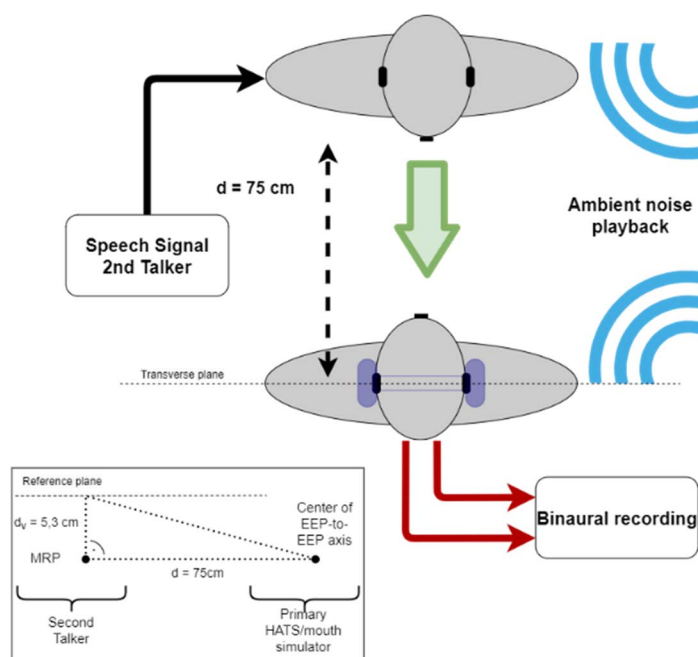


Figure 4.2.5.1-1: Setup for directional speech source

In the case of external speech generation by means of a secondary HATS/mouth simulator, the level of the test signal shall be calibrated as described in clause 4.5.1 at the MRP of the secondary HATS/mouth simulator. For measurements in the presence of BGN, the speech level is adjusted at the MRP to take into account the Lombard effect (see level adjustment calculation according to clause 4.5.1).

4.2.5.2 Directional loudspeaker system

For the directional loudspeaker system, a loudspeaker is placed with respect to the test equipment with variable horizontal distance, d_h , and vertical distance, d_v . Values for d_h and d_v are set such that the distance between the centre of the loudspeaker membrane and the centre of the primary test equipment's EEP-to-EEP axis, d , is equal to 50 cm. The loudspeaker shall be inclined or declined such that the main loudspeaker axis intersects the centre of the EEP-to-EEP axis. The loudspeaker shall be placed with a nominal azimuthal angle of incidence of 0° . Figure 4.2.5.2-1 demonstrates one acceptable orientation where $d_h = 40$ cm and $d_v = 30$ cm.

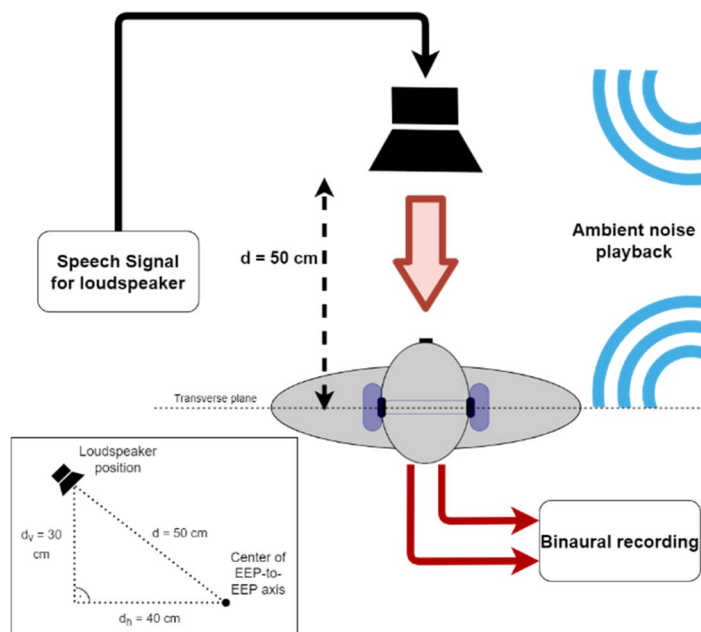


Figure 4.2.5.2-1: Setup for directional loudspeaker system

The loudspeaker system shall meet the following requirements:

- $\text{THD} \leq 0,5 \%$ when measured at 1 m on axis with an $85 \text{ dB}_{\text{SPL}}$ sinusoidal signal for frequencies ≥ 125 Hz
- Maximum long term SPL of $\geq 96 \text{ dB}_{\text{SPL}}$ when measured at 1 m on axis with the simulated programme signal defined in clause 7 of IEC 60268-1 [23]

The spectrum of the acoustic signal produced by the loudspeaker shall be equalized under free field conditions with a measurement microphone positioned on the main loudspeaker axis at a distance of $d_{\text{eq}} = 50$ cm from the loudspeaker membrane. The achieved free field equalized spectrum in $1/3^{\text{rd}}$ octave bands shall be within ± 1 dB from 100 Hz to 200 Hz and shall be within $\pm 0,5$ dB from 200 Hz to 20 kHz.

Unless otherwise specified, the level of an external speech test signal produced through the directional loudspeaker system for measurements in silence shall be calibrated to a nominal Active Speech Level (ASL) of $65 \text{ dB}_{\text{SPL}}$, calculated according to Recommendation ITU-T P.56 [9], at a distance of 50 cm from the loudspeaker membrane on the main loudspeaker axis. For measurements in the presence of background noise, the level of the test signal is additionally increased in order to provide an acceptable SNR as determined by equation 1 (inferred from clause 9.7.1 of Recommendation ITU-T P.1150 [i.1]). Similar as in clause 4.5.1, the A-weighted noise level N_A is used as an input variable here.

$$\text{ASL}(N_A) = \max(65 ; \min(46,183 + 0,404 \cdot N_A ; 85)) \text{ in } \text{dB}_{\text{SPL}} \quad (1)$$

4.2.6 Additional reference microphone

For certain measurements (e.g. clause 5.6), an additional reference microphone is required for test signal level calibration. The reference microphone shall meet the following requirements:

- Frequency range: 20 Hz – 20 kHz (± 2 dB from 125 Hz – 8 kHz)
- Dynamic range upper limit: > 130 dB_{SPL}
- Inherent noise: < 25 dB_{SPL(A)}

4.3 DUT access interfaces

4.3.1 Access interface for analog devices

This clause refers specifically to analog wired devices. There is no discussion of analog wireless devices. A general test setup configuration and access interface for an analog wired ANC device is shown in Figure 4.3.1-1.

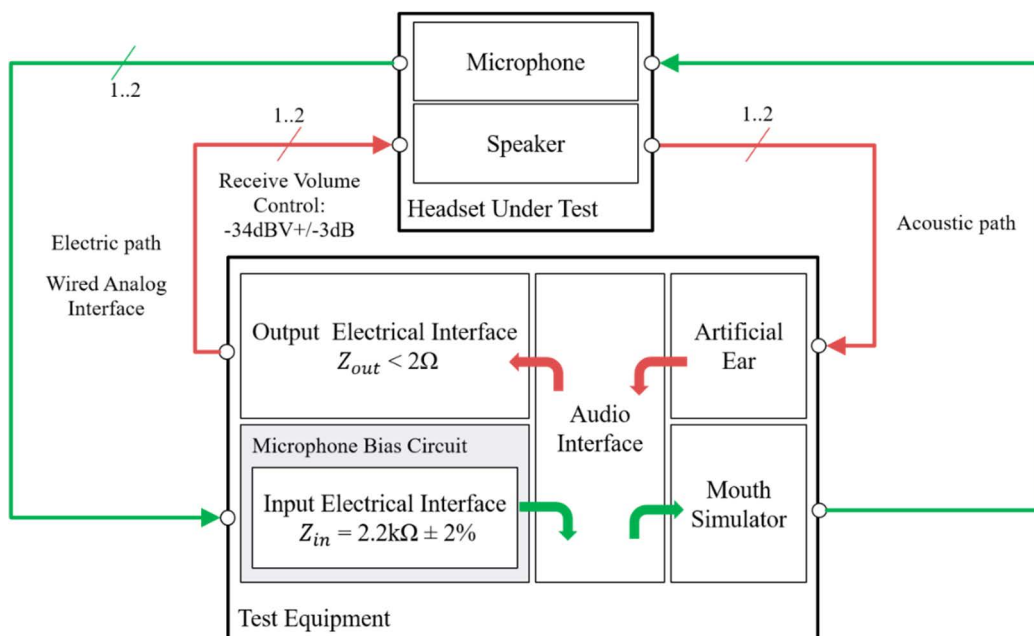


Figure 4.3.1-1: Test configuration for analog device

Specifications on input and output characteristics of the headset access interface are defined in Recommendation ITU-T P.381 [5], clause 8.1.1.1. Furthermore, details on the microphone bias circuit are provided in Figure 8-2 of the same recommendation.

4.3.2 Access interface for digital devices

The general setup configuration and access interface for a digital ANC device is described in Figure 4.3.2-1.

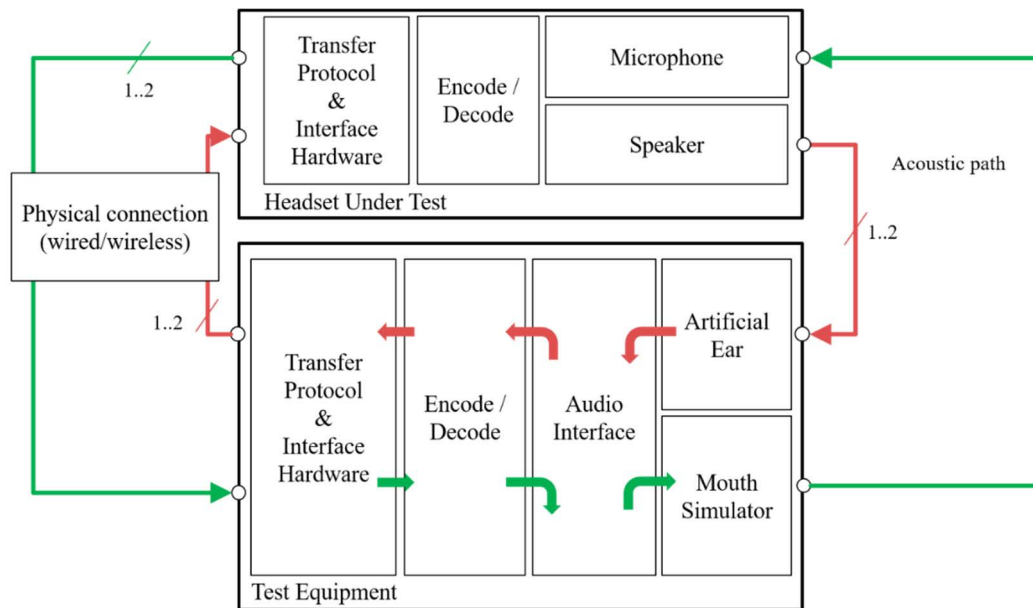


Figure 4.3.2-1: Test configuration for wireless digital device

For wireless digital devices, the short-range RF interface and wireless coding blocks are kept generic here to encompass any possible wireless connection between the access interface and the device. In some scenarios (e.g. if short-range RF interface transmits PCM audio), the signal sent and received by the test equipment and device under test may not require coding.

Voice call performance tests defined in clause 7 shall use the device access interfaces and test setups defined in the referenced recommendations.

4.4 Setup of ANC device and test equipment

The test room shall meet all requirements in clause 4.1.

The acoustic test equipment defined in clause 4.2.1 is positioned in the centre of the test room. The equalization procedure of the background noise system described in clause 4.2.4 shall be conducted at this position.

The ANC device is positioned on the left and right artificial ears, in a manner corresponding to the most typical envisioned usage for the product.

If not specified otherwise, for measurements which are to be conducted with the ANC functionality *disabled*, the ANC device shall be in a switched on (active) state, with the ANC functionality purposely disabled.

If not specified otherwise, for measurements which are to be conducted with the ANC functionality *enabled*, the amount of noise cancellation shall be set to a maximum, if such an option is provided by the ANC device. Furthermore, any additional ANC settings applied for testing shall be reported.

If not specified otherwise, for measurements which are to be conducted with pass-through functionality *enabled*, the amount of amplification shall be set to a maximum (e.g. maximum sound pressure level passed-through to the user), if such an option is provided by the ANC device. Furthermore, if not specified otherwise, all other signal processing functionality (e.g. equalization, beamforming, stationary noise rejection, etc.) shall be disabled, if such options are provided by the ANC device. Lastly, any additional pass-through settings applied for testing shall be reported.

If not specified otherwise, all measurements shall be repeated five times by removing and replacing the DUT on the artificial ears. The average and standard deviation across these repetitions shall be reported.

NOTE: The present document attempts to reduce fitting induced variability through the averaging of repeated measurements with multiple positioning attempts. To reduce inter-lab variability, test labs should be provided with a fitting strategy for a particular device.

4.5 Speech test signals

4.5.1 Overview

Several tests described in the present document require the playback of a speech signal from the mouth simulator. The following clauses provide information on these signals.

Unless otherwise specified, all test signals shall be calibrated to an active speech level according to Recommendation ITU-T P.56 [9] of:

- -4,7 dB_{Pa} at the MRP for playback via mouth simulator.
- -16 dB_{m0} at the POI for playback via digital electrical interface (assuming an overload point of 3,14 dB_{m0} according to Recommendation ITU-T G.100.1 [10]).
- -37 dB_V at the POI for playback via analog electrical interface.

For measurements in the presence of background noise which include speech reproduction, the output level of the mouth simulator is increased to account for the *Lombard effect*. The Lombard effect refers to the change in speaking behaviour caused by acoustic noise as perceived by the talker. The level is increased by 3 dB for every 10 dB that the long-term A-weighted noise level N_A exceeds 50 dB_{SPL(A)}. This relationship is shown equation 2:

$$I(N_A) = \begin{cases} 0 & \text{for } N_A < 50 \\ 0,3 \cdot (N_A - 50) & \text{for } N_A \geq 50 \geq 77 \\ 8,1 & \text{for } N_A > 77 \end{cases} \quad (2)$$

Where:

$I =$ The increase in dB of mouth output level due to noise level.

$N_A =$ The long-term A-weighted noise level measured at the talker's ears.

As an example, if the background noise measures 70 dB_{SPL(A)}, then the output of the mouth would be increased by 6 dB. No gain is applied for noise levels below 50 dB_{SPL(A)}. The maximum amount of gain that can be applied is 8,1 dB.

4.5.2 Speech sequence with long pauses

For a speech sequence with long pauses, the four FB British English double-sentences provided in Annex C of Recommendation ITU-T P.501 [8] shall be concatenated and used. The sequence is 32 s in duration and includes eight unique sentences (two male speakers and two female speakers, each with two sentences).

This signal is typically used for the evaluation of e.g. speech quality or listening effort.

4.5.3 Short speech sequence

For a short speech sequence, the two-sentence, FB British English sequence defined in Annex D (clause D.2.2) of Recommendation ITU-T P.501 [8] shall be used. The sequence is 6,5 s in duration and includes two unique sentences (one male speaker and one female speaker).

