

Current Practices in Modern Probe Microphone Measurement (PMM)

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When evaluating what is considered best clinical practice for fitting hearing instruments, Probe Microphone Measurement (PMM) must not be ignored (EUHA, ASHA, AAA, BSA & BAA). PMM is not only the gold standard for hearing instrument verification, but it is also a powerful business tool. Studies have suggested that satisfaction relates to clinical care more than the level of amplification provided and comprehensive protocols including PMM enhances the client experience both clinically and leads to better client outcomes (Consumer Reports, 2009; Kochkin, 2010; Hougaard and Ruf, 2011).



Terminology and procedures

There exist several classic references which describe traditional real-ear measurement terminology, explain how the measurements are conducted, and what they are used for (Dillon, H. 2001; Mueller et. al., 1992). A brief description of common terminology follows here.

Probe microphone measurement is generally described in terms of response (SPL in the ear canal) or gain (derived by subtracting response at the reference microphone from the response at the eardrum). Real Ear Unaided Response (REUR) is the sound pressure level (SPL) as a function of frequency measured in the open ear canal for a given input signal. Real Ear Unaided Gain (REUG) is the natural amplification provided by the ear canal and pinna resonances and is derived by subtracting the input spectrum from the REUR. Real Ear Occluded Response (REOR) is the frequency response in SPL, measured in the ear canal, with the hearing instrument in place and turned off. Real Ear Occluded Gain (REOG) is obtained by subtracting the input spectrum from the REOR. Real Ear Aided Response (REAR) is the frequency response at the eardrum with the hearing aid (and its acoustic coupling) in place and turned on. Real Ear Aided Gain (REAG) is the difference between the input spectrum and the REAR measured at the ear canal. Real Ear Insertion Gain (REIG) is the amount of gain provided by the hearing instrument and is obtained by subtracting the REUG from the REAG.

Real Ear Saturation Response RESR is the frequency response of a hearing instrument measured in the ear canal with an input signal that is intense enough to cause a hearing instrument to function at its maximum output level (MPO). Real Ear to Coupler Difference (RECD) is the difference in dB between the real-ear SPL and that measured in a 2cc coupler. Real Ear Dial Difference (REDD) is the difference in dB between the real-ear SPL and the audiometer dial setting that produced the signal.

Probe tube calibration and proper probe tube placement are essential basic procedures which influence the success of PMM. Probe tube calibration removes the acoustic effects the probe tube and microphone introduce during real-ear measurement, thereby making the probe tube

and the microphone 'acoustically invisible' (Pumford and Sinclair, 2001). This is the essential first step for all PMM procedures.

Proper probe tube placement is also a key for ensuring accuracy of PMM. The typical recommendation is to place the probe tube within 5 mm of the client's tympanic membrane in order to avoid standing waves.

General guidelines as described by Pumford and Sinclair (2001) suggest: For adult females, insert the probe tube 28 mm past the intertragal notch. For adult males, insert the probe tube 30-31 mm past the intertragal notch. For children, insert the probe tube 20-25 mm past the intertragal notch. To assist with placement, most manufacturers provide probe tube markers and/or probe tube supports to set the appropriate distance (e.g., 30 mm) from the probe tube's open end and insert the probe tube into the ear canal until the mark approaches the intertragal notch.

Probe tube placement can also be assisted via acoustical positioning procedures (ANSI, 1997; ISO 12124:2001). A simplified method is through visualization and repositioning based on the REUG curve, monitoring particularly the frequency region above 4000 Hz.

1. Insert the probe tube less than half way into the ear canal while presenting a 65 dB pink noise signal.
2. A notch in the gain curve above 4000 Hz is likely to be observed.
3. Gently insert the probe tube deeper while keeping an eye on the notch which is moving towards higher frequencies.
4. The probe tube is located correctly as soon as the notch is no longer dragging the gain curve down (-5 dB) in the high-frequencies.
5. Once the measurement is stabilized move the probe tube marker into position or to attach the probe tube to the probe tube support.

Traditional REM with modern hearing instruments

The terminology and measurements themselves haven't changed much over the years. But the signals we use and the way we apply them clinically has changed. Historically, PMM have been made with either tonal or noise signals (Mueller et. al., 1992). These signals were adequate for measuring the gain of linear hearing instruments. But as devices became more sophisticated the need for more dynamic signals and multiple input levels became apparent (Mueller 2006).

