Test Conditions, Stimuli, and Calibration Values for Sound Field Testing

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At present there is no Canadian or American standard for sound field audiometric testing. Appropriate stimulus characteristics and conditions for sound field testing have been summarized by Walker, Dillon, and Byrne (1984), and Skinner (1988). Reference equivalent threshold sound pressure levels (RETSPL) for sound field stimuli have been suggested for pure tones (ISO R226-1961), frequency modulated stimuli (Morgan, Dirks, & Bower, 1979), and narrow bands of noise (Pascoe, 1975 in Skinner, 1988). The available stimuli on clinical audiometers have different characteristics from those proposed by the various research groups, which leaves the clinician with some question as to the appropriate calibration values and stimulus parameters to be used. Several factors which should be considered when determining the appropriate test conditions, stimuli, and calibration values to be used for sound field audiometry will be discussed. As well, a general technique for evaluation of sound field calibration is suggested.

Theoretical Considerations

Sound Field Stimuli

Various stimuli are available for use in sound field audiometry, including: pure tones, frequency modulated (FM) tones, and narrow bands of noise. It is desirable that measurements obtained with stimuli transduced by loudspeaker in a sound field can be related to (i.e., have equivalence with) measurements made using pure tones transduced by an earphone. Therefore the test signal must be reasonably frequency specific (Dillon & Walker, 1982). In order to compensate for possible head movement, it is also desirable that the sound pressure level of the test signal be stable (i.e., low variability) in the region of the sound field that would normally be occupied by the subject's head (Dillon & Walker, 1982).

The ideal signal for frequency specific testing is a pure tone. However, pure tone stimuli are not satisfactory for sound field testing in the reverberant field because standing waves are produced which may, in combination with small changes in head position and/or drifts in the test signal frequency, result in significant changes in sound pressure level at the subject's ear (Walker, Dillon, & Byrne, 1984). When the subject and sound source are in close proximity (i.e., direct field) then the signal level follows the inverse square law. In the direct field the sound pressure level is only minimally influenced by room acoustics so that pure tones may be used for testing (Duffy, 1978). If pure tones are used in the direct field, then it is crucial to maintain a fixed distance between the sound source and the subject's ear because the signal level is highly dependent on distance from the source (Walker, Dillon, & Byrne, 1984). In the typical clinical environment, audiometric testing is not performed in an anechoic chamber, the subject is not held rigidly in position, and the subject may or may not be placed in the direct field portion of the test booth sound field. Therefore, in order to reduce the influence of sound pressure level variability on the psychoacoustic measures obtained in sound field, it is necessary to use a complex stimulus for testing.

Complex stimuli, composed of several frequency components centered around the test frequency, are used to overcome some of the problems inherent in using pure tone stimuli for testing in a sound field. In particular, complex stimuli are used to overcome standing wave effects by reducing the contribution of any one frequency component to the measurement of threshold. The principle is that, if the sound pressure level for a series of adjacent frequencies (which contain a sound pressure minimum or peak resulting from a standing wave) are measured in sound field, then the resulting average is not influenced to a large extent by the sound pressure level variation of any one frequency component (Walker & Dillon, 1983). The total sound pressure level of the test signal is the average of a larger number of frequency components, of which only a small proportion are influenced by the standing wave. Therefore the individual's threshold will be reflected by the average sound pressure level of the signal rather than the largest level. In order for a complex signal to reduce measurement variability in the sound field, the stimulus bandwidth must be wide enough and contain sufficient acoustic energy to overcome the influence of the standing wave. As well, the sound pressure level of frequency components within the frequency band of the signal must have a reasonably uniform distribution of energy, with no prominent peaks or troughs. Frequency modulated tones and narrow bands of noise are the most commonly used stimuli for clinical sound field testing, and both are considered to be acceptable stimuli under certain conditions (Morgan, Dirks, & Bower, 1979; Walker, Dillon, & Byrne, 1984; Skinner, 1988).

