EHIMA

EUROPEAN HEARING INSTRUMENT MANUFACTURERS ASSOCIATION

Testing hearing aids with a speech-like signal

Version 4.1

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051104	1	Added comments & remarks from ISMADHA group
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1 Preface

This **working document** was prepared by the EHIMA Technical Committee. The main purpose of this working document is to provide a method for characterisation of hearing aids using a speech-like signal. A hearing aid can to some extent no longer be seen solely a stand alone device. Today a hearing aid is more a combination of the physical hearing aid & the fitting software which accompanies it. The method described is to be seen as a supplement to existing standards, not a replacement.

2 Table of contents~~Table of contents~~

3

4 Scope

This proposal provides a method characterizing the hearing aid including the fitting software and based upon this characterise the hearing aid performance. I.e. all tests described are intended to be accompanied with a setting of the hearing aid that is comparable to the setting in the hearing aid worn by the end-user.
The purpose of this document is to define a test method with which it enables characterization of hearing aids using a modulated speech like signal. The described method provides conformity when measuring the performance of hearing aids using a speech-like signal. The method is particularly useful for characterisation of speech gain in hearing aids.. Furthermore the method provides simulated real ear measurement results which are comparable to results, obtained when measuring on a real person.
The requirements regarding the input signal, measurement setup and data processing are described in the document.

5 Setup

5.1System overview

The goal of the method is to provide simulated real-ear measurements of the insertion gain which are comparable to results when measuring on a real person. Furthermore it provides the option to measure coupler gain.

The next three figures shows an overview of the method .

- Figure 1 shows the measurement of the open ear response for the insertion gain.
- Figure 2 shows the measurement of the H.I response for the insertion gain.
- Figure 3 shows the measurement of the H.I response and the free-field response for the coupler gain

For the insertion gain, the real-ear equivalent is also shown.

5.1.1 Insertion gain

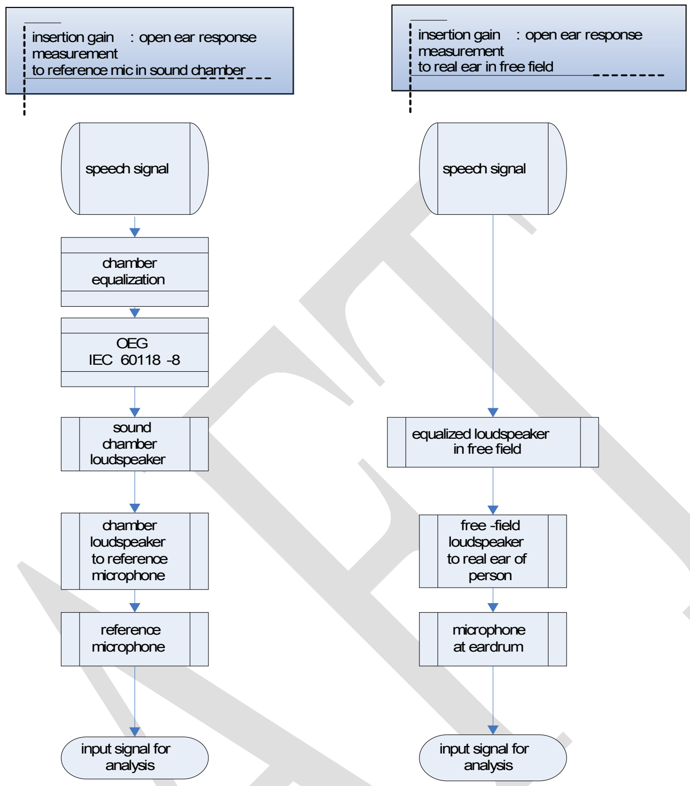

Figure 1: **Measurement set-up for open ear response for Insertion gain: Input signal.**
Left: proposed measurement method in sound chamber ,
Right: equivalent measurement method for real-ear measurement (not proposed).

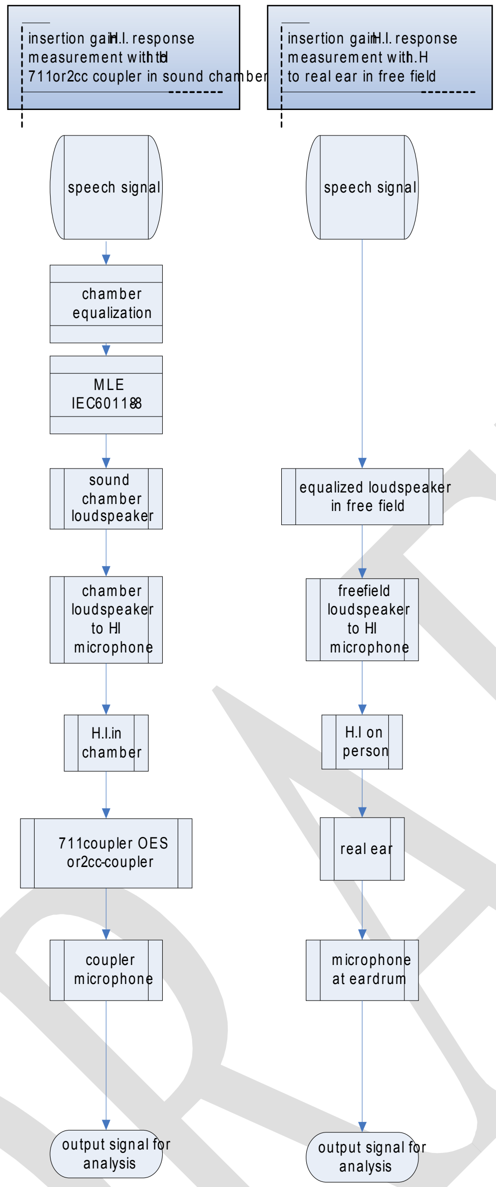

Figure 2: **Measurement set-up for H.I. response for Insertion gain. Blocks with vertical lines are actual physical parts of the measurement set-up. B with horizontal lines are pre- and post-processing steps in software.**
Left: proposed measurement method in sound chamber with HI to 711 or 2cc coupler.
Right: equivalent measurement method on real-ear (not proposed)

5.1.2 Coupler gain

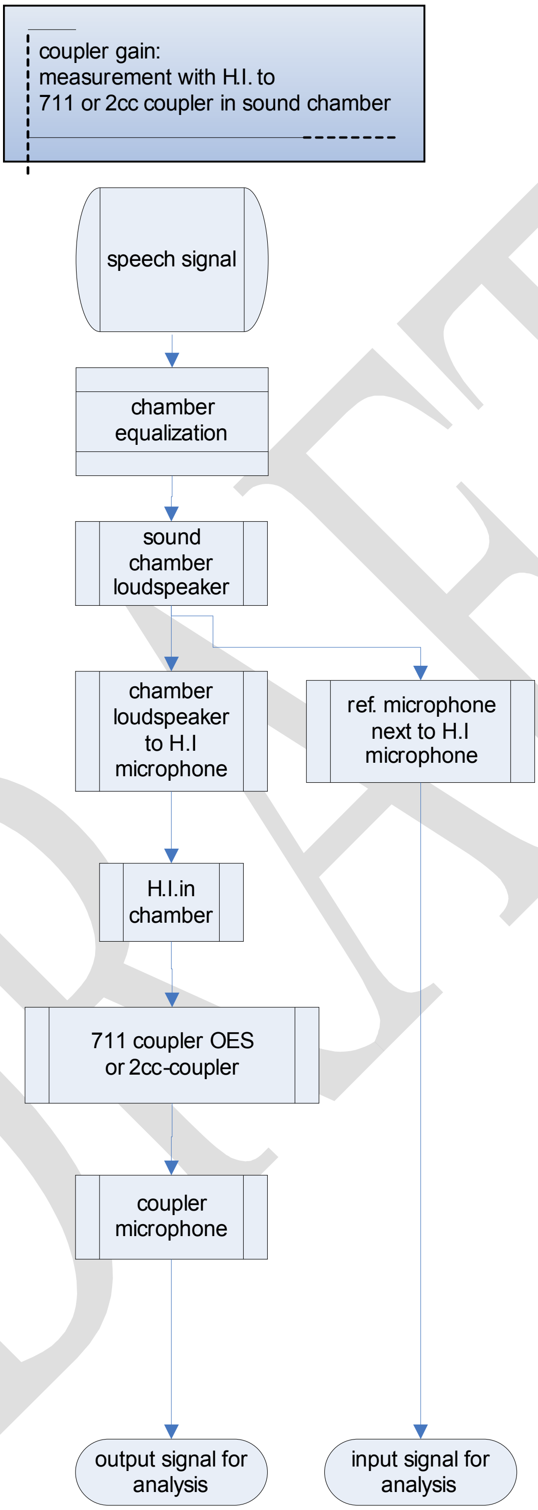

Figure 3 **Measurement set-up for Coupler gain (H.I response as well as free-field response). Proposed measurement method in sound chamber with HI to 711 coupler or 2cc coupler**