Digital Signal Processing

The use of digital signal processing (DSP) in hearing aids has allowed hearing aid manufacturers to employ very complex algorithms for compression, noise reduction and frequency shaping, and many manufacturers now have their own proprietary fitting algorithms. A major advantage of this technology is increased flexibility in programming capabilities, which allows audiologists to more accurately fit a hearing aid to a prescriptive target (Fabry, 2003). The counter to the enhanced flexibility is that complex signal processing created some challenges when working with traditional signals. Devices might interpret composite signals as "noise" and thus reduce gain, resulting in an underrepresentation of actual every day gain realized by the client in normal use.

The stimulus type must have certain spectral and dynamic properties in order to be processed correctly by the hearing instrument and relate to the prescriptive target. A signal that is as close to real speech as possible (and which has a similar shape to the signal used to develop the prescriptive method being verified) is preferred. The introduction of modulated broadband signals and the use of real speech signals now make it possible to conduct measurements with the noise suppression algorithms and other advanced hearing instrument features active.

In an effort to make an internationally applicable signal, which allows for reproducible measurement conditions and which features all or most relevant properties of natural speech. (e.g. the modulation spectrum and the fundamental frequency as well as its harmonics) the International Speech Test Signal (ISTS) was developed

(Holube, et. al, 2010). Furthermore, some systems make it possible to use alternative signals with prescriptive targets by creating custom targets. This is achieved by applying a correction for the input signal spectrum to the target generated by the chosen fitting prescription.

Validated fitting prescriptions versus proprietary manufacturer fitting prescriptions

The wide variability of fittings amongst hearing instrument manufacturers, whether programmed based on a validated fitting prescription such as NAL-NL1 or DSL 5, or the manufacturer's own proprietary algorithm underscores the need for verification of fittings (Smeds and Leijon 2001; Keidser et. al., 2003.) One of the primary reasons for utilizing PMM is that several studies have confirmed that the manufacturer's initial-fit algorithm often is an inadequate amplification prescription (Aarts N, Cafee C. 2004; Aazh H, Moore BC. 2007; Bentler R. 2004; Hawkins & Cook; 2003), sometimes providing less-than-prescribed gain in the high frequencies by as much as 20 dB .

It has also been pointed out that earlier versions of validated prescriptive methods might not have accounted for various the features which impact the output of modern, non-linear hearing instruments (Bretz, 2006). Fortunately, enhancements to generic validated fitting prescriptions have been made to handle parameters such as the number of channels in the hearing aid, the bandwidths of each channel, the compression threshold, the attack and release times and noise reduction programs (Dillon 2006.; Scollie, et. al. 2005).

Traditional REM and “Open Fittings”

Failure to mention “Open fittings” and micro-behind-the-ear (BTE) instruments would be an omission of one of the biggest trends in modern hearing instrumentation. Whether coupled to the ear with a thin-tube or a receiver placed in the ear canal, they have allowed dispensing professionals to offer occlusion-free devices and options such as directional microphones without compromising cosmetic appeal (Switalski, 2011).

Because the open ear advantage remains intact, truly open fittings present measurement challenges; particularly related to prescriptive targets and equalization of the input signal. There exists the belief by some that validated prescriptive targets are inappropriate for open fittings. However, as Mueller (2006) pointed out, “there would seem to be little reason to abandon the validated fitting methods that have proved to be successful over many years, simply because the ear canal is open. As expressed by Dillon (2006), there is no reason why changing the size of the venting should change the ear canal SPL that is optimal for a given individual.”

Equalization of the input signal plays a critical role in the ability to accurately verify open fittings. When speaking of equalization for PMM there are historically two approaches described in the literature; the substitution and pressure methods.

Substitution method

When the substitution method is applied, the calibration is performed without the subject in the room and with the microphone placed in the subject’s test position (American National Standards Institute, 1997). The calibration is stored and is applied as a reference for the rest of the measurement process. The substitution method is by nature a stored equalization method. However there are errors associated with this method. Indeed, the absence of the subject makes the precision of this reference uncertain, and even if the subject is positioned in the exact same location of the calibration, any movement will decrease the measurement accuracy. Certainly the head and body of the subject will influence the calibrated sound field (Hawkins & Mueller, 1992).

Modified Pressure Method

When equalization occurs automatically during the measurement by monitoring the reference microphone it is called “Modified Pressure Method” (American National Standards Institute, 1997). We can furthermore distinguish two variants of this method:

The Modified Pressure method using Concurrent Equalization (MPCE)

This method does not require calibration or equalization prior to positioning the client. The reference microphone is monitored in order to adjust the signal level to continuously produce a constant sound pressure level at the subject’s ear. Meaning that if the subject moves during the measurement the loudspeaker will alter stimulus intensity accordingly. The MPCE takes into account the head and body effects (reflection and diffraction). This method has typically been recommended for PMM (Dillon, 2001; Hawkins & Mueller, 1992). Lantz et al.(2007) have noted an underestimation of gain (REIG) when using the MPCE with open fittings; attributing this mainly to the pollution of the reference microphone by the amplified sound leaking from the ear.

