Current Issues In Probe Tube Microphone Measurements

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Introduction

In many ways, the field of real ear measurement is still in its infancy. While it is true that computerized probe tube microphone systems have gained acceptance and are being used fruitfully with all patient populations, there are many subtle features and test protocols which are still poorly defined. Other than the issue of terminology which seems to be changing monthly, the following are some issues which require serious attention. All have direct clinical ramifications and should be understood in order to maximize validity and minimize test artifact. The issues to be discussed include appropriate acoustic environment, reliability and validity, the transition region, the shape of the spectral stimulus, a comparison of real time versus stored calibration techniques, and brief descriptions of non-standard clinical applications of assessing telephone use, FM/hearing aid interactions, and hearing protection.

There have been many excellent recent publications in the area of real ear measurement. The reader may want to examine *Ear and Hearing, (Vol. 8, (5) Suppl.*, Oct. 1987) and *Hearing Instruments, (Vol. 39, (7)* July, 1988).

An Appropriate Acoustic Environment

There are three major areas in which the acoustic testing environment may alter or prevent the assessment of real ear measurement: (1) altering the calibrated equalized field, (2) reducing the usefulness of the noise reduction algorithms, and (3) level of background noise.

The first stage of operation of any real ear measurement device is to calibrate and equalize the sound field. Generally the probe tube effects are negated and the acoustic characteristics of the test room assessed in order to provide a well defined sound field. This is sufficient as long as the probe tube length does not change by more than 5 mm between calibrations and as long as the content of the room does not change significantly. Clearly this depends on the size and the reverberation characteristics of the test room, so caution should be exercised if the number and position of people near the client changes from test to test. If unsure, a new calibration run should be performed. The errors will tend to be more in the higher frequencies if the room calibration is altered.

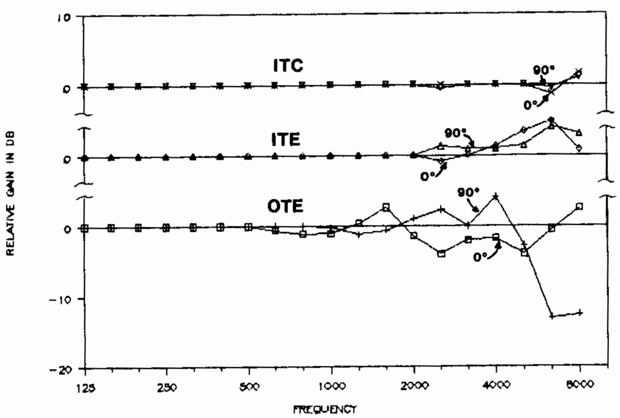
Many manufacturers tend to incorporate noise reduction algorithms in their software in order to reject or identify unwanted noise. Some manufacturers utilize phase data between the test and reference microphones; others use more or less sophisticated methods. Clearly, depending on the methods used, room acoustics will have a varying effect on the utility of these algorithms. As a rule of thumb, these algorithms start to become less optimal as the distance between the test and the reference microphone increases. The reduction in efficiency will be more noticeable in reverberant, noisy rooms than in sound treated enclosures.

Finally, the level of background noise will be more of a factor with insertion gain tests since low levels are chosen. Background noise is less important when SSPL-90 estimates are desired since levels of 85 or 90 dB may be utilized. (See also Hawkins, 1988.)

Reliability and Validity

A major issue in the study of real ear measurement is the determination of expected reliability. Figure 1 is from Killion and Revit (1987) and shows the standard deviations for different speaker angles and azimuths. The repeatability for five replications of insertion gain is shown for a speaker location straight above (90°,0°); for one directly in front (0°,0°); for 45° off to the hearing aid side, but on the horizontal plane (0°,45°); and for 45° off to the side, and 45° above the horizontal plane (45°,45°). As can be seen in Figure 1, the two conditions where the speaker is 45° off to one side have the lowest variability associated with them. The poorer reliability at 4000 Hz for the (0°,45°) condition with respect to the (45°,45°) condition is artifact. There is no reason why the 45° elevated condition should be any better than the non-elevated one. It probably is due to the small number of replications (Killion, 1988). The poorer reliability in the over the head condition (90°,0°) at 700 Hz is real, and this is due to "shoulder bounce" where sound is reflected off the shoulder (Killion, 1988). This should be of concern if the stimulus azimuth of choice exceeds 60°. It should also be noted that the reliability at 45° is on the order of a $\sqrt{2}$ increase in *S.D.* from the data for the ear canal SPL