This signal is typically used for the evaluation of e.g. sidetone or external speech transmission.

4.5.4 Long speech sequence

For a long speech sequence, the uncompressed British English single-talk test sequence described in clause 7.3.2.1 of Recommendation ITU-T P.501 [8] shall be used. The sequence is 35,4 s in duration and includes twelve unique sentences (six male speakers and six female speakers).

This signal is typically used for the evaluation of e.g. sensitivity, loudness rating and frequency response in voice call scenarios.

5 Noise cancellation test methods and performance requirements

5.1 Insertion loss

5.1.1 Overview

Insertion loss is a measurement (in dB) of the reduction in background noise caused by a DUT measured at the listener's eardrum. Insertion loss may be due to the physical occlusion of a device (*passive*), algorithmic noise reduction (*active*), or a combination of the two (*total*). Measurement methods for each of these three metrics are defined in this clause.

Each insertion loss metric is calculated as a difference spectrum between two measured spectra. Therefore, diffuse-field equalization is not required when conducting the acoustic measurements for insertion loss metrics.

For all insertion loss metrics, labs shall report results with 1/3rd octave frequency resolution from 20 Hz to 20 kHz. Labs may also report results with 4096-point FFT resolution for a more detailed representation.

Along with frequency dependent insertion loss metrics, labs should report the maximum insertion loss (with corresponding frequency), frequency range of insertion loss (e.g. where loss is ≥ 6 dB), and average insertion loss over all frequencies.

NOTE: Insertion loss metrics report positive numbers when the DUT reduces the measured background noise level at the listener's eardrum. As a convention, insertion loss metrics are often plotted with a flipped y-axis such that numerical increases in insertion loss are visualized as dips in the plotted response.

5.1.2 Total insertion loss

Purpose:

The *Total Insertion Loss* quantifies the attenuation of an ambient sound field when using an ANC device with noise cancellation enabled.

Definition:

The *Total Insertion Loss* (IL_{total}) is the difference between the equivalent continuous sound levels of the noise source measured at the listener's eardrum:

- 1) without wearing the ANC device (L_{OPEN}); and
- 2) when wearing the ANC device with ANC functionality *enabled* (L_{ON}).

The IL_{total} is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and noise type n as:

$$IL_{total}(f, i, n) = (L_{OPEN}(f, i, n) - L_{ON}(f, i, n)) \text{ in dB}, \forall i \in (l, r), \forall n \in (1: N) \quad (3)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{ON}(f, i, n)$ in dB_{SPL}.
- 6) The ANC device is removed from the acoustic test equipment.

- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, in the absence of the ANC device, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{OPEN}(f, i, n)$ in dB_{SPL}.
- 10) The IL_{total} as a function of frequency f is determined for each ear i and each noise type n by equation 3.

NOTE: If the test setup and environment has not changed, the $L_{OPEN}(f, i, n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.1.3 Passive insertion loss

Purpose:

The *Passive Insertion Loss* quantifies the portion of Total Insertion Loss attributed to the physical occlusion of the user's auditory system by an ANC device (e.g. through plugging of the ear canal).

Definition:

The *Passive Insertion Loss* ($IL_{passive}$) is the difference between the equivalent continuous sound levels of the noise source measured at the listener's eardrum:

- 1) without wearing the ANC device (L_{OPEN}); and
- 2) when wearing the ANC device with ANC functionality *disabled* (L_{OFF}).

The $IL_{passive}$ is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and noise type n as:

$$IL_{passive}(f, i, n) = (L_{OPEN}(f, i, n) - L_{OFF}(f, i, n)) \text{ in dB}, \forall i \in (l, r), \forall n \in (1: N) \quad (4)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *disabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC *disabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{OFF}(f, i, n)$ in dB_{SPL}.
- 6) The ANC device is removed from the acoustic test equipment.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, in the absence of the ANC device, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{OPEN}(f, i, n)$ in dB_{SPL}.
- 10) The $IL_{passive}$ as a function of frequency f is determined for each ear i and each noise type n by equation 4.

NOTE: If the test setup and environment has not changed, the $L_{OPEN}(f, i, n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.1.4 Active insertion loss

Purpose:

The *Active Insertion Loss* quantifies the portion of Total Insertion Loss attributed to an ANC device's active noise cancellation.

Definition:

The *Active Insertion Loss* (IL_{active}) is the difference between the equivalent continuous sound levels of the noise source measured at the listener's eardrum:

- 1) when wearing the ANC device with ANC functionality *disabled* (L_{OFF}); and
- 2) when wearing the ANC device with ANC functionality *enabled* (L_{ON}).

The IL_{active} is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and noise type n as:

$$IL_{active} = (L_{OFF}(f, i, n) - L_{ON}(f, i, n)) \text{ in dB}, \forall i \in (l, r), \forall n \in (1: N) \quad (5)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ON}(f, i, n)$ in dB_{SPL}.
- 6) The ANC device functionality is *disabled*.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, with ANC *disabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{OFF}(f, i, n)$ in dB_{SPL}.
- 10) The IL_{active} as a function of frequency f is determined for each ear i and each noise type n by equation 5.

NOTE: If this test is run after the *Total* and *Passive Insertion Loss* tests for the same device, the $L_{OFF}(f, i, n)$ and $L_{ON}(f, i, n)$ measurements may be re-used from previous measurements.

5.2 Noise loudness level reduction

5.2.1 Overview

Noise loudness level reduction is a single value perceptual measurement (in phon) of the attenuation of background noise at the listener's eardrum caused by a DUT. Noise loudness level reduction may be due to the physical occlusion of a device (*passive*), algorithmic noise reduction (*active*), or a combination of the two (*total*). Measurement methods for each of these three metrics are defined in this clause.

Loudness level shall be calculated for each of the following noise loudness level reduction metrics as follows (using a diffuse-field compensated binaural recording as input):

- 1) The loudness vs. time in unit *sones* is calculated according to the time-varying Zwicker loudness model in ISO 532-1 [14] for the left and right channels using the diffuse sound field calculation method.

- 2) A single value loudness in *some* for each ear is determined using the N5 statistical method, which returns the *some* threshold surpassed by 5 % of analysis frames.
- 3) Because ISO 532-1 [14] assumes diotic sound presentation of each individual channel, the overall loudness in unit *some* is determined as the average N5 loudness between the two ears.
- 4) The loudness level in *phon* is calculated with the overall loudness according to the transformation described in clause 5.3 of ISO 532-1 [14].

Test labs shall report the overall loudness level reduction for each of the following metrics. Additionally, test labs should report loudness level reductions for the left and right ears individually.

NOTE 1: Prior to binaural summation of loudness in step 2 from above, test labs should verify that left and right ear loudness measurements are comparable. Substantial differences in loudness between ears may be due to DUT fit or measurement errors rather than DUT performance.

NOTE 2: This clause enumerates the steps to calculate *noise* loudness level according to ISO 532-1 [14]. Other clauses in the present document compute *speech* loudness level according to Recommendation ITU-T P.700 [18]. Each approach provides a perceptual measurement of loudness (in phon) suitable for noise and speech stimuli, respectively.

5.2.2 Total noise loudness level reduction

Purpose:

The *Total Noise Loudness Level Reduction* quantifies the reduction in loudness of an ambient noise field when using an ANC device with noise cancellation enabled.

Definition:

The *Total Noise Loudness Level Reduction* (LLR_{total}) is the difference between the ISO 532-1 [14] loudness level of the noise source measured at the listener's eardrum:

- 1) without wearing the ANC device ($Loud_{OPEN}$); and
- 2) when wearing the ANC device with ANC functionality *enabled* ($Loud_{ON}$).

The LLR_{total} is expressed for each noise type n as:

$$LLR_{total}(n) = (Loud_{OPEN}(n) - Loud_{ON}(n)) \text{ in phon, } \forall n \in (1:N) \quad (6)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 5) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14] to determine $Loud_{ON}$ as described in clause 5.2.1.
- 6) The ANC device is removed from the acoustic test equipment.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, in the absence of the ANC device, is recorded for the duration of noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 9) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14], to determine $Loud_{OPEN}$ as described in clause 5.2.1.
- 10) The LLR_{total} is determined for each noise type n by equation 6.

NOTE: If the test setup and environment has not changed, the $Loud_{OPEN}(n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.2.3 Passive noise loudness level reduction

Purpose:

The *Passive Noise Loudness Level Reduction* quantifies the portion of Total Noise Loudness Level Reduction attributed to the physical occlusion of the user's auditory system by an ANC device (e.g. through plugging of the ear canal).

Definition:

The *Passive Noise Loudness Level Reduction* ($LLR_{passive}$) is the difference between the ISO 532-1 [14] loudness level of the noise source measured at the listener's eardrum:

- 1) without wearing the ANC device ($Loud_{OPEN}$); and
- 2) when wearing the ANC device with ANC functionality *disabled* ($Loud_{OFF}$).

The $LLR_{passive}$ is expressed for each noise type n as:

$$LLR_{passive}(n) = (Loud_{OPEN}(n) - Loud_{OFF}(n)) \text{ in phon, } \forall n \in (1: N) \quad (7)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *disabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC disabled, is recorded for the duration of noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 5) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14] to determine $Loud_{OFF}$ as described in clause 5.2.1.
- 6) The ANC device is removed from the acoustic test equipment.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, in the absence of the ANC device, is recorded for the duration of noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 9) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14], to determine $Loud_{OPEN}$ as described in clause 5.2.1.
- 10) The $LLR_{passive}$ is determined for each noise type n by equation 7.

NOTE: If the test setup and environment has not changed, the $Loud_{OPEN}(n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.2.4 Active noise loudness level reduction

Purpose:

The *Active Noise Loudness Level Reduction* quantifies the portion of Total Noise Loudness Level Reduction attributed to an ANC device's active noise cancellation.

Definition:

The *Active Noise Loudness Level Reduction* (LLR_{active}) is the difference between the ISO 532-1 [14] loudness level of the noise source measured at the listener's eardrum:

- 1) when wearing the ANC device with ANC functionality *enabled* ($Loud_{ON}$); and

- 2) when wearing the ANC device with ANC functionality *disabled* ($Loud_{OFF}$).

The LLR_{active} is expressed for each noise type n as:

$$LLR_{active}(n) = (Loud_{OFF}(n) - Loud_{ON}(n)) \text{ in phon, } \forall n \in (1: N) \quad (8)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC enabled, is recorded for the duration of noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 5) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14] to determine $Loud_{ON}$ as described in clause 5.2.1.
- 6) The device's ANC functionality is *disabled*.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, with ANC *disabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 9) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14], to determine $Loud_{OFF}$ as described in clause 5.2.1.
- 10) The LLR_{active} is determined for each noise type n by equation 8.

NOTE: If this test is run after the *Total* and *Passive Noise Loudness Level Reduction* tests for the same device, the $Loud_{OFF}(n)$ and $Loud_{ON}(n)$ measurements may be re-used from previous measurements.

5.3 Self-noise

5.3.1 Overview

The self-noise of an ANC-enabled device is caused by amplification of the electrical system noise floor resulting from ANC processing, which becomes audible in quiet environments. DUT self-noise is characteristically low level (nearing the threshold of audibility) and therefore requires minimal idle noise in the test environment and equipment. Test setup idle noise requirements are defined in clause 4.2.3.

Because DUT self-noise may approach or fall below test equipment idle noise in certain frequency bands, test labs shall report test equipment idle noise spectrum and idle-noise loudness level alongside DUT *Self-Noise Spectrum* and *Self-Noise Loudness Level*, respectively.

Test labs shall report the *Self-Noise Spectrum* with 1/3rd octave frequency resolution from 20 Hz to 20 kHz. Labs may also report results with 4096-point FFT resolution for a more detailed representation.

Along with the frequency dependent *Self-Noise Spectrum*, labs should report the maximum self-noise and its corresponding frequency bin.

5.3.2 Self-noise spectrum

Purpose:

The *Self-Noise Spectrum* quantifies the noise generated by an ANC device with noise cancellation enabled. Such noise may be perceptually relevant when an ANC device is used in a quiet environment.

Definition:

The *Self-Noise Spectrum* (N_{self}) is the diffuse-field corrected equivalent continuous A-weighted sound level of the self-noise generated by the ANC device, measured at the listener's eardrum when wearing the ANC device with the ANC functionality *enabled*.

The N_{self} is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i as $N_{self}(f, i)$.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The noise signal at the left and right artificial ears, with ANC *enabled* and diffuse-field correction *enabled*, is recorded for 10 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ON}(f, i)$ in dB_{SPL}.
- 6) A-weighting according to IEC 61672-1 [15] shall be applied to $L_{ON}(f, i)$, resulting in the spectrum $N_{self}(f, i)$.

5.3.3 Self-noise loudness level

Purpose:

The *Self-Noise Loudness Level* quantifies the loudness of noise generated by an ANC device with noise cancellation enabled. Such noise may be perceptually relevant when an ANC device is used in a quiet environment.

Definition:

The *Self-Noise Loudness Level* ($Loud_{self}$) is the ISO 532-1 [14] loudness level of the self-noise generated by the ANC device, measured at the listener's eardrum when wearing the ANC device with the ANC functionality *enabled*.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The noise signal at the left and right artificial ears, with ANC *enabled* and diffuse-field correction *enabled*, is recorded for 10 s with a 48 kHz sampling rate.
- 5) The recorded noise signal is processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14], to determine $Loud_{self}$ as described in clause 5.2.1.

5.4 Wind noise insertion gain

5.4.1 Overview

Wind noise insertion gain is a measurement (in dB) of the increase in the acoustic noise caused by a turbulent wind flow measured at the listener's eardrum when using an ANC device. Wind noise insertion gain may be caused by the interaction of the wind flow and the physical characteristics or placement of the device (*passive*), noise cancellation functionality (*active*), or a combination of the two (*total*).

Test labs should take the considerations in Annex A into account when selecting a turbulent wind generator.

The wind generator shall be positioned such that the height of the centre of the exit port is equal to the height of the acoustic test equipment reference plane. The distance between the centre of the exit port and the centre of the acoustic test equipment EEP-to-EEP axis shall be 30 cm. Lastly, the axis running through the centre of the exit port shall pass through the centre of the acoustic test equipment EEP-to-EEP axis, creating a relative angle of incidence θ with the test equipment vertical plane. The use of an automated turntable is recommended for changing the relative angle θ between the acoustic test equipment and the wind angle of incidence. See Figure 5.4.1-1 for more details.