Narrow bands of noise are described by the center frequency of the band, the bandwidth of the noise (e.g., 1/3 octave narrow band noise) and the steepness of the filter slope (e.g., 60 dB/octave) or out-of-band rejection rate. The bandwidth of a 1/3 octave narrow band of noise is a constant percentage (23%) of the center frequency. The bandwidth (in Hz) of a 1/3 octave band can be calculated by multiplying the center frequency of the noise band by the bandwidth percentage. For example, if the center frequency is 1000 Hz and the noise band is 1/3 octave wide (23%), then the bandwidth is 230 Hz. The steepness of the filter slope determines the degree to which the signal is reduced outside of the bandwidth limits. A filter slope of 30 dB per octave is shallow and the narrow band noise signal thus generated will contain significant acoustic energy outside of the bandwidth limits, making the signal a poor choice for sound field testing. A filter slope of 60 dB per octave or greater is steep, and the narrow band noise signal thus generated will contain less acoustic energy outside of the bandwidth limits, making the test signal a better choice for frequency specific sound field testing.

Frequency modulated tones are described by the frequency deviation (e.g., ±5%) of the tone and the modulation rate (e.g., 10 Hz). When the frequency deviation is expressed as a constant percentage, then the bandwidth of the signal can be calculated and is the frequency sweep times the center frequency (Dillon & Walker, 1982). For example, if the center frequency is 1000 Hz and the frequency deviation is ±5%, then the total frequency sweep is 10% and the bandwidth is 100 Hz. The modulation rate determines the spacing, and therefore the number of frequency components that will appear in the resulting signal (Dillon & Walker, 1982). If the modulation rate is 5 Hz, then frequency components will occur at 5 Hz intervals. If the modulation rate is 20 Hz, then frequency components will occur at 20 Hz intervals. Therefore, a higher modulation rate will provide for less distribution of the signal across the frequency spectrum. If the modulation rate is large enough (i.e., approaches the frequency deviation), then only one frequency component will contain all of the acoustic energy and the signal will effectively be a pure tone (Dillon & Walker, 1982).

The shape of the modulation signal (i.e., waveform) used to generate a frequency modulated tone also has an effect on the resulting signal. There are four basic types of waveforms used for modulation: square wave, ramped wave (i.e., sawtooth), triangular wave, and sine wave. The waveform affects the out-of-band rejection rate and the degree of amplitude modulation within the band. Both the square waveform and the ramped waveform are poor for use in the generation of frequency modulated signals for sound field testing because the resultant waveform has shallow out-of-band rejection of power (i.e., significant energy outside of the bandwidth). Either a sinusoid or triangular wave is suitable for use in the generation of frequency modulated tones for sound field testing. The sinusoid waveform generates a signal with steep out-of-band rejection rate but has less uniform amplitude within the band than does the signal generated by triangular waveform modulation. The triangular waveform generates a frequency modulated tone with a less steep out-of-band rejection rate than does the signal generated by a sinusoid waveform, but the signal generated by a triangular wave has a more uniform amplitude within the band limits (Dillon & Walker, 1982).

For narrow bands of noise the instantaneous frequency spectrum and amplitude varies randomly over time. For frequency modulated tones the instantaneous frequency spectrum and amplitude varies systematically (i.e., regularly) over time. Generally, the instantaneous fluctuations in amplitude are slightly larger for narrow bands of noise than they are for frequency modulated tones, when both signals have the same bandwidth (Dillon, Walker, & Byrne, 1984). As well, the individual frequency components of both narrow bands of noise and frequency modulated tones are equally affected by room acoustics (i.e., standing waves) and loudspeaker nonlinearities, which may produce intensity alterations in the test signal. However, when the signal bandwidth is sufficiently wide, the effect of intensity alterations, to individual frequency components, on the overall level of the test signal is minimized.