Note: For dynamic signals most PMM systems employ a variant of the MPCE in that the “concurrent” equalization precedes the measurement rather than adjusts the signal throughout. A brief automatic calibration signal is played and the level is set immediately before each measurement. The standard calibration can be used whenever the fitting is closed/occluded (as indicated by a loss of ear canal resonance when comparing unaided and occluded measurement curves). This just-in-time calibration of the stimulus ensures the correct level and spectrum of the stimulus in the current acoustical environment.

The Modified Pressure method with Stored Equalization (MPSE)

This method is a hybrid of the substitution and the modified pressure methods. The approach has been recommended for use when verifying open non-occluding hearing instruments (Lantz et al., 2007). With this method, the sound field is equalized and the level is adjusted with the client present, and the reference microphone at the client's ear. This method takes the

client's effect on the environment into account, and uses a stored signal calibration for consecutive measurements. When fitting a hearing instrument with more open coupling, you must turn off the hearing instrument before calibrating so that sound leaking from the ear does not interfere with the calibration and influence the measurement. MPSE has been shown not to introduce clinically significant errors in real-ear measurements when verifying hearing instrument fitting (Shaw 2010).

Modern PMM for modern hearing instruments

Traditional PMM hasn't always translated well when counseling clients. Measurements are precise but not intuitive from a client stand point. There has been an obvious need to make standard clinical procedure more understandable and perhaps even enjoyable for laymen and clinicians alike. What follows is a three-phase fitting process which is sure to add value and meaning to the fitting. The process includes a check for openness of fit and appropriate procedural adjustments, Dynamic REM with percentile analysis, and finally validation of key hearing instrument features.

Openness of fit

Despite the commonly used term "open-fit BTE," these devices encompass far more variance than the term encompasses. Simply stated, not all micro-BTE fittings are open. The product's evolution warrants modification of our processes. The main goal of this type of fitting—maximizing access to speech and environmental sounds despite a compromised auditory system—is not fundamentally different from traditional fittings. However, the wide array of options that micro-BTEs offer should be applied deliberately in order to maximize their effectiveness and provide greater client benefit and satisfaction.

The following procedure can be used, and only takes about 30 seconds. This is essentially comparing the client's individual ear canal resonance (or gain) with and without the hearing aid in place.

1. Present 5 seconds of pink noise and measure the client's Unaided Response.
2. Place hearing instrument in the ear. Turn it OFF or on MUTE.
3. Present 5 seconds of pink noise, and measure the Occluded Response.
4. Compare the two curves to determine if and how the ear canal resonance has been affected by the placement of the hearing instrument dome or receiver in the ear canal

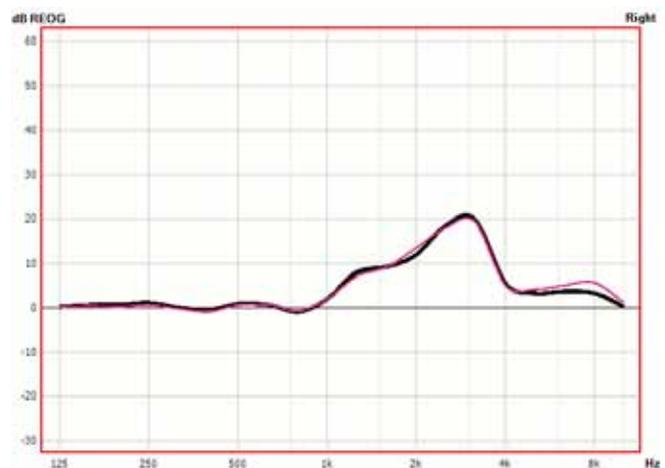


Figure 1. Open fitting. The REUG (black curve) and REOG (pink curve) are nearly identical, indicating the fit is largely open and open ear gain is retained.

If the fitting is truly open, the two curves should overlay one another, indicating the responses are virtually the same. In this case, placement of the dome or the receiver in the ear canal does not create occlusion.

When this is good: If you have planned for a truly open fitting for a client with normal low-frequency thresholds.

When this is not good: If your amplification goals require low frequency gain or greater power than can be easily offered in the open configuration, this result suggests the need to change to a larger or more occluding "power" dome or consider a custom mold.

If the fitting is partially occluded (Figure 2) or occluded (Figure 3), the occluded response curve will fall below the initial (unaided) curve. The effect can range from slight to significant.

