5 AUDIO HELMET-MOUNTED DISPLAYS

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A helmet is a device covering the head and intended to protect the user from hazards to the head. Helmet-mounted display (HMD) systems are generally described as display devices worn on the head as a part of the helmet assembly to provide video information directly in front of the eyes. Sometimes these devices are referred to as head-mounted display systems or head/helmet-mounted display systems (Rash, 2006) since they can be worn on the head both with and without a helmet. In addition, the above display systems are not limited to visual projections and may also include projections of auditory and, potentially, tactile (haptic) signals. Various display systems may be used together providing multimodal information enhancing the user's situation awareness. For example, it has been extensively reported that combined audio/video displays provide a significant increase in visual target detection performance and greatly reduce visual search time (Bergault, Wenzel and Lathrop, 1997; Bolia, D'Angelo and McKinley, 1999; Nelson et al, 1998; Pinedo, Yound and Esken, 2006).

Signals are variable quantities by which information is transmitted from a source to a receiver (Isaacs, 1996). Auditory signals are the acoustic and vibratory signals that create an auditory image of the external world; whereas tactile signals are mechanical pressure signals that are perceived as pressure on the skin (see Chapter 18, *Exploring the Tactile Modality HMDs*).

Auditory signals can arrive to a listener from natural sound sources surrounding the listener or from electroacoustic transducers converting recorded or synthesized electric signals into acoustic waves. The electric signals that are being converted to acoustic signals are called audio signals. The audio signals can be generally defined as audible acoustic signals recorded or generated in an electric form and emitted to the environment as acoustic signals by electroacoustic transducers (audio sources). The audio signals can be as simple as a beep or a spoken word or they can create complex immersive environments such as auditory virtual reality. The system of audio sources projecting auditory signals is called audio display system or, in short, audio display. In other words, an audio display system is a system converting audio signals into acoustic or mechanical (vibration) signals that elicit auditory sensations. Complex auditory sensations result in a perceptual representation of acoustic environment acting on the auditory system of the listener. This perceptual representation is called the auditory image of the environment. Acoustic or vibratory signals radiated by an audio display system can create their own acoustic environment or they can augment an existing auditory image of the natural environment. Numerous applications of audio display technology extend from radio communication, auditory navigation, hearing enhancement (e.g., hearing aids), and music, to fully immersive auditory virtual reality used for training and entertainment purposes. There are also some medical devices (e.g., cochlear implants) that can directly stimulate the auditory nerve and create auditory images but they are not considered in this book.

Audio display systems need to be differentiated from its product, auditory display, in the same way as video display systems need to be differentiated from its product, visual display. Visual displays can be produced by video display systems or can be a result of a specific visible behavior or arrangement of visible objects in the natural environment. In the same way an auditory display can be produced by an audio display system or it can be an arrangement of natural sounds entering the ear (Letowski et al., 2001). In other words, an auditory display is the sum of acoustic signals generating a perception of a particular acoustic environment. The auditory display may consist of a variety of intentional or unintentional sounds, such as speech communications; natural and synthetic sound effects; music, combat-related sounds and urban and vehicle sounds, as well as ambient noise.

The ambient noise is a sum of all unwanted continuous and repetitive sounds that blend together and create an all-encompassing acoustic background at a given location (ANSI, 1994).

The audio HMD system is an audio interface worn as a part of the helmet assembly and providing auditory stimulation. Although the name implicates that the system is helmet-mounted, in reality it is more often a head-mounted system (also abbreviated HMD system), which typically rests on the head of the user even if the system is fully integrated with the helmet. In addition, many military and civilian operations require uninterrupted access to a head-worn audio HMD system even when no helmet is worn. Therefore, in a large number of cases the audio HMD system is not integrated with the helmet and can be worn with or without the helmet as a part of the modular headgear.

A fully-featured audio HMD system must fulfill three major functions providing (1) audio display for radio communication and for other audio-supported functions, (2) hearing protection against harmful high intensity sounds, and (3) means for preserving an effective auditory awareness of the environment. In addition, an audio HMD system is usually equipped with a head-worn or boom microphone as well as a head tracker (Rash, 2006), which need to be incorporated in the design. Thus, the design of an audio HMD system needs to consider appropriate input and output transducers, wiring, connectors, switching systems, signal processing devices, electric impedance matching, padding, isolation issues, and low-power interfaces to the equipment processing the stimuli. Note: An audio head-mounted display (HMD) system is an audio communication system worn on the head or mounted in the helmet of the user. The system may or may not be equipped with a speech communication microphone.

The above requirements for audio HMD systems primarily address the needs of the dismounted Warfighter operating in constantly changing conditions. Mounted Warfighters and aviators may not need as much auditory situation awareness as dismounted Warfighters and their operational (encapsulating) helmets provide some degree of hearing protection. Therefore, the specific focus on individual features of audio HMD systems should depend on the military platform (e.g., dismounted operations, tank, or helicopter) and operational environment in which the system is intended to be used. It is however, important to stress that this system always needs to provide adequate hearing protection. Note that hearing loss is the most common disability of military personnel and the number of hearing loss cases rises rapidly during military conflicts. In addition, under emergency conditions auditory situation awareness may become equally important to all users regardless of the platform and always needs to be taken into account in audio HMD system design. It has to be recognized that all audio HMD systems that protect (cover) the ears introduce some degree of uncertainty in localizing outside sound sources and are detrimental to natural speech communication. Thus, determining the optimal balance between the three main functionalities of the HMD systems discussed above is the most challenging task facing design and selection of HMD systems for specific operations.

This chapter defines the acoustic environment and the concept of an auditory signal together with descriptions of various modes and techniques of audio signal delivery. The background information is followed by a discussion of technical and operational factors affecting design and selection of audio HMD systems including hearing protection and auditory awareness issues. This discussion is combined with an analysis of system requirements for military applications. Advantages, disadvantages, and salient characteristics of each of the audio display design options are discussed to help the reader understand the trade-offs involved in creating or selecting a functional, effective, and reliable audio HMD system that serves its intended purpose and works in concert with the visual HMD and other headgear. The anatomy and physiology of the hearing organ and psychoacoustic of sound perception are not discussed in this chapter since they are addressed in-depth in Chapters 8 (*Basic Anatomy and Structures of the Human Ear*), 9 (*Auditory Function*) and 11 (*Auditory Perception and Cognitive Performance*).

Acoustic Environment

The auditory system is the sensory system responding to a mechanical disturbance of the elastic medium that propagates though the medium as a longitudinal wave. This wave is called an acoustic wave and is perceived as a sound. The term sound is also used in the literature to describe an auditory sensation created by an acoustic wave or mechanical vibration. Therefore, the term sound has dual formal definitions and refers to both the acoustic wave and the auditory sensation (ANSI, 1994).

Opposition of the medium to wave (sound) propagation is called acoustic impedance. Acoustic impedance relates two most fundamental properties of the acoustic wave: acoustic pressure and particle velocity. This relation can be written as:

$$p = Z \times v, \qquad \text{Equation 5-1}$$

where p, Z, and v indicate acoustic pressure, acoustic impedance, and particle velocity, respectively. Acoustic pressure is a change in the atmospheric pressure due to a mechanical disturbance of the medium. Particle velocity is the velocity of the oscillatory movement of a particle caused by wave propagation. The product of acoustic (sound) pressure and particle velocity is called sound intensity (I) and it defines the acoustic power of a vibrating particle. Since acoustic pressure and particle velocity are related according to Equation 5-1, sound intensity is proportional to the square of acoustic pressure.

Sound intensity of everyday sounds varies over several magnitudes and therefore it is customary to express its values on the logarithmic scale called sound intensity level. Sound intensity level is defined as:

$$i = 10\log_{10}(\frac{I}{I_0}), \qquad \text{Equation 5-2}$$

where i and I mean sound intensity level and sound intensity respectively. A unit of a sound intensity level is the decibel (dB) (ANSI, 1995b). A 10-times change in sound intensity results in an increase of sound intensity level by 10 dB. The value I_0 is the reference point in relation to which sound intensity level is calculated. In many applications of acoustics and audio the value of I_0 is standardized and equal 10^{-16} Watts/centimeter^2 or 10^{-12} Watts/meter^2. This value is also called the zero level and corresponds roughly to the threshold of human hearing at 1000 Hz. Since sound intensity is proportional to the square of acoustic pressure, the Equation 5-2 can be also written as

$$i = 20\log_{10}(\frac{p}{p_0}), \qquad \text{Equation 5-3}$$

where p is actual acoustic pressure and p_0 is the reference acoustic pressure. The standardized value of the reference acoustic pressure p_0 corresponding to $I_0 = 10^{-12}$ W/m^2 is equal to $2\text{x}10^{-5}$ Pa. When the sound level is calculated in reference to $p_0 = 2\text{x}10^{-5}$ Pa, this level is called the sound pressure level (SPL) and is written as dB SPL (ANSI, 1995b).

Sounds can physically differ in a number of parameters including sound intensity, spectrum, and sound duration. One common classification of sounds based on their duration divides them into continuous (steady-state) sounds and impulse sounds. Continuous sounds are stationary or slightly varying sounds that are longer than a period of observation. Examples of such sounds are sounds of power generators, moving vehicles, and waterfalls. Relatively uniform traffic noise and cafeteria noise can be also considered continuous sounds. Impulse sounds are short sounds that have rapid onset and decay. Such sounds include explosions, weapon fire, and door slams.

Obviously, these two classes of sounds are just the extreme points of a physical continuum encompassing all the sounds, which normally include both stationary and impulse components.

Many sound sources, especially low-frequency sound sources, radiate sound in all directions. Such sources are called omnidirectional sound sources. The sources that radiate most of their energy in one or few distinct directions are called directional sound sources. Examples of such sources are unidirectional and dipole sound sources having beam-like and figure-of-eight radiation patterns, respectively.

Acoustic waves propagating though a medium are absorbed, reflected, dispersed (diffused), and diffracted by space boundaries and various objects located within the medium (space). The distribution of sound energy emitted by sound sources located in the space and modified by boundary effects of the space is called the sound field. A sound field observed at a specific point in the space is called an acoustic image of the field. The acoustic image acting on the listener's ears is the auditory display. The properties of the sound field greatly depend on the amount and distribution of reflected energy and its rate of decay after termination of sound source activity. This rate is called reverberation and is usually expressed as reverberation time (RT) defined as the time needed for the sound energy to decrease by 60 dB.

