ASP Testing

with the FONIX 6500 I/O Option

by Larry Revit, hearing scientist

Frye Electronics continues its commitment to provide the tests required to analyze ASP (Automatic Signal Processing) hearing aids. ASP circuits are designed to decrease low-frequency gain as the level of low-frequency noise in the environment increases. One of the best ways to check this function of an ASP circuit is to run an input/output (I/O) curve using a low-frequency signal. An I/O curve will show you the kneepoint of ASP action (the SPL required to trigger the circuit) and the amount of gain reduction provided as the level of low-frequency noise increases.

However, the I/O test conditions required for ASP circuits are different from those for regular AGC circuits. AGC circuits have fast attack times, because they are designed to protect against any unwanted loud sounds, sudden or otherwise. ASP circuits have slow attack times. They are designed to be triggered by long-term, as opposed to transient, low-frequency energy. For example, you would want an ASP circuit to be triggered by the "drone" of a motor, but not by the "thud" of a door closing. This avoids the annoying "pumping" sounds you would get with a fast attack time. Because of the slow attack time of ASP, in order to get a proper measure of the action of an ASP circuit, longer stimulus times must be used. That is, there must be a delay between the onset of the test signal and the acquisition of the measurement of the response: You have to give the ASP circuit a chance to react before measuring it.

With software Version 2.38, the FONIX 6500 I/O Option allows you to select the delay time between stimulus onset and acquisition of measurement necessary for testing ASP. Delay is selectable via the I/O menu. The default setting is one-half second, which is long enough for many circuits. But some circuits take longer to react. With the 6500 I/O Option, the delay may be varied between 0.1 and 3.0 seconds. (The shorter times are for regular, fast-acting, AGC circuits.)

Real Ear Measurements for Prescription of Hearing Aids?

by Larry Revit, hearing scientist

Can hearing-care professionals use probe measurements in the prescription process, to improve the chances for a successful fit?

Yes, we can.

We are living at a time when communications and technology are showing some real benefits for everyone. Our field is no exception. The word is out: individual ears are different from one another. And those differences can largely affect the benefit a hearing aid provides to the wearer.

Fortunately, we now have a fast and reliable way to measure real-ear differences. In a few short minutes, we can check the real-ear insertion gain of a hearing aid using probe equipment.

But, can't we use probe measurements before the fitting stage, to prescribe the hearing aid reponse that we know will work better when the aid is first placed in the client's ear?

The answer is most certainly "yes". Since the earcanal resonance (a natural peak) is lost when an aid is inserted in the ear, a hearing aid must compensate for the lost resonance before it can provide any benefit to the wearer. Conversely, if the hearing aid has a peak at a frequency that does not correspond to the peak of the earcanal resonance, the fitting could be far too strong at the frequency of the "misplaced" peak.

We can easily get useful information about the client's earcanal resonance before ordering a hearing aid. Measuring the client's unaided ear canal resonance with Fonix probe equipment is fast and easy. We don't need precise placement of the probe tube; the probe tip just has to be a few millimeters into the earcanal to measure the primary resonance peak (in the region of 3 Khz). An effective rule of thumb, then, is to specify that the primary peak of the ordered hearing aid should match the frequency of the measured earcanal resonance. (The above does not necessarily apply to IROS fittings.)

Beyond this rule of thumb, we can look forward to conversion factors that effectively translate target insertion gain into prescribed coupler gain. Several sets of conversion factors have been proposed, but the extent of their clinical effecitiveness has not yet been demonstrated. When there is a proven set of conversion factors available, Frye Electronics will put them into Fonix real-ear analyzers. And, only when manufacturers can match our prescribed coupler responses, can any prescription routine be effective. Even then, (and all researchers agree on this), every fit will need to be confirmed by real-ear measurements.
Fortunately, we now have the technology and the know-how to improve hearing aid prescriptions and fittings. And soon we will have the means to make the process even easier.

Proper Use of Telecoils: It's as easy as 1,2,3...4!

by Larry Revit, hearing scientist

Since many of you are hearing-aid dispensers, and since many others have clients that are hearing aid dispensers, here's some information that can be passed on to the end useres: those hearing-aid wearers who use inductive telephone pickups, or "telecoils." Ususally, instructions for using telecoils call for three steps:

1. Move the switch on the hearing aid to "T".
2. Turn up the gain (volume) control.
3. Move the telephone receiver (earphone) around the hearing aid until the loudest signal is heard.

If the telephone is far away from any stray magnetic fields, that set of instructions is fine. But power wires, fluorescent lights, electric motors, video monitors, etc., all emit electromagnetism which can seriously interfere with telecoil use. Such interference often causes loud humming or buzzing in the hearing aid when the instrument is set to "T". Fortunately, electromagnetic fields are directional, and the proper orientation of the telecoil within such fields can minimize the detrimental effects. Just as moving the telephone receiver around the hearing aid can minimize the desired coupling of the telephone's magnetic signal to the telecoil pickup in the hearing aid, moving the hearing aid around in a stray magnetic field can minimize the undesired magnetic coupling of the hum or buzz from extraneous environmental sources. Therefore, instructions for the use of telecoils should contain four steps:

1. Move the switch on the hearing aid to "T".
2. Turn up the gain (volume) control.
3. If there is a hum or buzz in the hearing aid, move your head up and down and side to side until the buzz is the most quiet.
4. Move the telephone receiver (earphone) around the hearing aid until the loudest signal is heard.

As a hearing-aid wearer, myself, I almost always have to use the above four-step process. Often, people pass me in telephone booths with questioning stares... wondering why I'm standing with my head tilted over to the side. Sometimes it's inconvenient, but that tilt of the head can make the difference between disability and ability in using the telephone.