Figure 1. Test-retest variability (average within-subjects standard deviation for five replications) in insertion gain for the four loudspeaker locations.



From Killion, M.C., & Revit, L.J. Insertion gain repeatability versus loudspeaker location: You want me to put my loudspeaker WHERE? *Ear and Hearing, 8, Suppl.*, 1987, p. 72S, reprinted with permission.

developed by an insert ear phone. This "indicates that about half the variance is coming from variations in closed-ear impedance and half from across- subject differences in the external ear effects" (Killion & Revit, 1987).

Another factor which affects reliability is the proximity of the end of the probe tube from the ear drum. Since standing waves are generated in the ear canal (and can be calculated approximately with a quarter wavelength model), taking a measurement in a region where the slope of the standing wave is great may cause large excursions in two seemingly identical measures. That is, a small change in probe tube location may result in a significant difference in the sound pressure level. If caution is used to ensure an identical position in the unaided and aided conditions, the insertion gain measurement (which is the difference) would be accurate. If there is a tendency, however, for slight movements to be made between these two conditions, it is advisable to do both measurements near the ear drum. The magnitude of the standing wave is at a minima at this relatively high impedance juncture.

The data from Stinson, Shaw, and Lawton (1982) suggest that if the impedance of the ear drum/middle ear system is pathologically high (as in otosclerosis), then measurements should be made as close to the ear drum as possible. As the impedance increases, there is more ear drum reflection and therefore the magnitude of the standing wave becomes greater. Killion and Revit (1987) also show that head movement in the horizontal plane of ± 10° results in insignificant changes up to 8000 Hz, and head movement in the vertical plane is less than 1 dB up to 5000 Hz, but may be 3 dB above 6000 Hz. This would be less of a problem with broader bandwidth stimuli.

In summary, the probe tube location which will yield the most reliable measure is one which is near the ear drum. Additionally, the loud speaker location which provides the best reliability is 45° off to the side. But is this a valid speaker position? Killion and Monser (1980) in reviewing some studies in the area of reflection of sound and perception note that with music in a typical listening environment as much as 90% of the energy reaching a listener's ear may be reflected; and this is true to a lesser extent with speech. The further a speaker is from the listener, the greater the proportion of reflected energy. They suggest that a random incidence field (or a diffuse field) may be a good representation of the way speech is heard in a typical reverberant environment.

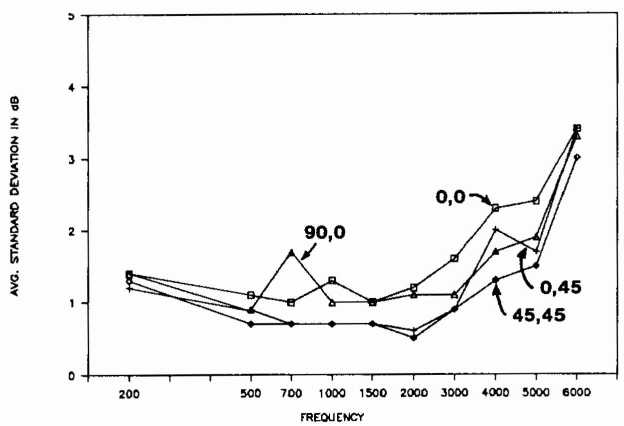
Figure 2 (from Killion & Revit, 1987) shows the difference between a random incidence field and one with a 0° incidence and a 90° incidence, for three styles of hearing aids–in the canal, in the ear, and behind the ear (OTE). Note that for the in the canal aid, as long as the microphone is seated deeply, for up to 5000-6000 Hz, a random incidence field is approximately equal to a 0° or 90° (and 45° but not shown) incidence. For in the ear aids, there is a minimal difference starting above 2000 Hz, and perhaps a correction factor can be used in the higher frequencies to express "in the ear insertion gain" as "in the ear random incidence insertion gain." For behind the ear aids (OTE), Killion and Monser (1980) suggest that in order to increase validity, one could "average the 0° and 90° incidence data for hearing aids whose response extends into the 5 to 10 kHz region."