5.2Test equipment

In general the test equipment, test conditions & test enclosure used for the measurements shall fulfil the following requirements, which follow the requirements listed in IEC 60118-0 2nd edition 1983 and in ANSI S3.22- 2003:

1) The test enclosure used shall provide essential free field conditions in the frequency range 200Hz-8kHz.
2) Calibration of SPL measurement system must (excluding the IEC 60711 Ear simulator) be accurate to within +/- 0,5dB at a specified frequency.
3) The pressure sensitivity of the measuring microphone shall be frequency-independent within +/- 1dB in the frequency range 200Hz - 3kHz and within +/- 2dB in the range 3kHz to 8 kHz relative to the pressure sensitivity level at 1kHz.
4) The free field calibration of the sound pressure levels shall be accurate within +/- 0.5 dB at a specified frequency. The free field sensitivity level of the measuring microphone shall be frequency independent within +/- 1dB in the frequency range 200Hz-5kHz and within +/- 1.5 dB in the frequency range 5kHz to 8kHz relative to the free-field sensitivity level at a specified frequency.
5) The sound field at the test point is to be accurate within +/- 1.5 dB within the frequency range 200Hz to 3kHz and accurate within +/- 2,5 dB from 3kHz to 8kHz.
6) The hearing must be positioned to reflect a sound incidence from 0 degrees, i.e. frontal sound incidence is assumed
7) Measurement frequency range are in 1/3 octave bands with center frequencies from 250Hz to 6.3kHz.

5.3Input signal

The signal to be used for the measurement is described in [14] and it consists of concatenated real female speech without noise with normal vocal effort.
The overall RMS level of the input signal will be measured within a bandwidth of 200Hz-5kHz, see Section 6.5.2.1 of [14].

On the figure and table below the free field 1/3 octave band 30% percentile, 99% Percentile & LTASS levels of the female speech-like with an overall LTASS of 65dBSPL (BW 200Hz-5kHz) are shown.

XX Figure & Table to be updated when test signal is ready. It may turn out that the low and high percentiles are to be changed, this depends upon the signal characteristics.
Ranking of short-time levels:

Speech with pauses **Speech without pauses**

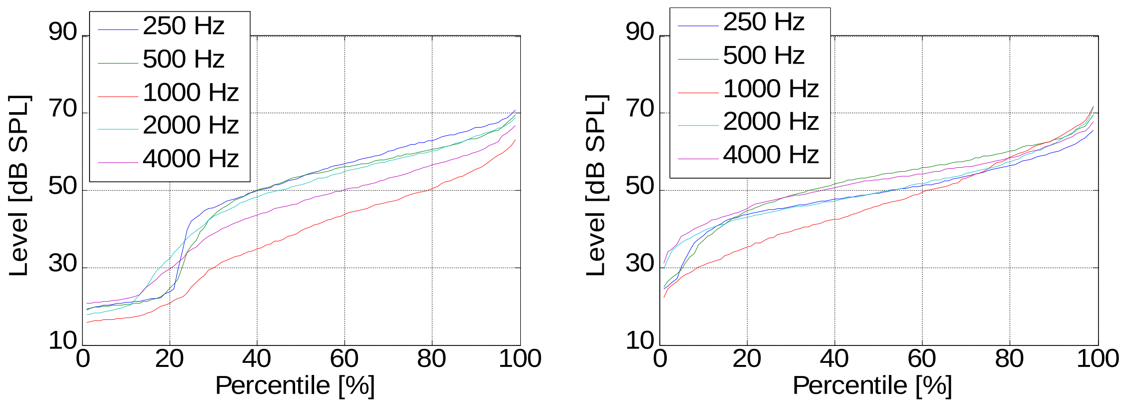

Figure 4: Percentile distribution of the input signal

Illustration of the dynamics of the input signal:

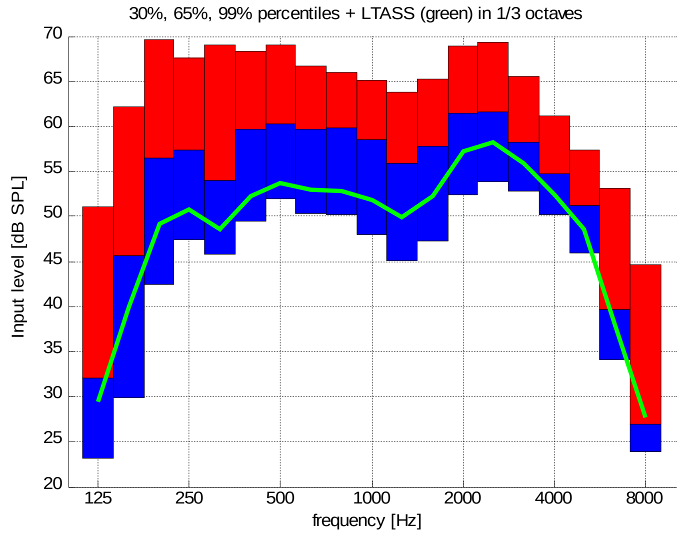

Figure 5: Input signal 30%,65%, 99% percentiles & LTASS vs. 1/3 octave frequency band

Data	250	315	400	500	630	800	1000	1250	1600	2000	2500	3150	4000	5000	6300
30% percentile															
65% percentile															
LTASS															
99% Percentile															

Table 1: 1/3 octave band levels in dBSPL of the female reference input signal

5.3.1 Shaping the input signal

The measurement method has two options: the insertion gain (preferred option) and the coupler gain. Depending on the option, the input signal has to be shaped in a specific manner.

5.3.1.1 Preferred: insertion gain

The input signal accompanying traditional measurement methods is usually specified under free field conditions. As described in the scope, the proposed method shall obtain measurement results which are comparable to results which would be obtained when measuring on a person. I.e. when the hearing aid is positioned on a person the free field condition no longer applies. This will be accounted for using the free field to head transformation.

For the hearing aid response measurements, this transformation is also known as the microphone location effect, the MLE. I.e. the free field input signal, is to be shaped/filtered with the MLE transfer functions. Data used are specified in IEC 60118.8, Annex A, Table 1.See data reproduced in below:

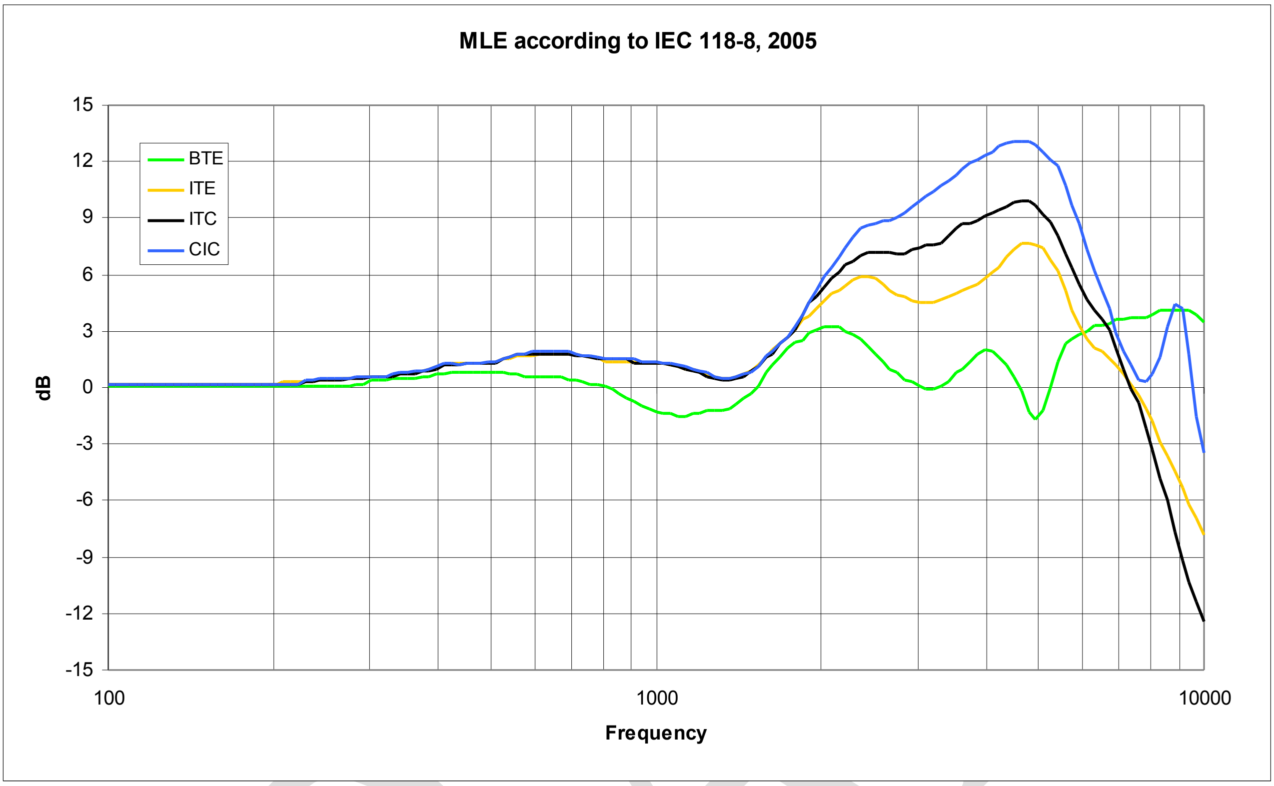

Figure 6: MLE according to IEC 60118-8, 2005

If the input signal is not shaped acc. to this, the used data sets shall be stated. The MLE-data appropriate for the actual hearing aid shall be used.