For frequency modulated tones, modulated at a rate of 5 Hz, any single frequency component, and the resultant instantaneous alteration in amplitude, is present for a short period of time relative to the integration time (approximately 200 ms) of the normal human auditory system (Walker & Dillon, 1983). The temporal integration time of individuals with a sensorineural hearing impairment may be significantly shorter than that of normal listeners, and therefore needs to be considered when selecting modulation rate (Walker & Dillon, 1983). The modulation rate should be high enough that the amplitude fluctuations can be fully integrated by the abnormal temporal integration time of hearing impaired listeners. However, the modulation rate should not be made too large because increasing the modulation rate also increases the out-of-band acoustic energy and reduces the number of frequency components within the band. Walker and Dillon (1983) recommend that the modulation rate be at least 20 Hz, but that at the same time the modulation rate should not be more than 1/3 of the frequency deviation.

Appropriate stimulus characteristics for frequency specific sound field testing represent a trading relationship between the need for frequency specificity (narrow bandwidth) and low sound field variability (wide bandwidth). The variability in a sound field, when measured at points of 20 cm distance around the center calibration point, has been shown to decrease when the percentage bandwidth of the signal increases, from 1% to 20% (Dillon & Walker, 1982). Variability is reduced with increasing bandwidth for all types of stimuli, including narrow band noise and frequency modulated tones. Dillon, Walker, and Byrne (1984) recommend stimulus bandwidths which should reduce the sound pressure level variability in the sound field to within 2 dB of the nominal value. The suggested bandwidths for sound field stimuli vary by center frequency and are on the order of 29% at 250 Hz to 8% at 4000 Hz for standard audiometric testing (Walker, Dillon, & Byrne, 1984).

A disadvantage of increasing stimulus bandwidth, to reduce errors in measurement arising from sound field variability, is that the individual's hearing loss may be underestimated at certain frequencies. Unless the individual's threshold occurs at the same level across the frequency spectrum, it is possible that some of the stimulus energy will fall into frequency regions where the subject has better hearing than at the test center frequency. When this occurs, the measured threshold may be better than the actual threshold is at the test center frequency (Walker, Dillon, & Byrne, 1984; Popelka & Mason, 1987; Skinner, 1989). In particular, this may occur when the individual has a high frequency steeply sloping (e.g., 75 dB per octave or greater) hearing loss (Popelka & Mason, 1987). When the difference in hearing threshold between the adjacent test frequencies—1000-1500 Hz or 2000-3000 Hz—is 45 dB or greater, then the threshold measured with a 1/3 octave narrow band of noise having an out-of-band rejection rate of 90 dB per octave will differ significantly from the threshold measured with a pure tone (Popelka & Mason, 1987). Similarly, when the difference in hearing threshold between the adjacent test frequencies 1500-2000 Hz is 30 dB or greater, then the threshold measured with a 1/3 octave narrow band of noise having an out-of-band rejection rate of 90 dB per octave will differ significantly from the threshold measured with a pure tone (Popelka & Mason, 1987). Thus, the bandwidths recommended by Dillon, Walker, and Byrne (1984) represent a compromise between maintaining low sound field variability and being able to test individuals with a high frequency steeply sloping hearing loss.

Table 1. Variability in sound pressure level (dB), measured at the eardrum, resulting from ±15° head rotation, for stimuli transduced by loudspeakers at five positions of azimuth (from Shaw & Vaillancourt, 1985).

Frequency (Hz)	Azimuth 0°	30°	45°	60°	90°
250	0.8	0.8	0.7	0.5	0.1
500	2.2	2.1	1.6	1.1	0.2
750	2.9	2.2	1.8	1.4	0.2
1000	3.3	2.4	2.0	1.5	0.3
1500	2.9	2.2	1.9	1.5	0.3
2000	2.5	1.6	1.0	0.4	2.1
3000	4.5	2.7	1.1	0.8	3.9
4000	4.7	1.9	0.7	2.2	4.2
6000	5.2	4.8	3.7	2.0	1.7
maximum	5.2	4.8	3.7	2.2	4.2
mean	3.2	2.3	1.6	1.3	1.4

Loudspeaker Position

The choice of loudspeaker test position (i.e., azimuth) can be made on the basis of two factors: (1) the effect of mild head rotation on signal level, and (2) the amount of signal attenuation to the nontest ear resulting from head shadow effects.