Sound pressure measurements are usually reported as the sound pressure levels in dB SPL. However, a specific sound pressure level does not necessarily mean that the sound is loud or even perceived by human hearing. Acoustic waves of very low and very high frequencies that fall outside of the range of the human hearing do not contribute to the loudness of the sound. Therefore, if someone wants to assess perceptual effects of sound, the measurement needs to take into account the properties of the human ear (see Chapter 8, *Basic Anatomy and Structure of the Human Ear*). There are several weighting curves that when applied to dB SPL data provide information about potential auditory effects of the specific sound. Most commonly used weighting curves are A-, B-, and C-weighting. These curves are mirror images of average frequency-dependent equal-loudness curves of human hearing in the 0-40, 40-70, and 70-120 phon range, respectively. They mainly represent the way in which the frequencies below 1000 Hz are filtered by the ear at different SPLs. The SPL data processed with these weightings are written as dB (A), dB (B), and dB (C) (ANSI, 1995b).

Auditory Signals and Display Formats

The process of perceiving sound is called hearing or audition (see Chapter 9, *Auditory Function*). The sensation of sound can be created by acoustic waves arriving at the ears of the listener (air conduction) or by direct vibration applied to the head (bone conduction). The auditory system acquires, interprets, selects, and organizes simple and complex auditory stimuli and creates an auditory image of the physical environment surrounding the listener. The field of science devoted to the human perception of sound is called psychoacoustics. In order to understand how humans perceive sounds one must know what the human hears, and which portions of the perceived sounds are considered to be useful information (signal) and which portions are considered to be distracting background (noise). While the sound pressure levels presented to the human ear may be precisely measured, it is difficult to determine exactly an auditory effect of the stimulation. These auditory effects may depend on a person's expectations, attention, health, and multi-faceted environmental conditions. They also depend on the relative importance assigned to the specific sounds by the listener. For example, Warfighters rely heavily on auditory information carried by environmental sounds when they are on patrol or on search missions and on sound signatures of weapons, helicopters, and vehicles when they are in a combat situation. The importance of auditory information increases many-fold when visual information is obscured by smoke, fog, or darkness.

Auditory signals can be generally defined as an acoustic or vibratory stimulus received by the hearing system and converted into auditory information. Both intentional sound messages and unintentional sounds can be signals. If specific auditory information is not considered useful and degrades perception of auditory signals it becomes an interfering noise. Auditory noise may have internal (physiological) and external (acoustic) origins. The effect of noise on the perceived signal is usually quantified as a signal-to-noise ratio (SNR). The SNR is the ratio of some measured aspect of a signal to a similar measure of a concurrent noise expressed in a logarithmic form (Letowski et al., 2001).

Auditory signals can be projected by distal and proximal display systems. The distal auditory display systems are those where the actual sound sources are located away from the listener's ears. Examples of distal audio display systems are all real-world environments and various loudspeaker-based sound projection systems. Proximal audio display systems are display systems located close to the listener's ears. All audio HMD systems are proximal audio display systems since they are mounted to the listener's head at or close to the listener's ears. A small loudspeaker mounted on the shoulder strap is usually sufficiently far away from the listener's ears to consider it a distal display.

When a listener is placed in an acoustic environment that contains several sound sources surrounding the listener, all sounds arrive at the both ears of the listener regardless of the location of the sources. This situation is shown in Figure 5-1. The sounds may arrive at different times and with different intensities and they may arrive directly from the sound sources (Figure 5-1) or after being reflected from surfaces in the surrounding environment. Regardless of the specific pathways, the sounds from each sound source will arrive at both ears of the listener.

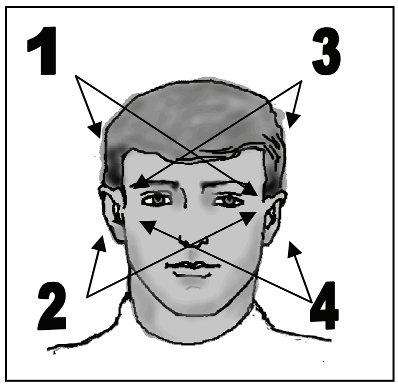


Figure 5-1. Auditory display created by distal sound sources (1 through 4).

Auditory signals may be presented to the human in various forms and by various techniques. The main classification of the auditory displays (Letowski et al. 2001) is shown in Table 5-1.

Table 5-1.
Classification of auditory displays created by various types of input signals and signal projection systems (natural sound sources and/or audio display systems).

Input Signal	Signal Projection	
	One Ear (Monaural Listening)	Two Ears (Binaural Listening)
One Channel (Monophonic Signal)	Monotic	Diotic (Biaural)
Many Channels (Multi-channel Signal) (Stereophonic Signal)	Monotic (Monomic)	Dichotic (Spatial)

A monophonic signal is a single channel signal delivered to one or many transducers of the audio display system. Multi-channel (or stereophonic) signals are a group of uncorrelated (or correlated) signals delivered to individual transducers of the audio display system. Regardless of the type of the signal, the audio system can create an audio display that can be projected to one (monaural listening) or both (binaural listening) ears of the listener.

A monotic auditory display is created by auditory signals delivered only to one of the listener's ears. This type of display is also frequently described as a monaural display. In the case when we want to stress that several signals are combined together and delivered to a single ear, the monotic display can be called a monomic display. Figure 5-2 shows monotic or monaural sound presentation to the listener's left ear.

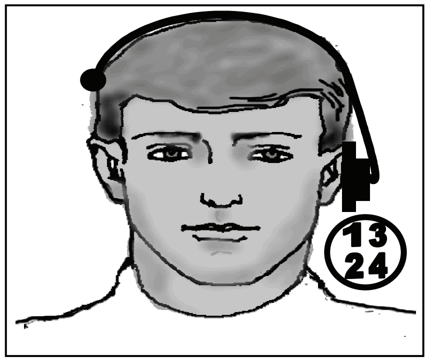


Figure 5-2. Monotic or monaural auditory display. All (1, 2, 3, and 4) sound sources are presented to a single ear of the listener.

When the same signal is presented to both ears of the listener the auditory display is called diotic or biaural. The biaural (diotic) display causes the image of the sound sources to appear within the listener's head instead of being located to the side of the head as in the case of monotic displays. Biaural listening improves speech intelligibility, especially in a noisy environment due to the increased perceived loudness provided by the biaural presentation and better spatial separation of the phantom signal source from extraneous noise sources. Figure 5-3 shows the concept of the biaural presentation method.

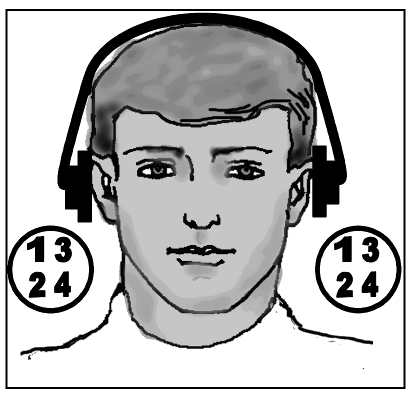


Figure 5-3. Diotic (biaural) auditory display. The same four sound sources are presented to each ear of the listener.

When different signals are delivered to each ear of the listener such an auditory display is called a dichotic or spatial display. Biaural (diotic) and dichotic (spatial) presentations are two forms of two-ear listening called binaural listening. The dichotic display is produced when independent (uncorrelated) auditory stimuli are delivered to the listener's right and left ear. This type of display takes place when the listener monitors two or more radio networks at the same time. Figure 5-4 shows the situation when uncorrelated stimuli are presented to each ear. Stimuli 1 to 4 are heard in one ear and stimuli a to d are heard in the other ear of the listener. When the signals delivered to the left and right ear of the listener are different but correlated they create a spatial auditory display in which phantom sound sources are distributed in space.

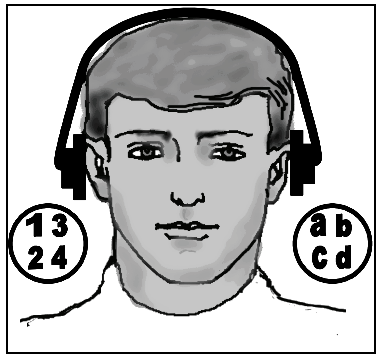


Figure 5-4. Dichotic auditory display. Different sets of sound sources are presented to each ear of the listener.

The display formats listed in Table 5-1 are the basic formats of audio displays. However, various combinations of the basic formats are possible through signal processing and audio switching techniques. For instance, two or more channels may be presented to one or both ears, and when a high priority signal requiring immediate attention arrives, it may be directed to one ear at a higher intensity level than the less important signals. With the proper control of the incoming signals the listener may cause the sound intensity of the selected channel (right or left) to be presented at a higher level than the other channel.

Auditory signals delivered though the audio HMD systems may provide different degrees of spatial information (i.e.; information to permit the listener to determine where the sound source is located in space) to the listener. Monophonic signals and uncorrelated multi-channel signals are localized as originating either in the ear or from within the listener's head. Such phantom location of the sound source is called internalization. If the monophonic signal delivered to the left and right ear has the same intensity the phantom sound source is located in the center of the listener's head. By changing the intensity ratio between the signals arriving at the left and right ears the phantom location of the sound source can be moved anywhere on the arc connecting left and right ears. Such displacement of the phantom sound source from its central position in the head is called lateralization. Figure 5-5 shows two examples of biaural auditory displays with the phantom sound source lateralized to one ear (Figure 5-5a) and internalized in the center of the head (Figure 5-5b).

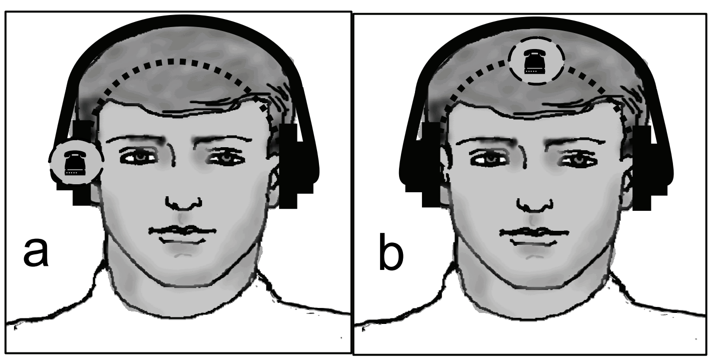


Figure 5-5. Biaural display with the phantom sound source completely lateralized to the right (a) and located in the center of the head (b). The dashed line shows an arc along which the phantom image lateralizes.