Use The In-Situ Option

by Larry Revit, hearing scientist

To Save Dispensing Time and Improve Initial Customer Satisfaction

The In-Situ Option lets the dispenser set the controls on a hearing aid to match a custom target insertion gain curve, before the client arrives! You create a target curve in Quik-Probe, and then use In-Situ with Multi-curve to set up the hearing aid to match. It's easy! Here's how you do it:

1. In Quik-Probe, enter the client's audiogram and select the appropriate target conversion. While still in Quik-Probe, use Multi-Curve to copy the target curve (CURVE 4) to REFERENCE 1.
2. Press [RESET] to go to the COMPOSITE WEIGHTED GAIN test-chamber mode.
3. Set the volume control on the instrument to a "use-gain" position. That would be about 10 to 15 dB below full-on gain. Occlude any vent and attach the instrument to the appropriate MZ coupler (preferably the MZ-1, using the client's custom earmold or shell).
4. Press [MENU] and then [INSITU] to call the In-Situ Menu. Select "ITE", "ITC", or "BTE" for SOURCE CORRECTION, and select "OES AND INSERTION GAIN" for OUTPUT CORRECTION. Press [START] to run the In-Situ Option. What you see is the estimated insertion-gain response of the instrument.
5. Set the source amplitude as will be used for verifying the insertion gain, once the client arrives. That would usually be about 65 dB SPL, the averaging long-term level of speech.
6. Now go to Multi-Curve and select DUAL SCALING. Choose CURVE 1 (the ongoing real-time chamber curve) to get the left scale and REFERENCE 1 (the target insertion gain curve) to get the right scale.
7. Now you're ready to adjust the hearing aid controls to match the target, while viewing the real-time esitmated insertion-gain curve.

HINTS:

- Adjust the overall (or mid-band) gain trimmer so the mid frequencies (1 to 2 KHz) match the target.
- Adjust the low and high tone (or band) trimmers so the low- and high-frequency slopes match the target.
- Adjust the resonant peak trimmer so the region between 2 and 4 KHz matches the target.
- Vary the source amplitude of the speech-weighted signal to see that any signal processing is responding properly.

URGENT: The important final step will be to verify the fit using Quik-Probe. Although the In-Situ Option can estimate the insertion-gain response for the average ear, for many ears you may find significant differences in the real-ear insertion response. Areas to watch out for are: Increased roll-off at low frequencies, because of venting and leakage; Sharpening of peaks near receiver resonances, also because of venting and leakage; A shifted or inverted resonant peak, because of differences in the unaided earcanal resonance; Overall gain shift, because of middle-ear impedance differences or post-operative conditions. In most cases, however, a pre-fitting setup of the hearing aid using the In-Situ Option should do a much better job than the traditional 2cc-coupler tests, for getting as close as possible to the prescribed fit before the client arrives. That should save the dispenser valuable time with the client, and the dispenser will look better by not having to fiddle with the controls too much. In many cases, where the client's ear is close to average, the fit will be perfect the first time the instrument is placed in the ear! Encourage your 6500 users to take full advantage of the valuable In-Situ Option.

"Why Do I NEED the Real-Time Composite Signal?"

by Larry Revit, hearing scientist

Although the following may sound more like an advertising supplement than an applications bulletin, there is an important issue to discuss: why the Fonix Real-Time Composite signal is a must for testing hearing aids in the nineties. For Speed. The first reason is the unbeleivable speed of testing, speed that can free the tester from waiting around for a frequency sweep to finish. Nearly as fast as you can place your eyes on the display screen, the complete response curve has already been measured. For Accuracy. In addition to sheer rapidity, high-speed testing carries important advantages. One of the requirements of accurate probe testing is that the test subject remain absolutely still and quiet. Trying to take a probe masurement with a swept tone or a swept band of noise is like trying to take a stop-action photograph with a slow shutter speed. Invariably, the subject moves; the results are blurred. Not so with the Fonix Real-Time Composite signal. Did you ever try to get a kid to sit still and quiet? For Utility. Another advantage of speed: The ongoing real-time display is an interactive link with the customer's hearing aid. As you change the hearing-aid settings, whether in real-ear or chamber modes, you see the results, as the changes are made. This is especially useful when trying to match a real-ear response curve, or when demonstrating setting changes to a potential customer for a programmable hearing aid. (And the programmables will take over... NO DOUBT. It's time to be prepared.) For Validity. Not only is the real-time composite signal fast, but it is right for testing hearing aids. To test how a hearing aid with signal processing (or even a linear hearing aid in saturation) performs in response to everyday inputs such as speech, you must use a broad-band signal such as the Fonix Real-Time Composite. Unlike swept-tone signals, the composite signal is not subject to the misleading result known as "the blooming effect. "1 The composite signal shows the true effect of signal processing. 2 And unlike swept-tone signals, the composite signal provides a clear indicator of a very nasty hearing-aid distortion called "intermodulation distortion." 3 (Intermodulation distortion can make voices sound garbled and harsh, especially the wearer's own voice!) The industry's leaders know all that. That is why an ANSI standards committee is busy drafting a new American standard for testing hearing aids with broad-band, speech-shaped signals, ...such as the Fonix Real-Time Composite signal. Can the competitors promise that their equipment will be upgradable to meet the new standard? We can.

What's more, the very latest independent research on hearing-aid testing4 points up the fact that speech is often a combination of broad-band and narrow-band information, implying that the best way to evaluate hearing-aid performance thoroughly is to use both broad-band and narrow-band test signals? Do our competitors offer that choice? We do.

Speed, accuracy, utility, validity... All things considered, the Fonix Real-Time Composite signal let's you do a better fitting job in a shorter time. Be prepared for today's hearing-aid testing, with a Fonix Real-Time Analyzer. Don't get left back in the eighties.