The change in sound pressure level at the eardrum associated with mild rotation (±15°) of an adult subject's head can be calculated from Shaw and Vaillancourt (1985) and is shown in Table 1. Variation in sound pressure level at the eardrum due to mild rotation of the subject's head is a function of both stimulus frequency and loudspeaker azimuth. The largest variation is observed for loudspeakers positioned at 0° azimuth (mean = 3.2 dB). The smallest variation occurs for loudspeakers positioned at 60° azimuth, where the variation is 2.2 dB or less for frequencies between 250-6000 Hz (mean = 1.3 dB). If testing is limited to frequencies up to and including 4000 Hz, then loudspeaker azimuths between 30° and 60° will reduce variation resulting from mild head rotation to below 3 dB.

The difference in sound pressure level between the ears of an adult subject, measured at the eardrum, for stimuli delivered from a loudspeaker positioned at azimuths between 0° and 90° can be calculated from Shaw and Vaillancourt (1985) and is shown in Table 2. Attenuation of the test signal to the contralateral ear is a function of both stimulus frequency and loudspeaker azimuth. Stimuli delivered from loudspeakers positioned at 0° azimuth are affected equally, at each ear, by the subject's head and pinnae. Therefore, for stimuli presented via a loudspeaker positioned at 0° azimuth,

Table 2. The difference in sound pressure level (dB) between the two ears, measured at the eardrum, for stimuli transduced by loudspeakers at five positions of azimuth (from Shaw & Vaillancourt, 1985).

Frequency (Hz)	Azimuth 0°	30°	45°	60°	90°
250	0.0	1.6	2.3	2.8	3.3
500	0.0	4.2	5.6	6.0	5.5
750	0.0	6.6	9.2	9.1	6.8
1000	0.0	6.6	9.2	9.1	6.8
1500	0.0	6.0	9.3	11.7	7.5
2000	0.0	5.0	7.3	9.3	8.9
3000	0.0	8.2	10.9	11.0	11.4
4000	0.0	9.3	13.3	15.2	12.9
6000	0.0	10.6	16.3	19.7	20.7
mean	0.0	6.5	9.3	10.4	9.3

there is no attenuation of the test signal at the nontest ear, when measured relative to the level at the test ear. The average attenuation of the test signal at the nontest ear is maximum for a loudspeaker test position of 60° azimuth (mean = 10.4 dB). Loudspeakers positioned at either 45° or 90° azimuth provide average attenuation across frequencies of similar magnitude to the attenuation resulting from head shadow effects for a loudspeaker positioned at 60° azimuth.

The data presented in Table 1 and Table 2 suggest that testing with a loudspeaker positioned at either 45° or 60° azimuth will maximize attenuation of the test signal to the nontest ear while minimizing the effects of mild head rotation. It should be remembered that these results are based on data collected in a free field with pure tones. The effects may be less pronounced in the typical sound field of an audiometric test booth when testing with complex stimuli.

Alternatively, the tester may choose to position the loudspeaker at 0° azimuth for sound field testing, as advocated by Skinner (1988). Having the listener face the loudspeaker at 0° azimuth simulates the typical position in which a person receives a message from a second person (i.e., normal conversational dyad). Another advantage of testing with a loudspeaker positioned at 0° azimuth is that only one loudspeaker is required for testing. One trade-off for testing with loudspeakers positioned at 0° azimuth is that there is an increase in test variability resulting from the effects of mild head rotation. As well, there is no attenuation of the test signal to the nontest ear for loudspeakers positioned at 0° azimuth.

Calculation of Minimum Audible Pressure

Calibration values for sound field stimuli will differ from the ANSI (S3.6-1969) values for pure tones transduced by earphones because of differences in calibration and stimulus presentation conditions for each test procedure. The ANSI (S3.6-1969) standard specifies the sound pressure level (dB re: 20μ Pa) an earphone should produce in a 6cc coupler. The sound pressure level developed at the listener's eardrum during stimulation via earphones is different from the 6cc coupler value (Shaw, 1966). Calibration of sound field is performed with the subject absent, and the measurement microphone positioned at the presumed center of the listener's head. The sound pressure level developed at the listener's eardrum during stimulation via loudspeakers is different from the sound field value obtained with no listener present (Shaw, 1974).