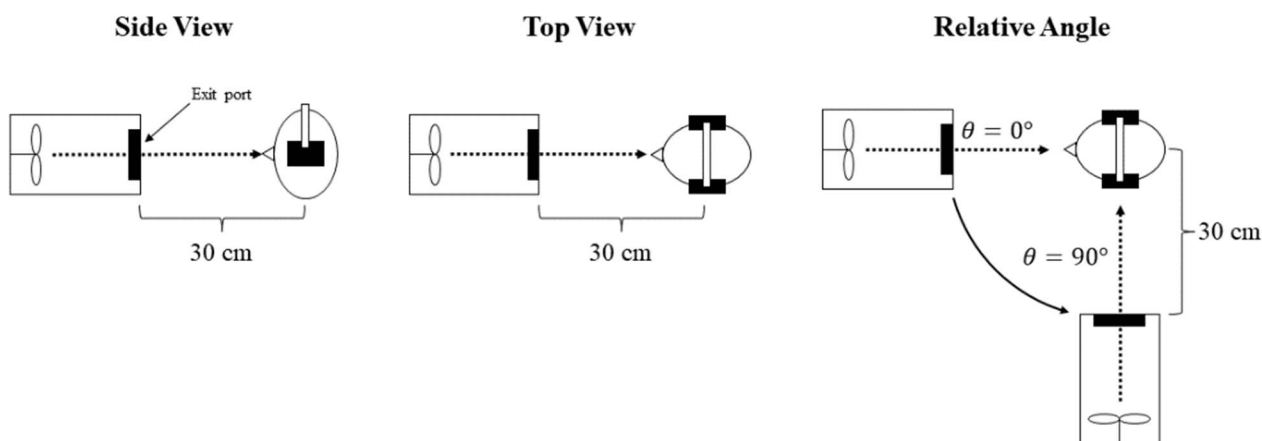


Figure 5.4.1-1: Wind generator position and relative angle

The wind generator wind speed shall be adjusted to 4,5 m/s for each measurement in clause 5.4. The wind speed shall be measured 30 cm from the centre of the wind generator with no acoustic test equipment (e.g. HATS) in place. This measurement point corresponds to the centre of the EEP-to-EEP axis when the acoustic test equipment is in place. Test labs should also test with lower (e.g. 3 m/s) and higher (e.g. 10 m/s) wind speeds. Test labs shall report all wind speeds used for testing.

For each measurement in clause 5.4, test labs shall record the wind noise for 15 seconds. A 10 second analysis window beginning 5 seconds after the start of the recording shall be used to allow for performance convergence.

NOTE: General guidance for the measurement of Wind Noise in microphones is given in IEC 60268-4 [1].

5.4.2 Total wind noise insertion gain

Purpose:

The *Total Wind Noise Insertion Gain* quantifies the amplification of wind noise when using an ANC device with noise cancellation enabled.

Definition:

The *Total Wind Noise Insertion Gain* ($IG_{WN,total}$) is the difference between the equivalent continuous sound levels generated by a turbulent wind flow measured at the listener's eardrum:

- 1) when wearing the ANC device with the ANC functionality *enabled* ($L_{WN,ON}$); and

- 2) without wearing the ANC device ($L_{WN,OPEN}$).

The $IG_{WN,total}$ is expressed as a function of the azimuthal angle of incidence θ of the turbulent wind flow and frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and a given wind speed v :

$$IG_{WN,total}(\theta, f, i) = (L_{WN,ON}(\theta, f, i) - L_{WN,OPEN}(\theta, f, i)) \text{ in dB, } \forall i \in (l, r), \forall \theta \in 0^\circ: 30^\circ: 330^\circ \quad (9)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The wind generator is positioned according to clause 5.4.1 with an initial angle of incidence of $\theta = 0^\circ$.
- 3) The wind generator is adjusted to a wind speed of $v = 4,5 \text{ m/s}$.
- 4) The device's ANC functionality is *enabled*.
- 5) The wind noise at the left and right artificial ears, with ANC *enabled*, is recorded for 15 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{WN,ON}(\theta, f, i)$ in dB_{SPL}.
- 7) The ANC device is removed from the acoustic test equipment.
- 8) The wind noise at the left and right artificial ears, in the absence of the ANC device, is recorded for 15 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{WN,OPEN}(\theta, f, i)$ in dB_{SPL}.
- 10) The $IG_{WN,total}$ as a function of frequency f is determined for each ear i and angle of wind incidence $\theta = 0^\circ$ by equation 9.
- 11) Steps 3 through 10 are repeated for counterclockwise angles of incidence from $\theta = 30^\circ$ to $\theta = 330^\circ$ in steps of 30° as depicted in Figure 5.4.1-1.

NOTE: If the test setup and environment has not changed, the $L_{WN,OPEN}(\theta, f, i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.4.3 Passive wind noise insertion gain

Purpose:

The *Passive Wind Noise Insertion Gain* quantifies the portion of Wind Noise Total Insertion Gain attributed to the physical occlusion of the user's auditory system by an ANC device (e.g. through plugging of the ear canal).

Definition:

The *Passive Wind Noise Insertion Gain* ($IG_{WN,passive}$) is the difference between the equivalent continuous sound levels generated by a turbulent wind flow measured at the listener's eardrum:

- 1) when wearing the ANC device with the ANC functionality *disabled* ($L_{WN,OFF}$); and
- 2) without wearing the ANC device ($L_{WN,OPEN}$).

The $IG_{WN,passive}$ is expressed as a function of the azimuthal angle of incidence θ of the turbulent wind flow and frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and a given wind speed v :

$$IG_{WN,passive}(\theta, f, i) = (L_{WN,OFF}(\theta, f, i) - L_{WN,OPEN}(\theta, f, i)) \text{ in dB, } \forall i \in (l, r), \forall \theta \in 0^\circ: 30^\circ: 330^\circ \quad (10)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The wind generator is positioned according to clause 5.4.1 with an initial angle of incidence of $\theta = 0^\circ$.

- 3) The wind generator is adjusted to a wind speed of $v = 4,5$ m/s.
- 4) The device's ANC functionality is *disabled*.
- 5) The wind noise at the left and right artificial ears, with ANC *disabled*, is recorded for 15 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{WN,OFF}(\theta, f, i)$ in dB_{SPL}.
- 7) The ANC device is removed from the acoustic test equipment.
- 8) The wind noise at the left and right artificial ears, in the absence of the ANC device, is recorded for 15 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{WN,OPEN}(\theta, f, i)$ in dB_{SPL}.
- 10) The $IG_{WN,passive}$ as a function of frequency f is determined for each ear i and angle of wind incidence $\theta = 0^\circ$ by equation 10.
- 11) Steps 3 through 10 are repeated for counterclockwise angles of incidence from $\theta = 30^\circ$ to $\theta = 330^\circ$ in steps of 30° as depicted in Figure 5.4.1-1.

NOTE: If the test setup and environment has not changed, the $L_{WN,OPEN}(\theta, f, i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.4.4 Active wind noise insertion gain

Purpose:

The *Active Wind Noise Insertion* quantifies the portion of *Wind Noise Total Insertion Gain* attributed an ANC device's active noise cancellation.

Definition:

The *Active Wind Noise Insertion Gain* ($IG_{WN,active}$) is the difference between the equivalent continuous sound levels generated by a turbulent wind flow measured at the listener's eardrum:

- 1) when wearing the ANC device with the ANC functionality *enabled* ($L_{WN,ON}$); and
- 2) when wearing the ANC device with the ANC functionality *disabled* ($L_{WN,OFF}$).

The $IG_{WN,active}$ is expressed as a function of the azimuthal angle of incidence θ of the turbulent wind flow and frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and a given wind speed v :

$$IG_{WN,active}(\theta, f, i) = (L_{WN,ON}(\theta, f, i) - L_{WN,OFF}(\theta, f, i)) \text{ in dB}, \forall i \in (l, r), \forall \theta \in 0^\circ: 30^\circ: 330^\circ \quad (11)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The wind generator is positioned according to clause 5.4.1 with an initial angle of incidence of $\theta = 0^\circ$.
- 3) The wind generator is adjusted to a wind speed of $v = 4,5$ m/s.
- 4) The device's ANC functionality is *enabled*.
- 5) The wind noise at the left and right artificial ears, with ANC *enabled*, is recorded for 15 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{WN,ON}(\theta, f, i)$ in dB_{SPL}.
- 7) The device's ANC functionality is *disabled*.
- 8) The wind noise at the left and right artificial ears, with ANC *disabled*, is recorded for 15 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).

- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{WN,OFF}(\theta, f, i)$ in dB_{SPL}.
- 10) The $IG_{WN,active}$ as a function of frequency f is determined for each ear i and angle of wind incidence $\theta = 0^\circ$ by equation 11.
- 11) Steps 3 through 10 are repeated for counterclockwise angles of incidence from $\theta = 30^\circ$ to $\theta = 330^\circ$ in steps of 30° as depicted in Figure 5.4.1-1.

NOTE: If this test is run after the *Wind Noise Total* and *Passive Insertion Gain* tests for the same device, the $L_{WN,OFF}(\theta, f, i)$ and $L_{WN,ON}(\theta, f, i)$ measurements may be re-used from previous measurements.

5.5 ANC Activation

5.5.1 Activation threshold level

Purpose:

The *Activation Threshold Level* quantifies the input level of an ambient noise field that triggers a device's ANC functionality. This test is only relevant for devices with automatic enabling/disabling of noise cancellation.

Definition:

The *Activation Threshold Level* (L_{thresh}) is the equivalent continuous sound level which causes the mean IL_{active} (measured according to clause 5.1.4) to be $> 0,5$ dB in the frequency range $125 \text{ Hz} \leq f \leq 1 \text{ kHz}$.

The L_{thresh} is expressed for each ear i as $L_{thresh}(i)$.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) An override gain G for the background noise generation system is initially set to a value of -60 dB.
- 3) The ANC functionality is *enabled*.
- 4) Pink Noise (first row of Table 4.2.4-1) is played back through the background noise generation system, with its level adjusted by G dB.
- 5) The noise signal at the left and right artificial ears, with ANC *enabled*, is recorded for 10 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ON}(f, i, 1)$ in dB_{SPL}.
- 7) The ANC functionality is *disabled*.
- 8) Pink Noise (first row of Table 4.2.4-1) is played back through the background noise generation system, with its level adjusted by G dB.
- 9) The noise signal at the left and right artificial ears, with ANC *disabled*, is recorded for 10 s with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 10) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{OFF}(f, i, 1)$ in dB_{SPL}.
- 11) The IL_{active} as a function of frequency f is determined for each ear i by equation 5 based on $L_{ON}(f, i, 1)$ and $L_{OFF}(f, i, 1)$.
- 12) If $\left(\frac{1}{20} \sum_{f,i} IL_{active}(f, i, 1)\right) \leq 0,5$ dB for $125 \text{ Hz} \leq f \leq 1 \text{ kHz}$ and $i \in (l, r)$, i.e. the ANC functionality is not *enabled*, the override gain G is increased by 5 dB and steps 3 to 12 are repeated.
- 13) If $\left(\frac{1}{20} \sum_{f,i} IL_{active}(f, i, 1)\right) > 0,5$ dB for $125 \text{ Hz} \leq f \leq 1 \text{ kHz}$ and $i \in (l, r)$, the measurement is stopped, and L_{thresh} is reported. L_{thresh} is the level of the Pink Noise from Table 4.2.4-1 plus the gain G .

5.5.2 Activation time

Purpose:

The *Activation Time* quantifies the time it takes for an ANC device to reach its nominal level of noise reduction after ANC is triggered. This test is only relevant for devices with automatic enabling/disabling of noise cancellation.

Definition:

The *Activation Time* ($t_{activation}$) is the time in ms required for the automatic ANC functionality to activate, following the incidence of a noise event at a level $> L_{thresh}$ as calculated in clause 5.5.1.

The $t_{activation}$ is expressed for each ear i as $t_{activation}(i)$.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The *Activation Threshold Level*, L_{thresh} , is determined according to clause 5.5.1.
- 3) The ANC functionality is *enabled*.
- 4) Pink Noise (first row of Table 4.2.4-1) is played back through the background noise generation system, at a level $L_{thresh} - 10$ dB for 10 s, immediately followed by playback at a level $L_{thresh} + 10$ dB for another 10 s. The level transition point is denoted as t_{10} in ms.
- 5) The noise signal at the left and right artificial ears, with ANC *enabled*, is synchronously recorded with the noise playback for 20 s with a 48 kHz sampling rate.
- 6) The equivalent continuous sound level over time at the left and right ears, $L_{ON}(t, i)$, is calculated using an integration time of 35 ms according to IEC 61672-1 [15].
- 7) The acoustic transmission delay from the background noise system to the left and right ear microphones is removed from the left and right ear recordings, respectively, such that $L_{ON}(0, i)$ is time aligned to the initial incidence of the Pink Noise test signal.
- 8) The $t_{activation}$ in ms is determined as the time elapsed between t_{10} and $t_{activated}$, where $t_{activated} > t_{10}$ is the time in ms at which point $L_{ON}(t)$ has reached its steady state of ANC activation. See Figure 5.5.2-1 for more detail.

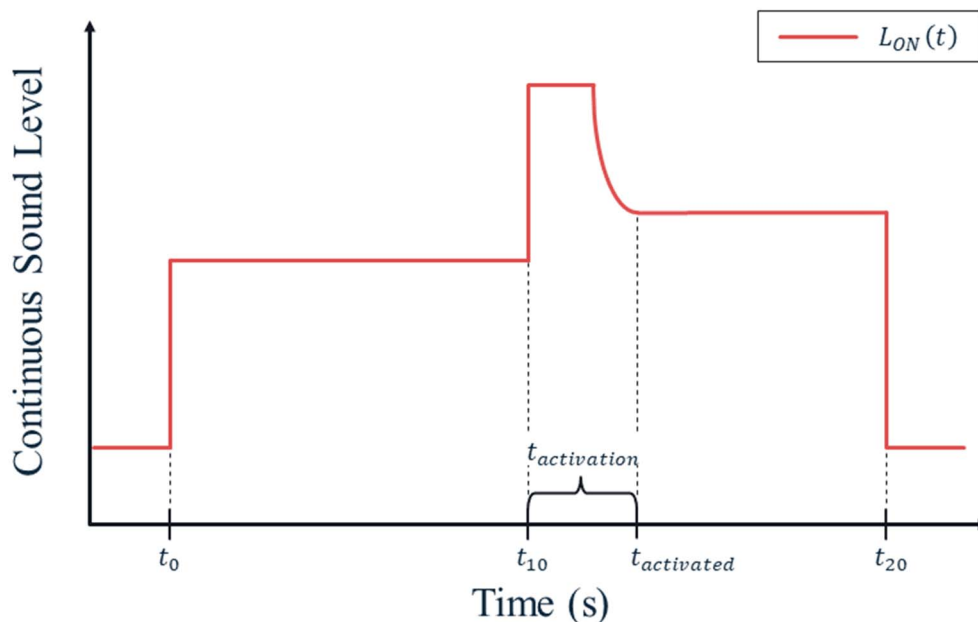


Figure 5.5.2-1: Illustration of Activation Time measurement

5.6 Acoustic overload point

Purpose:

The *Acoustic Overload Point* quantifies the point at which an external sound source overloads one or more component(s) of a device (e.g. microphone, speaker, etc.) and produces an undesirable level of distortion.