Simplified Aided Ear Customization for Target 2cc FOG Prescriptions

by Larry Revit, hearing scientist

As currently described in the 6500 manual, customizing a Target 2cc FOG response for the AIDED EAR requires a special insert earphone, a special adaptor, unplugging and plugging in these special devices... In other words, it's too much of a hassle for the average dispenser. That's too bad, because the AIDED EAR customization is the single most important thing a dispenser can do to ensure a successful fit the first time. So we've recently checked out a simplified procedure using a regular BTE hearing aid and no special apparatus. Here's the simplified, 2-step procedure:

1. Measure and store the 2cc coupler response of a BTE hearing aid.
    - This step can be done without the client present.
    - Attach an unvented, foam-type, stock earmold to a BTE.
    - Using Fun-Tak, attach the aid and earmold to an HA-1, 2cc coupler.
    - Set the volume control for a mild gain setting (about 20 dB), and then tape the volume control in that position.
    - Set the 6500 for a 70-dB-SPL, Composite Weighted Gain signal (the default setting), and set the Noise Reduction for 16.
    - Go to Multi-Curve and copy CRV 1 to REF 3.

Note: If you have CHAP (the Fonix Computer Hearing Aid Program), this coupler curve can be stored permanently and simply transferred to REF 3 whenever you need it.

2. Measure and store the Aided Response of the same BTE on the client.
   - It is probably best to be in MANUAL PROBE mode. Also, set the Noise Reduction for 16.
   - Using the same earmold as for the coupler response, put a convenient reference mark on a probe tube so that you will know the tube extends at least 5mm beyond the tip of the earmold when the earmold and probe tube are inserted in the ear.
   - Insert the probe tube and then the earmold, being careful that the probe remains in the desired position. Insert the earmold as closely as possible to the same depth as the client's final hearing aid will be.
   - With the aid in place on the ear, place the Reference Mic over the ear, so that the front grill is as close as possible to the mic of the BTE aid.
   - Level.
   - Turn the aid on (with the volume control in the same taped position as for the coupler measurement).
   - Using a 70-dB-SPL Weighted Composite signal (the default probe setting), measure the Aided Response in the normal way.
   - Copy the Aided Response to REF 2.

This is all you have to do. When you go to the Target 2cc FOG program, use the MENU to set "AIDED EAR" for "CUSTOM". The curves stored in REFs 2 and 3 will automatically be incorporated into the prescription conversion.

Fig. 1 — without damper

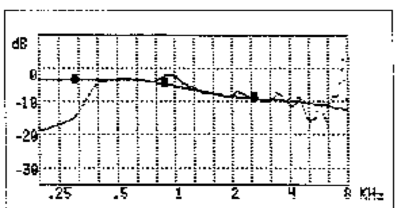
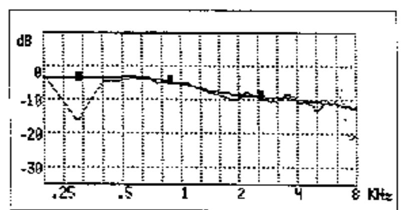


Fig. 2 — with 1000Ω damper



INSERT-EARPHONE METHOD

SIMPLIFIED METHOD

Shown in Figures 1 and 2 are examples of Aided-Ear-versus-2cc corrections (REF 3 minus REF 2) measured on KEMAR by the above sound-field procedure and by the insert-earphone procedure given in the 6500 manual. As you can see, the sound-field and insert-earphone curves match closely across most frequencies. The low-frequency variations are most likely caused by a "slit leak" in the Aided Response measurement. The high-frequency variations may be caused by the effects of receiver impedance, noise, and/or probe-position. These variations are shown to illustrate the "real-world" potential for such errors. This potential for error is the main reason why we have included the ability for the 6500 users to "Tune" the calculated Target 2cc FOG curve, so the perscription curve can be smooth and resonable. Use the "Course-" and "Fine-Tuning" controls as you see fit (described in Section 8.17.3.2).

A damped sound channel is recommended. The simplified-method curve in Figure 1 was done with no damping in the sound channel, whereas the simplified-method curve in Figure 2 was done with the same hearing aid but with a 1000-ohm damper in the earmold tube near the earhook. Since you can expect a cleaner measurement with damping in the sound channel, we recommend using either a damped earhook or a damper in the earmold tubing for these measurements.

The Aided-Ear-versus-2cc measurement is an important part of the custom prescription process. Including or not including the measurement in a prescription can "make or break" a hearing-aid fitting.

Now that the procedure has been simplified, we urge Fonix 6500 Quik-Probe II users to take advantage of this important feature of the Target 2cc program.