There is no difference in threshold for stimuli presented via earphones or loudspeakers when the actual sound pressure level (dB SPL) developed at the eardrum is considered (Killion, 1978). The sound pressure level of a signal which represents the listener's threshold is called the minimum audible pressure (MAP) when the sound pressure level is measured at the tympanic membrane of the listener (Sivian & White, 1933). In order to obtain the same sound pressure level at the eardrum (MAP) when testing with earphones and loudspeakers it is necessary to use different calibration values for the two conditions. The sound pressure level of a test signal, delivered via a particular transducer configuration, which will result in the same sound pressure level at the eardrum as that produced by earphones conforming to the ANSI standard (ANSI, S3.6-1969), is called the reference equivalent threshold sound pressure level (RETSPL).

Sound field RETSPLs for frequency modulated tones (with +5% frequency deviation, at 6 Hz modulation rate) transduced by loudspeakers positioned at 45° azimuth have been suggested by Morgan, Dirks, and Bower (1979). Sound field RETSPLs for 1/3 octave narrow band noise (filter slope greater than 50 dB per octave) transduced by loudspeakers positioned at 0° azimuth have been suggested by Pascoe (1975). Recent data by Cox (1986) and Shaw and Vaillancourt (1985) make it possible to calculate the minimum audible pressure (estimated MAP) for each test condition.

The data necessary for the calculation of estimated MAP are shown in Table 3. The estimated MAP for each test condition is obtained by adding the appropriate acoustic transfer function to the RETSPL value. The estimated MAP for testing with earphones is calculated by adding the 6cc to eardrum transform of Cox (1986) to the ANSI calibration values (ANSI, S3.6-1969). The estimated MAP for sound

Table 3. Data required for the calculation of estimated MAP (dB SPL at eardrum) values: a) for pure tones transduced by earphones (ANSI S3.6- 1969; Cox, 1986), b) for 1/3-octave narrow band noise transduced by loudspeakers positioned at 0° azimuth (Skinner, 1988; Shaw & Vaillancourt, 1985), and c) for 5% frequency modulated tones transduced by loudspeakers positioned at 45° azimuth (Morgan, Dirks, & Bower, 1979; Shaw & Vaillancourt, 1985). Values marked with an asterisk are interpolated.

a) Earphone

	Frequency (Hz)								
	250	500	750	1000	1500	2000	3000	4000	6000
ANSI	25.5	11.5	8.0*	7.0	6.5	9.0	10.0	9.5	15.5
transfer	-7.8	1.6	2.3*	2.8	3.4*	6.3	6.9*	5.1	1.2*
MAP	17.7	13.1	10.3	9.8	9.9	15.3	16.9	14.6	16.7

b) Loudspeaker 0°

	Frequency (Hz)								
	250	500	750	1000	1500	2000	3000	4000	6000
SF-0	15.0	11.0	7.5*	7.0	5.0	3.0	-2.0	-3.0	6.0
transfer	1.0	1.8	3.0*	2.6	5.1*	12.0	15.4	14.3	7.3
MAP	16.0	12.8	10.5	9.6	10.1	15.0	13.4	11.3	13.3

c) Loudspeaker 45°

	Frequency (Hz)								
	250	500	750	1000	1500	2000	3000	4000	6000
SF-45	20.2	7.8	4.2*	3.7	2.6	3.8	-2.9	-4.4	3.5
transfer	2.2	5.0	6.5*	6.4	8.7*	15.0	20.2	18.3	14.7
MAP	22.4	12.8	10.7	10.1	11.3	18.8	17.3	13.9	18.2

