Test signals and Measurement Procedures for the Evaluation of Modern Hearing Aids

Hans Wilhelm Gierlich
HEAD acoustics GmbH, 52134 Herzogenrath, Germany, H.W.Gierlich@head-acoustics.de

Summary

In modern DSP-based hearing aids a variety of signal processing procedures can be found. Basically they can be separated in algorithms having long term effects and algorithms having instantaneous effects, e.g. working on syllables of voice signals. Furthermore most of the procedures work in different frequency bands.

For the evaluation of the system performance -besides speech signals- different test signals are discussed which more reliably may be used in order to evaluate the system behavior. Different test signals, providing different speech like properties are introduced. Based on these test signals analysis techniques can be applied which are different to the ones used traditionally for hearing aids. The evaluation of hearing aids using the proposed test signals and procedures is shown.

Signal Processing in Hearing Aids

Modern hearing aids typically contain non-linear and time variant signal processing. Typically hearing aid concepts include filter banks which allow to adjust the individual parameters band-selective in order to meet the personal requirements of the hearing impaired. Within each band it is typically possible to adjust amplification and various types of compression/companding schemes. Long term and short term compression (effective within a syllable) may be realized. In addition algorithms for improving the background noise transmission performance of hearing aids (background noise reduction) are used. These may either work independently of the signal processing discussed before or the same bands as the other algorithms may be used. Background noise reduction further may be combined with acoustical measures such as directivity microphones. In addition all hearing aids have to provide suitable means to suppress/ avoid feedback between microphone and receiver. Again band selective attenuation, notch filters or specific types of echo cancellers might be used.

Test signals

Since modern hearing aids are designed to optimize the transmission of speech signals for hearing impaired listeners, test signals should represent to a high degree the characteristics of speech. On the other hand testing requires well defined signals easy to reproduce and leading to consistent and valid measurement results. Therefore the following properties of speech have to be taken into account [3], [4]:

- spectral distribution
- short and long term spectra (voiced/ unvoiced, fundamental frequency variation, formant filtering …)
- power density distribution
- time structures

From the measurement point of view the following parameters have to be taken into account:

- defined signal level
- defined spectra, spectral distribution as even as possible
- signal energy as high as possible

A variety of test signals taking into account the constraints discussed above have been constructed. Level and duration of the different test signals may vary depending on the application.

Test signal for determination of short term parameters

Fig. 1 gives an example for a test signal suitable for the determination of short term parameters. The signal is basically a composite source signal as described in ITU-T Recommendation P.501 [1]. The signal consists of 3 parts: voice sound (speech like), needed in order to activate speech controlled/ speech activated systems properly, followed by a pseudo random noise sequence which allows short term measurements of a system in a defined stage (quasi stationary observation of the system). This signals are followed by a pause which can be used to evaluate the background noise transmission or the system noise and which furthermore introduces an average amplitude modulation to the signal.

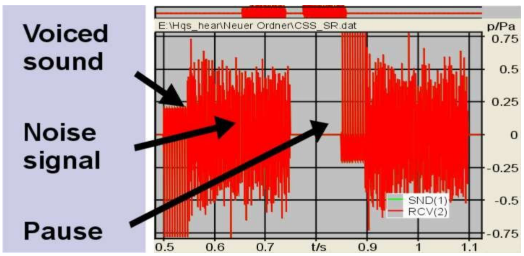

Fig. 1: Composite Source Signal [1]

Test signal for companding characteristics

The test signal for determining companding characteristics is constructed like the composite source signal, however the voiced sound is replaced by an artificial voice signal (1s duration). The main goal of this signal is to get a longer activation period for systems. The pseudo random noise sequence again can be used to determine reliably the insertion gain as a function of frequency and level. The spectral shape of the pseudo noise signal is similar to speech but provides sufficient signal energy at high frequencies for reliable test results (see fig. 2).

The test signal can be applied with various levels and with different level variations in order to determine the companding characteristics over the complete dynamic range.

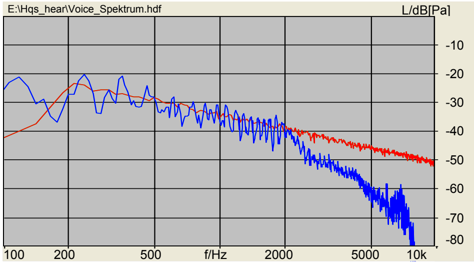

Fig. 2: Spectrum of the activation signal compared to the measurement signal
blue: spectrum of the artificial voice signal
red: spectrum of the pseudo random noise

Test signal for long term parameters

The test signal construction is similar, however, the artificial voice [2] is applied for a duration between 5s and 100s again followed by pseudo noise sequence. The main goal of this test signal is to check whether the system behavior changes after applying a speech like signal for a long period.