RMS Level vs. Spectrum Level in FONIX Composite Signals

by Larry Revit, hearing scientist

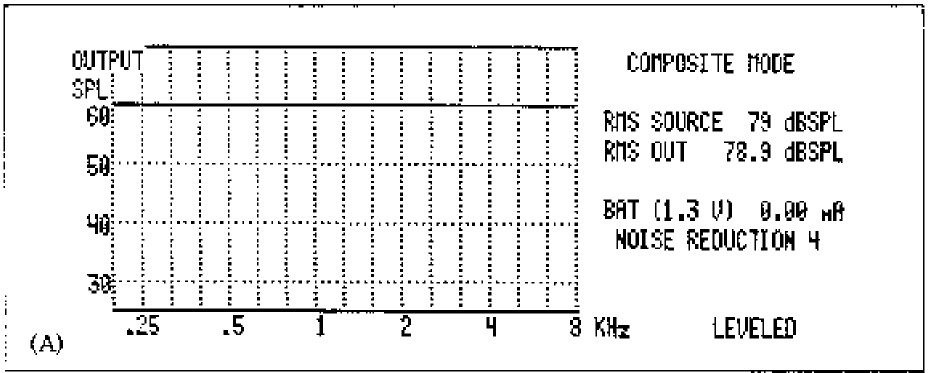
Many Fonix reps and users have questions about why the levels on composite frequency response curves are generally lower than the "RMS" levels displayed near the graphs. So herein lies an explanation. "RMS" (or "Root Mean Square") is a term referring to the overall level of a signal. "Spectrum level" refers to the level of each individual frequency component of a signal. With a pure-tone sweep, only one frequency is tested at a time: There is only one frequency component. With only one component, the overall, RMS level of the signal is the same as the spextrum level of the signal at each frequency. That's why there's little confusion with pure-tone tests. But the Fonix 6500 composite signal is a combination of 80 pure tones of different frequencies (79 for the FP40), presented simultaneously. The overall RMS level is a combination of all the spectrum levels of the 80 frequencies put together.1 The overall (RMS) level, therefore, is greater than the (spectrum) level of each component. Spectrum levels are what appear on the curve of a frequency response graph. So the levels on a composite frequency response graph will be lower than the numerical RMS level in the readout to the right of the graph. As an example, for flat-spectrum (unweighted) composite consisting of 80 pure tones, the spectrum level of each component is 19 dB down from the RMS level.2 The spectrum of the flat composite can be viewed by first "leveling" the sound chamber and then immediately observing a flat-weighted composite SPL ("POWER" or "OUTPUT") measurement, without moving the microphone from the calibration position in the chamber. {With the 6500 press[RESET], and then press [WEIGHT] twice; with the FP40, turn GAIN OFF with [F2] or [F3], and then select FLAT in the menu under COMPOSITE PARAMETERS.} An example is shown in graph "(A)". Note that the RMS SOURCE and OUT levels are essentially the same (because the microphone is at the calibration position), but the spectrum levels of the curve are 19 dB down from the RMS level. The "speech weighted" composite signal combines a high-frequency filter with the flat-spectrum composite. This filter causes a downward slope above 900 Hz. The spectrum of the speech-weighted composite can be viewed by observing a leveled, "WEIGHTED" SPL ("POWER" or "OUTPUT") measurement, without moving the microphone form the calibration position in the chamber. {With the 6500 press [RESET], and then press [WEIGHT] once; with the FP40, select SPEECH in the menu under COMPOSITE PARAMETERS.} An example is shown in graph "(B)". Note that the RMS SOURCE and OUT levels are, again, essentially the same (because the microphone is still at the calibration position), but the spectrum levels (levels on the curve) vary between about -10 to -30 dB relative to the RMS level.

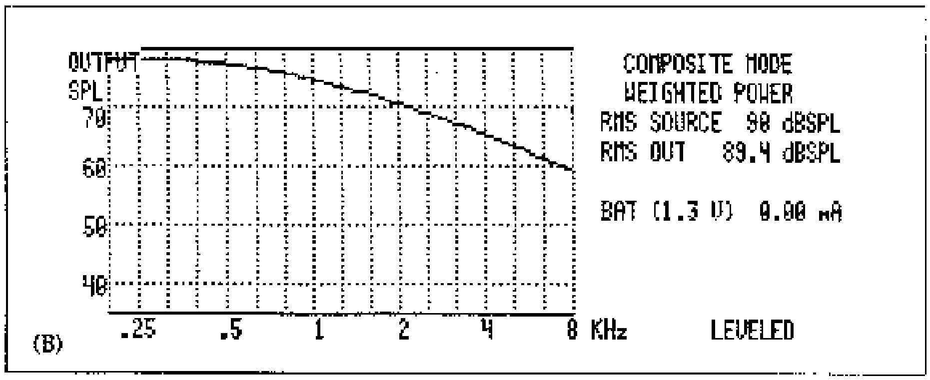
Graph "(B)" represents the spectrum of the signal that is presented whenever the speech weighting is enabled. Thus, when viewing a weighted SPL measurement, one can expect to see a downward slope (the slope of the test signal) superimposed on the frequency response of the instrument under test. The purpose of such a measurement would be to estimate the output of an instrument in response to a signal having the long-term, peak spectrum of speech.

On the other hand, when viewing a "WEIGHTED GAIN" measurement, the input spectrum is subtracted from the output spectrum before display. In this case, although the speech weighting is still part of the test signal, the frequency response is displayed with the speech weighting (downward slope) removed.

In summary, the "RMS" (or overall) level is a summation of all the frequency components of a composite signal. The levels at each frequency (or "spectrum levels") will be lower than the RMS level. The "WEIGHTED SPL" ("POWER" or "OUTPUT") test displays a composite frequency response of an instrument combined with the downward-sloping effects of speech weighting. The "WEIGHTED GAIN" test subtracts the speech weighting from the measured output (removes the downward slope) before displaying the results, and thus only the composite frequency response of the instrument is visible.

At present, the ANSI standards committee for hearing-aid testing is nearing completion of a new standard for broadband, speech-weighted tests (such as with the Fonix real-time composite). The Fonix composite signal is so useful, that as concerned professionals we must recognize that now is the time to become familiar the new ways of interpreting RMS and spectrum levels. (Call 800-547-8209 with questions. Ask for Larry.)





Footnotes

1. Mathematically, RMS is the square root of the mean of the squares of the values of each component:

$$\text{RMS} = \sqrt{\frac{1}{80}\sum_{i=1}^{80} x_i^2}$$

   Where "X" equals the voltage (or pressure) of each component, and there are 80 components.

2. For a flat-spectrum composite: Spectrum level =RMS level - 10 log (No. of frequency components) =RMS level - 10 log (80)=RMS level - (19).

The Articulation Index and Hearing Aid Fitting: The Bad News and the Good News

by Larry Revit, hearing scientist

Recently, our marketplace has expressed an interest in using the Articulation Index (AI) to assist in hearing aid fitting. Frye Electronics has been investigating efficacy of the AI in hearing aid fitting for over two-and-a-half years. Contained below is a summary of our findings, to date.
First the bad news, and then the good news.

