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A Geek and His Hearing Aids

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Self Test of Hearing Aid? For Geeks Only!

Every now and then one receives a communication from a reader that indicates that some people have a real concern about their hearing aid's performance and want to take it to the next level. Such communications show also that many of our hearing aid users have exceptional skills, often challenging those of us who fit hearing aids. This is the case of a reader who had stumbled across a post I had written on non-stability of directional microphones in hearing aids.

Reader Comments

My hearing got abruptly worse at around age 18, and I have worn hearing aids (several generations from the UK NHS – United Kingdom National Health Service) in both ears for the past 6?7 years (since I was in my early 30s). I have a 45?50 dB flattish loss. The instruments are behind-the-ear (BTE) units with open fit, although my newer ones are double-dome closed fit.

I do not use the directional program for everyday use, as I do not like the audio "blind spots" they create while in "normal" environments. I do sometimes manually switch to a directional option when in particularly noisy cafeterias etc. As a user, I've noticed firsthand that the degree of directionality – or rather the balance of apparent gain of the two aids (when in directional mode) gets worse over time (a year or three).

I also strongly suspect the gain/frequency response of a single microphone varies over that time period by enough to have a noticeable effect on my hearing performance. (I'm well aware that even partial obscuration of the tube with earwax changes the frequency response – and "clean" them regularly.)

The Request

I'd quite like to build a rig to be able to "test/calibrate" my hearing aids (HAs) from day to day – but never quite find the time. I’m a geek (sci-tech consultant designing – among other things ? medical devices in my day job). I’m a physics PhD, with an interest in sound and measurement.

I suspect there are long-term stability issues that are not well recognized by hearing-aid dispensers.

From my own experience there are also software issues which are not recognized by dispensers, such as being able to program (mine) aids with enough low-frequency gain to cause ghastly clipping distortions under certain listening conditions (e.g. church organs and amplified music in public buildings). I’ve also experienced very weird artifacts from the impulsive-noise suppression systems (audio dropouts when a distant door slams, hearing the reverb from a light-switch instead of the direct sound, my own keyboard noise being suppressed…) – and have had my audiologist turn this off as it was no real benefit to me, yet the artifacts were driving me insane.

Response

Two primary components would be required:

1. A coupler of some sort is needed to provide a constant load resistance for the HA measurement. Because the system isn’t likely to be a completely “Standards” based solution, it might be good to create something with a volume of 0.4 cc since that is the volume the HA industry is considering. The 2 cc coupler is not useful at frequencies higher than 8 kHz because the internal dimensions cause resonances above that range – hence the pending move to 0.4 cc. A 0.4 cc coupler results in approximately 14 dB higher gain and output levels in the high frequencies in comparison the current 2 cc volume standard.
2. A sound chamber to provide an input to the HA.

A major concern is one of calibration and stability. The measurement microphone technology will be of significance. The inexpensive Electret mic capsules are humidity and temperature sensitive. Knowles electronics makes MEMS mics that might be useful as the humidity stability is extremely good, but not sure on the temp side of things. The B&K or GRAS type lab-grade mics are perfect, but at $7K for a pair, including power supply and preamp, they make HAs look inexpensive. So, a more budget friendly solution is needed.

One way to reduce the drift issue, commented on earlier, is to use two microphones: one to monitor the HA output, and the other to capture the input to the HA microphone. The input mic should be placed in close proximity to the HA mic port. This way a differential measurement can be made that tends to be a bit more accurate if the mics have similar sensitivities to humidity and temperature.

The gain and frequency response of the HA is level sensitive so configuring the HA to be linear is best, but probably not easy unless the fitting software and programming hardware is available.

Of course, behind all this hardware discussion, a bunch of software is needed. The amount depends on the extent of characterization work intended.

Geek Reply

Thank you for your input. It seems that not much of what we do is easily managed, especially if repeatable measurements are to be made. I don't think that Joe Zwislocki will be happy with the 0.4 cc volume rather than the 0.2 cc volume of his original coupler. However, the difference should be very slight. The 0.4 cc should have similar response, including that at the high frequencies.

I bought some regular consumer-grade MEMS microphone modules a year or so ago with this project in mind. I was assuming to use them in a differential mode (which helps with room/enclosure calibration, but not if the microphone performances drift relative to each other – though in principle you could swap the mics for reference and output measurements mid-experiment).