For the open-ear response measurements, this transformation is known as the Open Ear Gain (OEG). I.e. the free field input signal, is to be shaped/filtered with the OEG transfer function. The preferred OEG data to be used is obtained from IEC 118-8, 2005, which is measured on a KEMAR mannequin see data below.

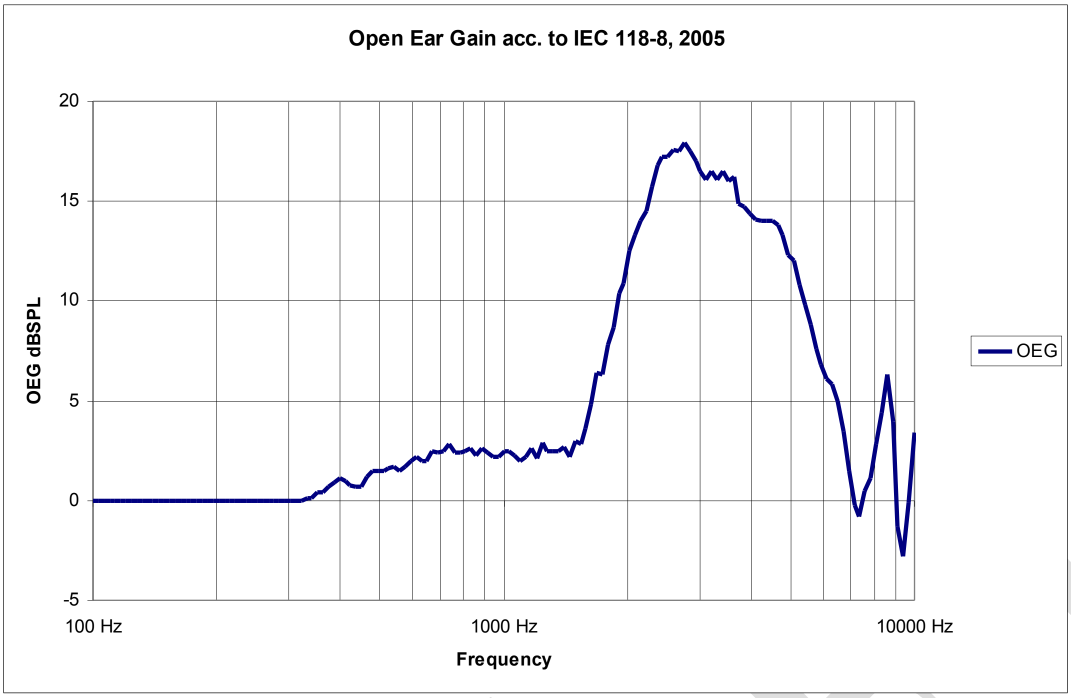

Figure 7: Open Ear Gain (OEG) according to IEC 118-8, 2005

If the OEG specified here is not used the used data, shall be stated.

5.3.1.2 Optional: coupler gain

The proposed method also supports coupler measurements: both the input signal applied to the hearing aid and the input to be subtracted from the hearing aid output is free field. Hence there are no transformations needed.

5.4Ambient noise

The validity of the measurements must be verified. The ambient noise in the frequency range shall be sufficiently low. The requirement is that measured 1/3 octave band levels within the measurement bandwidth 1/3 octave 250Hz to 6.3kHz should be 10 dB or more below the measured input & output values. If the condition can not be met the, validity of the measurement result is questionable.

Due to the probability of varying ambient noise level during the measurement period, any large ambient noise level variations during measurements must be avoided. Especially regarding the soft components, e.g. the 30% percentile input levels, the noise floor can affect the measurement result.

5.5Earphone coupler and attachments

For measurement of the hearing aid output, the ear simulator according to IEC 711 [14] is the preferred coupler. Furthermore the Zwislocki coupler [14] can be used as it yields results that are comparable to the IEC ear simulator. The IEC 711 ear simulator or the Zwislocki will provide an impedance termination of the hearing aid under test which is comparable to a real ear.

Standard coupler adaptors are to be used i.e. the ear simulator adaptors for ITE & BTE instruments acc. to the IEC 711 standard or the ANSI S3.25-1989.

If it is not possible to use the IEC 711 or Zwislocki coupler , a 2 cm^3 coupler, according to ANSI standards 3.7 [14] or IEC 60318-5 [14], can be used. The 2cc HA-1 coupler is used for ITE hearing aids and the 2cc HA-2 for BTE hearing instruments.

The used coupler and adaptor must be clearly stated, providing reproducibility of the measurement.

Results comparing 2cc and IEC711 coupler measurements will differ to each other mainly due to differences in receiver load. During the analysis, a correction will be applied for this difference, see Section 7.2.5. Section 10 (Annex 2) discusses in further detail the expected differences between 2cc coupler, IEC 711 coupler, and real-ear measurements.

N.B: The used setup must be specified in enough details to reproduce the complete setup.

5.61/3 octave filter specification

The used 1/3 octave band filters must fulfil the IEC 61260 “Electroacoustics – Octave-band and fractional-octave-band filters” class 2 requirements in the frequency range 200 Hz - 8 kHz, i.e. covering the 1/3 octave band filter in the specified frequency range from 250Hz to 6.3.kHz.

6 Programming the hearing aid

As stated in the foreword & scope, the main idea is to measure the hearing aid in a setting which compares well to the end user setting.
It must be clearly specified how the instrument has been programmed, i.e. which selections, features and trimmer actions are made in the fitting software.
There however might be exemptions to this, e.g. having a hearing aid only allowing open vent selections in the fitting software. The vendor might add a meta-purpose button to the fitting software which set's the fitting software up and program the hearing aid in its appropriate setting. The selections should be visible in the fitting software after activating this meta–purpose button.

In general all settings of the hearing aid, necessary to reproduce the measurement results, shall be clearly presented and be reproducible.

6.1 Audiograms and user experience

The manufacturer of the hearing aid shall state one appropriate audiogram from the audiograms stated below to be used in the fitting software, typical a best fit, within the hearing aids fitting range. Vendor should state which additional parameters are needed for setting up the fitting software to reproduce the hearing aids setting in which it was measured.
Recommended is a std. adult, experienced user. But the typical target group of the instrument may be used as well. Again the need for a unique description of the setup applies.

XX Karolinska Intistute, Sweden is to provide std. audiograms. Nicolai Bisgaard & Martin Dahlquist drives this task. Expect app. 5 std. audiograms which covers app. 60% of 19000 persons in database. Siemens & Frank Rosenberger has access to a database with data from app. 20000 persons. A benefit in combining these might be ideal?

XX Figure & table of audiograms needed.

6.2 Features

Noise reduction algorithms, feedback suppression systems, echo cancellation etc. should be set to end-user settings, i.e. the default settings from the fitting software. In some cases a special setting, may be recommended to increase reproducibility of the measurements.
Again: *In general all settings of the hearing aid, necessary to reproduce the measurement results, shall be clearly presented and be reproducible.*

6.2.1 Directionality

Hearing aids with directional microphones shall be set to omni directional mode if possible.

6.2.2 Vent selection

A closed vent is to be used for the insurance of a well defined measurement setup. In cases where this selection is not available the manufacturer can provide a special setting/setup handling this or choose to use an available vent selection. *The use vent selection must be clearly stated.*

I.e. this proposal focuses upon representation of simulated real ear gain of hearing aids programmed and measured using a closed vent. This is mainly due to expected large variances of the test results when measuring on an open system.

7 Measurements & analysis

7.1 Measurements

60 seconds of the test signal shall be used for every measurement. The first 15 seconds are used to stabilize the hearing aid. The manufacturer may specify a longer stabilisation time if needed, but the measurement time (45 seconds) is fixed, i.e. the time signal which is analysed, is fixed.
The recommended signal is the signal in Section 5.3. The overall RMS level of the free-field input signal will be calculated according to the Section 5.3.
The recommended overall RMS-level of the free-field input signal is 65 dB SPL.

A measurement at only one overall RMS level will not always result in a sufficient characterization of the hearing aid gain. A measurement at one level can only characterize the hearing aid as fast compressing or not. To characterize all hearing aids which may be characterized as linear, slowly compressing, fast compressing, and or expanding, the hearing aid also must be measured using the same input signal as described above, but with an overall RMS level of 80 dB SPL. Optionally the hearing aid may also be measured with an overall RMS level of 55 dB SPL. All levels are measured as specified in Section 5.3. This low level may however conflict with noise floor of the measurement system.