The Bad News

- The AI has not been validated as a hearing aid fitting method. Unlike the NAL, POGO, Berger, Lybarger, and Libby methods, no clinical study has shown the AI to be an effective fitting tool. To the contrary, even a very recent report1 shows the AI method to be either no better, or else poorer, than NAL or POGO in maximizing word-intelligibility scores. Keep in mind that almost any hearing aid will raise the AI score.

- Maximizing the AI score can easily cause problems in a hearing aid fitting. In most cases, maximizing the AI is accomplished either by raising the overall gain -- something which hearing aid users don't generally tolerate -- or by increasing the high-frequency response --something which, in some cases, may have merit, but in others, may cause discomfort and feedback problems, especially with ski-slope losses.
- The AI varies significantly for different listening tasks. The AI formula changes with every listening environment and set of speech-test materials. The AI's effectiveness in predicting speech intelligibility is inseparably tied to the particular, pre-selected listening conditions and materials accounted for in an AI formula. Thus, one hearing aid could have a high AI score and give good speech intelligibility for one listening condition, and the same hearing aid could have a high AI score and result in poor speech intelligibility for a different listening condition. Furthermore, acoustically different hearing-aids can have the same AI score. (which hearing aid is better???...)
- Estimating the AI accurately is no trivial matter. Estimating the AI accurately requires an accurate estimate of aided sound-field thresholds. Two currently available "easy" AI methods use a potentially inaccurate procedure to estimate aided sound-field thresholds. Both METHODS call for estimating aided sound-field thresholds by adding measured insertion gain to earphone thresholds. One factor determining the insertion gain is the real-ear unaided response (REUR). Yet there is no evidence that REUR is correlated with earphone thresholds. So mixing insertion gain with earphone thresholds to predict aided sound-field thresholds introduces an error associated with an extraneous variable: the individual variability of the REUR.

But there is some Good News:

The accuracy of AI may soon improve. It may yet be possible that an accurate estimate of aided sound-field thresholds can be made through the appropriate real-ear probe measurements and correction factors. We are working on this, and we're confident that an improved METHOD can be devised.
The AI can be an important counseling and selling tool. There can be no better way to sell a hearing aid than to display to the client that he or she will understand speech better with a hearing aid than without one. So displaying an unaided-versus-aided improvement on the AI (a speech-intelligibility-related index) should help the dispenser convey to the client the attractiveness of buying a hearing aid. That will be good for everyone. And that is why Frye Electronics is working on developing an accurate AI method.
But we won't put it in our analyzers until we know it's accurate.

Real-Time Spectrum Analysis: A New FONIX 6500 Tool for Improving a Hearing Aid Fitting

by Larry Revit, hearing scientist

"My Voice Sounds Like It's in a Barrel!" We Can Help

What is REAL-TIME SPECTRUM ANALYSIS?
"Spectrum Analysis" means that a complex sound, such as speech or noise, is broken into its individual frequency elements and then measured. The result is an amplitude-versus-frequency graph that tells the observer how much energy is present at each frequency. "Real-time" means it happens on the spot. The Fonix 6500 has done this all along, primarily using the Speech-Weighted Composite signal. But now you can use any signal you want, and get a Real-Time Spectrum Analysis from either the test chamber or the probe microphone.

How can REAL-TIME SPECTRUM ANALYSIS help in a hearing-aid fitting?

Here are some of the ways a dispenser can use Real-Time Spectrum Analysis:

- Take the guesswork out of the question: "How will the hearing aid perform under real-use conditions?" -- by measuring real-ear gain using live or recorded environmental sounds.
- Use real speech sounds to see that a hearing aid is producing them correctly. For example, you can analyze the "[f]" and "[s]" sounds, through a hearing aid, to see why a person might be having trouble discriminating between the words "fifty" and "sixty".
- Analyze the client's own voice in the earcanal. This is one of the most interesting uses of Real-Time Spectrum Analysis -- for two reasons:

First, it's a way to involve the client in the real-ear measurement process. The client can speak various speech sounds and watch the frequencies change on the screen. Since the sounds are being picked up in the client's own earcanal, this is an excellent way to familiarize a person with real-ear measurements. Plus, it's fun, and impressive.

But there's another important professional use of this new measurement capability: solving the "occlusion effect" problem. The "occlusion effect" is the cause of one of the most common complaints from new hearing-aid users. Almost every dispenser has heard the words, "My voice sounds like it's in a barrel!" But the example below clearly illustrates how Real-Time Spectrum Analysis can help solve the occlusion effect problem, and perhaps even save a hearing aid fitting:

The graph in Figure 1 shows the spectrum of a sustained "ee" sound from the client's own voice, measured in the client's earcanal.

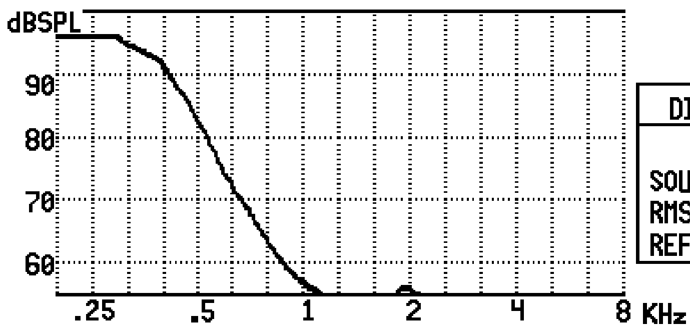


FIGURE 1 -- OCCLUSION EFFECT. Example spectrum of hearing-aid-wearer's own voice, sustained "ee" sound, in the ear canal, with a pin-hole vented ear mould in place, with the hearing aid turned off.