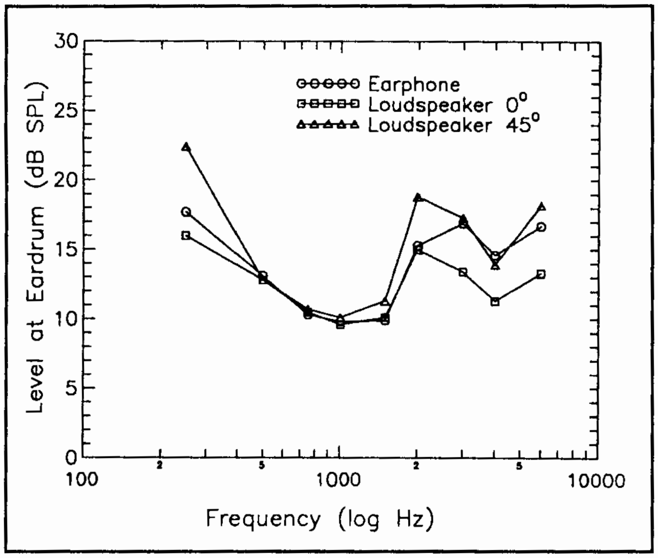
field testing with loudspeakers positioned at 0° azimuth is calculated by adding the sound field to eardrum transform from Shaw and Vaillancourt (1985) to the RETSPL values of Pascoe (Skinner, 1988). The estimated MAP for sound field testing with loudspeakers positioned at 45° azimuth is calculated by adding the sound field to eardrum transform of Shaw and Vaillancourt (1985) to the RETSPL values of Morgan, Dirks, and Bower (1979).

The calculated estimates of MAP (the sound pressure level produced at the listener's eardrum) for each transducer configuration are plotted in Figure 1. The average maximum difference, between each estimate of MAP, is 2.8 dB for the frequencies considered (i.e., 250-6000 Hz). The largest difference between the MAP estimates is 6.4 dB, and this occurs at 250 Hz. The RETSPL for sound field calibration of loudspeakers positioned at 45° azimuth might be high at that frequency; however, it should be remembered that the estimated MAPs are based on transfer functions which represent the average human ear. In general, agreement between thresholds obtained using the recommended RETSPL values with loudspeakers positioned at the correct azimuth and appropriate stimuli should be very good.

Practical Considerations

It is recommended that the sound field test environment be calibrated using procedures described by Walker, Dillon, and Byrne (1984), and Skinner (1988). Clinical audiometers generally offer stimuli of one constant percentage bandwidth: 10% for frequency modulated tones (i.e., ±5% frequency deviation and 5-10 Hz modulation rate) and 23% bandwidth (i.e., 1/3 octave) for narrow bands of masking noise. Further-

Figure 1. A plot of the three estimates of MAP calculated in Table 2. The MAP estimates are for: a) pure tone stimuli transduced by earphones (Earphone), b) narrow bands of noise transduced by loudspeakers positioned at 0° azimuth (Loudspeaker 0°), and c) frequency modulated tones transduced by loudspeakers positioned at 45° azimuth (Loudspeaker 45°).



more, the 1/3 octave narrow band noise typically is intended for use as a masking stimulus and may not have a sufficiently steep out-of-band rejection rate for use in frequency specific threshold determination, particularly at the higher frequencies (i.e., above 1000 Hz). In order to test with stimuli that approximate the bandwidths for standard audiometric testing (Walker, Dillon, & Byrne, 1984), it is suggested that 1/3 octave narrow band noise be used when testing with low frequency stimuli (i.e., 250-1000 Hz), and that frequency modulated tones be used when testing with high frequency stimuli (i.e., 1500- 6000 Hz). The use of 5% frequency modulated tones (5-10 Hz modulation rate) is not recommended for testing threshold at frequencies below 1000 Hz because the bandwidth is smaller, and the modulation rate is lower, than that recommended (Walker & Dillon, 1983; Walker, Dillon, & Byrne, 1984). As long as the narrow band noise has a bandwidth of 1/3 octave and has suitable out-of-band rejection (i.e., 60 dB/octave), it is recommended for testing threshold at frequencies from 250-1000 Hz. The bandwidth of ±5% frequency modulated tones is preferable to that of 1/3 octave narrow band noise for testing at frequencies from 1500-6000 Hz. For that reason the use of 1/3 octave band noise is not recommended when suitable frequency modulated tones are available. It is strongly recommended that the RETSPL values used for sound field audiometry by the clinic should be evaluated for equivalence to the ANSI (S3.6-1969) standard.