When the auditory signals that are delivered to both ears of the listener are stereophonic (correlated) signals such an audio display is frequently called a spatial, binaural, or stereophonic display. The phantom sound sources created by the stereophonic signals are internalized within the listener's head similarly to the diotic signals. However, specific location of the phantom sound source within the head is not only determined by the difference in the intensity of sounds delivered to the left and right ear but also by interaural cross correlation (ICC) associated with signals received by the two ears (White, 1987). The values of ICC can vary from -1 to +1, where -1 indicates identical signals are presented to both ears, but they are 180° out of phase; 0 indicates the two signals are unrelated to each other (dichotic) ; and +1 indicates the two signals are equal and in phase (diotic). Examples of phantom sound source locations dependent on the ICC values are shown in Figure 5-6. Note that when left and right signals are 180° out of phase the location of the sound source in the head is smeared and less defined (Figure 5-6a).

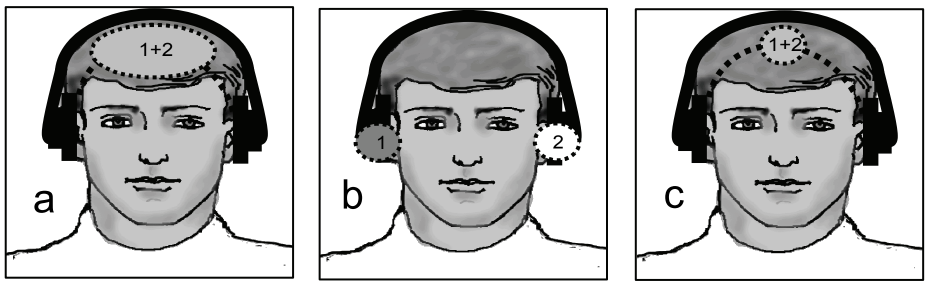


Figure 5-6. Phantom sound source locations created by different amounts of interaural cross correlation (ICC) between signals 1 and 2 delivered to the right and left ear: (a) ICC= -1, (b) ICC= 0, and (c) ICC= 1.

Head-Related Transfer Function (HRTF)

The human auditory system determines the location of the distal sound source in space by making use of binaural and monaural localization cues. In addition, familiarization with sound sources and head movements can enhance a person's ability to localize the direction of incoming sound.

Binaural cues are the differences in the intensity and the time of arrival of the sound wave from a particular sound source arriving at the left and right ear of the listener. These cues are called the interaural intensity difference (IID) and the interaural time difference (ITD). Binaural cues operate in the horizontal plane and allow the listener to determine the azimuth of the incoming acoustic signal. The amounts of IID and ITD are dependent on the size of the listener's head and for the signal source located on the side of the listener's head they can be as large as 15-20 dB and 0.6-0.8 ms, respectively. The amount of IID is frequency dependent whereas the amount of ITD is not. However, the ITD can be converted into an equivalent interaural phase difference (IPD), which is frequency dependent. Humans appear to use the IPD cues to localize sources of low frequency sounds and the IID cues to localize sources of high frequency sounds.

The monaural cues are related to the geometry of the listener's head and the shape and location of the pinna. These cues are used both to determine the azimuth and elevation of the sound source and to differentiate between the front and back directions of the incoming signals. Different monaural cues operate in different frequency ranges. For example, due to the small dimensions of pinnae as compared to the wavelengths of acoustic energy heard by humans, the pinna cues are most effective for localization of high frequency sound sources at and above 3000 Hz (Musicant and Butler, 1984).

The shape and intricacies of the human head and the upper torso generating monaural localization cues constitute a complex directional acoustic filter (equalizer) that relates the locations of the sound sources in the space to the specific characteristics of the signals arriving at the ears of the listener. This filter is called the head-related transfer function (HRTF). An HRTF is the ratio of the sound pressure at the ear of the listener to the sound pressure that would exist at this point if the listener was not present expressed as a function of frequency. Each location of the sound source in space is coded into a pair (left and right) of HRTFs that allow the listener to identify this location. The shape of the HRTF does not change with the distance from the sound source to the listener except for very short distances of less than 1 meter when the proximity effects and the sound bouncing between the head and the sound source need to be taken into consideration. A small set of HRTFs obtained for a group of listeners is shown in Figure 5-7. Please note that the shape of HRTF differs quite substantially among people and our listening experience is affected by the peaks and valleys of our individual HRTFs.

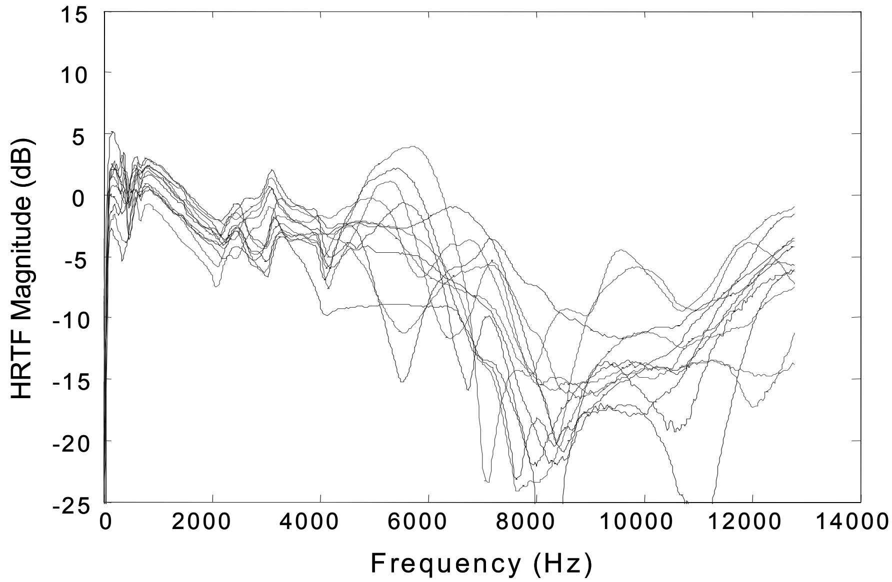


Figure 5-7. A set of HRTFs (magnitude) obtained for a small population of listeners at one specific position of a sound source (Vaudrey and Sachindar, 2003, used with permission).

The natural human ability to localize sound sources in space is lost when the auditory signals are presented directly to the human head by the proximal display systems, such as earphones. In order to restore an impression that the sound sources are located in space outside of the human head (externalized), the specific HRTFs of the listener need to be captured and incorporated into the signal delivered to the display system. A pair of HRTFs for the left and right ear represents the anatomical capabilities of a specific human that are used in identifying the specific direction toward a sound source. A set of HRTFs for various angles of incidence captures all binaural and monaural localization cues characterizing a specific individual and can be used to synthesize spatial perception in a virtual environment created by audio HMD system. Therefore, a monophonic sound recording convolved with a matched pair of HRTFs and played through earphones will result in the impression the sound source is located in space and outside of the head of the listener. In other words, when the recording of the natural distal environment is convoluted with a pair of matched HRTFs and played through earphones (proximal displays), the listener experiences externalized auditory images (Hartman and Wittenberg, 1996). Such audio displays are called 3D-audio displays or spatial audio displays. Figure 5-8 presents the differences between the display of real-world stimuli, the biaural display, and the spatial biaural display created with HRTFs.

In order for the listener to have natural externalization of the sound sources, the auditory signals must be convoluted with the listener's own HRTFs. Such HRTFs are called individualized HRTFs. Attempts to create average (non-individualized) HRTFs that can be effectively used for many listeners were unsuccessful and resulted in poor accuracy of the sound field recreation (Wenzel, Arruda, Kistler and Wightman, 1993).

The HRTFs can be recorded in a number of ways, but the most common method is to place miniature microphones in the openings of a listener's ear canals and to make multiple recordings of a standard test signal presented from various azimuths and elevations around the listener (Wightman and Kistler, 1989ab). The test signal can be a frequency-swept sine wave or a standard impulse signal such as maximum-length sequence (MLS) or a Golay code (Zahorik, 2000). Figure 5-9 shows a typical HRFT measurement system (left) and the microphone placement in the listener's ear (right). In order to determine the HRTFs for specific listeners and directions, the recordings of the standard signal are made in the manner shown in Figure 5-9 and compared to the similar recordings made with a single microphone located at the point corresponding to the center of the listener's head. The differences between these recordings for specific sound source locations constitute the set of HRTFs for a given listener. Such HRTFs can be applied to any acoustic signal through a process called convolution and delivered to the Audio HMD systems to create an impression of auditory stimuli arriving from the surrounding space.

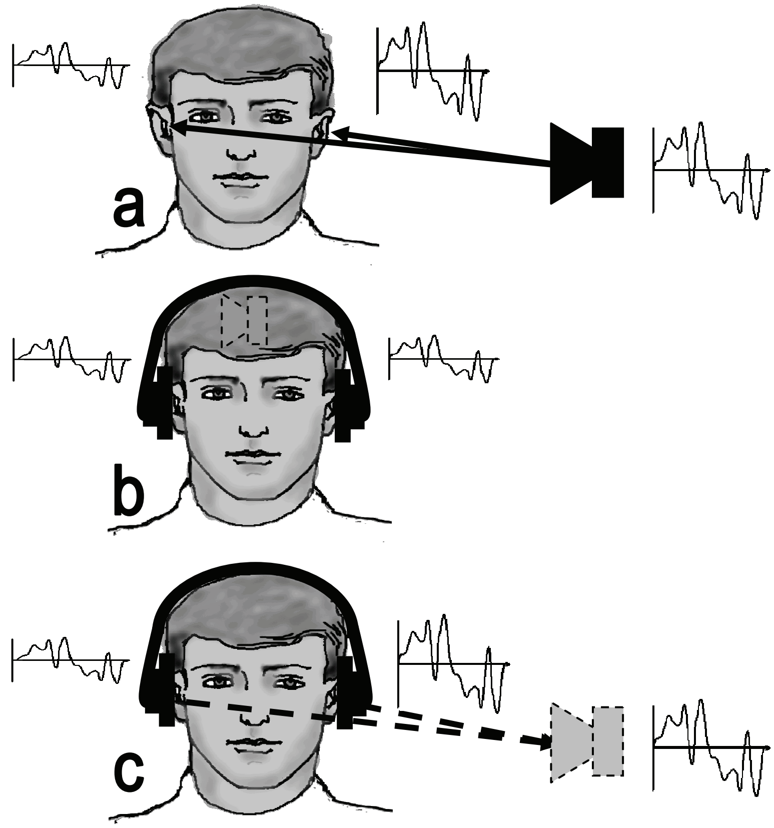


Figure 5-8. Natural hearing and spatial audio display created by using HRTFs. Part a shows spatial position of the distant sound source perceived by natural hearing. Part b shows a biaural display that creates the image of the sound source inside the head. Part c shows spatial position of the sound source faithfully recreated when the earphone-reproduced signals are filtered with an appropriate head related transfer function (HRTF). Signal waveforms shown in individual panels represent the signals emitted by the real loudspeaker (panel a), associated with phantom loudspeaker (panel c), and the signals heard by the listener (symbols next to the head in all three panels).