7.1.1 Frequency range

Measurements with 1/ 3 octave band level determination with centre frequencies from 250 Hz – 6.3 kHz.

7.2 Analysis

Both the input signal and hearing aid output are measured. An overview of the analy~~i~~sis method is shown in Figure 8.

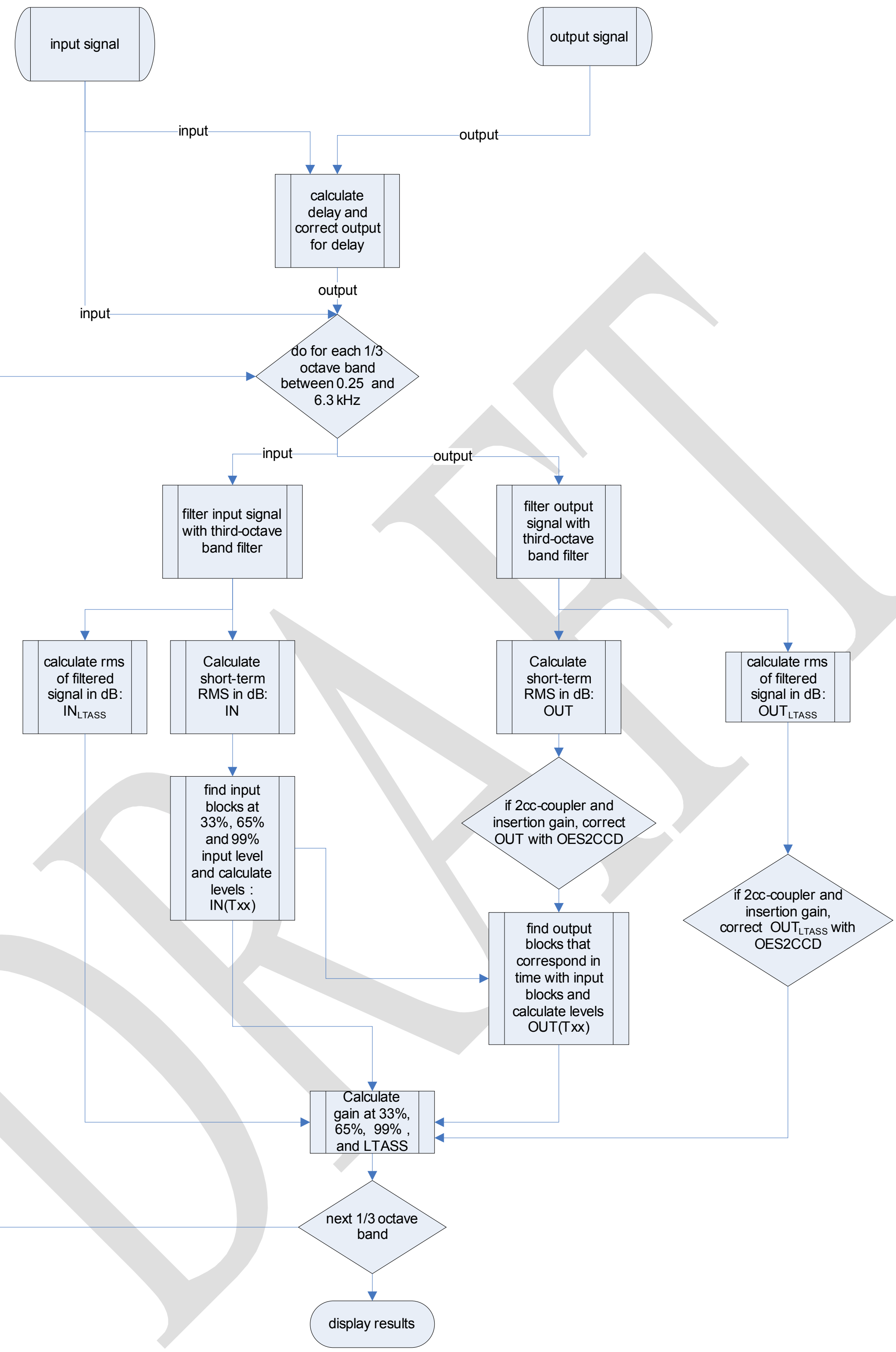

Figure 8: **Overview of analysis.**
Frequency indpendent time-alignment of output signal to input signal based upon a broad band cross correlation method (SectionTime alignment of input & output signal7.2.1) ,
Filter input and output signals in 1/3 octave bands (Section 7.2.2),
Calculation of LTASS of input and output (Section 7.2.3),
Determination of short-term RMS values for input and output by ~~time windowing~~segmenting in frequency independent ~~window~~ sections with a length of 125 ms (Section 7.2.4).

Correction of output signal for OES2CCD (Section 7.2.5),
Determination of percentile levels (Section 7.2.6)
Calculation of gain (Secton 7.2.7)

Below are some further details listed.

7.2.1 Time alignment of input & output signal

Due to a processing delay of most digital hearing aids the input and output must be time aligned before analysis. The method for this is based upon a broad band cross correlation method. I.e. the output is time aligned to the input signal.
The time alignment used, i.e. how much is output shifted must be specified for reproduction purposes and should be accurate within 10 ms. It is important that the time shift specified only represents the hearing aid, not the measurement system used.

For LTASS determination no time alignment is needed. Based upon the figures below the amount of time which signal the hearing aid output y(t) has to be shifted is represented. τ_{shift} represents the absolute value where the cross correlation has its maximum.

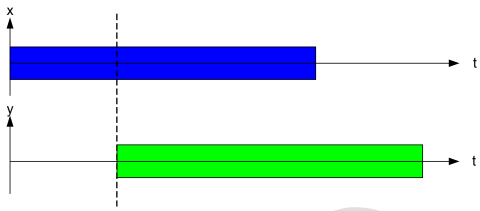

Figure 9: Input (blue), output (green) before shift

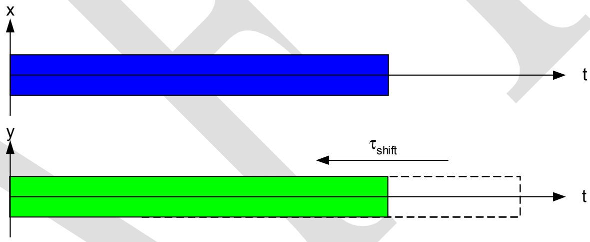

Figure 10: Input (blue), output (green) after shift

XX FR, Siemens to provide a updated graphical overview

7.2.2 1/3 octave filter

The input and output signal are filtered with 1/3-octave band filters for each 1/3 octave band from 250 Hz-6.3 kHz, where the 1/3 octave band filters are as specified in Section 5.6.

7.2.3 LTASS

Based upon the filtered time signals, the average RMS value of each complete 1/3 octave band time series are determined. The average RMS value of the input signal in dB is IN_{LTASS} and the average RMS value for the output signal in dB is OUT_{LTASS}.

7.2.4 Short term RMS values

Based upon the filtered time signals, the short-term RMS values of the input and output signals are determined. ~~The calculation of short term RMS values involves parameters such as integration period, overlap factor of windows, window lengths etc. These aspects are addressed below:~~
First the filtered time signal is segmented into sections as shown in Figure 11.