Procedure for Setting Up the Sound Field

The first step in setting up a sound field for audiometric testing is to decide upon the test conditions and stimuli to be used. The choice of stimuli is largely determined by what is available on the audiometer. If both frequency modulated tones and narrow bands of noise are used, then each stimulus type should be calibrated separately. The appropriate RETSPL values are determined by the loudspeaker test position (i.e., azimuth) and not the stimulus type. Therefore, once the test position is chosen, the RETSPL corresponding to the loudspeaker azimuth, independent of stimulus type, is used.

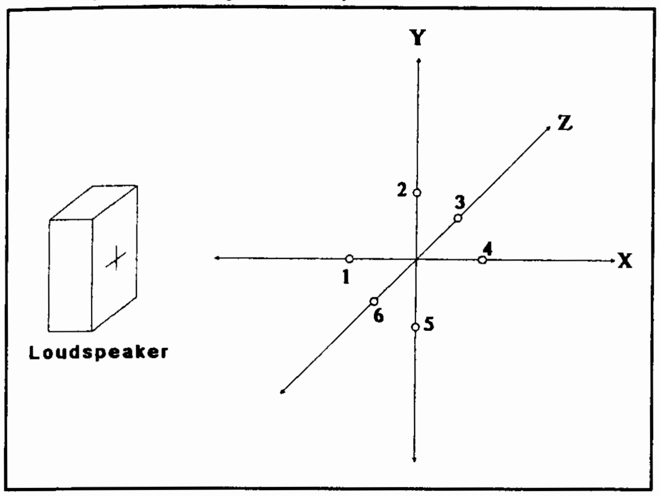
The subject test position should be selected so as to minimize the possibility of room reflections. The subject should be placed as far from the loudspeaker as possible while still remaining in the direct field, in order to minimize the effects of head movement (Walker, Dillon, & Byrne, 1984). Placing the subject near the center of the room is generally a good starting point for large sound booths. If possible the test position should be such that the subject is approximately 1 meter away from the loudspeakers and 1 meter away from the walls (Skinner, 1988). The presumed center of the subject's head (calibration point) should be marked in such a way that the subject can be easily and quickly positioned. Hanging a piece of string (with a small weight on the end) from the ceiling is an effective method for reliably determining the calibration point. The length of the string from the ceiling should mark the height to the center of the loudspeaker.

The loudspeaker(s) should be positioned in the room such that the plane of each loudspeaker front is perpendicular to the test azimuth. The center of the loudspeaker should be adjusted in height until it is level with the position of the average subject's ear when the subject is in the test position (i.e., seated). If the loudspeaker contains only one transducer (i.e., speaker cone), then the center of the speaker cone should be used as the center of the loudspeaker. If the loudspeaker contains several transducers, then the center of the loudspeaker enclosure should be used as the center of the loudspeaker.

Calibration

Once the sound field is assembled it is necessary to calibrate the test stimuli. Ambient noise levels in the sound booth should meet the ANSI standard (ANSI S3.1-1977) for monaural testing with the test ear uncovered. The sound field is then calibrated, at each frequency, by positioning the measurement microphone at the calibration point and adjusting the measured output (dB SPL) to the RETSPL value plus the output level (e.g., 70 or 80 dB HL) set on the audiometer. To

Figure 2. Schematic representation of the 6 locations around the calibration point which are recommended for measurement of sound field variability (modified from Walker, Dillon, & Byrne, 1984).



minimize artifacts (i.e., reflections) when using a sound level meter in the sound field, measurements should be made with no one present in the test booth. As well, the sound level meter microphone should be oriented at the proper angle for sound field measurement (determined by microphone type).

Sound Field Variability

Sound field variability around the calibration point should be determined for each frequency (Walker, Dillon, & Byrne, 1984; and Skinner, 1988). Sound field variability is determined with the measurement microphone placed a fixed distance on each of three axes (see Figure 2) around the calibration point (Walker, Dillon, & Byrne, 1984). If the sound field is to be used with adults, then a six inch perimeter is sufficient (Skinner, 1988). However, if the sound field is to be used with children, then it is probably better to use a 12 inch perimeter (Skinner, 1988). As a guideline sound field variability should be within 2 dB.