The hearing aid, with only a pinhole vent, is in place, but the aid is turned off. The earmold itself has a considerable amplification effect on the low frequencies of the wearer's own voice. Even with hearing aid off, the low frequencies are cooking away at over 95 dB SPL, and the overall RMS OUTPUT is more than 100 dB SPL! No wonder there's a "barrel" problem.

Figure 2 shows the solution.

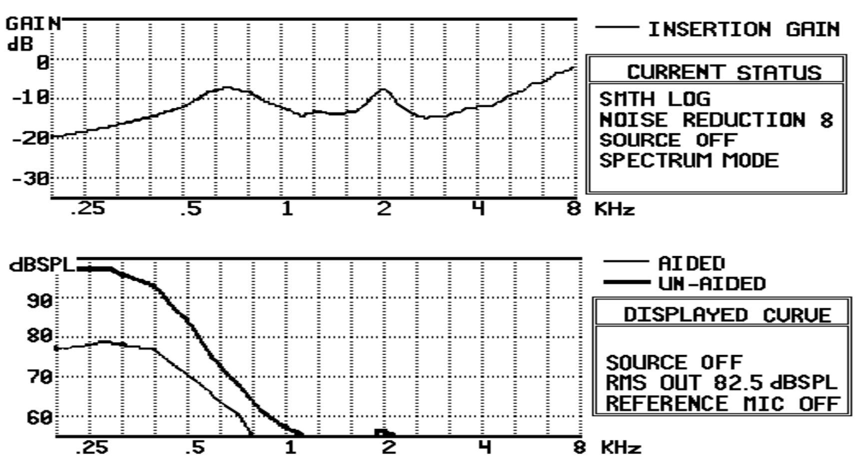


FIGURE 2 -- OCCLUSION EFFECT. Example spectrum of hearing-aid-wearer's own voice, sustained "ee" sound, in the ear canal. Thick curve in lower graph is with a pin-hole vented ear mold; thin curve in lower graph is with a wide-open ear mold , both with the hearing aid turned off. Upper graph shows the difference between the two lower curves.

The thick curve in the lower graph is the same curve as before. It has been saved as the "UN-AIDED" curve. The "AIDED" curve shows a similar measurement, but with the vent wide open. (Keep in mind that these curves are of the client's own voice, momentarily sustaining an "ee" sound while the real-time measurement is captured.) The curve in the upper graph, labeled "INSERTION GAIN", is the difference between the two lower curves. From either graph, it is clear that opening the vent in the earmold has drastically decreased the occlusion effect, by up to 20 dB!
The same set of measurements is possible with the hearing aid turned on, as illustrated in Figure 3.

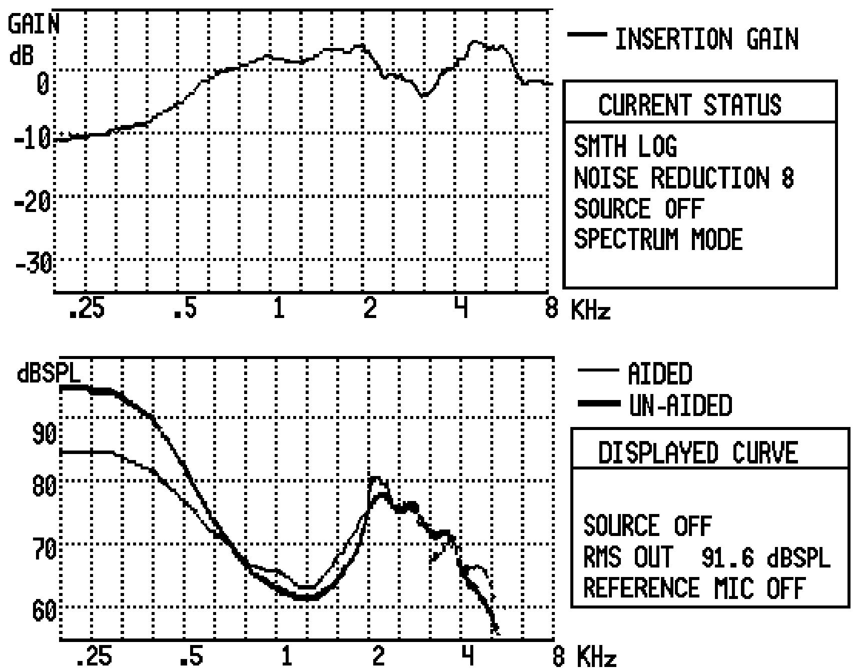


FIGURE 3 -- OCCLUSION EFFECT. Example spectrum of hearing-aid-wearer's own voice, sustained "ee" sound, in the earcanal. Thick curve in lower graph is with a pin-hole vented earmold; thin curve in lower graph is with amedium vented earmold, both with the hearing aid turned on. Upper graph shows the difference between the two lower curves.

Here we see a 10-dB improvement in the low frequencies, this time in going from a pinhole vent to a medium vent. The wide open vent was not possible with the hearing aid turned on, because of feedback. But a 10-dB improvement means that the low frequencies of the wearer's own voice (the "barrel" sound) will be only half as loud as before (a 10-dB change means half the loudness). This kind of observation can give the dispenser, and the client, confidence of an improved hearing-aid fit.
Step-by-step, "cook-book" procedures for all the Real-Time Spectrum Analysis tests mentioned above are given in the revised Chapter 10 of the 6500 Operator's Manual, shipped with Software Version 2.66. I'm confident this new feature of the Fonix 6500 can be an important addition for our customers.