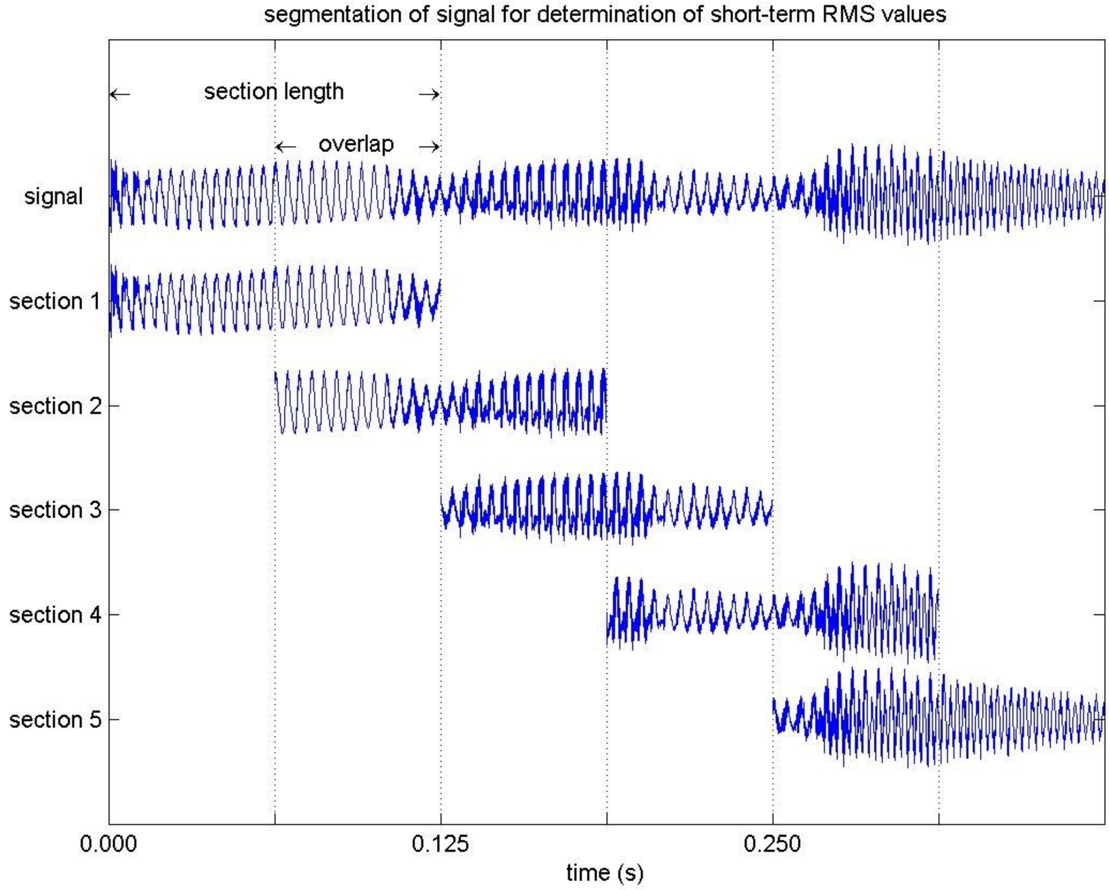

Figure 11: Segmentation of signal for determination of short-term RMS values.

The calculation of short term RMS values involves parameters such as integration period, overlap factor of sections, section lengths, window functions etc. These aspects are addressed below:

7.2.4.1 ~~Window~~ Section length

The ~~window~~ length of the sections ~~lengths~~ are frequency independent and for each 1/3 octave band from 250 Hz – 6.3 kHz equal to 125 ms [14]. ~~ee.~~

7.2.4.2 ~~Window~~ Section overlap

The sections overlap with 50%. ~~Overlap: 50%~~ I.e. 2 time ~~windows~~ sections pr. defined ~~window~~ section length.

7.2.4.3 Window ~~shape~~

A r~~R~~ectangular window is applied to each section.

7.2.4.4 RMS calculation

The short term RMS values are the RMS values of each section of the filtered time signal in dB.

The short term RMS values for the input signal is IN and the short-tem RMS-values for the output is OUT.

7.2.5 Correction for use of 2cc-coupler.

7.2.5.1 Preferred: Insertion Gain

Results comparing 2cc and IEC711 coupler measurements will differ to each other mainly due to differences in receiver load. The measured output signal will be corrected to simulated eardrum dB SPL by applying an OES~~D~~2CCD [1] transformation.

Table 2 shows the recommended OES~~D~~2CCD values.

f	250	315	400	500	630	800	1000	1250	1600	2000	2500	3150	4000	5000	6300
OES2CCD HA-1	4	4	4	4.2	4.3	4.5	5.2	6.1	6.6	8	9.3	10.5	12.2	13.6	14.7
OES2CCD HA-2	4	4	4	4.2	4.3	4.5	5.2	6.1	6.6	8	9.3	10.5	12.2	13.6	14.7

Table 2: Recommended OESD2ccCD values as function of third-octave frequencies.

The values for HA-1 are from [14],[14] and the values from HA-2 are from [14], [14].

If another OES~~D~~2CCD correction is applied to the measured output of a 2CC coupler, the correction must be specified.
More background information on this topic is available in Annex 2: Real Ear Measurement comments

The correction is as follows:

$$OUT = OUT + OES\cancel{D}2CCD;$$

and

$$OUT_{LTASS} = OUT_{LTASS} + OES2CCD.$$

where OUT are the short-term RMS values in dB for a specific 1/3 octave band of the output signal,OES2CCD is the OES2CCD value for that specific 1/3 octave band, and OUT_{LTASS} is the LTASS of the output signal of that specific 1/3-octave band in dB (Section 7.2.3).

7.2.5.2 Optional: Coupler gain

Coupler gain depends on the kind of coupler and therefore no correction will be applied.

7.2.6 Calculation of input levels of percentiles.

Based upon the short term RMS-levels of the input ~~analysed input data~~, i.e. IN(N) in each 1/3 octave band, the 30% percentile, LTASS, 65% and 99% (peak) percentile & levels are calculated.
From the ~~analysed~~ short term RMS-levels of the input ~~data levels~~, the ~~data~~ sections where the input levels are within +/- 3 dB from 30% percentile, 65%, 99% (peak) percentile levels are determined, T_{30}, T_{65} and T_{99}. I.e. IN(T_{30}), IN(T_{65}) IN(T_{99}) where T_{XX} is a subset of N.
The filtered OUT data of ~~at~~ the ~~equivalent input levels~~same sections are determined, i.e. the output levels OUT(T_{30}), OUT(T_{65}),OUT(T_{99}) are determined.
See graphical description below:
Look up ~~time slots,~~sections, where 30th, 65th and 99th percentile occur in the input signal:

[1] The difference between a real-ear measurement and coupler measurement is called a Real Ear Coupler Difference (RECD). In this case, it is a Occluded Ear Simulator 2cc Difference. Hence (OES~~D~~2CCD)

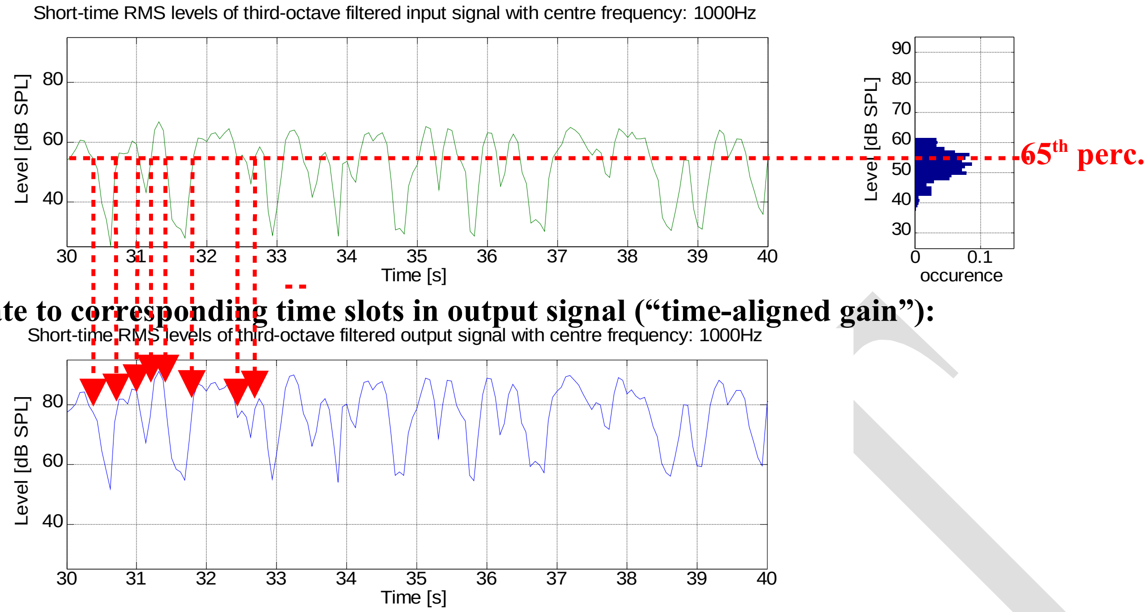

Figure 12: Illustration of the method for obtaining "time aligned gain"

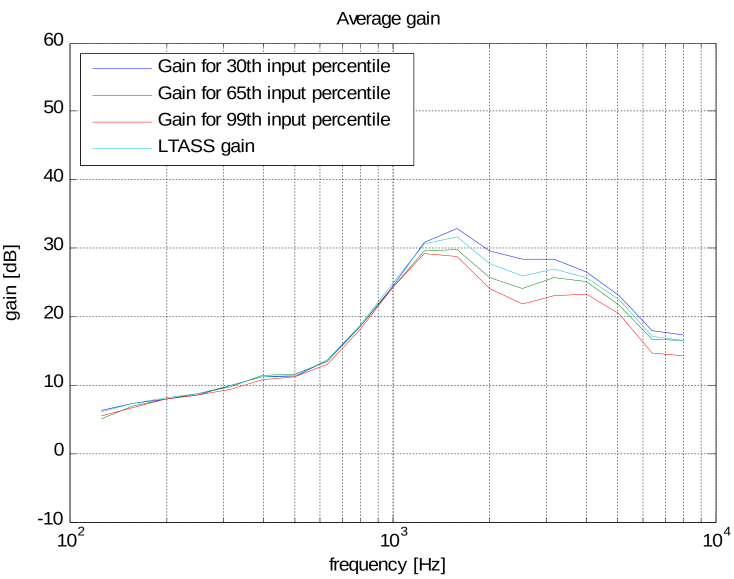

Figure 13: Average gain for 30th, 65th and 99th input percentile for each band

7.2.7 Gain calculation

7.2.7.1 Preferred: Insertion gain

The insertion gain is calculated using the input signal, as measured in Figure 1, and the output signal, as measured in Figure 2.
Based upon these input and output,~~s~~ the gain for the sections Txx is determined as:

$$GAIN(Txx) = OUT(Txx)-IN(Txx).$$

From this gain, the average gain at percentile xx is determined by averaging the gain of the sections Txx energy-wise:

$$Gain~~(T_{XX})~~ = 10*log10(mean(10^\wedge(~~OUT~~GAIN(T_{XX})/10~~- IN(T_{XX})~~))).$$