Definition:

The *Acoustic Overload Point* (expressed in dB_{SPL}), is the equivalent continuous sound level at the ANC noise microphone port entry that causes the Total Harmonic Distortion Plus Noise (THD+N) of the signal at the listener's eardrum to reach 10 %. The *Acoustic Overload Point* is assessed for discrete frequencies, f , with $125 \text{ Hz} \leq f \leq 8 \text{ kHz}$.

For this measurement, an extra loudspeaker shall be used, which meets the requirements described in clause 4.2.5.2. Furthermore, a reference microphone shall be used, which meets the requirements in clause 4.2.6.

NOTE: A reference microphone placed just outside the DUT is used for level calibration because physical access to the ANC microphone(s) is generally not available in commercial ANC devices.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The ANC functionality is *enabled*.
- 3) The extra loudspeaker shall be positioned 0,1 m from the acoustic test equipment at a 90° angle of incidence (directly to the left). The acoustic centre of the loudspeaker shall be facing the left ear simulator at the same height as the EEP.
- 4) A 1 second sine wave with a test frequency of 125 Hz is selected as the test signal, which is played back via the loudspeaker.
- 5) The reference microphone is placed along the axis running through the two ear simulators' EEPs 1 cm from the left ear DUT.
- 6) The test signal playback level is calibrated to an initial level, L , of 60 dB_{SPL} at the reference microphone.
- 7) The reference microphone is removed from the test setup.
- 8) The test signal is played back at the level L and recorded at the left ear simulator at 48 kHz sampling rate.
- 9) The THD+N of the recording is computed. If the THD+N is less than 10 %, L is increased by 5 dB and steps 8 and 9 are repeated. Otherwise, the level L is stored as $AOP(f) = L$, where f is the test frequency.
- 10) Steps 5 - 9 are repeated with test frequencies 250, 500, 1 000, 2 000, 4 000 and 8 000 Hz.
- 11) Steps 3 - 10 are repeated for the right ear.

5.7 Self-speech

5.7.1 Overview

The measurements described in clause 5.7 assess the performance of the ANC device in terms of transmission of speech sounds generated by the user of the device. The transmission of self-speech to a user's own ears is characterized by multiple paths. Recommendation ITU-T P.76 [i.4] defines four such paths, three of which are relevant to the transmission of self-speech for ANC devices:

- The bone conduction path through mechanical transmission within a user's head.
- The direct air path through acoustic transmission between a user's mouth and ear.
- The electrical path through a device's circuitry from the microphone(s) to the driver(s).

Due to the limitations of existing test equipment (e.g. HATS as defined in Recommendation ITU-T P.58 [3]), the bone conduction path of self-speech may not be reliably reproduced. Therefore, the subjective accuracy of tests described in clause 5.7 may be limited to the direct air and electrical paths of self-speech.

5.7.2 Self-speech insertion loss

Purpose:

The *Self-speech Insertion Loss* quantifies the attenuation of self-speech when using an ANC device with noise cancellation enabled.

Definition:

The *Self-speech Insertion Loss* (IL_{SS}), divided into *Passive Self-speech Insertion Loss* ($IL_{SS,passive}$) and *Total Self-speech Insertion Loss* ($IL_{SS,total}$), is the difference between the continuous sound level measured at the listener's eardrum without wearing the device, $L_{SS,OPEN}$, and the equivalent continuous sound levels measured at the listener's eardrum while wearing the ANC device with:

- 1) ANC functionality *disabled*, $L_{SS,OFF}$; and
- 2) ANC functionality *enabled*, $L_{SS,ON}$

respectively, for a test signal generated by the listener's mouth (e.g. the artificial mouth of the HATS wearing the ANC device).

The multiple IL_{SS} conditions are expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i as:

$$IL_{SS,passive}(f, i) = \left(L_{SS,OPEN}(f, i) - L_{SS,OFF}(f, i) \right) \text{ in dB}, \forall i \in (l, r) \quad (12)$$

$$IL_{SS,total}(f, i) = \left(L_{SS,OPEN}(f, i) - L_{SS,ON}(f, i) \right) \text{ in dB}, \forall i \in (l, r) \quad (13)$$

Measurement:

- 1) The test signal used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The test equipment shall be set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.
- 4) The device's ANC functionality is *disabled*.
- 5) The test signal at the left and right artificial ears, with ANC *disabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{SS,OFF}(f, i)$ in dB_{SPL}.
- 7) The device's ANC functionality is *enabled*.
- 8) The test signal at the left and right artificial ears, with ANC *enabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{SS,ON}(f, i)$ in dB_{SPL}.
- 10) The ANC device is removed from the acoustic test equipment.
- 11) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 12) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{SS,OPEN}(f, i)$ in dB_{SPL}.
- 13) The $IL_{SS,passive}$ and $IL_{SS,total}$ are determined as functions of frequency f for each ear i from equations 12 and 13, respectively.

NOTE: If the test setup and environment has not changed, the $L_{SS,OPEN}(f, i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.7.3 Self-speech loudness level reduction

Purpose:

The *Self-speech Loudness Level Reduction* quantifies the reduction in loudness of self-speech when using an ANC device with noise cancellation enabled.

Definition:

The *Self-speech Loudness Level Reduction* (LLR_{SS}), divided into *Passive Self-speech Loudness Level Reduction* ($LLR_{SS,passive}$) and *Total Self-speech Loudness Level Reduction* ($LLR_{SS,total}$), is the difference between the loudness level measured at the listener's eardrum without wearing the device ($Loud_{SS,OPEN}$), and the loudness levels measured at the listener's eardrum while wearing the ANC device with:

- 1) ANC functionality *disabled*, $Loud_{SS,OFF}$; and
- 2) ANC functionality *enabled*, $Loud_{SS,ON}$

respectively, for a test signal generated by the listener's mouth.

Due to the perceptual nature of LLR_{SS} and the lack of accurate bone conduction masking in state-of-the-art test equipment (e.g. HATS), no measurement procedure is provided until such equipment is made available.

Void (14)

Void (15)

Measurement:

Void.

5.8 External speech

5.8.1 Overview

The measurements described in clause 5.8 assess the performance of the ANC device in terms of transmission of speech sounds generated in the near end of the device. For external speech sound source generation, test labs may use either the directional speech generation system defined in clause 4.2.5.1 or the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 40$ cm and $d_v = 30$ cm. Test labs shall report which directional sound source generation system is used.

5.8.2 External speech insertion loss

Purpose:

The *External Speech Insertion Loss* quantifies the attenuation of external speech when using an ANC device with noise cancellation enabled.

Definition:

The *External Speech Insertion Loss* (IL_{ES}), divided into *Passive External Speech Insertion Loss* ($IL_{ES,passive}$) and *Total External Speech Insertion Loss* ($IL_{ES,total}$), is the difference between the continuous sound level measured at the listener's eardrum without wearing the device, $L_{ES,OPEN}$, equivalent continuous sound levels measured at the listener's eardrum while wearing the ANC device with:

- 1) ANC functionality *disabled*, $L_{ES,OFF}$; and
- 2) ANC functionality *enabled*, $L_{ES,ON}$

respectively, for a test signal generated by an external speech sound source as described in clause 4.2.5.

The multiple IL_{ES} conditions are expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i as:

$$IL_{ES,passive}(f, i) = (L_{ES,OPEN}(f, i) - L_{ES,OFF}(f, i)) \text{ in dB}, \forall i \in (l, r) \quad (16)$$

$$IL_{ES,total}(f, i) = (L_{ES,OPEN}(f, i) - L_{ES,ON}(f, i)) \text{ in dB}, \forall i \in (l, r) \quad (17)$$

Measurement:

- 1) The test signal used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The test equipment shall be set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.
- 4) The device's ANC functionality is *disabled*.
- 5) The test signal at the left and right artificial ears, with ANC *disabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ES,OFF}(f, i)$ in dB_{SPL}.
- 7) The device's ANC functionality is *enabled*.
- 8) The test signal at the left and right artificial ears, with ANC *enabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ES,ON}(f, i)$ in dB_{SPL}.
- 10) The ANC device is removed from the acoustic test equipment.
- 11) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 12) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ES,OPEN}(f, i)$ in dB_{SPL}.
- 13) The $IL_{ES,passive}$ and $IL_{ES,total}$ are determined as functions of frequency f for each ear i from equations 16 and 17, respectively.

NOTE: If the test setup and environment has not changed, the $L_{ES,OPEN}(f, i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.8.3 External speech loudness level reduction

Purpose:

The *External Speech Loudness Level Reduction* quantifies the reduction in loudness of external speech when using an ANC device with noise cancellation enabled.

Definition:

The *External Speech Loudness Level Reduction* (LLR_{ES}), divided into *Passive External Speech Loudness Level Reduction* ($LLR_{ES,passive}$) and *Total External Speech Loudness Level Reduction* ($LLR_{ES,total}$) is the difference between the loudness level measured at the listener's eardrum without wearing the device ($Loud_{ES,OPEN}$), and the loudness level measured at the listener's eardrum while wearing the ANC device with:

- 1) ANC functionality *disabled*, $Loud_{ES,OFF}$; and
- 2) ANC functionality *enabled*, $Loud_{ES,ON}$

respectively, for a test signal generated by an external speech sound source as described in clause 4.2.5. The loudness level is measured according to Recommendation ITU-T P.700 [18].

The multiple LLR_{ES} conditions are expressed for each ear i as:

$$LLR_{ES,passive} = (Loud_{ES,OPEN} - Loud_{ES,OFF}) \text{ in phon} \quad (18)$$

$$LLR_{ES,total} = (Loud_{ES,OPEN} - Loud_{ES,ON}) \text{ in phon} \quad (19)$$

Measurement:

- 1) The test signal used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The test equipment shall be set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.
- 4) The device's ANC functionality is *disabled*.
- 5) The test signal at the left and right artificial ears, with ANC *disabled*, is recorded with a 48 kHz sampling rate and the loudness level is calculated according to Recommendation ITU-T P.700 [18] to determine $Loud_{ES,OFF}$ in phon.
- 6) The device's ANC functionality is *enabled*.
- 7) The test signal at the left and right artificial ears, with ANC *enabled*, is recorded with a 48 kHz sampling rate and the loudness level is calculated according to Recommendation ITU-T P.700 [18] to determine $Loud_{ES,ON}$ in phon.
- 8) The ANC device is removed from the acoustic test equipment.
- 9) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate and the loudness level is calculated according to Recommendation ITU-T P.700 [18] to determine $Loud_{ES,OPEN}$ in phon.
- 10) The $LLR_{ES,passive}$ and $LLR_{ES,total}$ are determined for each ear i from equations 18 and 19, respectively.

NOTE: If the test setup and environment has not changed, the $Loud_{ES,OPEN}$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.8.4 External speech listening effort change

Purpose:

The *External Speech Listening Effort Change* quantifies the change in a user's listening effort when attending to an external speech source when using an ANC enabled device in a noisy environment.

Definition:

The *External Speech Listening Effort Change* (ΔLE_{ES}), divided into *Passive External Speech Listening Effort Change* ($\Delta LE_{ES,passive}$) and *Total External Speech Listening Effort Change* ($\Delta LE_{ES,total}$) is the difference between the listening effort measured at the listener's eardrum without wearing the device ($MOS-LEO_{ES,OPEN}$) and the listening effort measured while wearing the ANC device with:

- 1) ANC functionality *disabled*, $MOS-LEO_{ES,OFF}$; and
- 2) ANC functionality *enabled*, $MOS-LEO_{ES,ON}$

respectively, for a test signal generated by a directional sound source generation system as described in clause 4.2.5. The listening effort mean opinion score, $MOS-LEO$, is calculated as described in ETSI TS 103 558 [19] without a noise-only reference signal.

The multiple ΔLE_{ES} conditions are expressed for each noise type n from Table 4.2.4-1 as:

$$\Delta LE_{ES,passive}(n) = (MOS-LEO_{ES,OFF}(n) - MOS-LEO_{ES,OPEN}(n)), \forall n \in (1:N) \quad (20)$$

$$\Delta LE_{ES,total}(n) = (MOS-LEO_{ES,ON}(n) - MOS-LEO_{ES,OPEN}(n)), \forall n \in (1:N) \quad (21)$$

Measurement:

- 1) The test signal used for the measurements shall be the speech sequence with long pauses described in clause 4.5.2.
- 2) The test equipment shall be set up according to clause 4.4.
- 3) The device's ANC functionality is *disabled*.
- 4) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back through the directional generation system.
- 5) The test and noise signals at the left and right artificial ears, with ANC *disabled*, are recorded for the duration of the test signal with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{ES,OFF}(n)$.
- 6) The device's ANC functionality is *enabled*.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back through the directional generation system.
- 8) The test and noise signals at the left and right artificial ears, with ANC *enabled*, is recorded for the duration of the test signal with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{ES,ON}(n)$.
- 9) The ANC device is removed from the acoustic test equipment.
- 10) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back through the directional generation system.
- 11) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded for the duration of noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{ES,OPEN}(n)$.
- 12) The $\Delta LE_{ES,passive}$ and $\Delta LE_{ES,total}$ are determined for each noise type n from equations 20 and 21, respectively.

NOTE 1: Test labs should consider the limitations of prediction accuracy when interpreting the results of perceptual models. The current $MOS-LEO$ prediction accuracy for ANC device use cases is provided in clause D.5.2 of ETSI TS 103 558 [19].

NOTE 2: If the test setup and environment has not changed, the $MOS-LEO_{ES,OPEN}(n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

5.9 Downlink speech intelligibility index improvement

Purpose:

The *Downlink Speech Intelligibility Index Improvement* quantifies the improvement in downlink speech intelligibility when using an ANC device with noise cancellation enabled in a noisy environment.

Definition:

The *Downlink Speech Intelligibility Index Improvement* (ΔSII) is the difference between the Speech Intelligibility Index - as measured according to ANSI/ASA S3.5-1997 [21] - of a simulated downlink speech signal, when measured with the ANC algorithm enabled and disabled.

The ΔSII is expressed for each ear i and noise type n as:

$$\Delta SII(i, n) = (SII_{ON}(i, n) - SII_{OFF}(i, n)), \forall i \in (l, r), \forall n \in (1: N) \quad (22)$$

NOTE 1: This KPI is typically only relevant for handset ANC where the SNR with ANC disabled may be low enough to impact intelligibility. It may also be applicable for certain non-occluded ANC device designs.

NOTE 2: The Speech Intelligibility Index is calculated according to ANSI/ASA S3.5-1997 [21]. More information, as well as MATLAB and C code is available at <http://sii.to/programs.html>.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The ANC functionality is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with ANC *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{ON}(f)$.
- 6) The *Speech Intelligibility Index* with ANC *enabled*, SII_{ON} , is calculated according to ANSI/ASA S3.5-1997 [21] considering: (1) A long-term average speech spectrum (at normal level) according to Table 3 of ANSI/ASA S3.5-1997 [21], and (2) The noise spectrum given by $L_{ON}(f)$.
- 7) The ANC functionality is *disabled*.
- 8) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 9) The noise signal at the left and right artificial ears, with ANC *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 10) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{OFF}(f)$.
- 11) The *Speech Intelligibility Index* with ANC *disabled*, SII_{OFF} , is calculated according to ANSI/ASA S3.5-1997 [21] considering: (1) A long-term average speech spectrum (at normal level) according to Table 3 of ANSI/ASA S3.5-1997 [21], and (2) The noise spectrum given by $L_{OFF}(f)$.
- 12) The ΔSII is determined for each ear i and noise type n by equation 22.