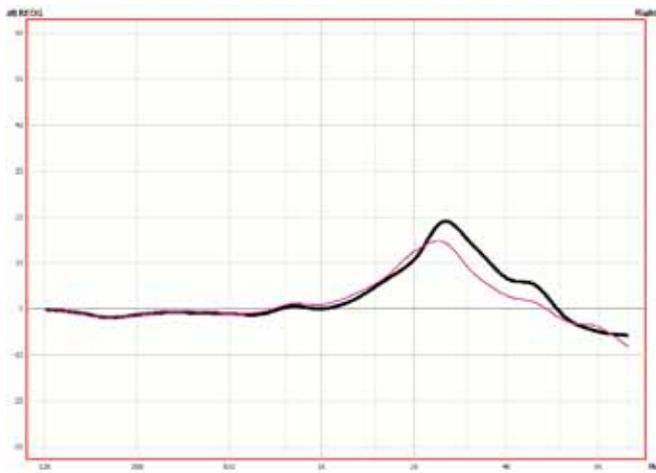


Figure 2. Partially open fitting. The REUG (black curve) and REOG (pink curve) are closely overlaid, but not identical indicating the fit is largely open and most of the open ear gain is retained.

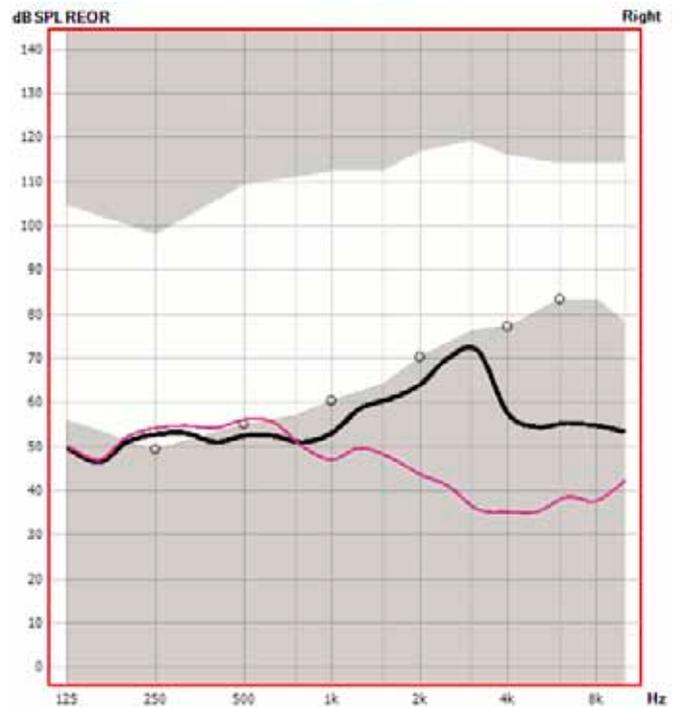


Figure 3. Occluded fitting. Shown in response view. The REUR (black curve) and REOR (pink curve) are significantly different, indicating a closed/occluded fitting.

When this is good: This is appropriate for fittings that require more power. The adequate seal is important for preventing leakage of sound directed into the ear. This minimizes the opportunity for feedback occurrences, and simultaneously maximizes available gain. Despite the increasingly sophisticated and effective feedback cancellation systems in today's hearing instruments, the most effective "feedback management" encompasses both physical fitting characteristics (optimized dome/ear canal match) in addition to digital instrument capabilities.

When this is not good: This finding may be problematic when you have recommended an open fitting for a client with narrow ear canals, and you started with the smallest dome available.

In cases where an open fitting is the goal, you can try a less occluding dome. However, this may not be necessary, especially if the effect is slight. You may still be clear to proceed with device programming using probe microphone measurements for verification because the probes are already in place once you have measured these responses. In these cases, make note of the finding to streamline potential "own voice" issues during the initial fitting or a later follow-up.

In cases where a closed/occluded fitting is the goal, a power dome, larger-size dome, or custom mold may be necessary to provide the necessary gain and output. If you are unable to more effectively close the ear canal, be aware that your control over delivering amplification in the low frequencies can be limited. In addition, any low frequency gain the client is receiving may not correspond with the manufacturer's fitting software indication. Relying on your probe microphone measurements will be vital to gauge the true amount of gain being delivered.

Comparing the unaided and occluded responses does not provide a direct measure of the "occlusion effect" that is often seen in fittings with limited venting (i.e., custom products or traditional ear molds). However, the openness of fit can guide you toward the most-appropriate option for resolution—or at least an acceptable level of improvement—when clients report issues with the perception of their own voice.