Figure 5-9. Measurement of HRTFs using the KEMAR manikin and a loudspeaker mounted on a robotic arm (left) and the HRTF microphone placement in the listener's ear (right). (U.S. Army Research Laboratory (left) and Vaudrey and Sachindar, 2003 (right) (used with permission).

An alternate method of recording HRTFs involves placing the sound source in the ear canal and placing an array of microphones around the listener (Zotkin et al., 2006). Such a recording configuration is shown in Figure 5-10. This method is based on the reciprocity principle with microphone and loudspeakers reversing their positions in space in comparison to the previous method. The advantage of the reversed setup is much shorter measurement time. However, such a system requires many identical calibrated microphones and is both expensive and difficult to maintain.

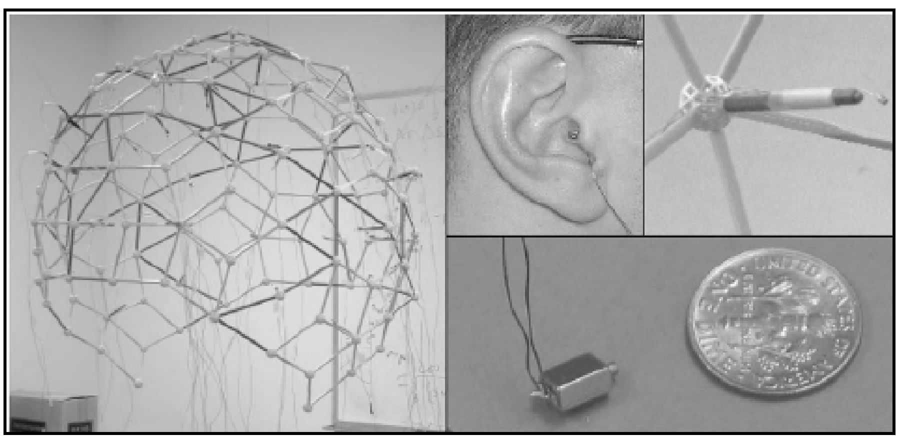


Figure 5-10. Left: The measurement mesh consisting of 32 microphones. Bottom right: The miniature loudspeaker. Top middle: The miniature loudspeaker inserted into the listener's ear. Top right: An enlargement of the one node of the measurement mesh (adapted from Zotkin et al., 2006).

Despite the natural differences in specific techniques used to record HRTFs in various laboratories, it seems that the properly measured HRTF sets obtained in various laboratories on the same people are operationally equivalent. A study conducted at the U.S. Army Research Laboratory at Aberdeen Proving Ground (MacDonald and Tran, 2008) compared the effects of four sets of KEMAR HRTFs on localization accuracy of the listeners in virtual auditory environments. The four sets were: (1) in-house KEMAR HRTFs measured with MLS signal and Tucker Davis Technology hardware (RoboArm), (2) HeadZap HRTFs from AuSim Inc., (3) Massachusetts Institute of technology (MIT) HRTFs from Media Lab at MIT, and (4) CIPIC HRTFs from CIPIC (Center for

Image Processing and Integrated Computing) Interface Lab at the University of California at Davis. These four sets were compared by a group of 16 listeners in a localization task using spatial stimuli presented over headphones. The task was limited to the horizontal plane and eight discrete phantom sound source locations. The mean absolute localization errors observed at each of the eight phantom source locations for the four sets of HRTFs are shown in Figure 5-11.

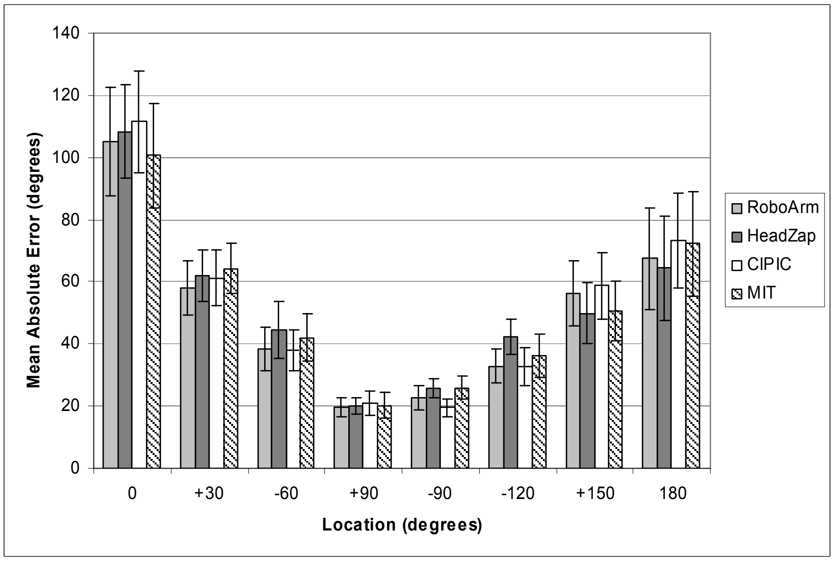


Figure 5-11. Result of the HRTF comparison study. Mean absolute errors for eight phantom sound source locations obtained with four HRTF sets (U.S. Army Research Laboratory).

The data in Figure 5-11 show that all four HRTF datasets resulted in functionally identical spatial audio simulations despite the differences in recording technologies, locations, and recording personnel for each of the datasets. Considered in terms of the localization performance of the listeners using the generalized (non-individualized) HRTFs, the four sets were nearly indistinguishable from one another.

Audio HMD System Specifications

The technical specifications of an audio HMD system depend on the complexity and multi-purpose character of the system and can substantially differ from one system to another. However, the minimum specifications of such a system should include the type of device, available operational modes, audio performance data, hearing protection data, weight, and the type of electric interface to the supporting platform. It is also assumed that each of these devices meets standard military requirements for ruggedness and required range of operational conditions. These requirements are specified in MIL-STD 810F *Department of Defense Test Method Standard for Environmental Engineering Considerations and Laboratory Tests* (DOD, 2001). All these parameters are important for proper operation of the system but the core requirements that characterize the quality of the audio interface and affect its potential range of applications are the audio performance data. These data should represent a number of specific electroacoustic characteristic of the system on the basis of which two or more systems can be compared. The list of basic electroacoustic characteristics of audio HMD systems is provided in Table 5-2. More information about specific transducers and other elements of the audio HMD systems is provided in the later sections of this chapter.

Table 5-2.
Basic operational characteristics of audio HMD systems.

Operational Characteristic	Definition
Sensitivity	System sensitivity is the effectiveness of the system (audio transducer) to convert input signal into the output signal. Three basic audio sensitivities are earphone sensitivity, loudspeaker sensitivity, and microphone sensitivity. Earphone sensitivity – or earphone efficiency - is the sound pressure level in dB SPL produced by an earphone in response to a 1 mW signal in a standardized coupler with a built-in microphone (IEC 60268-7). Unit: dB SPL/mW (dB SPL per milliwatt). Loudspeaker sensitivity – or loudspeaker efficiency - is the sound pressure level in dB SPL produced by a loudspeaker at 1 m distance in response to a 1 W signal (IEC 60268-5). Unit: dB SPL/W (dB SPL per watt). Microphone sensitivity is the voltage output in mV produced by a microphone in response to a 94 dB SPL signal (1 Pa) (IEC 60268-4). Unit: mV/Pa (milivolts per Pascal).
Frequency Bandwidth	Frequency bandwidth is the frequency range within which the system sensitivity does not change by more than a specific number of dB, usually 3 dB, from its nominal sensitivity level. Unit: Hz (Hertz). In digital systems the bandwidth is defined as the data transfer rate and measured in bits per second. Unit: kb / s (kilobits per second).
Nonlinear Distortions	Nonlinear distortions are new frequency components in the output signal that do not exist in the input signal but result from the presence of the input signal. Nonlinear distortions are typically measured as a percent of the total signal. The most common type of nonlinear distortions is harmonic distortions and intermodulation distortions. Unit: % (percent).
Maximum Input Power	Maximum input power (rated power) is the highest continuous power that the device can handle without producing excessive sound distortion or being damaged. Maximum input power is usually defined by a specific percent of nonlinear distortions that the system cannot exceed under normal operational conditions. Unit: mW (milliwatts) or W (watts).

Table 5-2. (Continued)
Basic operational characteristics of audio HMD systems.

Dynamic Range	The ratio of the highest undistorted input signal to the smallest input signal that produces output signal that is discernible from system noise. Unit: dB (decibels).
Headroom	Headroom is the level difference between the typical operating level of the device and the maximum operating level defined by the maximum input power or the onset of signal clipping. Unit: dB (decibels).
Electric Impedance	Electric impedance is the opposition of a system to the flow of the electric current. Unit: Ω (ohms). Electric impedance can be either the input impedance (earphone, loudspeaker) or the output impedance (microphone) of a system. Effective transmission of power requires that input impedance of the next system matches the output impedance of the previous system.

Audio HMD Transmitter Systems

An audio transducer is a device that converts electric energy into acoustic energy or acoustic energy into electric energy. The former type of transducer is called a transmitter and the latter is called a receiver. Common examples of audio transmitters and audio receivers are the loudspeaker and the microphone, respectively. Audio transmitters are the main elements of audio HMD systems converting audio signals into auditory stimuli. In addition, the system may be equipped with an audio receiver (boom microphone) for converting user's speech into audio signals that can be transmitted through radio communication equipment. The topic of audio receivers is briefly addressed in several places in this chapter but is beyond the scope of the chapter.

Common sound delivery methods used in audio HMD systems can be divided into two basic classes based on the sound transmission interface: air conduction transducers (earphones) and bone conduction transducers (bone vibrators). Each of these has their own distinct advantages and limitations and the selection of one or another needs to be dictated by specific operational requirements.