Speech Testing with Audiometeric Earphones

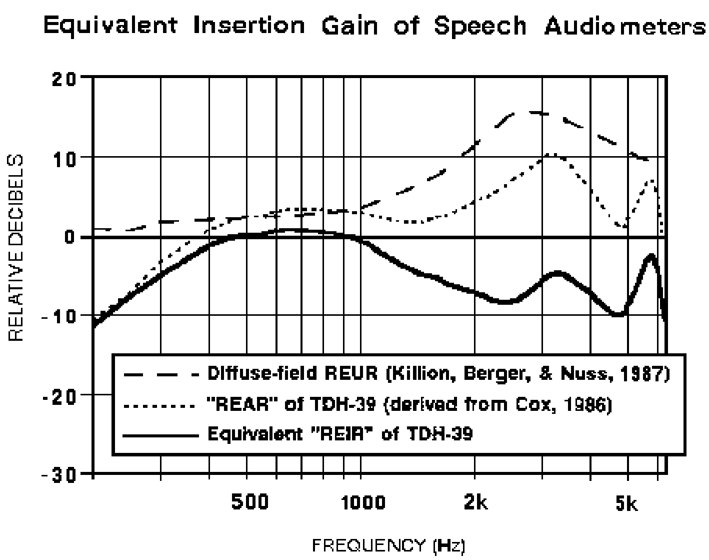
by Larry Revit, hearing scientist

New data suggests a possible need to revise standard practice.

Here's a clinical tidbit which will come as no surprise to some, but which may shock others. Because Frye Electronics has just introduced the FONIX FA-10 Hearing Evaluator, I have been assigned the task of devising instructions for using the "Hearing Aid Simulator" feature. Operating this feature requires presenting speech through the audiometer, so, being the "hearing scientist", I soon began asking questions -- like what, acoustically, is being presented to the client during speech audiometery, and how does it relate to hearing aids?

Because almost all the talk about hearing aids nowadays relates to insertion gain, the above questions boiled down to "What is the equivalent insertion gain, the above questions boiled down to "What is the equivalent insertion gain of a speech audiometer?" I found some data in JASA1 that let me calculate the sound pressure level (SPL) at the average eardrum, in response to a constant SPL into a speech audiometer having a flat frequency response and using a TDH-39 earphone. This is no more and no less than the "real-ear aided response" (REAR) of a speech audiometer: the audiometer, in this case serving as the hearing aid. Because "real-ear insertion-gain response" (REIR) is defined as the difference between the aided and unaided real-ear responses, I had only to find an appropriate real-ear unaided response (REUR) to subtract from the already known aided response. For this I used the mean, diffuse-field data from Killion, Berger, and Nuss2. The results are shown in the accompanying figure. The bold curve shows that the high frequencies are attenuated relative to the lower frequencies.

What is the practical implication of these results? The implication is that audiometeric speech tests are given under acoustic conditions that attenuate the important speech frequencies above 1000 Hz by as much as 10 dB (in the average ear). So we are giving our clients an additional 10 dB of high-frequency hearing loss before we determine their abilities to hear speech!



Recommendations: The hearing health care profesion might consider re-examining the validity of this practice, and could consider the possible alternatives of:

- a. Electronically compensating speech audiometers for the "insertion loss" of audiometric earphones, or else
- b. using earphones that reproduce the natural earcanal resonance.

In the meantime, for those interested in experimenting, using the "6dB" or "12dB" slope settings of the Hearing Aid Simulator in the mid to high frequencies than using the "Off" setting. Let me know what you come up with!

Estimating the Real-Ear Saturation Response
by Larry Revit, hearing scientist

The following procedure lets you estimate what the real-ear saturation response (RESR) of a hearing aid will be without subjecting the client to a high-level, real-ear frequency sweep. The procedure is a modification of the method described by Sullivan (Hearing Instruments, Vol 38, 10/87). (Italicized items refer to using the FONIX 6500, software versions 2.50 and higher.)
First you find the difference between the response of the hearing aid in the coupler and that in the wearer's ear, using a moderate signal level. This is called the "real-ear-to-coupler difference". Then, to estimate what the saturation response will be in the ear, you apply the real-ear-to-coupler difference to the SSPL-90 response measured in the coupler.

1. Have the client adjust the volume control (while wearing the instrument) to the normal use setting -- or, choose a low-gain setting. Secure the volume control in this position with tape or putty. This same setting will be used for all measurements, except in step #5.
2. Using a moderate signal level, measure the Aided Response of the hearing aid in the client's ear. Copy this curve into REF 2 of Multi-curve.
3. Using the same signal and level as for the real-ear case, measure the response of the hearing aid on the scc coupler in the test box. Copy this curve into REF 3.
4. Subtract the Aided Response (REF 2) from the coupler response (REF 3). The difference curve (automatically in REF 9) is the real-ear-to-coupler difference (RECD). Copy the difference curve from REF 9 to REF 4 for later use.
5. Now remove the tape from the volume control and set it for full on. Measure the response of the aid on the coupler in the test box, but with a pure-tone signal at 90 dB SPL.
6. Subtract the RECD curve of step #4 (REF 4) from the result of step #5 (CRV 1). This new difference curve (in REF9) is the estimated real-ear saturation response (ERESR).
7. Adjust the maximum power output of the hearing aid and re-test, until the desired estimated real-ear levels are achieved.

Use this procedure to ser the hearing aid in the sound chamber, before measuring the actual real-ear saturation response. Another way to set the hearing aid in the sound chamber before real-ear testing is to use the Target 2cc prescription software. But you will need frequency-specific loudness discomfort data to make a Target 2cc SSPL-90 curve. By using either the estimated RESR method or the Target 2cc method, you can be reasonably sure you won't cause excessive loudness in the client's ear.