Regarding LTASS determination, the complete measured time series (excl. stabilisation time) is filtered in 1/3 octave and the average RMS levels are determined. I.e. the corresponding LTASS formula looks like this:

$$Gain_{LTASS}= OUT_{LTASS}- IN_{LTASS}$$

7.2.7.2 Optional: Coupler gain

The coupler gain is calculated using the input and output signal as measured in Figure 3.
Based upon these input and output, the gain for the sections Txx is determined as:

$$GAIN(Txx) = OUT(Txx)-IN(Txx).$$

From this gain, the average gain at percentile xx is determined by averaging the gain of the sections Txx energy-wise:

$$Gain_{XX} = 10*log10(mean(10^\wedge(GAIN(T_{XX})/10))).$$

Regarding LTASS determination, the complete measured time series (excl. stabilisation time) is filtered in 1/3 octave and the average RMS levels are determined. I.e. the corresponding LTASS formula looks like this:

$$Gain_{LTASS}= OUT_{LTASS}- IN_{LTASS}$$

~~N.B: Section 5.3.1 has explained that IN and OUT have been measured with different signals depending on the measurement method~~

1. ~~Preferred: Insertion gain~~
   - ~~IN: the recording of the reference microphone to the OEG-filtered input signal~~
   - ~~OUT: the recording of the coupler microphone to the MLE-filtered input signal~~
2. ~~Optional: Coupler gain:~~
   - ~~IN: the recording of the reference microphone to the free-field (=unfiltered) input signal~~
   - ~~OUT: the response of the coupler microphone to the free-field (=unfiltered) input signal~~

8 Data presentation

The gain data calculated, i.e. gain at 30% percentile, LTASS, 65% & 99% percentile input levels are presented on the same graphs, clearly identifying each data set. See example below:

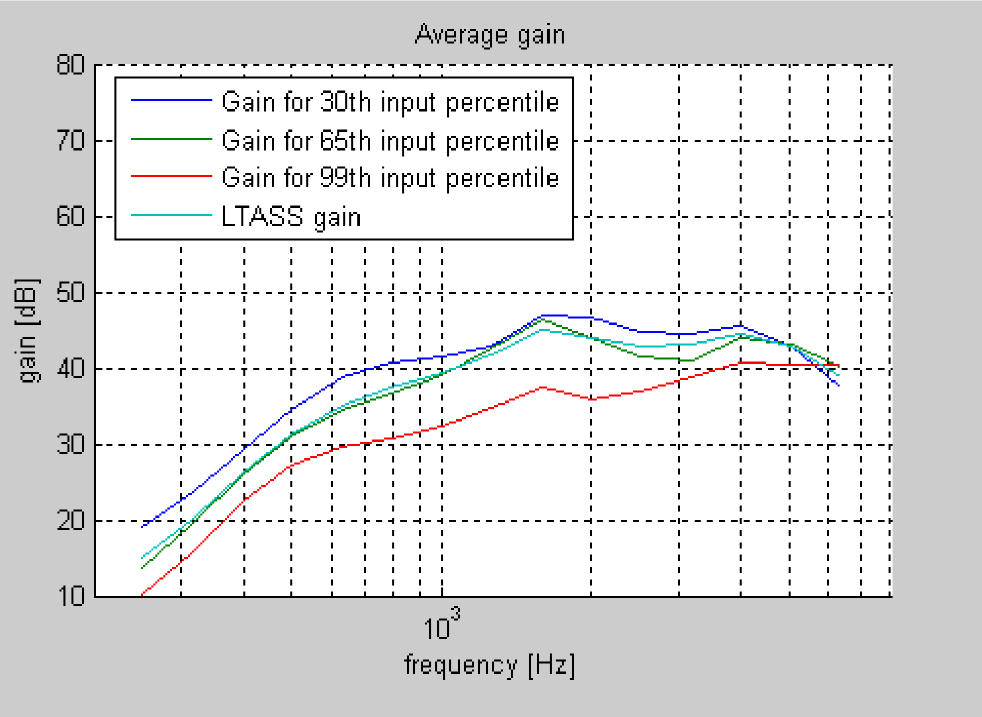

Figure 14: Gain illustration

It is recommend showing the LTASS gain using the different input signal levels simultaneously in the same graph. This will provide increase knowledge of compression system in the tested hearing aid.

8.1Which gain views can be expected?

In general a hearing aid with linear gain will show identical data for all data sets, and the larger the compression ratio the larger the spread of gain will become. A compressing hearing aid with slow time constants will show identical gain for all data sets. This compression type can be illustrated when comparing gain with different input signal levels.
The faster the time constants are in a compressing hearing aid the larger the gain spread for the different percentiles will be.

9 Annex 1: Pure tone vs. speech-like signal measurements

The frequency response of hearing aids has traditionally been obtained with a swept pure tone input signal whose level is held constant while the system output is monitored over the frequency range of interest. Some standards exist which utilises a time-stationary, steady-state broad-band noise, which is more typical of the complex input signals that hearing aids are required to process in non-laboratory real-world environments. However, to date, no standardized document exists that denotes procedures for testing hearing aids with a **modulated** broad-band input signal.

The intent of the present document is to describe the frequency response of hearing aids in their mode as presented to an average end-user, regardless of whether automatic signal processing such as AGC, noise reduction etc. is incorporated or not. Care in selecting the most appropriate methods for characterizing e.g. AGC systems is only one manifestation of a growing awareness in hearing aid measurements: the more sophisticated signal processing techniques are employed such as, for example, frequency-selective input or output compression, the more the selection of appropriate measurement signals and measurement techniques becomes crucial to realizing the goal of obtaining meaningful performance measures.

Frequency response curves developed for hearing aids with level dependent frequency response, or for AGC hearing aids with frequency dependent compression threshold using swept pure tones at varying input levels may not be representative of the response using complex signals. This occurs with the swept-tone method because only one frequency is presented at a time and the control system responds to each frequency individually. In the case of AGC hearing aids with frequency dependent compression threshold tested at high input levels, compression may vary with frequency, producing a flattened frequency response curve not representative of the response obtained with a complex input signal.

For an input signal more representative of real use conditions, such as speech, many frequency components are present simultaneously. The test signal specified in this standard has spectral characteristics similar to those of the short-term spectrum of speech and is representative of other important characteristics such as the temporal nature and the amplitude probability distribution of real speech. With this signal, e.g. an AGC detector will respond to a single level, from contributions at many frequencies, not to the individual frequency components. Thus, with this signal, the individual frequency components affected by an AGC loop will retain their relative amplitude relationships.

The new measurement method characterizes the hearing aid performance when applied with a signal with dynamic characteristics of real speech. This new method characterizes the hearing aid performance as used by the end user, to a greater extent than previously available in any proposals or standards.

10 Annex 2: Real Ear Measurement comments

The proposed measurement method can be adopted for real ear measurements. It is not supported in the proposal mainly due to the increasing possibility for variability of the measurement results. I.e. reproducibility decreases when applied in real life.

Some of the aspects involved in real ear measurements compared to the proposed simulated real ear coupler measurements are listed below:

0 degrees vs. 45 degrees
Real ear measurements are performed with both 0 and 45 degrees sound incidence. The used MLE and OEG in the proposed measurements method are based upon 0 degrees sound incidence. This will cause different input levels applied to the HI and a difference in the insertion and/or coupler gain obtained.

OES2CCD (i.e. Std. RECD) vs. real RECD
An individual real ear has differences to an ear simulator. The impedance termination/ load of the hearing aid will be different in the two situations. The RECD measured upon a real person will be different form the specified in this measurement proposal. According to [14], the standard deviation of RECD is 1.5 dB for frequencies below 1 kHz and it increases linearly to 5dB at 7 kHz.

Standard OES2CCD vs. hearing instrument specific OES2CCD
The OES2CCD of an individual hearing instrument can differ from the standard OES2CCD. Measurement has shown [14], [14], and [14] that variations up to +-4 dB can be expected.