The adaptation that occurs following consistent hearing aid use will often allow many side-effects, including those related to their own voice, to resolve on their own. However, in cases where resolution does not take place, or when more rapid client acceptance is needed, applying the information acquired using the above technique is invaluable in determining the best approach to take.

In a fitting that shows any level of occlusion, changing to a truly open configuration by selecting a smaller or more vented dome (if not contra-indicated by power needs) can have a dramatic and sudden positive effect. Conversely, in a fitting that is confirmed to be open, own-voice issues may require minor frequency response adjustments in the hearing instruments and/or counseling of the client.

Knowing which tool to reach for first—the dome kit or the programming cables —provides you with the right solution for each individual client. Additionally, the opportunity to quickly resolve an issue like this has an added benefit. Recent data indicates a high percentage of clients achieve "above-average success" with hearing aids that were fit in 1 to 2 visits versus the 4 to 6 visits reported by clients with "below-average success" (Kochkin, 2011).

OpenREM Calibration is essentially the MPSE method of equalizing the input signal spectrum. Once you have determined the openness of fit you can decide whether to use standard calibration procedures (MPCE) or OpenREM calibration. If the measurements reveal complete occlusion there is no need to use OpenREM calibration.

It takes about 20 seconds.

1. With probe tubes still in place and the hearing aids in the client's ears, select the "MUTE" or "OFF" setting in the manufacturer's fitting software. Select the "Use OpenREM Calibration" option in your PMM system.
2. A quick calibration stimulus will run, automatically measuring and setting the sound between the loudspeaker and the reference microphones, and then storing that level. The reference microphones are no longer monitored throughout the rest of the fitting process.
3. After turning the hearing aids back on in the manufacturer's fitting software, proceed with your fitting to set appropriate gain levels and frequency response.

While most clients remain relatively still during fitting and programming, if they happen to move (e.g., sitting back in their chair), this process can quickly be repeated to equalize the input signal again.

OpenREM Calibration for CROS and BiCROS fittings are also easily evaluated using PMM. Pumford (2006) provides a thorough description of the process of verifying CROS (contra-lateral routing of signals) or BiCROS (bilateral

CROS) hearing instrument fittings and which also provides a method of informing clients with unaidable unilateral hearing loss about the benefits they can expect from a CROS/ BiCROS hearing instrument. Three key principals for verifying CROS aids with PMM were outlined.

1. The probe-tube microphone must always be located in the ear canal of the better ear.
2. The reference microphone should be located on the same side of the client as the speaker. If the reference microphone cannot be separated from the measuring probe microphone, it should be deactivated and the substitution method used.
3. The loudspeaker can be moved within the range of $\pm 90^\circ$ relative to the front of the client, depending on the stage in the fitting process.

The use of MPSE as an equalization method allows the clinician to satisfy these conditions and reduces the error associated with the substitution method. Use of the OpenREM calibration protocol in a given PMM system provides an equally good starting point for verification of CROS and BiCROS fittings.

Dynamic REM with Percentile Analysis

One trend that has taken a foothold is fitting to prescriptive targets with the use of real speech stimuli and verifying to an REAR target in an SPL view rather than in a gain view. This allows one to examine the impact of the hearing instrument signal processing on the dynamics of speech via percentile analysis for multiple input levels. Percentile analysis is a statistical method which evaluates the dynamic properties of the measured signal displaying the LTASS, the 99th and 30th percentiles (EUHA, 2011). The 99th percentile curve shows levels which are exceeded by 1% of the signal measured at the eardrum, and is commonly referred to as the peaks of speech. The 30th percentile curve illustrates the levels which are exceeded by 70% of the signal and is commonly referred to as the valleys of speech. Also, the percentile and LTASS curves (commonly referred to as the speech envelope, speech banana, or speech spectrum) are measured at the eardrum, and thereby reflect fully the dynamics of the aided response. Because the spectrum of a speech sample varies over time, the minimum measurement duration of 10 seconds is required in order for the Long Term Average Speech Spectrum (LTASS) measurement to result in a stable, repeatable result (Olsen 1988).