5.10 Informative measurements

5.10.1 Acoustic leak robustness

Purpose:

The *Acoustic Leak Robustness* quantifies the impact on noise cancellation performance of an ANC device, for scenarios in which the acoustic leak is increased compared to the device's optimal fit on the user. This impairment may be caused by physical objects located between the device and the ears, such as hair/facial hair or eyeglasses/sunglasses, or an improper fit of the device on the user.

Definition:

The *Acoustic Leak Robustness* (ALR_x) is the result difference of a measured attribute x (e.g. total insertion loss, passive insertion loss, active insertion loss, total noise loudness reduction, etc.) between:

- 1) A measurement with the ANC device set up according to clause 4.4; and
- 2) A measurement with the ANC device set up according to clause 4.4, which is additionally and intentionally impaired by an acoustic leak.

The multiple ALR_x conditions should be calculated for the various measured attributes or quantities x described in clauses 5.1 to 5.9, and are expressed as a difference (Δx) according to:

$$ALR_x = (x_{impaired} - x_{normal}) \quad (23)$$

where the various attributes or quantities x may or may not be functions of the form $x(f, i, n)$, $x(f, i)$, $x(i, n)$, or $x(i)$, for each frequency f , with $20\text{Hz} \leq f \leq 20\text{kHz}$, each ear $i \in (l, r)$ and each noise type n from Table 4.2.4-1.

The resulting measurement unit of ALR_x is the measurement unit of the attribute or quantity x under test.

Measurement:

- 1) The ANC device is set up according to clause 4.4.
- 2) A given measurement is conducted as described in clause 5.1 to 5.9 and the result is reported as x_{normal} .
- 3) The ANC device is mounted on the ear simulators with an intentional impairment on the earphones' leak.
- 4) The measurement is repeated one more time with the acoustic leak impairment and the result is reported as $x_{impaired}$.
- 5) The Acoustic Leak Robustness ALR_x of the given measured attribute or quantity x is determined from equation 23.

NOTE: At the time of publication of the present document, no ear simulator with a defined earphone leak control was available for this test. As long as such equipment is under study, a pair of eyeglasses/sunglasses mounted between the artificial ears and the ANC device may be used for this measurement. Similarly, the acoustic seal of in-ear devices may be impaired through purposeful placement of the device in a non-occluded fashion.

6 Media playback test methods and performance requirements

6.1 Level range

Purpose:

The *Audio Playback Level Range* quantifies the impact, if any, of an ANC device's noise cancellation functionality on the output level range of playback audio.

Definition:

The *Audio Playback Level Range (APLR)* is the difference between the equivalent continuous sound levels over the frequency range of 20 Hz to 20 kHz, measured at the listener's eardrum while wearing the ANC device with the playback volume of the ANC device set to:

- 1) the maximum (highest) volume setting, APL_{max} ; and
- 2) the minimum (lowest) volume setting, APL_{min}

for a test signal played back via the ANC device. *APLR* is determined with ANC functionality enabled ($APLR_{ON}$), and disabled ($APLR_{OFF}$) as the average of left and right ear:

$$APLR_{OFF,AVG}(f) = \frac{1}{2}(APLR_{OFF}(f, l) + APLR_{OFF}(f, r)) \text{ in dB} \quad (24)$$

$$APLR_{ON,AVG}(f) = \frac{1}{2}(APLR_{ON}(f, l) + APLR_{ON}(f, r)) \text{ in dB} \quad (25)$$

The multiple *APLR* conditions are expressed for each ear i as:

$$APLR_{OFF}(f, i) = (APL_{max,OFF}(f, i) - APL_{min,OFF}(f, i)) \text{ in dB}, \forall i \in (l, r) \quad (26)$$

$$APLR_{ON}(f, i) = (APL_{max,ON}(f, i) - APL_{min,ON}(f, i)) \text{ in dB}, \forall i \in (l, r) \quad (27)$$

Measurement:

- 1) The test signal used for the measurements shall be programme simulation noise as defined in EN 50332-1 [17].
- 2) The ANC device is set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.

- 4) The device's ANC functionality is *disabled*.
- 5) The device's playback volume is set to the minimal (lowest) setting.
- 6) The test signal at the left and right artificial ears, with ANC *disabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 7) This FFT representation is used to calculate a 1/3rd octave resolution spectrum in $APL_{min,OFF}(f, i)$.
- 8) The device's playback volume is set to the maximal (highest) setting.
- 9) The test signal at the left and right artificial ears, with ANC *disabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 10) This FFT representation is used to calculate a 1/3rd octave resolution spectrum in $APL_{max,OFF}(f, i)$.
- 11) The *Audio Playback Level Range* $APLR_{OFF}(f, i)$ is determined from equation 26.
- 12) The device's ANC functionality is *enabled*.
- 13) Steps 5 to 11 are repeated with ANC functionality *enabled* in order to determine the corresponding playback levels $APL_{min,ON}(f, i)$ and $APL_{max,ON}(f, i)$, as well as the $APLR_{ON}(f, i)$ as given by equations 26 and 27.
- 14) The left and right ear average results are calculated for each of the *APLR* conditions as $APLR_{OFF,AVG}(f)$ and $APLR_{ON,AVG}(f)$ as given by equations 24 and 25.

6.2 Frequency response deviation

Purpose:

The *Audio Playback Frequency Response Deviation* quantifies the impact, if any, of an ANC device's noise cancellation functionality on the output frequency response of playback audio.

Definition:

The *Audio Playback Frequency Response Deviation* (ΔFR_{AP}) is the difference (if any) between the audio playback frequency response measured while wearing the ANC device with:

- 1) the ANC functionality *enabled* ($FR_{AP,ON}$); and
- 2) the ANC functionality *disabled* ($FR_{AP,OFF}$)

for a test signal played back via the ANC device at the same playback volume setting.

Each frequency response $FR_{AP,ON}$ and $FR_{AP,OFF}$ is determined separately for each earphone for frequencies from 20 Hz to 20 kHz, inclusive, by measuring the equivalent continuous sound level at the listener's eardrum and referring it to the equivalent continuous sound level of the test signal, averaged over the complete test sequence length.

The ΔFR_{AP} is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i as:

$$\Delta FR_{AP}(f, i) = \left(FR_{AP,ON}(f, i) - FR_{AP,OFF}(f, i) \right) \text{ in dB, } \forall i \in (l, r) \quad (28)$$

Measurement:

- 1) The test signal used for the measurements shall be programme simulation noise as defined in EN 50332-1 [17].
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.
- 4) The device's ANC functionality is *enabled*.
- 5) The test signal is recorded with ANC functionality *enabled* at the left and right ears of the acoustic test equipment with a 48 kHz sample rate and the audio playback frequency response, $FR_{AP,ON}(f, i)$, is determined separately for each ear i as a function of frequency f .

- 6) The device's ANC functionality is *disabled*.
- 7) The test signal is recorded with ANC functionality *disabled* at the left and right ears of the acoustic test equipment with a 48 kHz sample rate and the audio playback frequency response, $FR_{AP,OFF}(f, i)$, is determined separately for each ear i as a function of frequency f .
- 8) The $\Delta FR_{AP}(f, i)$, is determined as a function of frequency f for each ear i by equation 28.

6.3 Loudness level change

Purpose:

The *Audio Playback Loudness Level Change* quantifies the impact, if any, of an ANC device's noise cancellation functionality on the output loudness of playback audio.

Definition:

The *Audio Playback Loudness Level Change* ($\Delta Loud_{AP}$) is the change (if any) in the loudness level measured at the listener's eardrum while wearing the ANC device with:

- 1) ANC functionality *enabled*, $Loud_{AP,ON}$; and
- 2) ANC functionality *disabled*, $Loud_{AP,OFF}$

for a test signal played back via the ANC device at the same playback volume setting. The loudness level is calculated according to ISO 532-1 [14].

The $\Delta Loud_{AP}$ is expressed as:

$$\Delta Loud_{AP} = (Loud_{AP,ON} - Loud_{AP,OFF}) \text{ in phon} \quad (29)$$

Measurement:

- 1) The test signal used for the measurements shall be programme simulation noise as defined in EN 50332-1 [17].
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.
- 4) The device's ANC functionality is *enabled*.
- 5) The test signal is recorded with ANC functionality *enabled* at the left and right ears of the acoustic test equipment with a 48 kHz sampling rate. The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14] to determine $Loud_{AP,ON}$ as described in clause 5.2.1.
- 6) The device's ANC functionality is *disabled*.
- 7) The test signal is recorded with ANC functionality *disabled* at the left and right ears of the acoustic test equipment with a 48 kHz sampling rate. The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14] to determine $Loud_{AP,OFF}$ as described in clause 5.2.1.
- 8) The $\Delta Loud_{AP}$, is determined by equation 29.
- 9) The left and right ear average result is calculated as $\Delta Loud_{AP,AVG}$.

6.4 Audio playback listening effort change

Purpose:

The *Audio Playback Listening Effort Change* quantifies the change in a user's listening effort when attending to playback audio when using an ANC device with noise cancellation enabled in a noisy environment.

Definition:

The change in *Audio Playback Listening Effort Change* (ΔLE_{AP}), is the change (if any) in listening effort measured at the listener's eardrum while wearing the ANC device with:

- 1) ANC functionality *enabled* ($MOS-LEO_{AP,ON}$); and
- 2) ANC functionality *disabled* ($MOS-LEO_{AP,OFF}$).

The listening effort mean opinion score, *MOS-LEO*, is calculated as described in ETSI TS 103 558 [19] without a noise-only reference signal.

The ΔLE_{AP} is expressed for each noise type n as:

$$\Delta LE_{AP}(n) = \left(MOS-LEO_{AP,ON}(n) - MOS-LEO_{AP,OFF}(n) \right), \forall n \in (1:N) \quad (30)$$

Measurement:

- 1) The test signal used for the measurements shall be the speech sequence with long pauses described in clause 4.5.2.
- 2) The test equipment shall be set up according to clause 4.4.
- 3) The device's ANC functionality is *disabled*.
- 4) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back via the left and right side of the DUT access interface.
- 5) The test and noise signals at the left and right artificial ears, with ANC *disabled*, are recorded for the duration of the test signal with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{AP,OFF}(n)$.
- 6) The device's ANC functionality is *enabled*.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back via the left and right side of the DUT access interface.
- 8) The test and noise signals at the left and right artificial ears, with ANC *enabled*, are recorded for the duration of the test signal with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{AP,ON}(n)$.
- 9) The ΔLE_{AP} is determined for each noise type n from equation 30.

NOTE: Test labs should consider the limitations of prediction accuracy when interpreting the results of perceptual models. The current *MOS-LEO* prediction accuracy for ANC device use cases is provided in clause D.5.2 of ETSI TS 103 558 [19].

7 Voice call test methods and performance requirements

7.1 Overview

In addition to ambient noise cancelation and media playback use cases, ANC devices also often provide voice call functionality (typically realized via analog or digital connection to another communication device). Clause 7 describes the test methods and performance requirements for voice call functionality with an ANC device, many of which are available in other ETSI specifications or Recommendations from ITU-T. Since the ANC functionality is often closely coupled to (or even share) the speech signal processing used for telephony, the test methods are intended to identify any possible undesired interaction between these two subsystems, including electro-acoustical interfaces (microphones, loudspeakers) of the device. In most cases, these tests assume that the speech communication performance is not degraded by enabled ANC functionality.

Beyond the cited methods and requirements, there are additional requirements on the voice call behaviour with both ANC functionality enabled and disabled.

If not specified otherwise, all tests described herein shall be conducted with the device's ANC functionality *enabled*. Furthermore, all requirements in clause 7 shall be met with the device's ANC functionality *disabled*.

7.2 Tests only for analog interface

7.2.1 Sensitivity

7.2.1.1 Sensitivity in send

Purpose:

The *Sensitivity in Send* evaluates if the signal processing of the enabled ANC functionality applies an undesired level change in send.

Definition:

The *Sensitivity in Send* direction is defined according to clause 8.1.2 of Recommendation ITU-T P.381 [5] and is measured from the MRP to the sending input of the device access interface.

Measurement:

The *Sensitivity in Send* measurements is relevant only for analog (wired) devices, connected to the test equipment as described in clause 4.3.1:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in clause 8.1.3 of Recommendation ITU-T P.381 [5].

Requirement:

The limits for the *Sensitivity in Send* direction shall comply with the requirements according to clause 8.1.3 of Recommendation ITU-T P.381 [5].

7.2.1.2 Sensitivity in receive

Purpose:

The *Sensitivity in Receive* evaluates if the signal processing of the enabled ANC functionality has any influence on the frequency response in receive.

Definition:

The *Sensitivity in Receive* direction is defined according to clause 8.1.3 of Recommendation ITU-T P.381 [5] as the diffuse-field equalized sound pressure at the listener's eardrum, and is measured from the receiving output of the device access interface to the to the drum reference position (DRP) with diffuse-field correction.

Measurement:

The *Sensitivity in Receive* measurements are relevant only for analog (wired) devices, connected to the test equipment as described in clause 4.3.1. For testing ANC devices, the sensitivity in receive direction is divided further into S_{ON} , measured with the device's ANC functionality enabled, and S_{OFF} , measured with the device's ANC functionality disabled:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in clause 8.1.2 of Recommendation ITU-T P.381 [5], in order to obtain the sensitivity in receive, S_{ON} .
- 5) The device's ANC functionality is *disabled*.
- 6) The test is repeated according to the measurement method as described in step 4, but with the ANC functionality disabled, in order to obtain the *Sensitivity in Receive*, S_{OFF} .

Requirement:

The limits for the *Sensitivity in Receive* direction for the analog ANC device with ANC functionality both enabled and disabled shall comply with the requirements according to clause 8.1.3 of Recommendation ITU-T P.381 [5].

7.3 Tests only for digital interface

7.3.1 Loudness rating

7.3.1.1 Send loudness rating

Purpose:

The *Send Loudness Rating* evaluates if the signal processing of the enabled ANC functionality applies an undesired level change in send.

Definition:

The *Send Loudness Rating (SLR)* is defined, depending on the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.3.1.1-1.

Table 7.3.1.1-1: References to send loudness rating measurements

Transmission Bandwidth	Specification	Clause
NB	ETSI TS 103 737 [11]	6.2.1
WB	ETSI TS 103 739 [12]	6.2.1
SWB, FB	ETSI TS 102 924 [13]	6.3.2

Measurement:

The *SLR* measurements are relevant only for digital (wired and wireless) devices, connected to the test equipment as described in clauses 4.3.1 and 4.3.2, respectively.

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the corresponding specification listed in Table 7.3.1.1-1, depending on the provided bandwidth of the ANC device.

Requirement:

The *SLR* limits shall comply with the requirements according to the corresponding specification from Table 7.3.1.1-1, depending on the provided bandwidth of the ANC device.