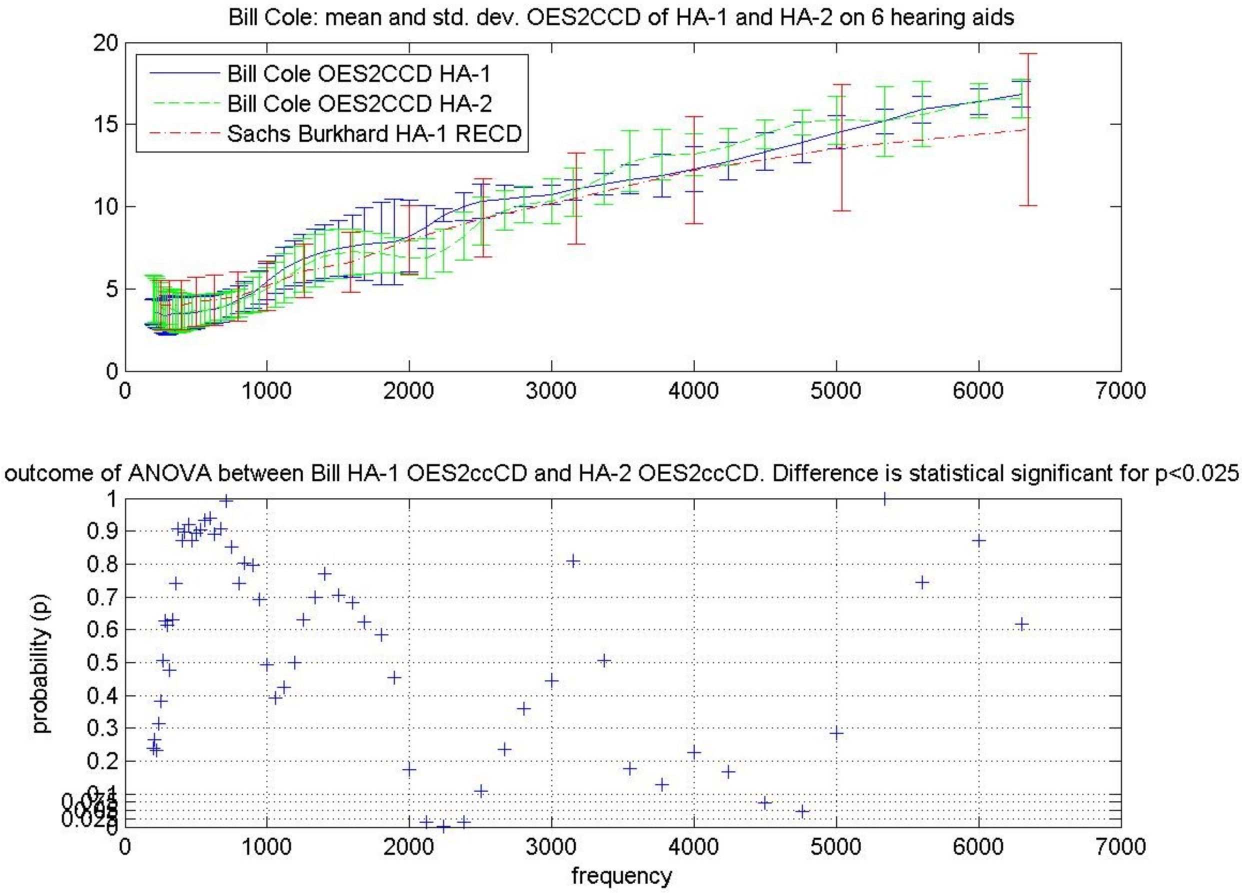

Figure 15: Mean and standard deviation for OED2CCD (from [14]) and RECD (from [14]).

<u>MLE variations average vs. real persons</u>

Individual persons and individual HI from different manufacturers will produce exactly the same MLE data as used in this proposal. I.e. this will cause different input levels applied to the HI and a difference in the gain obtained.

<u>OEG variations</u>

Individual OEG can be different from the OEG used in the proposal. I.e. real ear insertion gain measurements may show differences, often due to different peak positions of the OEG.

<u>Vent effects</u>

The measurement proposal assumes a 100% closed system. Real ear fittings all have some sort of leakage, either from an intentional vent and/or from leakage around the mould. Output SPL from the hearing aid at especially low frequencies will be reduced due to leakage. The measurement method proposes using a closed vent in the fitting software (if available), so that if any vent compensation effects applied in the fitting software will not show up on the measurements when measured in a coupler.

<u>Insertion effects</u>

Custom (ITE, ITC, CIC) hearing aids are positioned in a (IEC 711 or 2cc) coupler such that the earmould or earshell terminates in the reference plane of the coupler. The reference plane is a plane at 13 mm from the eardrum. Hearing aids that are inserted in actual use beyond this reference plane (i.e. closer to the eardrum) will behave differently on a real person than on a Occluded Ear Simulator or 2cc coupler. (from [14])

Bottom-line is that individual measurements can show large deviations if results are compared to measurements with the proposed method. This fact is un-avoidable with the proposed measurement method.

11 Annex 3: Calculation of percentile levels

Below is a Matlab function (http://www.tech.plym.ac.uk/spmc/links/matlab/matlab_statistics.html) of how to determine the level that represents a given percentile.

function perc = percentile(X,Percentile)
"X" is the data series, i.e. the short term 1/3 octave band RMS values
"Percentile" is the wanted percentile

Nx = length(X); %Determine sample size

% STEP 1 - rank the data
y = sort(X);

% STEP 2 - find k% (k /100) of the sample size, Nx
k = Percentile/100;
result = k*Nx;
perc= y(round(result));

% STEP 1 - rank the data
y = sort(x);

% STEP 2 - find k% (k /100) of the sample size, n.
k = Percentile /100;
result = k*Nx;
perc = y(round(result));

12 Annex 4: Questions and answers

Questions and Answers on the ISMADHA method for testing hearing aids with a speech-like signal Version 1.0 ; 21 June 2006

General

1. Q: How is the new procedure (and text signal) better than existing procedures that use ANSI or ICRA noise?
   A: The method can show that gain in some HI's is time variant, it can illustrate compression and the influence of different time constants. The largest benefit of the method is that a new, "best ever" input signal is provided, that makes existing signals redundant for test purposes with modulated signals. Furthermore the proposed method uses the setting of the hearing aid as worn by the end-user.

2. Q: How is the new testsignal an improvement over other testsignal such as ICRA noise?
A: None of the test-signals used in the past have the same properties in all regards as real speech. These properties include characteristics like voiced/unvoiced, speech pauses, formant transitions, modulation spectrum and many more. The new signal will also be representative of multiple languages.

3. Q: How is the new procedure an improvement over current IEC and ANSI test standards?
   A: The existing test procedures are primarily testing electro-acoustical properties of the hearing aids and not testing the sophisticated signal processing offered by modern hearing aids. The new procedure also includes the fitting software that is an integral part of the system and reflects how speech is amplified to the user of the hearing aid.

4. Q: How is this procedure to be used in view of existing standards such as ANSI and IEC?
   A: The new procedure is primarily intended as a supplement to other standards to offer more pertinent characterization of the hearing aids. It could be applied also for verification purposes, but for a quick quality check of a hearing aid the existing procedures are still very useful.

5. Q: Does the new procedure take into account the fact that some hearing aids have frequency responses that are dependent on the spectrum of the signal that they amplify? **A: Not as such, but such hearing aid functionality is mostly directed towards adapting the response to different types of input signals such as traffic noise, party noise etc. There could be individual differences between speakers as well, but such differences are more subtle and should not cause significant changes in hearing aid behaviour.**

Application

1. Q: Will it be possible to use this procedure in the free field using KEMAR or real ears?
   A: Yes REM system manufacturers may support this usage for the new method. It will not be part of the proposed norm here, mainly due to reproducibility issues.
   For REM usage you have to set MLE=0 and make sure that the SPL at the head is correct (the loudspeaker shall be free fleld equalized).

2. Q: How to measure for instruments intended for open ear mould?
   A: This is not directly supported. The instrument will be characterized as being occluded. The main reason is that the effect of an open ear-mould can not be measured correctly in a test-chamber
   Alternatives to get data for open ear mould:
   - **Make measurements using REM equipment on KEMAR.**
   - **Measure the response using an open earsimulator. By that the vent-loss is considered correctly, but there will be a small error in the direct sound contribution. This error is due to an incorrect MLE for the direct sound path to the open mould.**
   - **Measure the response on the occluded ear simulator. From that correct for the vent-loss and the direct sound input. This would require a reasonable model to calculate the vent-loss and direct**

sound (refer to REOG).

3. Q: The procedure specifies omnidirectional mode only, but many people will want to use it to assess directional performance using two signal sources in the sound chamber, one for speech the other for noise?
A: This is not the intended measurement because no noise has been defined and directionality performance validation is not the objective of the method. However such such method could be possible.

4. Q: Can the procedure be used for speech mixed with noise?
A: Not the intended measurement. But will be possible in principle. However the the results will be inconclusive, as it will depend on the type of noise and on the effects of special signal processing algorithms.