The aim is to restore the speech peaks of a soft input to audibility, the full dynamics of an average speech signal within the client's dynamic range and to confirm the 99th percentile for a loud speech input will not exceed the client's uncomfortable loudness level (UCL). The most commonly used input levels are 50, 65 and 80 dB SPL, often based on the gain handles within the various fitting software programs. Clinicians will commonly make all adjustments for the average input and then

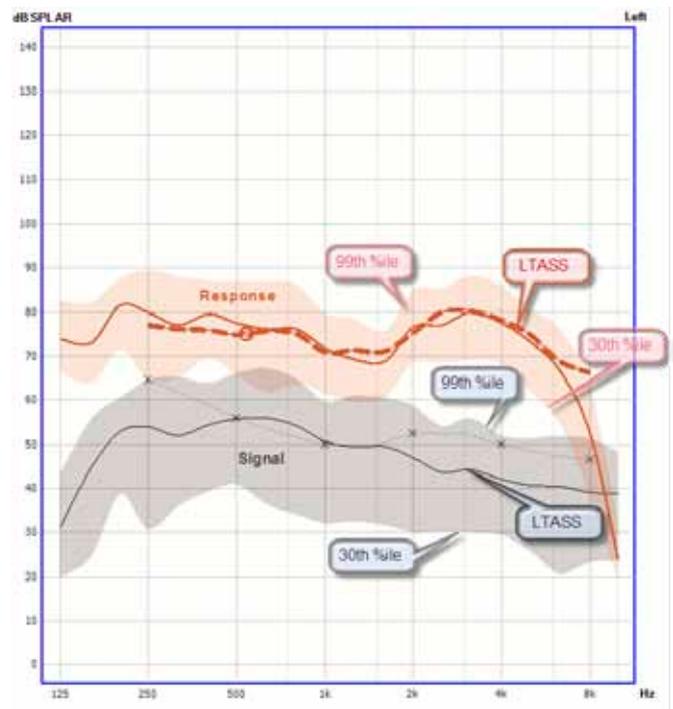


Figure 4. Percentile curves for 65 dB ISTS. The black curve is the LTASS for the input signal; the accompanying gray shaded is the input speech spectrum. The orange curve and accompanying shaded area are the amplified LTASS and speech spectrum measured at the eardrum.

run a sequence of measurements to document that all other levels have come into place based on the average settings. Finally, an 85 or 90 dB tone sweep is presented for adjusting the Maximum Power Output (MPO) of the hearing instrument to ensure loud sounds do not exceed the client's UCL. On principle, it is important to find a balance between speech intelligibility and acceptance. Full restoration of audibility may result in reduced acceptance and even reduced intelligibility (EUHA, 2011).

Percentile analysis enhances prescriptive fitting. The most recent versions of the two primary validated fitting prescriptions consider client experience type (adult versus pediatric- DSL5.0b; new versus experienced- NAL-NL2). This has resulted in somewhat lower “adult” prescriptions at some frequencies for some hearing loss configurations. Confusion can arise when in certain instances the LTASS target falls at or even below the clients hearing threshold levels. Percentile analysis can be quite useful in these situations. Understanding that restoration of LTASS audibility of high frequencies is not always desirable, one can visualize that the peaks of speech are usually restored above the client’s threshold when reasonable. Verification to match target without percentile analysis would not provide this meaningful perspective.

Further, it is possible to compare the measured speech spectrum with that of the known stimulus input spectrum to see the impact of the amplification on the dynamics of speech. Adjusting individual hearing instrument parameters such as number of channels, compression speed, compression threshold, etc... will result different outputs which can easily observed with percentile analysis (Mueller, 2006). This approach also helps to make the verification process meaningful for the client and they can easily see that the dynamics of speech are delivered at appropriate levels (Cunningham et. al, 2002). The clients understand that their amplification is tuned precisely and the need for multiple follow-up sessions for “tweaking” and adjustments is reduced or eliminated.

Key feature validation

Some of the more “fun” applications of PMM involve the use of the tool less formally, such as in the validation of specific hearing instrument features and as a means to evaluate client comfort. While the jury is still out on the true real-world benefit of many advanced hearing instrument features, it is important to have some idea of the impact these features have on the signal arriving at the client’s eardrum (Bentler and Mueller, 2006). Dynamic signals make it possible to quickly and easily demonstrate whether features such as digital noise reduction, directionality, feedback suppression, and frequency lowering function as expected. What follows are simple steps for using PMM to conduct basic assessment of function.