Clearly for all head worn hearing aids, the differences are in the higher frequencies and are minute. Clinically, we may be more interested in the difference between one aid and another, rather than the absolute value of insertion gain.

The Transition Region

In occluded (aided) ear canals, due to the sound erupting from a small duct into a larger one (i.e., the hearing aid tubing [1-2 mm] into the ear canal [6-8 mm]), radial waves are produced which interact with the planar (longitudinal) waves, causing a non-uniform field to develop. Because radial waves tend to be damped quickly, this field only exists for 4-5 mm beyond the end of the earmold tip. The extent of this field (also called the transition region or region of spreading inertance) depends on frequency, ear canal geometry, and on the nodal characteristics of the standing wave associated with the radial wave (Burkhard & Sachs, 1975). There is some evidence that the field may be shorter for larger aperture earmolds and therefore may be less significant for Libby Horn earmolds (Chasin, 1985). Clinically, if the probe tube terminates at a point at least 5 mm beyond the earmold tip, there is no danger of being in the transition region.

Figure 2. Expected shape of measured insertion-gain curve for 0° and 90° loudspeaker locations for three types of hearing aids, each having flat insertion gain frequency response curve when measured in a diffuse (random incidence) sound field: ITC (canal) aids, ITE (in-the-ear) aids, OTE (over-the-ear) aids.



From Killion, M.C., and Revit, L.J. Insertion gain repeatability versus loudspeaker location: you want me to put my loudspeaker WHERE? *Ear and Hearing, 8, Suppl.*, 1987, p. 70S, reprinted with permission.

Spectral Shape of the Stimulus

Currently little is known about the "optimal stimulus" for probe tube microphone measurements. We do know, however, what it should not be! A stimulus is not appropriate if standing waves cannot be controlled. This includes a sweep sinusoid signal, an overly narrow warble (constant percentage) or woble (constant Hz), and a band of sinusoids. Several manufacturers market a stimulus which is a bank of eighty puretones, each separated by 100 Hz. The higher frequency sinusoids within this bank will have standing waves associated with them (typically above 3000 Hz).

A complex noise may be better for assessing ASP hearing aids (although we should first decide what it is we want to assess about them). Caution should be used in order to ensure that the level does not saturate the aid. With this in mind one should know the RMS value of the stimulus as well as the peak value of the stimulus (the difference being the crest factor). If a complex noise (or bank of sinusoids) is used, either a Fast Fourier Transform (FFT) or a swept narrow band tracking filter is needed to get frequency specific data (Preves, 1987). Some FFT algorithms improperly take a sample and cause high frequency components to enter into the calculation. This easily can happen if a manufacturer's FFT is used with different stimuli than what was selected to be used with the FFT.

Needless to say, this is still a very controversial area. While most manufacturers are moving to offer a broader range of stimulus types, primarily for reasons of marketing competition, this should not be confused with a resolution of this issue. This may be one of the last issues in this field to be standardized. An interim suggestion would be to use a warbled sweep, since this is an easy stimulus to control, is relatively free of test artifacts, and artifacts when they do occur are clearly evident.