5. Q: In which extend the measurement method may give different results when using different settings/methods of noise-suppression, also when a pure speech test signal will be used?
A: This will depend on the type of noise reduction algorithm (fast / slow acting) or soft squelch. The results can be different to a situation without Noise reduction. An ideal noise canceler should not influence the measurement results for a clean speech signal.

Measurement

1. Q: How is the acoustic signal presented so that it represents an exact copy of the CD signal? This requires the equalization of the speaker source in both frequency response and in transient response in order to achieve results that are repeatable from one location to another.
A: The frequency response can be equalized very well with FIR filters that keep the phase characteristics intact. The phase effects due to speaker characteristics and sound propagation should be kept minimal; These effects cannot be controlled in practice.
Note also the following:
   - **Any phase distortion within the sound field is the same for the input and the output signal.**
   - **Phase responses have no big influence on the results, as only RMS level values are analyzed.**

2. Q: How is the correction for the microphone position made?
A: The measurement method includes the use of the Microphone Location Effects (MLE) for BTE, ITE and CIC according to IEC 118-8(2005).

3. Q: Why are the input levels 55, 65, and 80 dB. ?
A: The choice of the input levels has been based on the following requirements.
1. input levels should include an LTASS level which is based on a well known reference.
2. input levels should be spread over a good range so that the level-dependency of a compression system can be measured while still not being too sensitive to noise or clipping.
3. input levels should correspond to the notion of soft, normal, and loud speech which is frequently used in fitting software.

Ad 1. Levels of female speech in quiet range from 55 to 60 dB (see [14], [14], [14], and [14]). It should be noted that there is a large spread in the levels and a large standard deviations between recordings.
Ad 2.: Previous measurement have shown that 55 dB is the lowest level at which measurements can be performed reliably without too much noise interference. The difference between levels should be at least 10 dB to get a sufficient range
Ad 3.: In Pearsons study [14], the levels of casual (50), normal (55) , raised (63) , loud (71), and shout (82) speech have been measured.
In fitting software, there is also a notion of soft, normal, and loud sounds, where soft is between 40-50 dB, normal is 65, and loud is 80-90 dB>

From the requirements and observations, the input levels have been chosen to be 55, 65, and 80 dB.

4. Q: What type of analyzer is to be used? While a 1/3-octave band analyzer in some ways approximates the hearing system in humans, it does not lend itself to good transient analysis. The short term FFT is also limited in this respect.

A:The transient response is not used for the characterization of hearing aids. Only the short-time RMS levels of input and output signal are analyzed. The amount in which the signals are distorted in phase due to the analysis method is the same for both the input and output signals.

5. Q: Will the analysis be real-time or FFT-processed?
   A: For commercial implementations also an FFT approach can be used as long the filter charactersitics are conform the specified charateristics.
   Note: The filters as used in the reference matlab implementation("filter.m") are in the time-domain. That filter is a direct form II transposed implementation of the standard difference equation, which is documented in, Oppenheim, A. V. and R.W. Schafer. Discrete-Time Signal Processing, Englewood Cliffs, NJ: Prentice-Hall, 1989, pp. 311-312.

6. Q: Why using rectangular windows?
   A: Rectangular is fine with filter-banks. Hanning or other windowing is needed for FFT analysis. The Ismadha reference implementation is filtered in the time-domain, after which 125 ms portions are "snapped" (rectangular window) to perform RMS analysis on.
   When using FFT filtering the error from the window function used before FFT analysis should be corrected to be within the required filter specifications. The choice of a given window function is a matter of pros and cons, with respect to stop band attenuation and compared to pass gain.

7. Q: Is there any issue of phase distortion when filtering with filter banks?
   A: Mainly due to the use of RMS analysis and using 125 ms ~~windows~~ sections of the filtered signals, the phase is not an issue

8. Q: Why we need overlap processing? Is this really needed when using rectangular windows and using gain/output calculations only?
   A: The overlap processing of 50% is included in the procedure to increase the number of RMS measurements, so the increase accuracy without increasing measuring time. By 50% overlap the signals are reanalyzed with a shift of 0.5 x 125 ms = 62.5 ms.

9. Q: Will a special test-mode be required and why?
   A: No, the proposal specifies that all HI setup should be available as in standard fitting scenario. I.e. only fitting software handles that normally are available. However the proposal allows to disable special functionality such as noise reduction etc. when it affects reproduceability of the measurements.

10. Q: How strong is the advice to use the OES coupler? How much error will be made by using a 2cc coupler with standard RECD correction (OED2CCD) in comparison to using OES?
    A: Both a 2cc coupler, a IEC ear simulator and a Zwislocki coupler can be used. The "errors" obtained when comparing the measurements data methods is expected to be within +/- 4 dB. Reproduceability is of essence. I.e. all details regarding measurmentent setup, settings and signals must be provided together with the results.

11. Q: Why are there no different OES2CCD corrections for HA-2 and HA-1 2cc coupler?
    A: Although literature ([14], [14], [14]) shows differences between HA-1 and HA-2 couplers, this difference is not statistically significant in comparison with the difference due to the individual hearing instruments.

12. Q: Why a fixed ~~window~~ section length of 125 ms is used. What would be the effect when using a different ~~window~~ section length?
    A: The ~~window~~ section length of 125 ms is a psychoacoustically (loudness perception) motivated value [14]. Lowering the ~~window~~ section length will (e.g. for fast HI time constants) increase the measured output variations, which will show up as larger percentile gain variations.

Test-signal

1. Q: Why a female test signal will be recommended for testing?
   A: female test signal will be recommended because its LTASS/pitch is between male and a child.

2. Q: How and in which extent the results will depend on the chosen signal (in particular when presentation is based on signal percentiles)?
   A: The results will depend on the chosen signal, because the signal is used to measure a system that is (or can be) non-linear. The use of a specific signal will be part of the measurement method. We expect that this specific signal will give a good characterization of the most common multi-channel compression systems with regards to linear/non-linear (compression and expansion), fast/slow compression. This is also one of the reasons that we will use this signal only to characterize the hearing aid and NOT to verify or validate the hearing aid.

3. Q: For comparison purposes, can the measurement procedure itself be applied to ANSI or ICRA noise or will it require the use of the new signal only?
   A: Yes, it would be possible to use other signals, but the results will be dependent on that signal. The use of other signals is not supported by the new measurement procedure. Note that the use of a non-modulated signal would result in gain percentiles that will be very coincident, making not much sense to use that kind of signals.

Presentation and Results

1. Q: What do you do with the results? How will "good" results be defined? If "bad" results require that the aid be adjusted, how do you know when satisfactory results have been achieved? Will the data be quantified by something like an SII or only by gain and output curves?
   A: This method is not intended to be used for verification or validation of a hearing aid. So whether the results are good or bad is not the issue, the main purpose of the method is to show the gain of the HA and give an impression of how the HA works for speech sounds. The data will not be quantified by an SII.

2. Q: Why the used audiogram will not be presented in simulated REAR graphs?
   A: Only gain curves are shown with this method. This removes focus from audiograms and thresholds and puts focus on how the HA works for a speech signal.
   For adding the audiogram it would also require some agreement on which corrections to be used when converting 1/3 octave broad band levels to pure tone audiogram levels.

3. Q: Why using percentiles instead of input SPL levels?
   A: With the use of percentiles one does not need to define absolute SPL levels for the different parts of the speech signal. These levels are estimated automatically with the defined percentiles. Once the input signal is a fixed signal it is possible to show the corresponding input SPL level.

4. Q: Why not present REAR using output percentiles?
   A: If only the output percentiles where used, one would not be able to relate a given output percentile to a given input level. The information about how different input levels result in different outputs is lost using output percentiles. Also if the output percentiles were determined without using the link to the input percentiles, the gain would not be "time aligned". I.e. the gain represented would not shot the actual gain in the HI.

5. Q: Why using 30% percentile instead of say 5%; also 95% instead of 99%; also why there is a difference of 50% and LTASS?
   A: This answer depends on the final input signal. The 30% percentiles give a good detection of the soft/weak parts of the speech. If a too low percentile of say 5% is used the pauses would be represented instead. This would result in an output showing the gain of the pauses instead of the soft/weak parts of the speech. By definition the 99% percentiles are used for detection of the speech peaks. Pauses in the signal is the main reason for differences between 50% percentiles and LTASS.

6. Q: Is it possible to use this method for hearing aids using frequency shifting?

A: No, this method cannot be used for hearing aids where information in one or more frequencies are shifted/moved to other frequencies. For hearing aids where frequency shifting is optional this must be turned off.

13 Annex 5: Definitions

Term	Description
LTASS	Long Term average Speech spectrum
Percentile	A 30% percentile used in the documents context indicates the level, which the remaining 70% of the measured levels are higher than. I.e. a 99% percentile is a peak indicator.
OEG	Open Ear Gain
OES2CCD	Occluded Ear Simulator 2CC Gain
RECD	Real Ear Coupler Difference

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