Test signal for determination the background noise transmission performance

This test signal is constructed using a period of 10 seconds speech simulating noise (random noise, spectrally shaped like speech) followed by 1s of artificial voice. The test signal can be used to evaluate the background noise reduction performance of a hearing aid as well as the ability of the hearing aid to react immediately on a speech signal after playing background noise.

Test signal for modulation distortions

Typically distortion problems occur when high background noise level is presented simultaneously with a speech signal. The test signal proposed for this condition is a time and frequency multiplexed test signal. A narrow band noise signal (100 Hz - 500 Hz) is applied during the whole test simultaneously with a speech like test signal (2.3 sec and followed by a sign wave (500 ms) for testing modulation distortions.

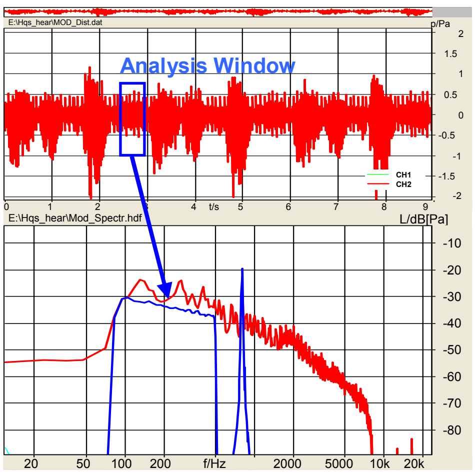

Fig. 3: Time signal (upper) and overall spectrum (red) of the test signal compared to the measurement signal (blue - sine wave with background noise)

The test signal guarantees a reliable activation of the hearing aid combined with a background noise signal and a sine wave test signal to measure the modulation distortion.

Analysis Examples

Companding Characteristics

A very important test for hearing aids is the evaluation of the level and frequency dependant amplification. A test signal as shown in Fig. 2 can be used. The signal is repeated with decreasing the average level of the individual signal bursts by 5 dB each and then increasing the level again by 5 dB for each step. The signal level at the hearing aid varies between 55 dB_{SPL} (A) and 80 dB_{SPL} (A).

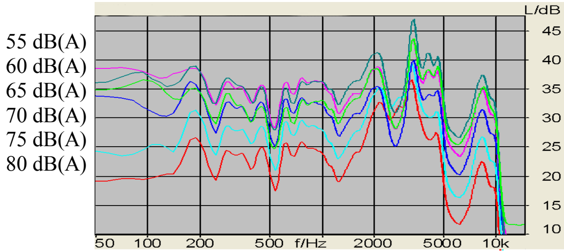

Fig. 4: Level and frequency dependant insertion gain for different speech signal levels

The response of one hearing aid setting on this test signal is shown in Fig. 4. The level and frequency dependant amplification can be seen clearly. At levels lower than 60 dB_{SPL} (A) the amplification is kept constant. The frequency dependant companding characteristics can be seen e.g. when comparing the insertion gain difference between 200 – 500 Hz (10 dB) to 5-10 kHz (15 dB).

Modulation distortion

Fig. 5 demonstrates the result of modulation distortion measurements using the test signal shown in Fig. 3, again for different signal processing settings of a hearing aid. It can be seen clearly, that the low frequency noise may introduce significant distortion components in the frequency range above 500 Hz. The distortion introduced to the sine wave in this example can be seen but is still acceptable.

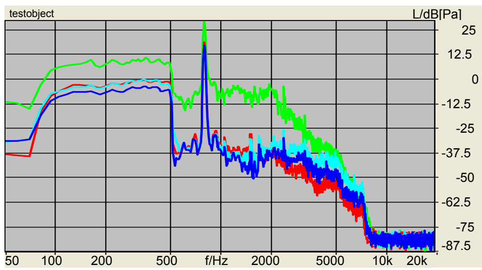

Fig. 5: Modulation distortion for different settings of algorithms of one hearing aid

References

[1] ITU-T Recommendation P.501: Test signals for use in telephonometry
[2] ITU-T Recommendation P.50: Artificial Voices
[3] Gierlich, H. W.: Adaptierte Messignale und Messverfahren für Hörgeräte, Erlanger Kolloquium, 6.-7.3.2003
[4] Gierlich, H. W.: A Measurement Technique to Determine the Transfer Characteristics of Hands-Free Telephones, Signal Processing, Vol. 27, Issue 3, 1992, S. 281-300