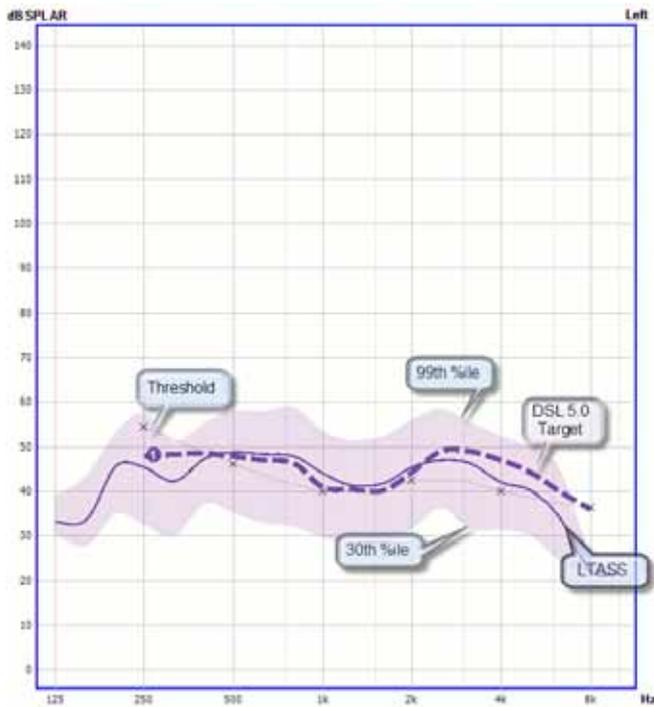


Figure 5. Soft speech target and aided response for mild hearing loss. The target curve is at the level of threshold for some frequencies. Soft speech peaks are made audible throughout the spectrum.

Digital Noise Reduction (DNR) can easily be evaluated in approximately 30 seconds for a single setting, a bit longer when comparing different settings.

1. With DNR active, present a 65 dB noise input signal for up to 25 or 30 seconds.
2. An initial short term average curve is taken immediately after the signal is started continue stimulus presentation until DNR kicks in, taking another snapshot curve. This can range anywhere from 5-25 seconds.
3. Compare the two curves. If DNR is working the 2nd curve will demonstrate less gain and output than the curve initiated before DNR is engaged.



Figure 6. DNR measurement curves. The pink curve in the top graph is the short term average (STA) curve taken before DNR is activated. The yellow curve is the STA taken after DNR engaged. There is a noise reduction of 5 dB RMS observed.

The difference between the two curves can be used to determine if the noise reduction setting is appropriate for the client's signal-to-noise ratio (SNR) loss (as established via QuickSIN or other speech in noise procedure).

Directionality can be quickly assessed in 30-60 seconds.

1. Face the client away from the speaker.
2. Make an initial LTASS measurement for a 65 dB speech signal with the hearing instrument in omni-directional mode.
3. Make a second measurement with fixed-directionality engaged. (Or have fixed directionality engaged for both measurements and make measurement one with the client facing the speaker and measurement two facing away from the speaker.)

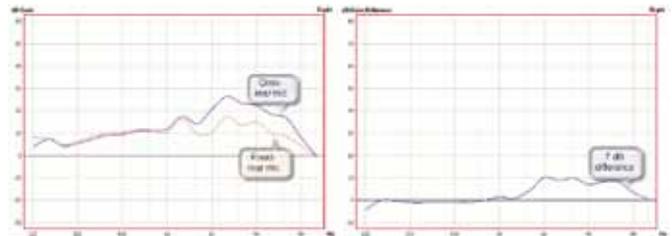


Figure 7. Omni versus Fixed Directionality measurement curves. The purple curve in the left graph is the response from the rear microphone in omni-directional mode. The orange curve is the response from the rear microphone in fixed directional mode. There is a difference of 7 dB RMS observed.

In either scenario, the rear fixed directionality condition should result in the lesser output. Again, these measurements can be related to the client's SNR loss.

Feedback suppression takes an additional 30 to 60 seconds to check. Depending on the slope of the hearing loss and the hearing instrument model being fit the amount of available stable gain before feedback can be quite high.

1. With the hearing instrument programmed to meet prescriptive targets turn off the feedback suppression algorithm.
2. Make an LTASS measurement for a 50 dB speech signal while increasing the gain programmed in the hearing instrument until feedback occurs. This is the stable gain before feedback.
3. Engage the feedback suppression algorithm and repeat the process. The second measurement is the added stable gain.

The difference between the two curves is an indication of how much headroom is available in the instrument and a quick check of feature performance.

Frequency lowering is now available on several of the major manufacturer devices and is very easily demonstrated in 10-45 seconds.

1. Present a 50 dB high-frequency pure-tone (6kHz, 7kHz, or 8kHz, for example) both with
2. and without frequency lowering active.
3. Alternatively, conduct the measurements using Ling sounds /s/ and /sh/.