Evaluation of Sound Field

After the sound field is calibrated and sound field variability is determined to be acceptable, it is desirable to evaluate the validity of the RETSPL values used. This can be done by comparing the thresholds obtained in sound field for a group of subjects to their thresholds obtained by earphone. To reduce the potential interference of ambient noise on the sound field threshold measurement, subjects with a flat mild hearing loss should be used. Also, the subject's nontest ear should be occluded during sound field testing.

To evaluate the RETSPL values used for calibration in a particular sound field, the monaural threshold of each subject, at each frequency, is obtained for test stimuli transduced by earphone and by loudspeaker. It is preferable to use a smaller step size than the standard 5 dB used for audiometric threshold determination. As well, test sequence (i.e., between sound field and earphones) should be randomly varied between subjects. For each subject and each frequency the difference (d) between the earphone (TDH) and sound field (SF) threshold (dB HL) is determined by subtracting the SF value from the TDH value. The mean difference [mean(d)] and standard error of the difference [*SE*(d)] is then calculated for each frequency separately using the following formulae:

$$mean(d) = \frac{SUM(d)}{n}$$

$$SE(d) = \frac{SUM(d^2) - \frac{SUM(d)\,)^2}{n}}{n(n-1)}$$

where:
d = difference (TDH-SF)
n = number of ears tested
SUM is the summation of values within the brackets

The mean difference of each frequency can then be tested for statistical significance by the formula:

$$STAT = \frac{mean(d)}{SE(d)} \qquad t_n\text{–}1$$

The test statistic is compared to a critical value for t (with n-1 degrees of freedom). If twenty ears are tested and an alpha of 0.01 (i.e., one percent error rate) is selected, then t = 2.861. If STAT is larger than this, the obtained difference is statistically significant. For differences that are statistically significant and clinically relevant (5.0 dB), it is necessary to either adjust the RETSPL value or not perform sound field testing in that sound field. When the difference is not statistically significant but is considered to be clinically relevant (i.e., >5 dB), a larger group of subjects should be evaluated to increase the reliability of the sound field evaluation.

Summary

The selection of sound field stimuli bandwidth represents a trade- off between the desire to obtain frequency specific information and the need to reduce sound field variability. A signal of relatively wide bandwidth will provide a more uniform sound field than the same signal presented with a narrower bandwidth. When stimuli with a bandwidth wider than a pure tone are used, then the signal must have high out-of-band rejection. The suggested bandwidths for sound field stimuli vary by center frequency and are on the order of 29% at 250 Hz to 8% at 4000 Hz, for standard audiometric testing (Walker, Dillon, & Byrne, 1984). Frequency modulated tones should be either triangularly or sinusoidally modulated and should have a relatively high (i.e., 20 Hz) modulation rate (Walker, Dillon, & Byrne, 1984). Narrow band noise should have a sufficiently steep (i.e., greater than 60 dB) filter slope, otherwise the subject may be responding to the out-of-band portion of the signal. It is recommended that the sound field test environment be calibrated using procedures described by Walker, Dillon, & Byrne (1984), and Skinner (1988). Clinical audiometers generally provide stimuli of one constant percentage bandwidth. Furthermore, the 1/3 octave narrow band noise is typically intended for use as a masking stimulus and may not have a sufficiently steep filter slope for use in frequency specific threshold determination. In order to test with stimuli that approximate the bandwidths suggested by Walker, Dillon, and Byrne (1984) for standard audiometric testing, it is suggested that 1/3 octave narrow band noise be used when testing with low frequency stimuli (i.e., 250-1000 Hz), and that frequency modulated tones be used when testing with high frequency stimuli (i.e., 1500-6000 Hz). It is recommended that the RETSPL values used for sound field audiometry by the clinic be evaluated for equivalence to the ANSI (S3.6-1969) standard using the procedure outlined.

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As of April 14, 1990 the phone number of the *JSLPA/ROA* office will change from 424-1268 to 494-1268.