7.3.1.2 Receive loudness rating

Purpose:

The *Receive Loudness Rating* evaluates if the signal processing of the enabled ANC functionality applies an undesired level change in receive.

Definition:

The *Receive Loudness Rating (RLR)* is defined, depending on the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.3.1.2-1.

Table 7.3.1.2-1: References to receive loudness rating measurements

Transmission Bandwidth	Specification	Clause
NB	ETSI TS 103 737 [11]	6.2.3 binaural Headset
WB	ETSI TS 103 739 [12]	6.2.3 binaural Headset
SWB, FB	ETSI TS 102 924 [13]	6.4.4

Measurement:

The *RLR* measurements are relevant only for digital (wired and wireless) ANC devices, connected to the test equipment as described in clauses 4.3.1 and 4.3.2, respectively. The *RLR* is adjustable via the volume control of the ANC device. For testing ANC devices, the *RLR* is divided further into RLR_{ON} , measured with the device's ANC functionality enabled, and RLR_{OFF} , measured with the device's ANC functionality disabled.

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.3.1.2-1, depending on the provided bandwidth of the ANC device. The volume setting of the ANC device is adjusted in order to achieve a RLR_{ON} , within the limits of the nominal RLR value.
- 5) The device's ANC functionality is *disabled*.
- 6) The test is repeated according to the measurement method described in the corresponding specification listed in Table 7.3.1.2-1 depending on the provided bandwidth of the ANC device, at the same volume setting of the ANC device as adjusted in step 4, in order to obtain the RLR_{OFF} .

Requirement:

The limits for the nominal *Receive Loudness Rating* shall comply with the requirements according to the corresponding specification from Table 7.3.1.2-1 depending on the provided bandwidth of the ANC device, for both measured values of *Receive Loudness Rating*, RLR_{ON} and RLR_{OFF} , with ANC functionality enabled and disabled, respectively.

7.3.2 Sidetone performance

7.3.2.1 SideTone Masking Rating (STMR)

Purpose:

The *Sidetone Masking Rating* evaluates if the signal processing of the enabled ANC functionality has any undesired impact on the perceived self-speech when using the device.

Definition:

The *Sidetone Masking Rating (STMR)* is defined, depending on the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.3.2.1-1.

Table 7.3.2.1-1: References to sidetone masking rating measurements

Transmission Bandwidth	Specification	Clause
NB	ETSI TS 103 737 [11]	6.3.1
WB	ETSI TS 103 739 [12]	6.3.2
SWB, FB	ETSI TS 102 924 [13]	6.3.7

Measurement:

For testing ANC devices, the *STMR* is divided further into $STMR_{ON}$, measured with the device's ANC functionality *enabled*, and $STMR_{OFF}$, measured with the device's ANC functionality *disabled*.

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.3.2.1-1, depending on the provided bandwidth of the ANC device, in order to obtain the $STMR_{ON}$.
- 5) The device's ANC functionality is *disabled*.
- 6) The test is repeated according to the measurement method described in the corresponding specification listed in Table 7.3.2.1-1 depending on the provided bandwidth of the ANC device, in order to obtain the $STMR_{OFF}$.

Requirement:

The limits for the *STMR* shall comply with the requirements according to the corresponding specification from Table 7.3.2.1-1 depending on the provided bandwidth of the ANC device, for both measured values of *Sidetone Masking Rating*, $STMR_{ON}$ and $STMR_{OFF}$, with ANC functionality enabled and disabled, respectively.

7.3.3 Quality of echo cancellation

7.3.3.1 Terminal coupling loss

Purpose:

The *Terminal Coupling Loss* evaluates if the signal processing of the enabled ANC functionality has any undesired impact on the echo cancellation performance.

Definition:

The *Terminal Coupling Loss (TCL)* and weighted *TCL (TCL_w)*, for NB transmission only) are defined, depending on the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.3.3.1-1.

Table 7.3.3.1-1: References to terminal coupling loss measurements

Transmission Bandwidth	Specification	Clause
NB	ETSI TS 103 737 [11]	6.6.2
WB	ETSI TS 103 739 [12]	6.7
SWB, FB	ETSI TS 102 924 [13]	6.3.9

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.3.3.1-1, depending on the provided bandwidth of the ANC device, in order to obtain the *TCL*.

Requirement:

TCL and *TCL_w* shall fulfil the requirements according to the corresponding specification from Table 7.3.3.1-1 depending on the provided bandwidth of the ANC device.

7.3.4 Double talk performance

7.3.4.1 Attenuation range in send direction during double talk

Purpose:

The *Attenuation Range in Send Direction During Double Talk* evaluates if the signal processing of the enabled ANC functionality has any undesired impact on the double talk performance.

Definition:

The *Attenuation Range in Send Direction During Double Talk (A_{H,S,dt})* is defined, depending on the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.3.4.1-1.

Table 7.3.4.1-1: References to sidetone masking rating measurements

Transmission Bandwidth	Specification	Clause
NB	ETSI TS 103 737 [11]	6.7.1
WB	ETSI TS 103 739 [12]	6.8.2
SWB, FB	ETSI TS 102 924 [13]	6.4.10.2

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.3.4.1-1, depending on the provided bandwidth of the ANC device, in order to obtain the $A_{H,S,dt}$.

Requirement:

The $A_{H,S,dt}$ shall fulfil the requirements according to the corresponding specification from Table 7.3.4.1-1 depending on the provided bandwidth of the ANC device.

7.3.5 Background noise performance

7.3.5.1 Speech quality in presence of background noise

Purpose:

The *Speech Quality in Presence of Background Noise* evaluates if the signal processing of the enabled ANC functionality has any undesired impact on the noise suppression performance in send.

Definition:

The *Speech Quality in the Presence of Background Noise* is defined, depending on the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.3.5.1-1.

Table 7.3.5.1-1: References to Speech quality in the presence of background noise measurements

Transmission Bandwidth	Specification	Clause
NB	ETSI TS 103 737 [11]	6.9.2
WB	ETSI TS 103 739 [12]	6.10.2
SWB, FB	ETSI TS 102 924 [13]	6.4.12.2

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's ANC functionality is *enabled*.
- 3) Noise 4 from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.3.5.1-1, depending on the provided bandwidth of the ANC device, but only for the background noise scenario as described in step 2.

Requirement:

The *Speech Quality in the Presence of Background Noise* shall fulfil the requirements according to the corresponding specification from Table 7.3.5.1-1 depending on the provided bandwidth of the ANC device.

7.4 Tests for analog and digital interface

7.4.1 Frequency response

7.4.1.1 Send Frequency Response

Purpose:

The *Send Frequency Response* evaluates if the signal processing of the enabled ANC functionality has any impact on the spectral shaping of the speech to be transmitted in send.

Definition:

The *Send Frequency Response (SFR)* is defined, depending on the type of access interface and the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.4.1.1-1.

Table 7.4.1.1-1: References to send frequency response measurements

Access Interface	Transmission Bandwidth	Specification	Clause
Digital (wired/wireless)	NB	ETSI TS 103 737 [11]	6.1.1
	WB	ETSI TS 103 739 [12]	6.1.1
	SWB, FB	ETSI TS 102 924 [13]	6.3.1
Analog	NB	-	-
	WB, SWB, FB	Recommendation ITU-T P.381 [5]	8.1.4

In the NB voice call scenario for analog (wired) devices, connected to the test equipment as described in clause 4.3.1, the *SFR* is measured from the MRP to the sending input of the device access interface.

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.4.1.1-1, depending on the type of access interface and the provided bandwidth of the ANC device, in order to obtain the *SFR*.

In the NB voice call scenario for analog (wired) devices, the test signal used for the measurements shall be the long speech sequence as described in clause 4.5.4. The measured power density spectrum at the MRP is used as the reference power density spectrum for determining the sending sensitivity. The send sensitivity is determined in third octave intervals as given by IEC 61260-1 [16] for frequencies between 100 Hz and 4 kHz inclusive. In each third octave band, the level of the measured signal is referred to the level of the reference signal, averaged over the complete test sequence length. The sensitivity is determined in dBV/Pa.

Requirement:

The measured frequency response shall be within the limits as defined in the corresponding specification from Table 7.4.1.1-1 depending on the type of access interface and the provided bandwidth of the ANC device.

In the case of NarrowBand (NB) voice call scenario for analog (wired) devices, connected to the test equipment as described in clause 4.3.1, the measured frequency response shall be within the limits as defined in Table 7.4.1.1-2.

Table 7.4.1.1-2: Tolerance mask for the narrowband send frequency response

Frequency (Hz)	Upper limit	Lower limit
200	4	$-\infty$
300	-	-4
1 000	-	-4
3 000	4	-
4 000	-	-12
5 000	2	$-\infty$

NOTE: All sensitivity values are expressed in dB on an arbitrary scale. The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) - logarithmic (Hz) scale.

7.4.1.2 Receive frequency response

Purpose:

The Received *Frequency Response* evaluates if the signal processing of the enabled ANC functionality has any impact on the spectral shaping of the speech in receive.

Definition:

The *Receive Frequency Response (RFR)* is defined, depending on the type of access interface and the provided bandwidth of the ANC device, in the corresponding specification listed in Table 7.4.1.2-1.

Table 7.4.1.2-1: References to receive frequency response measurements

Access Interface	Transmission Bandwidth	Specification	Clause
Digital (wired/wireless)	NB	ETSI TS 103 737 [11]	6.1.2
	WB	ETSI TS 103 739 [12]	6.1.2
	SWB, FB	ETSI TS 102 924 [13]	6.4.2
Analog	NB	-	-
	WB, SWB, FB	Recommendation ITU-T P.381 [5]	8.1.5

In the NB voice call scenario for analog (wired) devices, connected to the test equipment as described in clause 4.3.1, the *RFR* is defined as the diffuse-field equalized sound pressure at the listener's eardrum, measured with narrowband speech from the receiving output of the device access interface to the DRP with diffuse-field correction.

Measurement:

For testing ANC devices, the *RFR* is divided further into RFR_{ON} , measured with the device's ANC functionality enabled, and RFR_{OFF} , measured with the device's ANC functionality disabled.

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *enabled*.
- 4) The test shall be carried out according to the measurement method described in the corresponding specification listed in Table 7.4.1.2-1, depending on the type of access interface and the provided bandwidth of the ANC device, in order to obtain the RFR_{ON} .

In the case of narrowband voice call scenario for analog (wired) devices, connected to the test equipment as described in clause 4.3.1, the test signal used for the measurements in receive shall be the long speech sequence as described in clause 4.5.4, band-limited in the narrowband frequency range. The band limitation is achieved by bandpass filtering in the frequency range between 100 Hz and 4 kHz using a bandpass filter providing ≥ 24 dB/octave. The test signal level applied to the access interface shall be -16 dBm₀, measured according to Recommendation ITU-T P.56 [9]. The receive sensitivity is determined in third octave intervals as given by IEC 61260-1 [16] for frequencies between 100 Hz and 4 kHz inclusive. In each third octave band, the level of the measured signal is referred to the level of the reference signal, averaged over the complete test sequence length. The sensitivity is determined in dB_{Pa/V}.

- 5) The device's ANC functionality is *disabled*.
- 6) The test is repeated according to the measurement method as described in step 4, but with the ANC functionality *disabled*, in order to obtain the RFR_{OFF} .

Requirement:

The measured frequency response shall be within the limits as defined in the corresponding specification from Table 7.4.1.2-1 depending on the type of access interface and the provided bandwidth of the ANC device, for both measured frequency responses, RFR_{ON} and RFR_{OFF} , with ANC functionality enabled and disabled, respectively.

In the NB voice call scenario for analog (wired) devices, connected to the test equipment as described in clause 4.3.1, the measured frequency response shall be within the limits as defined in Table 7.4.1.2-2.

Table 7.4.1.2-2: Tolerance mask for the narrowband receive frequency response

Frequency (Hz)	Upper limit	Lower limit
200	10	$-\infty$
300	9	-10
400	-	-6
1 000	6	-6
2 000	8	-6
3 000	8	-6
4 000	8	-12
5 000	8	$-\infty$

NOTE: All sensitivity values are expressed in dB on an arbitrary scale. The limits for intermediate frequencies lie on straight lines drawn between the given values on a linear (dB) - logarithmic (Hz) scale.

7.4.2 One-way speech quality in receive change

Purpose:

The *One-way Speech Quality in Receive Change* evaluates if the signal processing of the enabled ANC functionality has any impact on the playback quality of received speech signals.

Definition:

The *One-way Speech Quality in Receive Change* (ΔSQ_{VC}) is the difference (if any) between the objective Mean Opinion Score of the listening quality ($MOS-LQO$) measured at the listener's eardrum while wearing the ANC device with ANC functionality *enabled*, $MOS-LQO_{VC,ON}$, and ANC functionality *disabled*, $MOS-LQO_{VC,OFF}$, during a voice call.

The listening speech quality $MOS-LQO_f$ is calculated according to Recommendation ITU-T P.863 [20] in fullband mode for voice call transmissions in WB, SWB and FB. For NB transmissions, $MOS-LQO_n$ is calculated according to Appendix III of Recommendation ITU-T P.863 [20].

The ΔSQ_{VC} is expressed for each ear i as:

$$\Delta SQ_{VC}(i) = MOS-LQO_{VC,ON}(i) - MOS-LQO_{VC,OFF}(i), \forall i \in (l, r) \quad (31)$$

Measurement:

- 1) The test signal used for the measurements shall be the long speech sequence described in clause 4.5.4.
- 2) The test equipment shall be set up according to clause 4.4. The background noise generation system is *disabled*.
- 3) The device's ANC functionality is *disabled*.
- 4) The test signal at the left and right artificial ears is recorded with 48 kHz sampling rate and calibrated to an active speech level of 73 dB_{SPL} according to Recommendation ITU-T P.56 [9]. The listening quality is calculated according to the aforementioned definitions to determine $MOS-LQO_{VC,OFF}(i)$.
- 5) The device's ANC functionality is *enabled*.

- 6) The test signal at the left and right artificial ears is recorded with 48 kHz sampling rate and calibrated to an active speech level of 73 dB_{SPL} according to Recommendation ITU-T P.56 [9]. The listening quality is calculated according to the aforementioned definitions to determine $MOS-LQO_{VC,ON}(i)$.
- 7) The $\Delta SQ_{VC}(i)$ is determined from equation 31 for each ear i .

Requirement:

For each ear i , $\Delta SQ_{VC}(i)$ shall be greater or equal to -0,1 MOS.

7.4.3 Voice call listening effort change

Purpose:

The *Voice Call Listening Effort Change* evaluates if the signal processing of the enabled ANC functionality provides any improvement of the perceived listening effort by, e.g. cancelling the ambient noise or a modifying playback of the received speech signal.

Definition:

The *Voice Call Listening Effort Change* (ΔLE_{VC}) is the difference between the listening effort ($MOS-LEO$) measured at the listener's eardrum while wearing the ANC device with ANC functionality *enabled*, $MOS-LEO_{VC,ON}$, and ANC functionality *disabled*, $MOS-LEO_{VC,OFF}$, during a voice call in the presence of background noise.