Earphone systems

The terms earphones, headphones and headsets are frequently used interchangeably but to the audio community each of these terms defines a unique system of technologies. The relation between these systems is shown in Figure 5-12.

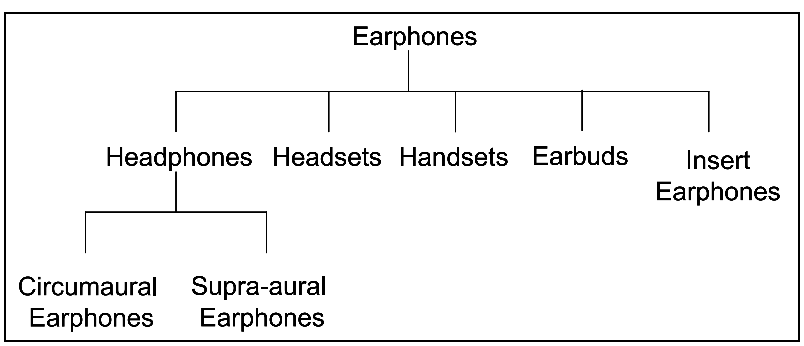


Figure 5-12. Types of earphones.

Earphones are all audio transducers that are directly coupled to the ear of the listener in such a manner that the reproduced sound waves are delivered to the eardrum through the air in the ear canal. In some professional literature (e.g., telephony, audiology) earphones are sometimes referred to as earpieces. Headphones are the earphones applied outside of the ear and supported by a headband placed over the head whereas insert earphones are the earphones inserted directly into the ear canal. The earphones mounted in the concha (earbuds) are not insert earphones and are considered a separate class of the earphones. Headsets are the headphones equipped with an additional microphone for speech communication and handsets are earphone-microphone combinations applied to the head by hand such as the telephone handsets.

The headphones are generally divided into circumaural and supra-aural headphones (details below). They are usually referred to as the circumaural and supra-aural headphones when considered as a system together with the supporting headband. Otherwise, they may be simply referred to as circumaural and supra-aural earphones.

Circumaural earphones are large earphones with earcups (earmuffs) that surround the outer ear resting against the head with little or no contact with the pinna. The audio transducer is loosely coupled to the ear with a relatively large volume of air under the earcup (ANSI, 1995). An important characteristic of circumaural earphones is that their earcup encloses and extends the cavity of the ear canal lowering natural resonances of the canal and adding new resonances of the earcup volume.

The circumaural earphones can be either closed-air (closed- back) or open-air (open-back) earphones. The closed-air earphones have earcups that completely separate the output from both sides of the earphone membrane from the external sound environment. This type of earcup attenuates external noise and minimizes sound leakage from the earphone. The noise attenuation is typically 10 dB to 15 dB at mid-frequencies. The closed back of the earphone results in extended and more resonant (boomy) low-frequency response of the earphones due to more pronounced resonances of the earcup space and earphone enclosures. Open-air earphones have an open grille at the back and in some cases the openings on the side of the ear cup. Modern open-back earphones use a semi-open design in which the open grille at the back of the earcup is replaced by several small, well defined openings that are used to tune the frequency response of the earphone. These earphones are often referred to as the semi-open earphones. An example of such an earphone is the AKG K240 headphones. The open-air and semi-open earphones do not isolate the listener from the external environment as much as the closed-air earphones and they leak the sound to the environment. However, open back design reduces the effects of earcup and earphone resonances making the frequency response smoother and reproduced sounds more natural. Examples of closed-air and open-air circumaural headphones are shown in Figure 5-13.

Figure 5-13. Examples of circumaural earphones: Sennheiser Model HD201 (closed-air [left]) and Sennheiser Model HD 595 (open-air [right]) (Courtesy of Sennheiser USA).

In general, circumaural headphones produce high sound quality and provide some hearing protection against external noise. They are used for studio listening and when some degree of isolation from the environment is required. However, they are bulky and expensive when compared to other styles and may become uncomfortable after prolonged use due to the lack of air circulation under the earmuffs. In military operations the closed-air circumaural earphones are frequently mounted in the aviator or tanker helmets, which need to provide protection against the platform noise and where environmental hearing (auditory awareness of the environment) is not critical. Figure 5-14 shows an example of a circumaural audio HMD system designed for use in the U.S. Army's combat vehicles.

Supra-aural earphones rest on the external ears pressing against the pinnae. An audio transducer is coupled to the ear through a foam (soft) or rubber (hard) cushion. The earphones provide virtually no isolation from external noise and leak some sound to the environment. In some cases the audio transducer is specially located at a distance from the ear to provide a more uniform sound. Examples of such earphones are the Sennheiser HD 414 that is separated from the ear by a thick foam cushion covering the ear and the AKG 1000 that features a pair of loudspeakers radiating into the ears without any physical coupling.

Figure 5-14. Closed-air circumaural earphones mounted in the Combat Vehicle Crewman (CVC) Helmet used in the U.S. Army combat vehicles. The CVC Helmet is shown with protective ballistic shell attached at its top (Courtesy of the Bose Corporation).

Similarly to the circumaural earphones, the supra-aural earphones can be open-air (open-back) or closed-air (closed-back) systems providing various trade-offs of sound quality, comfort, and sound isolation. They are smaller and typically much lighter than circumaural earphones. Traditionally, supra-aural earphones are open-air types; however, here are also some supra-aural earphones that are the closed-air type (e.g., Sony MDR V700). Two examples of supra-aural headphones are shown in Figure 5-15.

Earbuds are small bowl-like earphones with or without small earcups that are placed in the concha or at the entrance of the ear canal without fully covering it. They are usually open-air type. Earbuds can be secured in place with a headband or ear clips that clip on the outer ears. This is the most common type of earphone used with portable devices such as cellular phones, iPods®, and MP3 players. In general, earbuds are inexpensive and lightweight (usually weighting less than 10 grams) but are not always comfortable and in general provide an audio display of lower quality than other types of earphones. However, despite their small size they are designed to produce quite high sound levels comparable to those of high power circumaural earphones in order to compensate for their lack of isolation from external noise. For example, the AKG K12P earbud-type earphones are rated as producing 127 dB SPL/V. Two examples of earbud-type earphones are shown in Figure 5-16.

Figure 5-15. Lightweight Supra-aural Earphones: Sony Model MDR 410LP (left) (Photo courtesy of the Sony Corporation); Beyer Dynamic Model DT 231PRO (right) (Courtesy of Beyer Dynamic Corp).

Figure 5-16. Sony MDR A34L earbud earphones with a headband (left) and Sony MDR EX71SL earbuds (right) (Courtesy of the Sony Corporation).

The last basic type of air conduction audio devices used in audio HMD systems are insert earphones, also known as canal earphones, canalphones, in-the-ear monitors, or in-the-ear earphones. Insert earphones use very small audio transmitters that fit inside the ear canal or are coupled to the ear by acoustic tubing ending with eartips (ear adapters). The eartips may be either disposable foam plugs or differently-sized permanent tips. Some manufacturers also offer custom molded insert earphones. Such earphones offer high noise isolation and have the potential for perfect fit to the shape of the ear canal. However, custom fitted canal earphones are expensive and cumbersome to replace.

The degree of comfort depends greatly on how the earphones fit into the ear canals of the listener. If the earphones fit perfectly, they provide excellent ambient noise attenuation and are comfortable for long term use. Their sound quality is usually very good and comparable to that of supra-aural and circumaural earphones. However, in some cases the users can notice the effect of occlusion due to modified acoustic properties of the ear canal blocked by the earphone. The occlusion effect is an increased audibility of low-frequency bone conducted sounds. This effect results as an additional amplification of the talker's own voice, which is heard by the talker as louder and stronger in low frequency energy (darker) than normal voice. Some occlusion effect is also present when the ear is covered by closed circumaural earphones. Some people with sensitive skin can also experience skin irritation if the insert earphones are worn for an extended period of time. Another disadvantage of insert earphones is their high maintenance; special care is needed due to progressive accumulation of earwax in the eartips of the earphones. The earwax closes the output port of the earphone and degrades the quality of the sound. To minimize this effect some of the insert earphones, e.g., Etymotic ER 6, use a special replaceable filter at the tip of the earphone. Finally, insert earphones, especially those that are not custom made, have a tendency to move in the ear when the user is running or moving heavily. These movements produce some noise in the ear canal and changes in the quality of sound. An external view of Shure E 4G insert earphone is shown in Figure 5-17.

Figure 5-17. Shure E4 insert earphone (right); shown with a custom molded silicon-gel sleeve (left) (Courtesy of Sensaphonics, Inc.).

A special application of an audio display system incorporating insert earphones is hearing aids that are designed to compensate for various types of hearing impairment. The primary difference between an insert earphone and a hearing aid is electronic circuitry of the hearing aid that includes built-in microphone, amplifier, some signal processing circuitry, and a battery to power the system. Hearing aids are self contained in a very limited space. Depending on the location of the system circuitry, hearing aids are classified as (1) behind-the-ear (BTE), (2) in-the-ear (ITE), (3) in-the-canal (ITC), and (4) completely in-the-canal (CIC) hearing aids. The BTE hearing aid is placed behind the pinna and is coupled to the ear though acoustic tubing terminated with an earplug. Conversely, the CIC hearing aid is deeply inserted in the canal and completely hidden from a casual observer in the shadow of the canal. Figure 5-18 shows typical shapes of various types of hearing aids.

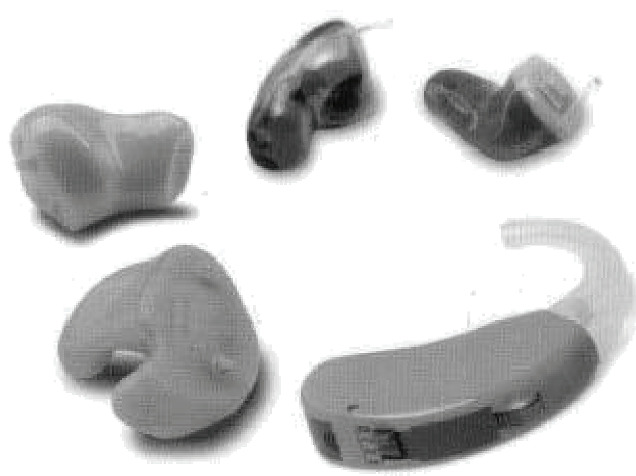


Figure 5-18. Various types of hearing aids (clockwise from the top: left and right completely-in-the-canal (CIC) hearing aids, behind-the-ear (BTE) hearing aid, in-the-ear (ITE) hearing aid, in-the-canal (ITC) hearing aid (Courtesy of Beckner Hearing Aides, Inc.).