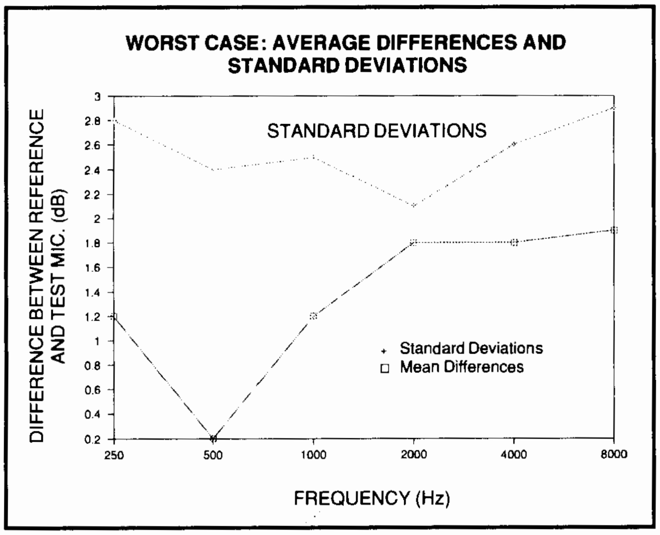
Real Time and Stored Calibration Techniques

The most frequently used *real time* calibration method is the pressure method. In this method, the reference microphone is alive and is constantly updating or changing the calibrated field. Patients are allowed to move during this test (within limits), but head diffraction and body baffle effects cannot be computed explicitly. Since the reference microphone is in the diffracted field, it chooses the diffracted level as the new 0 dB calibration point. Strictly speaking this method is referred to as the "modified" pressure method, since the reference microphone is not necessarily at the hearing aid microphone location. However, the reference microphone position can be moved at least 10 cm from the hearing aid microphone port, and no statistically significant differences are observed up to 8000 Hz (Chasin, 1988). Figure 3 shows this lack of difference based on 20 runs of the same subject with a 10 cm spacing between the reference and the test microphones. Note that the respective standard deviations are higher than the means, attesting to the lack of statistical significance.

The most frequently used *stored* calibration technique is substitution. In this method, a static point in space is calibrated, and this information is stored. The patient is substituted into the calibrated field with the hearing aid in place (and the probe tube is at or very near the ear drum), and the difference is computed. This method allows the reference microphone to be disabled during the hearing aid run. Because the reference microphone is disabled, the head diffraction and the body baffle effects can explicitly be computed and will be observed in the ear canal resonance curve and the in situ curve. Because the insertion gain is the difference between two similar comparable measures, head diffraction and body baffle effects will be subtracted out and therefore will not be observed in the insertion gain curve. If baffle and diffraction effects are of interest, only a stored calibration technique such as the substitution method can be used. An advantage of a stored method is that leakage from an acoustic vent cannot bias the reference microphone (since it is disabled). This is a problem however with a real time pressure method. These differences will be observed in the lower frequencies (Sullivan, 1987) but may be observed at 4500-5500 Hz in cases of earmolds with diagonal (or Y-) vents (see also Cox, 1979).

A final characteristic of the substitution method is that the results from the in situ curve can be used directly to estimate

Figure 3. Mean differences and standard deviations are shown for 20 repetitions on one subject. There is a 10 cm spacing between the reference and test microphones.



the 2 cc coupler SSPL-90, because of where the probe tube has been situated and because the diffraction and baffle effects are evident in the substitution in situ curve (Preves, 1987). In summary, if an insertion gain estimate is all that is required as is the case in most clinical settings, then use of either the pressure or the substitution method will yield identical results.

Non-Standard Hearing Aid Work

The following is a brief description is some non-standard clinical applications of computerized probe tube microphone systems.

Telephone/Telecoil Testing

Using the substitution calibration method, a telephone handset is connected through the speaker output of the probe tube microphone. A 150Ω (1/2 watt) resistor is placed in series, in order to approximate the characteristic impedance of the telephone set (Arndt, 1985). Since the reference microphone cannot be biased by the telephone handset, three conditions of gain can be assessed: (1) hearing aid on the "T" position; (2) hearing aid on the "M" position; and (3) without the hearing aid in place. The first two will yield insertion gain results, and the third condition is done as an in situ measurement. The configurations of the gain for all three methods are directly comparable and can be used expediently to counsel your patient on the best telephone method to use.