The listening effort mean opinion score, $MOS-LEO$, is calculated according to ETSI TS 103 558 [19] without a noise-only reference signal.

The ΔLE_{VC} , is expressed for each noise type, n , from Table 4.2.4-1 as:

$$\Delta LE_{VC}(n) = MOS-LEO_{VC,ON}(n) - MOS-LEO_{VC,OFF}(n), \forall n \in (1:N) \quad (32)$$

Measurement:

- 1) The test signal used for the measurements shall be the speech sequence with long pauses described in clause 4.5.2.
- 2) The test equipment shall be set up according to clause 4.4.
- 3) The device's ANC functionality is *disabled*.
- 4) Noise 4 from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back via the left and right side of the DUT access interface.
- 5) The test and noise signals at the left and right artificial ears, with ANC *disabled*, are recorded with a 48 kHz sampling rate and the listening effort according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{VC,OFF}(n)$.
- 6) The device's ANC functionality is *enabled*.
- 7) Noise 4 from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back via the left and right side of the DUT access interface.
- 8) The test and noise signals at the left and right artificial ears, with ANC *enabled*, are recorded with a 48 kHz sampling rate and the listening effort according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{VC,ON}(n)$.
- 9) The ΔLE_{VC} , is determined from equation 32.

NOTE: Test labs should consider the limitations of prediction accuracy when interpreting the results of perceptual models. The current $MOS-LEO$ prediction accuracy for ANC device use cases is provided in clause D.5.2 of ETSI TS 103 558 [19].

8 Pass-through mode test methods

8.1 Overview

Many ANC devices include functionality beyond ambient noise reduction. One such mode, referred to as pass-through mode in the present document, reproduces external sound sources and self-speech with selectively applied noise reduction, amplification, and/or other signal enhancements. Clause 8 describes test methods for ANC devices supporting pass-through mode.

NOTE: Currently no performance requirements are provided for the pass-through mode test methods defined in clause 8. Such requirements will be included in a future revision.

8.2 Total insertion gain

8.2.1 Overview

Total insertion gain is a frequency dependent measurement (in dB) of the pass-through amplification of a sound source or sound field presented to the user when using an ANC device in pass-through mode. It is computed as the difference between a test signal measured at the DRP with pass-through mode enabled and the same signal measured with an open ear canal. Clause 8.2 defines total insertion gain for ambient noise fields, directional external speech, and self-speech.

Because total insertion gain is a difference spectrum, diffuse-field equalization is not required when conducting the acoustic measurements.

For each total insertion gain metric, labs shall report results with 1/3rd octave frequency resolution from 20 Hz to 20 kHz. Labs should also report results with 4096-point FFT resolution for a more detailed representation.

8.2.2 Total ambient noise insertion gain

Purpose:

The *Total Ambient Noise Insertion Gain* quantifies the amplification of an ambient noise field when using an ANC device in pass-through mode.

Definition:

The *Total Ambient Noise Insertion Gain* ($IG_{noise,total}$) is the difference between the equivalent continuous sound levels of a diffuse sound field measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* (L_{PT}); and
- 2) without wearing the ANC device (L_{OPEN}).

The $IG_{noise,total}$ is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i and noise type n as:

$$IG_{noise,total}(f, i, n) = (L_{PT}(f, i, n) - L_{OPEN}(f, i, n)) \text{ in dB, } \forall i \in (l, r), \forall n \in (1: N) \quad (33)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's pass-through mode is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with pass-through *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{PT}(f, i, n)$ in dB_{SPL}.

- 6) The ANC device is removed from the acoustic test equipment.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, in the absence of the ANC device, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum, $L_{OPEN}(f, i, n)$ in dB_{SPL}.
- 10) The $IG_{noise,total}$ as a function of frequency f is determined, for each ear i and noise type n from equation 33.

NOTE: If the test setup and environment has not changed, the $L_{OPEN}(f, i, n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

8.2.3 Total external speech insertion gain

Purpose:

The *Total External Speech Insertion Gain* quantifies the amplification of directional external speech when using an ANC device in pass-through mode. This measurement is useful for assessing the existence and impact of additional directional processing in a device's pass-through mode.

Definition:

The *Total External Speech Insertion Gain* ($IG_{ES,total}$) is the difference between the equivalent continuous sound levels of a directional external speech source measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* ($L_{ES,PT}$); and
- 2) without wearing the ANC device ($L_{ES,OPEN}$).

The $IG_{ES,total}$ is expressed as a function of the directional sound source counterclockwise azimuthal angle of incidence θ frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, and for each ear i as:

$$IG_{ES,total}(\theta, f, i) = \left(L_{ES,PT}(\theta, f, i) - L_{ES,OPEN}(\theta, f, i) \right) \text{ in dB, } \forall i \in (l, r), \forall \theta \in 0^\circ: 15^\circ: 345^\circ \quad (34)$$

For the measurement of $IG_{ES,total}$, test labs shall use either the directional speech generation system defined in clause 4.2.5.1 or the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 50 \text{ cm}$ and $d_v = 0 \text{ cm}$. Test labs shall report which directional sound source generation system is used. If the directional loudspeaker generation system is used, the far field criteria in clause 4.1.1 shall be met.

Measurement:

- 1) The test signal used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The device's pass-through mode is *enabled*.
- 4) The azimuthal angle of incidence is set to $\theta = 0^\circ$.
- 5) The test signal at the left and right artificial ears, with pass-through *enabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ES,PT}(0, f, i)$ in dB_{SPL}.
- 7) Steps 5 and 6 are repeated for azimuthal angles of incidence from 15° to 345° in steps of 15° .
- 8) The ANC device is removed from the acoustic test equipment.
- 9) The azimuthal angle of incidence is set to $\theta = 0^\circ$.
- 10) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).

- 11) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ES,OPEN}(0, f, i)$ in dB_{SPL}.
- 12) Steps 10 and 11 are repeated for azimuthal angles of incidence from 15° to 345° in steps of 15°.
- 13) The $IG_{ES,total}$ for each azimuthal angle of incidence θ and frequency f is determined for each ear i from equation 34.

NOTE 1: The use of an automated turntable is recommended for changing the relative angle θ between the acoustic test equipment and the directional sound source.

NOTE 2: If the test setup and environment has not changed, the $L_{ES,OPEN}(\theta, f, i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

8.2.4 Total self-speech insertion gain

Purpose:

The *Total Self-speech Insertion Gain* quantifies the amplification of self-speech when using an ANC device in pass-through mode.

Definition:

The *Total Self-speech Insertion Gain* ($IG_{SS,total}$) is the difference between the continuous sound level of a test signal generated by the listener's mouth (e.g. the artificial mouth of the HATS wearing the ANC device) measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* ($L_{SS,PT}$); and
- 2) without wearing the ANC device ($L_{SS,OPEN}$).

$IG_{SS,total}$ is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, for each ear i as:

$$IG_{SS,total}(f, i) = \left(L_{SS,PT}(f, i) - L_{SS,OPEN}(f, i) \right) \text{ in dB}, \forall i \in (l, r) \quad (35)$$

Due to the limitations of existing test equipment (e.g. HATS as defined in Recommendation ITU-T P.58 [3]), the bone conduction path of self-speech may not be reliably reproduced. Therefore, the subjective accuracy of tests described in this clause may be limited to the direct air and electrical paths of self-speech. Refer to clause 5.7 for more information.

Measurement:

- 1) The test signal used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The background noise generation system is *disabled*.
- 4) The device's pass-through functionality is *enabled*.
- 5) The test signal at the left and right artificial ears, with pass-through *enabled*, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 6) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{SS,PT}(f, i)$ in dB_{SPL}.
- 7) The ANC device is removed from the acoustic test equipment.
- 8) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{SS,OPEN}(f, i)$ in dB_{SPL}.
- 10) The $IG_{SS,total}$ is determined as functions of frequency f for each ear i from equation 35.

NOTE: If the test setup and environment has not changed, the $L_{SS,OPEN}(f, i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

8.3 Total loudness level gain

8.3.1 Overview

Total loudness level gain is a single value perceptual measurement (in phon) of the pass-through amplification of a sound source or sound field presented to the user when using an ANC device in pass-through mode. It is computed as the difference between the loudness level of a test signal measured at the DRP with pass-through mode enabled and the same signal measured with an open ear canal. Clause 8.3 defines total loudness level gain for ambient noise fields, directional external speech, and self-speech.

For ambient noise fields (clause 8.3.2), loudness level shall be calculated according to ISO 532-1 [14] with the steps presented in clause 5.2.1. For speech signals (clauses 8.3.3 and 8.3.4), loudness level shall be calculated according to Recommendation ITU-T P.700 [18].

8.3.2 Total ambient noise loudness level gain

Purpose:

The *Total Ambient Noise Loudness Level Gain* quantifies the increase in loudness of an ambient noise field when using an ANC device in pass-through mode.

Definition:

The *Total Ambient Noise Loudness Level Gain* ($LLG_{noise,total}$) is the difference between the ISO 532-1 [14] loudness levels of a diffuse sound field measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* ($Loud_{noise,PT}$); and
- 2) without wearing the ANC device ($Loud_{noise,OPEN}$).

The $LLG_{noise,total}$ is expressed for each noise type n as:

$$LLG_{noise,total}(n) = (Loud_{noise,PT}(n) - Loud_{noise,OPEN}(n)) \text{ in phon, } \forall n \in (1:N) \quad (36)$$

Measurement:

- 1) The ANC device and test equipment shall be set up according to clause 4.4.
- 2) The device's pass-through functionality is *enabled*.
- 3) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 4) The noise signal at the left and right artificial ears, with pass-through *enabled*, is recorded to include analysis start and duration for noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 5) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14] to determine $Loud_{noise,PT}$ as described in clause 5.2.1.
- 6) The ANC device is removed from the acoustic test equipment.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system.
- 8) The noise signal at the left and right artificial ears, in the absence of the ANC device, is recorded for the duration of noise n indicated in Table 4.2.4-1 with a 48 kHz sampling rate.
- 9) The recorded noise signal is diffuse-field corrected and processed with a Zwicker Loudness calculator, conforming to ISO 532-1 [14], to determine $Loud_{noise,OPEN}$ as described in clause 5.2.1.
- 10) The $LLG_{noise,total}$ is determined for each noise type n by equation 36.

NOTE: If the test setup and environment has not changed, the $Loud_{noise,OPEN}(n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

8.3.3 Total external speech loudness level gain

Purpose:

The *Total External Speech Loudness Level Gain* quantifies the increase in loudness of an external speech source when using an ANC device in pass-through mode. This measurement is useful for assessing the existence and impact of additional directional processing in a device's pass-through mode.

Definition:

The *Total External Speech Loudness Level Gain* ($LLG_{ES,total}$) is the difference between loudness levels of a directional external speech source measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* ($Loud_{ES,PT}$); and
- 2) without wearing the ANC device ($Loud_{ES,OPEN}$).

External speech loudness level is measured according to Recommendation ITU-T P.700 [18]. The $LG_{ES,total}$ is expressed as a function of the speech source counterclockwise azimuthal angle of incidence θ as:

$$LLG_{ES,total}(\theta) = \left(Loud_{ES,PT}(\theta) - Loud_{ES,OPEN}(\theta) \right) \text{ in phon, } \forall \theta \in 0^\circ: 15^\circ: 345^\circ \quad (37)$$

For the measurement of $LLG_{ES,total}$, test labs shall use either the directional speech generation system defined in clause 4.2.5.1 or the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 50$ cm and $d_v = 0$ cm. Test labs shall report which directional sound source generation system is used. If the directional loudspeaker generation system is used, the far field criteria in clause 4.1.1 shall be met.

Measurement:

- 1) The test signal used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The device's pass-through mode is *enabled*.
- 4) The azimuthal angle of incidence is set to $\theta = 0^\circ$.
- 5) The test signal at the left and right artificial ears, with pass-through *enabled*, is recorded with a 48 kHz sampling rate and the loudness level is calculated according to Recommendation ITU-T P.700 [18] to determine $Loud_{ES,PT}(0)$.
- 6) Step 5 is repeated for azimuthal angles of incidence from 15° to 345° in steps of 15° .
- 7) The ANC device is removed from the acoustic test equipment.
- 8) The azimuthal angle of incidence is set to $\theta = 0^\circ$.
- 9) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate and the loudness level is calculated according to Recommendation ITU-T P.700 [18] to determine $Loud_{ES,OPEN}(0)$ as described in clause 5.2.1.
- 10) Step 9 is repeated for azimuthal angles of incidence from 15° to 345° in steps of 15° .
- 11) The $LLG_{ES,total}$ for each azimuthal angle of incidence θ is determined by equation 37.

NOTE 1: The use of an automated turntable is recommended for changing the relative angle θ between the acoustic test equipment and the directional sound source.

NOTE 2: If the test setup and environment has not changed, the $Loud_{ES,OPEN}(\theta)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

8.3.4 Total self-speech loudness level gain

Purpose:

Total Self-Speech Loudness Level Gain quantifies the increase in loudness of self-speech when using a pass-through enabled device.

Definition:

The *Total Self-Speech Loudness Level Gain* ($LLG_{SS,total}$) is the difference between the loudness levels of a test signal generated by the listener's mouth measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* ($Loud_{SS,PT}$); and
- 2) without wearing the ANC device ($Loud_{SS,OPEN}$).

Due to the perceptual nature of $LLG_{SS,total}$ and the lack of accurate bone conduction masking in state-of-the-art test equipment (e.g. HATS), no measurement procedure is provided until such equipment is made available.

Void (38)

Measurement:

Void.

8.4 Pass-through self-noise

8.4.1 Overview

The pass-through self-noise in quiet is a function of several things including the device's electrical system noise floor (as described in clause 5.3.1) and the device's amplification of room noise through pass-through mode signal processing. See clause 5.3.1 for test equipment idle noise considerations with respect to self-noise measurements.

8.4.2 Pass-through self-noise spectrum

Purpose:

The *Pass-Through Self-Noise Spectrum* quantifies the noise generated by an ANC device in pass-through mode. Such noise may be perceptually relevant when an ANC device is used in a quiet environment.

Definition:

The *Pass-Through Self-Noise Spectrum* ($N_{self,PT}$) shall be measured using the procedure described in clause 5.3.2 with pass-through mode *enabled*.

8.4.3 Pass-through self-noise loudness level

Purpose:

The *Pass-Through Self-Noise Loudness Level* quantifies the loudness of noise generated by an ANC device in pass-through mode. Such noise may be perceptually relevant when an ANC device is used in a quiet environment.

Definition:

The *Pass-Through Self-Noise Loudness Level* ($Loud_{self,PT}$) shall be measured using the procedure described in clause 5.3.3 with pass-through mode *enabled*.

8.5 External speech listening effort change

Purpose:

The *External Speech Listening Effort Change* quantifies the change in a user's listening effort when attending to an external speech source while using an ANC device in pass-through mode in a noisy environment.