An audio HMD system can incorporate any type of the audio transmitters described above. For example, both the closed-air circumaural earphones and the insert earphones have been incorporated in U.S. Army aviator helmets (e.g., HGU-56/P and SPH-4). Some proposed military helmet assemblies included small supra-aural earphones or loudspeakers built into the helmet's ballistic shell (e.g., Land Warrior Integrated Helmet Assembly Subsystem [IHAS]). Similar systems are also offered for the motorcycle helmets. The advantages and disadvantages of various earphones types used for audio displays are summarized in Table 5-3.

Table 5-3.
Advantages and disadvantages of various types of earphones used in audio HMD systems.

Earphone type	Advantages	Disadvantages	Examples
Closed-air circumaural headphones 	-Comfortable -Excellent acoustic isolation	-Bulky -Accumulate heat -Low frequency resonances	-AKG K271 -Beyer DT 250 -Beyer DT 770 -Bose QuietComfort 2 -Sennheiser HD 265/280 Sony MDR 7506
Open-air circumaural headphones 	-Comfortable -Sound natural	-Bulky -Low acoustic isolation	-AKG K240 -AKG K701 -Beyer DT 990 -Sennheiser HD 595/600/650
Closed-air supra-aural headphones 	-Less bulky, lighter weight compared to circumaural -Good acoustic isolation	-Accumulate heat -Resonance	-AKG K171 -Beyer DT 231 -Bose QuietComfort 3 -Sennheiser HD 25 -Sennheiser PX 200 -Sony MDR V700 -Telephonics TDH 39
Open-air supra-aural head-phones 	-Light weight -No acoustic isolation	-No acoustic isolation	-AKG K141 -AKG K1000 -Beyer DTX 30 -Grado RS1 -Grado SR 325 -Sennheiser HD 414/465 -Sony MDR 410

Table 5-3. (Cont)
Advantages and disadvantages of various types of earphones used in audio HMD systems.

Earbuds 	-Low cost -Very light weight -No acoustic isolation	-Low sound quality -No acoustic isolation	-AKG K14P -Altec Lansing AHP 131 -Beyer DTX 20 -Sony MDR A34L -Sony MDR EX81 -Sony MDR E82
Insert earphones 	-Very light weight -Good acoustic isolation	-Need good fit -High maintenance -May cause skin Irritation and occlusion effect -Good earphones are expensive	-Bose TriPort IE -Etymotic Research ER 3A/ER4/ER6 -Shure E3/E4 -Westone UM2

One problem caused by earphone-based audio display systems is that such systems occlude the ear and adversely affect auditory awareness of the environment. In many cases, such as air or mounted operations such awareness is not essential. However, during dismounted operations in quiet or in urban environments such awareness is critical to Warfighter's safety and effectiveness. In these cases earphone-based systems need to be equipped with environmental microphones and combine the function of a traditional audio display (audio communication, GPS-based navigation) with audio-processed monitoring of the nearby environment (talk-through system). Such systems are commonly called Communication and Hearing Protection Systems (C&HPSs). Several C&HPSs, such as the Nacre QuietPro, Bose ITH, ATI QuietCom, and Silynx QuietOps™ are available for both military and civilian applications. However they are quite expensive and can each cost several hundred dollars or more. In addition, all of them share some operational limitations including poor directional properties that affect the user's ability to identify the direction of incoming sound and a scaling of the distance to the sound source due to the amplification of external sounds. Despite these limitations, they effectively combine three required functions of audio HMD systems and will be discussed in-depth after the section on hearing protection devices.

Bone conduction systems

An alternative to earphone-based audio displays are bone conduction displays that transmit audio information through the bones of the skull without occluding the ears. Bone conduction systems utilize mechanical vibrators that deliver the audio signals to the listener by vibrating the bones of the skull. When pressed against the skull, the vibrator excites the bones and soft tissues of the head, transmitting the auditory signals through the mechanical pathways of the head into the cochleae. In addition to bone conduction transmitters, audio HMD systems can also use bone conduction receivers (microphones) to convert skull vibrations produced during speech emission into audio signals. Some advantages of the bone conduction microphones over air (boom) microphones are that they are not sensitive to environmental noise and can be placed inconspicuously at almost any place on the head.

Bone conduction audio HMD systems differ from earphone-based audio HMD systems only in the use of bone vibrators in place of earphones as the audio transmitters projecting auditory signals. Such systems can serve as separate audio communication systems or can be built into the helmet. In either case they need to be used with some form of noise protection system when used in noisy environments. Bone conduction systems are very effective audio displays when they operate in quiet and in moderate noise levels (up to approximately 80 dB (A)) without compromising the wearer's auditory awareness of the environment. At higher intensity noises they can be worn with an independent hearing protection system without affecting its operation. The use of hearing protection extends the operational range of bone conduction systems to approximately 110 dB (A). Bone conduction systems can be effectively used by both dismounted and mounted Warfighters because with proper mounting on the Warfighter's head they are not sensitive to vehicle vibrations (Henry and Mermagen, 2004). In addition, when transmitting signals convoluted with the wearer's HRTFs, they can provide directional resolution similar to that of the earphone-based display systems (MacDonald, Henry, and Letowski, 2006). As such they are a viable alternative to other types of audio HMD systems when auditory awareness is of critical concern.

Bone conduction displays, when used in quiet and moderately noisy environments, are inconspicuous, easy to hide, and have minimal effect on situation awareness of the surrounding acoustic environment. Bone vibrators and bone microphones can be used in situations where the listener must monitor acoustic activity in the surrounding environment and does not want anyone to be aware of the use of the communication system. The primary disadvantage of using bone conduction audio display systems in quiet environments is that they can produce some amounts of aerial leakage or excite the device to which they are mounted (i.e., a helmet). Fortunately, aerial leakage is preventable and the designs of military bone conduction audio displays need to significantly reduce or eliminate this leakage.

In high-noise environments, noise has less of an effect on the perception of bone-conducted messages than on the perception of messages emitted through a distal audio display (Knudsen and Jones, 1931). When the auditory signal and the noise are both emitted in the surrounding space their auditory images overlap spatially. When the auditory signal is transmitted through the bones, its phantom source is located inside the head and spatially separated from the noise sources located outside of the head. Spatial separation between the signal and noise sources improves the detection and clarity of the signal.

When a bone conduction system is used in a high noise environment its real value lies in its ability to be worn without interfering with the use of hearing protection devices (hearing protectors). In fact, the presence of hearing protection causes the sounds transmitted by bone conduction to be perceived as louder than when the ears are open. This effect is due to the sound amplification by the cavity of the external ear when it is closed by a hearing protector (Henry and Letowski, 2007). Thus, bone conduction systems worn with hearing protection devices can be used to communicate in noise levels up to approximately 100-110 dB (A) (Letowski, Henry and Mermagen, 2005; Letowski et al., 2004).

The quality of bone conduction displays greatly depends on where and how the vibrators are coupled to the bones of the skull. In general, locating the vibrator close to the cochlea improves the cochlea's response to stimulation (Stenfelt, Håkansson and Tjellström, 2000). However, the effectiveness of stimulation is also affected by the orientation of the axis of stimulation and the interconnections between bones and cartilages of the skull. McBride, Tran, and Letowski (2005) examined eleven locations around the skull to determine the most sensitive head locations for bone vibrator placement. Among the locations tested, the condyle (the bony portion directly in front of the opening to the ear canal) was found to be the most sensitive, followed by the jaw angle, mastoid bone, vertex (top of the head), and temple locations. Figure 5-19 shows relative average differences in the head sensitivity.

In addition to selecting an appropriate location for the vibrator placement it is also important to make a secure and stable contact between the vibrator and the head. The skin lies fairly loosely over the bones of the skull and provides some damping of the skull vibration caused by the bone conduction vibrator. The same damping effect is caused by hair and body fat on the head. In addition, large vibration magnitude and the curved surface of the contact area may cause intermittent and weak contact between the vibrator and the skull. Therefore, the vibrator needs to be pressed against the head with some minimal static force to transmit its vibrations effectively.

The greatest effect of skin damping on vibration transmission is at low frequencies and for the same static force pressing the vibrator against the head the damping effect decreases with an increase in frequency of stimulation. Békésy (1939) reported that at low frequencies around 200 Hz a force of 250G[1] applied over a contact area of 0.5 cm^2 is sufficient to transmit vibrations through the skin without an excessive loss of the transmitted signal. Békésy (1939) also reported that vibration transmission at frequencies above 7000 Hz is no longer affected by the static pressure as long as the pressure exceeds a certain minimum value.

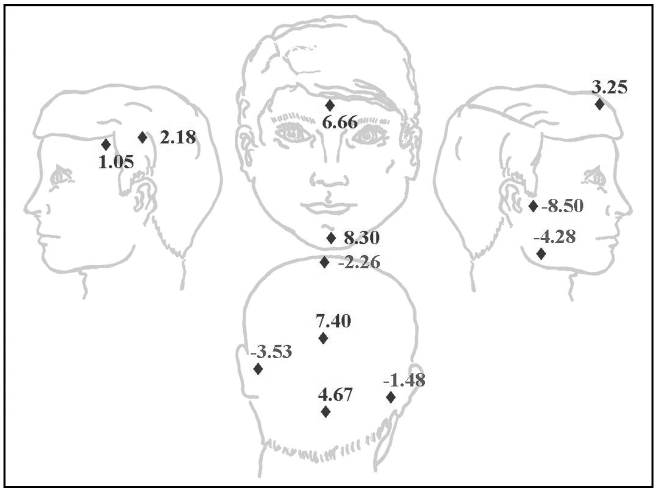


Figure 5-19. Average differences in coupling efficiency of a bone conduction vibrator at various locations on the human head. The locations with negative values are recommended for future applications. Units are in dB (HL) (adapted from McBride, Tran, and Letowski, 2005).