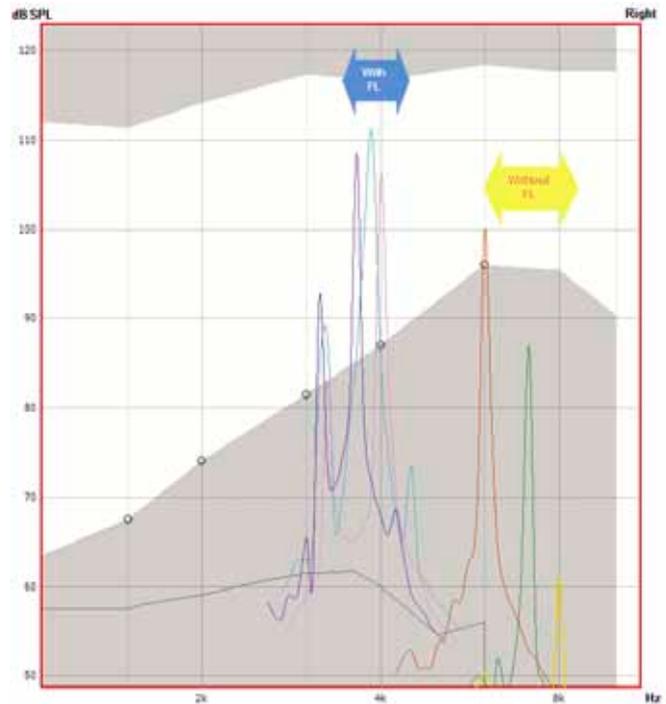


Figure 8. Frequency Lowering. High-frequency pure-tone signals as measured with and without frequency lowering enabled. The orange, green, and yellow curves are without frequency lowering; only 6 kHz is audible. The purple, blue, and pink curves are with frequency lowering enabled; all three frequencies are shifted into the client's dynamic range and made audible, while remaining spectrally distinguishable from one another.

If frequency lowering is functioning properly the signal's frequency will be shifted down in frequency depending on the settings. Tones or phonemes which are inaudible without frequency lowering enabled may become audible when frequency lowering is enabled.

The process can also be repeated with an LTASS measurement for speech to assess the dynamics of speech with and without frequency lowering active.

Modified Loudness Scaling and Loudness comfort can be achieved by using the PMM system as a sound level meter to present speech and “obnoxious sounds” in a controlled and calibrated manner. For details regarding the procedure for the Loudness Contour test the reader is referred to Cox, et. al. (1997). The essential aim of this testing is to adjust gain until 60-65 dB falls in the comfortable range and 85 dB is perceived as loud but ok or comfortable but slightly loud.

#7	Uncomfortably Loud
#6	Loud, But Okay
#5	Comfortable, But Slightly Loud
#4	Comfortable
#3	Comfortable, But Slightly Soft
#2	Soft
#1	Very Soft
#0	Cannot Hear

Figure 9. *The Cox loudness scale.*

5	Very Loud
4	Slightly Soft
3	Comfortable
2	Slightly Soft
1	Very Soft

Figure 10. *Shows a modified version of the scale used by the second author of this paper.*

Bentler, et. al. (2006) investigated annoyance and aversiveness of sounds and the impact of DNR strategies. Their results suggest the need for counseling clients about realistic expectations related to annoyance and aversiveness of sounds at the time of hearing aid fitting. They noted the importance for the hearing aid user to understand that the noises perceived as annoying through the hearing aid are also perceived as annoying by normally hearing listeners. Switalski (2011) describes a practical procedure using the RESR measurement to adjust the hearing instrument MPO and to communicate this to the hearing instrument user.

1. Perform an MPO measurement with the hearing instrument in place, both with and
2. without the instrument turned on.
3. The perception of loudness is compared in the two conditions and with that of their accompanying significant other (if applicable).

This provides an opportunity to reinforce the reality of loudness for both hearing-impaired and normal-hearing individuals. Furthermore, open canal fittings can often result in too high MPO because of the retained ear canal resonance. Performing comparative MPO measurements allows the clinician to identify this as a potential source of client discomfort and make appropriate adjustments.

In closing

As hearing instrument technology becomes more and more sophisticated the need for useful PMM procedures to assess performance becomes more and more critical. The ability to provide a competent, high-quality standard of care is an absolute must in order to compete in this age of high technology and of savvy consumers whose career spans continue to lengthen. The key is choosing techniques which add value to the fitting process, thereby ensuring audibility goals are met and promoting client satisfaction. This will ultimately elevate the client's perception of the clinical services provided, of their amplification, and of our industry in general. Modern PMM instrumentation provides nearly a blank canvas for validation and verification of modern hearing instru-

mentation and is not limited to the procedures contained in this paper, but also inclusive of the many advances and features which are likely to come.

Sections of this paper are based in part on the article "Three Probe Microphone Measurement Techniques to Enhance Open Fittings" which appeared in the October 2011 Hearing Review available in the HR Archives at www.hearingreview.com.

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