Definition:

The *External Speech Listening Effort Change* for a pass-through enabled device ($\Delta LE_{ES,PT}$) is the difference between the listening effort measured at the listener's eardrum with:

- 1) pass-through mode *enabled* ($MOS-LEO_{ES,PT}$); and
- 2) without wearing the device ($MOS-LEO_{ES,OPEN}$)

for a test signal generated by a directional sound source as described in clause 4.2.5. The listening effort mean opinion score, $MOS-LEO$, is calculated as described in ETSI TS 103 558 [19] without a noise-only reference signal.

$\Delta LE_{ES,PT}$ is expressed for each noise type n from Table 4.2.4-1 as:

$$\Delta LE_{ES,PT}(n) = \left(MOS-LEO_{ES,PT}(n) - MOS-LEO_{ES,OPEN}(n) \right), \forall n \in (1:N) \quad (39)$$

Test labs may use either the directional speech generation system defined in clause 4.2.5.1 or the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 40$ cm and $d_v = 30$ cm. Test labs shall report which directional sound source generation system is used.

Measurement:

- 1) The test signal used for the measurements shall be the speech sequence with long pauses described in clause 4.5.2.
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The device's pass-through functionality is *enabled*.
- 4) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back through the directional generation system.
- 5) The test and noise signals at the left and right artificial ears, with pass-through *enabled*, are recorded for the duration of the test signal with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{ES,PT}(n)$.
- 6) The ANC device is removed from the acoustic test equipment.
- 7) Noise n from Table 4.2.4-1 is played back through the background noise generation system and the test signal is simultaneously played back through the directional generation system.
- 8) The test and noise signals at the left and right artificial ears, in the absence of the ANC device, are recorded for the duration of the test signal with a 48 kHz sampling rate and the listening effort score according to ETSI TS 103 558 [19] is calculated to determine $MOS-LEO_{ES,OPEN}(n)$.
- 9) The $\Delta LE_{ES,PT}$ is determined for each noise type n from equation 39.

NOTE 1: Test labs should consider the limitations of prediction accuracy when interpreting the results of perceptual models. The current $MOS-LEO$ prediction accuracy for ANC device use cases is provided in clause D.5.2 of ETSI TS 103 558 [19].

NOTE 2: If the test setup and environment has not changed, the $MOS-LEO_{ES,OPEN}(n)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

8.6 Loud speech output level

Purpose:

The *Loud Speech Output Level* illustrates the sound level produced when using an ANC device in pass-through mode in the presence of a loud speech signal. It is important to characterize the response of a device to a loud input signal to ensure that the acoustic levels experienced by the user do not reach or exceed harmful levels.

Definition:

The *Loud Speech Output Level* ($L_{ES,loud}$) is the equivalent continuous sound level of a device's pass-through mode output measured at the user's eardrum in response to a 90 dB_{SPL} ASL speech signal with DUT amplification set to its maximum (if amplification user controls are provided by the DUT manufacturer).

The $L_{ES,loud}$ is expressed as a function of frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$, and for each ear i as $L_{ES,loud}(f, i)$ in dB_{SPL}.

Test labs shall use the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 50 \text{ cm}$ and $d_v = 0 \text{ cm}$ for the measurement of $L_{ES,loud}$.

NOTE 1: The test signal level of 90 dB_{SPL} ASL is chosen to harmonize the measurement of $L_{ES,loud}$ with the Maximum Acoustic Output measurement in ANSI/CTA 2051-2017 [i.5] and the Output Sound Pressure Level 90 dB Input SPL (OSPL90) measurement in ANSI/ASA S3.22-2014 [i.6].

NOTE 2: A speech sequence is chosen as the test signal for the measurement of $L_{ES,loud}$ to bypass the impact of additional DUT signal processing, which may impact the measurement of $L_{ES,loud}$ through rejection of pure tone or stationary input signals.

Measurement:

- 1) The test signal used for the measurements shall be the long speech sequence described in clause 4.5.4.
- 2) The test equipment shall be set up according to clause 4.4 without the ANC device (i.e. open canal).
- 3) The test signal is calibrated to an ASL of 90 dB_{SPL} at the left artificial ear DRP, in the absence of the ANC device and with diffuse-field correction *disabled*.
- 4) The ANC device is positioned according to clause 4.5.4.
- 5) The device's pass-through functionality is *enabled*.
- 6) The test signal at the left artificial ear, with pass-through *enabled* and diffuse-field correction *disabled*, is recorded for the duration of the test signal with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 7) The FFT representation is used to calculate a 1/3rd octave resolution spectrum $L_{ES,loud}(f, l)$ in dB_{SPL}.
- 8) Steps 2 – 7 are repeated for the right ear.

8.7 Speech gain curves

8.7.1 Overview

The speech gain curves defined in clause 8.7 are divided into a single *Overall Speech Gain Curve* (clause 8.7.2) and several *Frequency Dependent Speech Gain Curves* (clause 8.7.3). Speech gain curves illustrate any non-linear gain characteristics of a pass-through enabled ANC device. Non-linear gain characteristics are the result of hardware components and/or signal processing features that adjust signal gain based on the signal input level (e.g. automatic gain control, dynamic range compression, hardware limiting, etc.).

If the ANC device provides user controls to enable or disable non-linear dynamic range control, such settings shall be enabled for all tests in clause 8.7.

Test labs shall use the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 50$ cm and $d_v = 0$ cm for all tests in clause 8.7.

NOTE: Input-output characteristics and gain curves are often measured with pure-tone or stationary noise signals [i.6]. A speech sequence is chosen for the measurements in clause 8.7 to bypass the impact of additional DUT signal processing, which may impact the measurement results through rejection of pure tone or stationary input signals.

8.7.2 Overall speech gain curve

Purpose:

The *Overall Speech Gain Curve* produces a single broadband gain curve. This curve illustrates the non-linear gain characteristics of a pass-through enabled ANC device when presented with speech at various input levels.

Definition:

The *Overall Speech Gain Curve* ($G_{PT,ASL}$) is the difference between the ASL (calculated according to Recommendation ITU-T P.56 [9]) of a directional external speech source measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* (ASL_{OUT}); and
- 2) without wearing the ANC device (ASL_{IN}).

The $G_{PT,ASL}$ is expressed as a function of the calibrated input level ASL_{IN} , with $50 \text{ dB}_{SPL} \leq ASL_{IN} \leq 90 \text{ dB}_{SPL}$ for each ear i as:

$$G_{PT,ASL}(ASL_{IN}, i) = (ASL_{OUT}(ASL_{IN}, i) - ASL_{IN}) \text{ in dB}, \forall i \in (l, r) \quad (40)$$

Measurement:

- 1) The test signal used for the measurements shall be the long speech sequence described in clause 4.5.4.
- 2) The test equipment shall be set up according to clause 4.4 without the ANC device (i.e. open canal).
- 3) The test signal is calibrated to $ASL_{IN} = 50 \text{ dB}_{SPL}$ ASL at the left artificial ear DRP, in the absence of the ANC device and with diffuse-field correction *disabled*.
- 4) The ANC device is positioned according to clause 4.5.4.
- 5) The device's pass-through functionality is *enabled*.
- 6) The test signal at the left artificial ear, with pass-through *enabled*, is recorded for the duration of the test signal with a 48 kHz sampling rate.
- 7) The ASL is calculated according to Recommendation ITU-T P.56 [9] as $ASL_{OUT}(50, l)$.
- 8) $G_{PT,ASL}(50, l)$ is computed according to equation 40.
- 9) Steps 2 – 8 are repeated for $55 \text{ dB}_{SPL} \leq ASL_{IN} \leq 90 \text{ dB}_{SPL}$ in steps of 5 dB SPL.
- 10) Steps 2 – 9 are repeated for the right ear.

8.7.3 Frequency dependent speech gain curves

Purpose:

The *Frequency Dependent Speech Gain Curves* produce a gain curve for each analysis frequency bin. These curves illustrate the frequency dependent non-linear gain characteristics of a pass-through enabled ANC device when presented with speech at various input levels.

Definition:

The *Frequency Dependent Speech Gain Curves* ($G_{PT,spectrum}$) are the difference between the equivalent continuous sound levels of a directional external speech source measured at the listener's eardrum:

- 1) when wearing the ANC device with pass-through mode *enabled* (L_{OUT}); and
- 2) without wearing the ANC device (L_{IN}).

The $G_{PT,spectrum}$ is expressed as a function of the calibrated input level ASL_{IN} , with $50 \text{ dB}_{SPL} \leq ASL_{IN} \leq 90 \text{ dB}_{SPL}$ and frequency f , with $20 \text{ Hz} \leq f \leq 20 \text{ kHz}$ for each ear i as:

$$G_{PT,spectrum}(ASL_{IN}, f, i) = (L_{OUT}(ASL_{IN}, f, i) - L_{IN}(ASL_{IN}, f, i)) \text{ in dB}, \forall i \in (l, r) \quad (41)$$

Measurement:

- 1) The test signal used for the measurements shall be the long speech sequence described in clause 4.5.4.
- 2) The test equipment shall be set up according to clause 4.4 without the ANC device (i.e. open canal).
- 3) The test signal is calibrated to $ASL_{IN} = 50 \text{ dB}_{SPL}$ ASL at the left artificial ear DRP, in the absence of the ANC device and with diffuse-field correction *disabled*.
- 4) The test signal at the left artificial ear, in the absence of the ANC device, is recorded for the duration of the test signal with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 5) The FFT representation is used to calculate a $1/3^{\text{rd}}$ octave resolution spectrum $L_{IN}(50, f, l)$ in dB_{SPL} .
- 6) The ANC device is positioned according to clause 4.5.4.
- 7) The device's pass-through functionality is *enabled*.
- 8) The test signal at the left artificial ear, with pass-through *enabled*, is recorded for the duration of the test signal with a 48 kHz sampling rate and analysed with a 4096-point FFT resolution (Hann window with 50 % overlap).
- 9) The FFT representation is used to calculate a $1/3^{\text{rd}}$ octave resolution spectrum $L_{OUT}(50, f, l)$ in dB_{SPL} .
- 10) $G_{PT,spectrum}(50, f, l)$ is computed according to equation 41.
- 11) Steps 2 – 10 are repeated for $55 \text{ dB}_{SPL} \leq ASL_{IN} \leq 90 \text{ dB}_{SPL}$ in steps of 5 dB_{SPL} .
- 12) Steps 2 – 11 are repeated for the right ear.

8.8 Pass-through latency

Purpose:

The *Pass-Through Latency* quantifies the time it takes for an external sound source, captured by a pass-through enabled device's microphone(s), to be processed and reproduced for the user. Excessive pass-through latency may result in audible artifacts such as comb filtering.

Definition:

The *Pass-Through Latency* (*Latency*) is the difference between the time delay (ΔT) of an external speech source measured at the listener's eardrum:

- 1) with pass-through mode *enabled* (ΔT_{PT}); and
- 2) without wearing the device (ΔT_{OPEN}).

The ΔT is estimated according to equation 42. $R_{y,x}$ denotes the cross-correlation of two discrete-time signals (y and x), m is the lag of the cross-correlation computation in samples, and f_s is the sampling rate of x and y .

$$\Delta T = \underset{m}{\operatorname{argmax}}(|R_{y,x}(m)|)/f_s \text{ in seconds} \quad (42)$$

The *Latency* is expressed for each ear i as:

$$\text{Latency}(i) = \Delta T_{PT}(i) - \Delta T_{OPEN}(i) \text{ in seconds, } \forall i \in (l, r) \quad (43)$$

For the measurement of *Latency*, test labs shall use the directional loudspeaker generation system defined in clause 4.2.5.2 with $d_h = 50$ cm and $d_v = 0$ cm.

The temporal resolution of ΔT and *Latency* are proportional to f_s . The nominal $f_s = 48$ kHz used in this clause produces a resolution of 20,8 microseconds per sample. Test labs that require a higher resolution should increase f_s (e.g. $f_s = 192$ kHz produces a temporal resolution of 5,2 microseconds per sample).

NOTE 1: The computation of ΔT_{PT} may underestimate the true pass-through time delay if the level of the amplified pass-through signal is not sufficiently higher than the level of the acoustic leak path determined by the DUT passive attenuation. This potential underestimation is mitigated when pass-through amplification is set to a maximum.

NOTE 2: The output of a DUT pass-through mode may have a group delay response that varies with frequency (i.e. a non-linear phase response). In this case, the precise ΔT_{PT} computed via cross-correlation may be sensitive to the spectral characteristics of the test signal. For this reason, a speech sequence is chosen as the test signal to reflect the spectral characteristics of a typical pass-through use case.

Measurement:

- 1) The test signal (x) used for the measurements shall be the short speech sequence described in clause 4.5.3.
- 2) The ANC device and test equipment shall be set up according to clause 4.4.
- 3) The device's pass-through functionality is *enabled*.
- 4) The test signal at the left and right artificial ears, with pass-through *enabled*, is recorded with a 48 kHz sampling rate (y_{PT}).
- 5) The cross-correlation of y_{PT} and x is computed for each ear and used to estimate the pass-through time delay according to equation 42 as $\Delta T_{PT}(i)$.
- 6) The ANC device is removed from the test equipment.
- 7) The test signal at the left and right artificial ears, in the absence of the ANC device, is recorded with a 48 kHz sampling rate (y_{open}).
- 8) The cross-correlation of y_{open} and x is computed for each ear and used to estimate the open ear time delay according to equation 42 as $\Delta T_{open}(i)$.
- 9) The *Latency* for each ear i is determined according to equation 43.

NOTE 3: If the test setup and environment has not changed, the $\Delta T_{open}(i)$ measurement may be re-used from previous measurements since the measurement is independent of the ANC device under test.

Annex A (informative): Turbulent wind generation considerations

Wind noise testing necessitates the generation of a repeatable turbulent wind flow. This is important for both inter- and intra-lab repeatability of data. In this annex, several considerations are presented, which test labs may use to guide their selection of an appropriate low-noise turbulent wind generator.

Unless otherwise specified, all measurements (both wind speed and acoustic level) are made 30 cm from the centre of the wind generator's exit port with no acoustic test equipment in place.

- The wind generator's exit port size, measured as the smallest edge length (if rectangular), smallest semi-axis (if elliptical) or diameter (if circular), should be large enough to produce a wind flow that covers the acoustic test equipment and DUT.
- The wind generator should produce a turbulent wind flow with adjustable speeds to allow for fine tuning of the wind speed prior to testing. Wind noise testing defined in clause 5.4 calls for a wind speed of 4,5 m/s. Higher wind speeds should be achievable to facilitate testing in more adverse conditions.
- The wind generator should produce minimal acoustic noise (e.g. $< 40 \text{ dB}_{\text{SPL}}(\text{A})$), measured in third octave bands from 20 Hz to 20 kHz using a reference microphone with appropriate wind screening.
- The wind generator should produce a temporally consistent wind speed, measured as the absolute average error between the target wind speed and the measured wind speed over 15 seconds.
- The wind generator should produce a spatially consistent wind speed, measured as the absolute average error between the target wind speed and the wind speed measured for 15 seconds at the points coinciding with the acoustic test equipment EEPs when positioned with a relative angle of 0° .

History

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