Software-wise I'm more than capable – see also http://techmind.org/audio/ and http://techmind.org/audio/specanaly.html. Since (unfortunately) I don't have hearing-aid programming interface and software, I'd have to work-around the aids' "smart" features. Among other things, that probably means I'll need to measure frequency response using pseudo-speech (or actual recorded speech) rather than pure tones (or white/pink noise), to avoid interaction with the feedback-canceling system or background-noise suppression functionality.

Without having thought through the coupling in detail (e.g. "2 cc coupler"), I was thinking along the lines of using a short length of rubbery tube, perhaps 5?6 mm inside-diameter to couple the HA output "dome" to the microphone module. I suppose a metal tube may be more stable with respect to time and temperature though.

Hopefully I'll get back to this at some point. The issue of course is that you need to collect data when everything is "fine," so you've got reference for when you suspect something's "not right." Software-wise I'm more than capable.

Our Response

Based on your website you have enough science background to easily handle your goal of measuring the HAs.

Depending on which MEMS mics you have you may already have the basis for a very stable setup. The Knowles MEMS mics are very stable, and suspect the Analog Devices (inverSense) are as well. The challenge for generic measurement use is headroom ... on a 2 cc coupler some power aids will produce 140 dB SPL which will over drive most of the readily available devices. Going to 0.4 cc will make the problem 14 dB worse, which is a concern for sure. If you test at reasonable levels and check that the mics aren't clipping you should be good to go.

Your idea for a simple coupler made of a section of tubing sounds straightforward. It is suggested that you strive to only do measurements on a closed/sealed tube such as would happen if you use the double flanged domes you spoke of. An open system will complicate the measurements ... there is a reason all the HA standards use closed coupler configurations.

Pure tones etc. are being used simple as QC (quality control) measures these days. They provide little information on the actual performance of the HA.

ANSI S3.22 is the test standard to which the US FDA mandates HAs be tested to. There was a time when HA processing was simple enough that pure-tone test results in conjunction with some level of knowledge by the clinician was enough to tell how a particular HA performed. Sadly, both those assumptions aren't usually true anymore ? the later being of greater concern.

The S3.22 working group had members of the audiology community participating in an effort to provide meaningful measurements. But, with the introduction of many DSP (digital signal processing) products, the sound processing as measured by pure tones is pretty far from what a speech stimulus would produce. The latest version of the standard no longer requires attack and release time measures as those datasheet numbers tell you next to nothing about the underlying processing.

Back in the analog days attack and release values served two purposes: (1) help define the speed of compression, and (2) test that the time constant capacitor had been soldered in during manufacturing, and was approximately the correct value. Lots of capacitors are still used in a HA, but not for time constant type things. Today, compression speed is determined by a numerical coefficient, and can vary depending on the number of channels, the overall level of the incoming signal, and the properties of the incoming signal itself. The complexity of the interaction drove the S3.22 Working Group to drop the measurement as a requirement, you can still do it if you want though.

So, S3.22 and its IEC equivalent simply verify that all the components are installed and functioning within some parametric margin of normal. At some level, this is one of your questions. But, to get to the controlled/consistent state you have to put the HA into a simple mode (linear is ideal), but not easy in your case.

To address the shortcomings of testing using stationary signals, the ANSI and IEC technical groups have developed a speech based testing paradigm. IEC 118-15 and ANSI S3.42 part 2 are identical in method. During development, IEC took the lead on this standard and ANSI adopted it verbatim. Likewise, ANSI took the lead on real ear measurements with IEC adopting those verbatim. Everyone is trying to harmonize in an effort to save work during future product development.

The speech based testing standard says that the performance of a HA to a speech signal is a combination of the test stimuli, the HA algorithm, and the fitting software (which produces a set of parameters for the algorithm), which in turn is based on the audiogram input to the fitting software. It is a systems approach and called the ISTS (International Speech Test Signal).

Of course, the signal processing within the HA can be different depending on previous inputs, so in anticipation of this the stimuli is 60 seconds long and only the last 45 seconds is analyzed. This gives the HA 15 seconds to “warm up” so to speak (adapt in reality). If a manufacturer knows 15 seconds isn't sufficient time then the test stimuli can be concatenated with itself to create a 120 second long test file, but again, only the last 45 seconds is used for measurement.

The test signal is freely available at EHIMA's website in the documents section http://www.ehima.com/documents/. The specific link is http://ivo.ehima.dev02.accedo.dk/wp-content/uploads/2014/03/international_speech_test_signal_plus_terms_of_use_ists_v2.zip. Contained in the zip file are a few WAV files, including a calibration tone and a description of how the ISTS signal was created.

In your case you want to look at the stability of the results over time, well at least for a first step. It sounds like your interests will take you further then where you planned.