The effect of static force on transmission loss of a 2500 Hz tone is shown in Figure 5-20. For static forces above 500G and a skin thickness of 2.5 mm, the skin attenuates vibration by only approximately 2 dB for frequencies up to approximately 10 kHz. According to data provided by Békésy (1960) the static force of 250-300G is adequate for proper operation of bone conduction display systems, Other authors have recommended the use of similar or slightly higher forces: 200 to 400G (Harris, Haines, and Myers, 1953; Watson, 1938), 300 to 600G (Goodhill and Holcomb, 1955), 350 to 750G (Whittle, 1965). Although large static forces may be desirable for reliable and repeatable coupling of the transducer to the scull, forces exceeding 400-500G can cause physical discomfort for the listener and are therefore not practical for long term use and the force of 250-300G seems to be an acceptable compromise between quality of display and comfort of use.

The area of contact between the vibrator and the skull is another factor affecting effectiveness of bone conduction transmission. Khanna, Tonndorf and Quellar (1976) reported that the perception of vibrations improves with an increase in the area of contact. However, Goodhill and Holcomb (1955) observed better reliability of the threshold data with a vibrator having a contact area of 1 cm^2 than with a comparative vibrator having a contact area of 3.2 cm^2. Thus it seems that the optimal contact area is dependent on the stimulation location. The effect of contact area is also dependent on the signal frequency. Watson (1938) and Nilo (1968)

[1] Letter **G**, used to describe the amounts of static force in this chapter, stands for “Gram-of-force” as opposed to letter “**g”** meaning “gram-of-mass”. Gram-of-force (G) is a standard SI metric unit of force. Please note that in this context letter “**G”** does not mean gravity.

observed that the changes in the contact area from $1.1cm^2$ to $4.5cm^2$ had only minimal effect on hearing thresholds at low frequencies but hearing thresholds improved with larger contact areas for frequencies above 2000 Hz (2000 to 7000 Hz). Watson (1938) also noted that a smaller area, on the order of 0.5 cm^2, was uncomfortable to the wearer even with a relatively small (375G) amount of contact force. The concentration of pressure on a smaller area increases the wearer's discomfort and needs to be avoided.

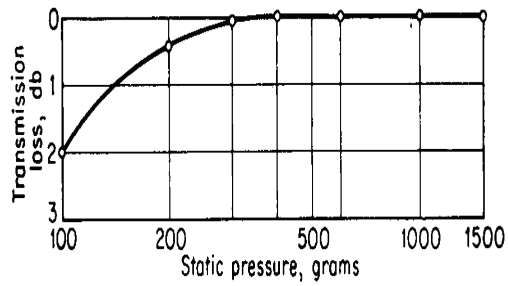


Figure 5-20. The effect of static force (in grams) on transmission loss (dB) of 2500 Hz vibrations transmitted through the skin (Békésy, 1939).

From an operational point of view bone vibrators can be divided into three main categories: head-mounted, in-the-ear, and dental transducers. Head-mounted vibrators are designed to be placed on the surface of the head and secured by a headband or other headgear-type retention system. They are commercially produced by a handful of companies (e.g., Percom, Temco, Sensory Devices, and Oiido) and may have various shapes and sizes. Examples of head-mounted bone vibrators are the Percom 31MIT and Teardrop vibrators shown in Figure 5-21.

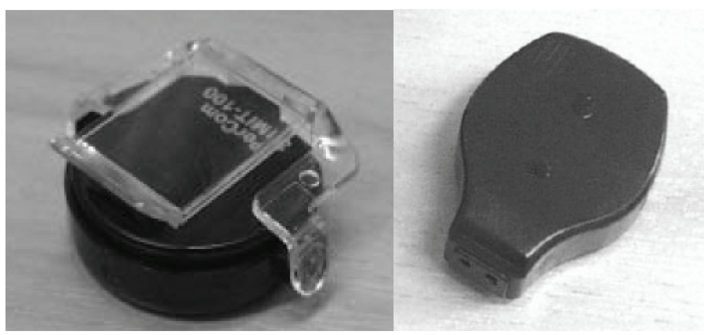


Figure 5-21. Percom 31MIT (left) and teardrop (right) head-mounted vibrators (Courtesy of Percom, Inc.).

Bone conduction systems can also be designed to operate in the ear canal (in-the-ear vibrators). An example of in-the-ear bone conduction display system is the TransEar, developed by Ear Technology Corporation, which has a bone vibrator embedded in the earmold. Such devices are similarly unobtrusive as ITC or CIC hearing aids and may be the devices of choice for creation of spatial bone conduction displays since their point of ear stimulation coincides with the entrance to the ear canal. However, they occlude the ear which negates the primary advantage of bone conduction systems over the earphone-based systems.

The dental vibrator is specially designed for placement in direct contact with the user's teeth. The vibrator can be attached to the listener's tooth or made to be clamped between the teeth as shown in Figure 5-22. Bone vibrators clamped between the teeth have been used as audio display systems by Navy Seals and recreational divers. Mounting a vibrator on a tooth is a challenging operation requiring dental skills and vibrating the teeth for an extended period of time may be harmful to the dental structure. In addition such wired systems are not operationally practical because of the cable connection running through the mouth. However, the concept of a dental vibrator with a wireless connection appears attractive for some special applications (e.g., stealth operations). This technology is still in its early stages of development.

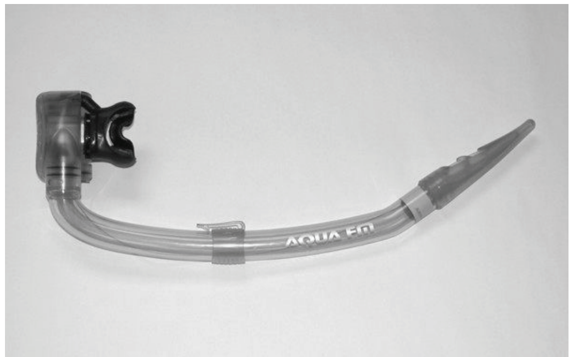


Figure 5-22. Vibrator embedded into the mouthpiece of snorkel (Aqua FM snorkeling system) (Courtesy of AMPHICOM®).

Various forms of bone conduction systems are used by hearing impaired people and serve as alarm devices, hearing aids, and assistive listening devices. Their use as general purpose audio displays is still in its infancy although several types of bone conduction audio HMD systems have been developed for use by firefighters, law enforcement agencies, and special operations forces. The main limitations of the existing systems are excessive nonlinear distortions at high operational levels and lack of optimized head interfaces. The list of major manufacturers of bone conduction devices and their main products is shown in Table 5-4.

A comparison of the main advantages and disadvantages of air conduction (earphone-based) and bone conduction audio systems is provided in Table 5-5. It is evident from Table 5-4 that the general problem in selecting an audio HMD system is the proper balance between auditory awareness of the environment and needed hearing protection. Since this balance is affected by the type of hearing protection device incorporated in the audio display system, it is important to consider all available options.

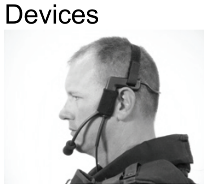
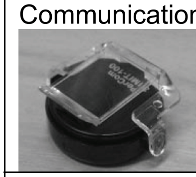
Hearing Protection Devices

Susceptibility to noise varies among people, but exposure to high levels of noise can cause permanent hearing impairment (see Chapter 11, *Auditory Perception and Cognitive Performance*).

Exposure to continuous noise at levels exceeding 85 dB (A) for 8 hours or more causes noise-induced hearing loss (NIHL). Such continuous noise sources include aircraft, tracked vehicles, and power generators. The resulting hearing loss is initially temporary and becomes gradually permanent after prolonged exposure. Similarly, exposure to impulse noise, such as weapon fire or bomb explosion, with peak sound pressure level exceeding 140 dB can cause permanent damage (acoustic trauma) to the hearing system. Very high impulse levels can also mechanically damage the tympanic membrane and soft tissue organs such as the lung or liver.

According to Department of the Army Pamphlet 40-501 *Hearing Conservation* (DA PAM 40-501, 1998), the first indication of the early state of noise induced hearing loss is decreased sensitivity in the range of frequencies above 2 kHz. Other symptoms include tinnitus (ringing in the ear), temporary muffling of sound, and a feeling of fullness in the ear, stress, and fatigue. More information on hearing loss is included in Chapter 11. The reader is also referred to the U.S. Army Center for Health Promotion and Preventive Medicine (USACHPPM) website (USACHPPM, 2006) that is an excellent source of information on hearing conservation.

Table 5-4.
Commercially available bone conduction communication systems.

Company	Product Name	Uses	Contact Information
Temco 	-HG-16 (boom microphone) -HG-17 (bone microphone) -HG-21 -HG-21D -FM-200 (gas mask) -Band-aid BC System	Stealth communication Gas masks Construction workers	www.temco-j.co.jp
Radioear / Sensory Devices 	-Radioear B-70/ 71/72 -BC System RE-1	Hearing testing Hearing aids Special forces Law enforcement	www.sensorydevices.com
Aliph 	-Jawbone	Mobile wireless communication (Bluetooth)	www.jawbone.com
PerCom / Audio Communications 	-Teardrop MIT -31MIT -17 MIT	Firefighters Law enforcement	www.audiocomms.co.nz
Vonia, Pegaso 	-BH-10/20/80 -EZ-500 -EZ-2000 -EZ-2000 -EZ-3000 -EZ-4200 -EZ-10/20/80	Telemarketing Entertainment Stealth communication Underwater communication	www.vonia.co.kr www.pegaso.co.jp

Military environments are predominantly high noise environments. Common military vehicles and equipment generate very high noise levels requiring hearing protection as specified by Department of Defense Instruction 6055.12 (DOD, 1991) and DA PAM 40-501 (1998). Continuous noise levels in armored personnel carriers (M113 and Bradley Fighting Vehicle) are the highest noise levels among Army vehicles. Internal noise in an idling vehicle is approximately 92 dB (A). Noise levels in these vehicles increase with speed and can reach 118 dB (A) at 40 mph. Similarly, noise levels inside military helicopters are higher than 100 dB (A). In helicopters, such as the Blackhawk and Apache, noise levels at the pilot's position can reach 106 and 104 dB (A), respectively.

Table 5-4. (Continued)
Commercially available bone conduction communication systems.