Functional Gain and Insertion Gain

by Larry Revit, hearing scientist

When are they equal; when are they not?
Functional gain is the difference between behavioral soundfield thresholds under aided and unaided conditions. Insertion gain is the difference between the sound pressure level in the earcanal under aided and unaided conditions. Several studies have indicated that, on average, functional gain and insertion gain are indeed equivalint measures. Yet many users of probe-tube systems find significant differences between functional gain and insertion gain when the two are measured with the same hearing aid on the same individual. Why? Two reasons. 1] The equivalence of the two measures holds true only for linear hearing aid operation. 2] Even though the two measures may be equivalent on average, individual cases may vary considerably for many cases.
Let's look at case #1 first. The problem with anything other than linear hearing aid operation is this: Normally, the signal levels used for the "aided" parts of the two measures are quite different. The "aided" part of functional gain is the aided soundfield threshold. This measurement deals with relatively low signal levels -- at the (aided) threshold of hearing. In contrast, the "aided" part of insertion gain is the real-ear aided response (or REAR). This measurement deals with signal levels that may be considerably above the client's hearing threshold. So what happens if we're measuring functional and insertion gain on a hearing aid having automatic

gain control (AGC)? The signal levels are different for the two measures, so the gain of the hearing aid (controlled by AGC) can be different for the two measures -- thus, a disparity between the two measures.

Now let's look at several other sources of individual variability. a] Soundfield requirements: For soundfield threshold measures, as used for functional gain, the test environment must be designed according to stringent requirements; the bandwidths of the signals used must be carefully controlled, and the soundfield must be painstakingly calibrated. If these conditions are not met, considerable error is possible in soundfield threshold measurements. b] Extraneous noise: If there is any audible extraneous noise present during an aided threshold will be masked by the extraneous noise. Thus the functional gain measurement will be influenced by the effective masking level of the extraneous noise. c] Slit leaks (this one affects insertion gain): If there is a change in earmold venting caused by leakage between the probe tube and the earmold, the low-frequency gain registered by the REAR might be lower than what it would be with no probe tube in place -- thus, an error. And in addition to these sources of differences, we have the usual variability associated with thresholds and soundfield measurements: loudspeaker location, head movements, probe-tube position, audiometer step-size, etc.

If the hearing aid you are working with is operating linearly under all test conditions, and if you use a carefully controlled soundfield for thresholds, and if environmental and hearing aid noise do not mask the aided soundfield thresholds, and if no significant extra venting is caused by the probe tube,... then both insertion gain and functional gain will be equal, within the normal constraints of measurement error. Otherwise, there will be differences.

So which is the better measure? Certainly, each has important advantages. Functional gain involves the entire auditory system, including the client's behavioral response. Insertion gain is much faster to do, it measures many more frequencies, calibration is easy to control, accuracy is better, and the results can tell you more about what happens under normal conversational conditions. The choice is yours. But be aware of the normal differences.

How to Evaluate Hearing Aid Fittings for Quiet and Noisy Listening Environments

by Robert Martin

Introduction:

Most hearing aid circuits that reduce background noise do so by cutting the low frequency gain and limiting the output. This paper suggests measurements you can make to see whether or not the hearing aid circuits are performing their job (increasing intelligibility) for the patient. These measures evaluate the speech information cues (real-ear gain) available to the patient for word understanding ability in quiet and noisy listening environments.

Summary

Give the patient the typical hearing tests. Then evaluate the patient's ability to understand filtered speech. Select the hearing aids and make initial adjustments. Next, using a real ear system, generate an NAL target. Simulate a quiet listening environment by using a soft input level and do real ear tests to compare REIR (real ear insertion response) to the NAL target. Simulate a noisy environment by using a loud input level; again, compare the REIR to the NAL target. Inspect both sets of data for adequacy of amplification. Be sure the amplified sound is comfortable. Do sound box tests to evaluate the "cleanness" of the signal. Confirm this procedure by checking discrimination ability in a noisy listening environment.

Step-by-Step

1. Do a typical audiometric evaluation and find the patient's maximum speech discrimination ability.
2. Since many "noise reduction" circuits cut the gain in the lower frequencies, evaluate the patient's word-understanding ability for similar conditions. Use a master hearing aid unit (FONIX FA-10) and measure the patient's speech discrimination ability through the -6, -12, -18, and HFE settings. Plot these scores. These filtered speech tests show how much the low frequency gain can be reduced without damaging word-understanding ability significantly.
3. Select, adjust, and fit the hearing aids.
4. Simulate a quiet listening environment by using a 50-dB composite noise input level. Overlay the REIR on the NAL target. Check for adequate amplification. The REIR values should match or exceed the NAL values.

5. Simulate a noisy listening environment by using an 80-dB composite noise input. If the hearing aid has a "noise reduction" switch or program key, activate it now. Next, run real ear tests and overlay the REIR on the NAL target. Check for adequate amplification in the 1000-6000 Hz zone, the zone most critical for word understanding. Adjust trim pots, digital code, etc. to best match the NAL target in the higher frequencies while reducing the gain in the frequencies below 1000 Hz.
6. Compare the response curve and RMS values obtained with the 80-dB input to the patient's uncomfortable listening thresholds, UCLs. The response curve values give information at specific points; the RMS value indicates "total energy" in the signal. Some tolerance problems are observed in specific zones; other times the overall signal is just too loud.
7. It is important that the amplified sound have low harmonic and inter modulation distortion values. To evaluate the "cleanliness" of the amplified sound take the aid off of the patient and place it in the sound box, using an 80-dB input signal. The presence of inter modulation distortion can be observed by inspecting the response curve for smoothness. The amount of harmonic distortion observed at the use-gain setting can be quantified using a 70-dB pure tone input signal and sweeping the spectrum.
8. Confirm the above procedure by measuring speech discrimination ability in a noisy listening environment.

All the above Contents are subjected from the prints of the Web and edited for your reference by :

Biswajeet Sarangi
Audiologist & Speech Therapist
India