Hearing Aid/FM System Testing

Using either the substitution or the pressure method, the FM transmitter microphone is suspended from the ceiling on a string. The transmitter microphone is at +45° from the loud speaker, and the student is sitting at -45° at the same distance. It is crucial that the transmitter and the student are at the same distance from the speaker for reasons of calibration. The hearing aid is turned on with the probe tube microphone in place in order to assess the insertion gain at 1000 Hz. The FM transmitter is attached to the hearing aid, and the aid's microphone is disabled. An 80 dB stimulus is used in order to assess the gain of the system on all hearing aid volume settings and on all FM volume settings. Based on gain requirements and the maximum level before saturation, the appropriate volume settings are determined.

Real Ear Attenuation of Hearing Protection

Using a probe tube microphone, the ear canal resonance is measured, and the insert ear protector is inserted. The difference is dB between the unplugged and the plugged condition is the frequency specific protection derived from the earplug, when inserted correctly. For earmuffs, the in situ data is all that is required, since there is no insertion loss with circumaural ear muffs. Caution should be taken if the attenuation exceeds 40 dB at 2000 Hz, since a bone conducted route will enter the computations. Any real ear result will then overestimate the true protection (Berger, 1986).

Conclusion

We now know more about the optimal angle and azimuth of the speaker, the stimulus type, the relationship to validity, the facts and artifacts with commonly used calibration procedures, and the non-standard uses of probe tube microphone devices. We still do not have sufficient data and clinical knowledge of the characteristics of both the unoccluded and the occluded ears at frequencies above 8000 Hz. The optimal stimulus and the distance of the stimulus from the speaker still need to be researched.

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Call For Papers—CAA Meeting

Acoustics Week in Canada, October 1989

Acoustics Week in Canada will be held at the Chateau Halifax, Halifax, Nova Scotia from 16 to 19 October. The event will begin with two days of courses on topics including underwater acoustics, condominium acoustics, and sound intensity, followed by a full technical program for two days.

The meeting provides opportunities for members of the Canadian Acoustical Association and other interested parties to exchange and share information about all aspects of acoustics. The convener of the meeting is Mr. Bob Cyr, Nova Scotia Power corporation, P.O. Box 910 [or street address, 5261 Duke St., Duke Tower, Suite 418, Scotia Square] Halifax, Nova Scotia, B3J 2W5 (902) 428-6589. Miss Margaret Cassidy, who is serving as the Secretariat, may also be contacted for further information at the same address, phone (902) 428-6214; FAX (902) 428-6100; Telex.: 019-21736.

Technical Program

Contributed Symposia, individual papers, and posters are invited from all areas of acoustics including but not restricted to architectural, underwater, industrial, environmental, physical, physiological, musical, sound recording, psychological, noise control, ultrasonic, and infrasonic acoustics. Highlighting the program are a special Plenary Session on Weather Observation Through Ambient Noise (WOTAN) and Invited Symposia on Underwater Acoustics, Speech Communication, and Physiological Acoustics.

All accepted one-paragraph abstracts will be published in *Canadian Acoustics*. A refereed conference proceedings will be published containing accepted full papers. It is not necessary to submit a full paper for participation in the program; an abstract is all that is required. For further information about submission of abstracts and proceedings contact the Technical Program Chair: Dr. A.J. Cohen, Department of Psychology, Dalhousie University, Halifax, Nova Scotia B3H 4J1, Telephone: (902) 424-8888. (Bitnet E-mail address is ACOHEN@DALAC).

Student Awards

There will be up to three awards of $500 made for the best contributions by students as judged by an Awards Committee chaired by Dr. Bruce Dunn, University of Calgary. The award is based on oral presentation and thus to be eligible for the award, student papers must be presented as a lecture. The paper may be co-authored, but the student must be the first author on the paper. Students must be currently enrolled in a graduate program and must complete a short form to indicate their candidacy.