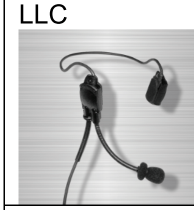
Company	Product Name	Uses	Contact Information
IntriCon 	-Bone conduction headset LV23	Warehouse operations Train operations Safety personnel Audio production crew	http://www.intricon.com.sg
Oiido Equipment 	-Bone conduction headset	Law enforcement Telemarketing	http://www.oiido.com/
Atlantic Signal LLC 	-Tactical Headset System MH180-H -Tactical Headset System MH180-S -Tactical Headset System MH-3	Military Law enforcement Gas masks	http://www.mhseriestacticalheadsets.com/index2.html
Tactical Command Industries 	-Tactical Assault Bone Conduction (TABC) Headset -SPEC-OPS II Binaural Bone Conduction Headset	Law enforcement Special forces	www.merchantmanager.com/tactical/headset_products.htm

Table 5-5.
Advantages and disadvantages of air conduction earphones and bone conduction vibrators.

Audio Display Systems	Advantages	Disadvantages
Air conduction systems	• Mature technology • Wide range of styles, quality, and prices	• Occlude ear canal • Interfere with hearing protection devices and must provide proper noise attenuation • Some systems provide insufficient ear ventilation and/or irritate the ear
Bone conduction systems	• Do not occlude ear canal • Inconspicuous to use (easy to hide) • Less susceptible to ambient noise effects than distal display systems; • Do not interfere with hearing protection devices	• Not quite mature technology, • Aerial sound leakage • Excessive static pressure may cause discomfort • Require a separate hearing protection system

Military personnel are also exposed to extremely high impulse noise levels from weapon firings. For example, USACHPPM (2006) reports at the gunner's position the peak sound pressure level for the Multi-Role Anti-Armor Anti-Personnel Weapon System (MAAWS) recoilless rifle is 190 dB SPL and for the Light Antitank Weapon M72A3 it reaches 182 dB SPL. Even small arms weapons like the M16 rifle and M9 pistol produce impulse noise levels reaching 157 dB (peak) at the shooter's ear, far above the hazardous level of 140 dB (peak). The methods of measuring impulse noise levels are the subject of the ANSI standard S12.7 (ANSI, 1986).

Military Standard MIL-STD 1474D, *Design Criteria Standard: Noise Limits* (DOD, 1997) is the governing noise control document for military materiel used by the U.S. Department of Defense. It specifies noise limits to equipment designers and manufacturers. It is intended to cover typical operational conditions. Required noise limits must not be exceeded if the materiel is to be acceptable. This standard is based upon requirements for hearing damage-risk, speech intelligibility, aural detection, state-of-the-art of noise reduction, and government legislation. The standard includes requirements for: steady-state noise, aural non-detectability, community annoyance, impulse noise, shipboard equipment noise, and aircraft noise (DOD, 1997).

Recent studies indicate that current noise exposure standards and design guidelines, as described in requirement 4 of MIL-STD 1474D (DOD, 1997) for impulse generating weapons are seriously in error. To overcome these limitations, the Army Research Laboratory (ARL) developed a mathematical model of the human auditory system which predicts the hazard from any free-field pressure and provides a visual display of the damage process as it is occurring. The model – Auditory Hazard Assessment Algorithm for the Human Ear (AHAAH) – is a powerful design tool which shows the specific parts of the waveform that need to be addressed in machinery and weapon design. This unique model is the only method of assessing noise hazard for the entire range of impulses relevant to the Army. This mathematical model calculates stress in the inner ear based on head orientation, hearing protection [manikin or Real-Ear-Attenuation-at-Threshold (REAT) measurements], aural reflex, and stapes displacement limitation. Risk is calculated based upon a hypothesis that damage to the hair cells in the cochlea correlates to a mathematical function of the number of and amplitude of basilar membrane displacements in a manner analogous to mechanical fatigue of solid materials. (Price, 2005; 2007) Additional information about the AHAAH is available at the AHAAH Website (http://www.arl.army.mil/ahaah/).

The primary means to protect Warfighters from harmful noise levels are hearing protection devices (HPDs). Communication earphones covering the ear or occluding the ear canal protect the user to some degree against the harmful effects of external noise. However, generally, the resulting level of protection is not satisfactory to prevent hearing loss in high level noise environments. In addition, in many operational situations the user may not have a communication system covering the ears. Therefore, in designing audio HMDs it is important to focus on hearing protection offered by both the communication systems and by the dedicated hearing protection devices alone.

Various HPDs and classifications of HPDs are available, but the two most important dichotomies of all HPDs divide them into passive and active devices and into linear and non-linear devices.

Passive linear HPDs are sound barriers that reduce the overall noise level by covering the entire ear or by insertion into the ear canal. They are generally in the form of an earmuff surrounding the ear or an earplug that blocks the ear canal and typically provide 25 to 35 dB of noise reduction, if worn correctly. They can also be shallow conic earplugs connected by a headband and providing some ear occlusion at the entrance to the ear canal. Such devices are called semi-aural HPDs or ear canal caps.

Earmuff cups (shells) may have various depths, shapes, sizes, and weights. The earmuff system consists of two earmuff cups supported by an over-the-head headband, attached to a safety hardhat, or mounted in a helmet. The positive features of the earmuff-type HPDs are their reliable and repeatable level of protection, easy fit, and comfort of wear. They are also easy to don and doff. However, they interfere with wearing glasses, protective masks, and other safety equipment (Wagstaff, Tvete and Ludvigsen, 1996).

Examples of passive earplug devices are foam earplugs, preformed flanged rubber earplugs, and custom-made rubber plugs. Each type of these devices has its own advantages and disadvantages. They are supplied in various sizes and have different sound attenuation characteristics. Foam earplugs may have different sizes and shapes (e.g., cylinder, cone). They are generally disposable HPDs and therefore require constant resupply. The flanged earplugs are soft rubber earplugs that may have one to four flanges (see Table 5-6). In addition, some of the flange earplugs may have built-in filters to equalize their frequency response (e.g., Earlove Earplugs, Musicians Earplugs ER-20) and can be custom-made for individual users (e.g., Musicians Earplugs ER-9, ER-15, and ER-25).

The semi-aural devices, or canal caps, offer less protection at medium frequencies than either earmuffs or earplugs and similar protection at high frequencies. Their typical average attenuation of noise is 10-15 dB. They are supported by a light headband that can be worn under the chin or behind the head, permitting semi-aural devices to be used together with various types of safety equipment. To some degree they facilitate speech communication in noise but are uncomfortable when used for long time periods due to the pressure exerted at the entrance to the ear canal.

Attenuation characteristics of linear passive HPDs are frequency dependent. Their attenuation increases with frequency due to increasing transmission loss through the solid material of the HPD and the elimination of ear canal resonances when earplugs are worn. The HPDs that attempt to provide uniform attenuation across a wide frequency range normally achieve it by some reduction of attenuation at middle and high frequencies. Typical passive earmuff-type HPDs provide attenuation of less than 15-20 dB at frequencies below 200 Hz and approximately 40-50 dB at frequencies above 3000 Hz for well-fitted HPDs. Passive earplug-type HPDs, especially foam earplugs, provide greater and more uniform attenuation at low frequencies but attenuation similar to earmuffs at high frequencies. It must be stressed that noise attenuation provided by earplug-type HPDs varies dramatically with the quality of fit and the depth of insertion. For example, poorly fitted flange earplugs may provide almost no attenuation at low frequencies. The overall protection provided by single or double hearing protectors cannot exceed 50-60 dB due to bone conduction pathways that bypass the ear protection.

Numerous field evaluations of HPDs suggest that the actual noise protection offered by HPDs in real operational environments is much less than the manufacturers' published data. This difference is mainly due to improper fit of the devices, wear and tear, and inappropriate size. Thus, selection of any HPD to be used as part of an audio HMD system must take into considerations both the official manufacturer's data and the field reports. Various types of passive HPDs, together with their advantages and disadvantages, are shown in Table 5-6.

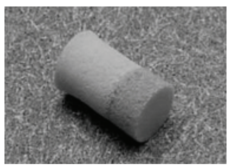
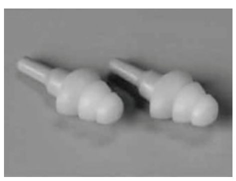
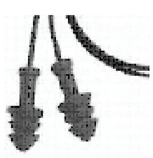
In general, passive linear HPDs are inexpensive (except for custom-made earplugs), easy to use, and effective if fitted correctly. They can be used separately or built into a helmet. For example, most military helmets with integrated communication and audio display systems, such as the HGU-56/P or SPH-4B flight helmets and combat vehicle crewman's (CVC) helmet, provide hearing protection with a closed-air (closed-back) circumaural earcup style of earphone. Additional protection against high levels of noise can be achieved by wearing a combination of earmuff and earplug devices (double hearing protection). Similarly, when using an earphone-based audio display, additional noise protection can be added by wearing the earplugs in combination with the earphones but such use will also reduce the level of the auditory signals generated by the earphones.

As described previously the major deficiency of earmuff-type HPDs is insufficient attenuation of low frequency industrial and military noise levels. Conversely, both earmuff-type and earplug-type HPDs provide relatively high attenuation of high frequencies, which adversely affects speech communication in quiet and in low levels of noise. In general, all passive linear HPDs interfere with speech communication and prevent detection of low-level sounds in the surrounding environment, thereby compromising situation awareness. This deficiency is addressed by different types of level-dependent HPDs. Level-dependent HPDs are a class of non-linear HPDs that significantly attenuate hazardous high intensity impulse sounds while passing low intensity sounds, such as conversational speech, with minimal attenuation. Level-dependent reduction of noise levels can be achieved by either passive or active reduction techniques.

Passive nonlinear HPDs are vented devices that have small orifices, diaphragms, or valves built into the HPD. These increase the protection provided against impulse noise as the noise level exceeds a certain threshold, usually 120 dB SPL (Shaw, 1982). Above this threshold, high noise levels result in a turbulent flow of acoustic energy through the non-linear element of the protector effectively closing the vent. At noise levels below this threshold, the protector acts as a regular vented earmold (earcup) providing usually less than 20 dB noise attenuation at high frequencies and none or very little attenuation at low and middle frequencies below 1000 Hz (normally less than 5 dB). Such protection characteristics of level-dependent HPDs facilitate speech communication and awareness of the environmental sounds in quiet and moderately noisy environments while protecting the Warfighters from high intensity impulse sounds from their own and enemy weapon fire. An additional feature of some passive level-dependent HPD is that they function as a pressure valve that slows down rapid changes of atmospheric pressure typically experienced during take-off and landing of aircraft. Examples of passive level-dependent HPDs include V-51R (American Optical), Bilsom ISL 655 (Bilsom), Ear Defender EP3 (EarPro), EarGuard (Cirrus), and the Combat Arms Earplug (Aearo).

The Combat Arms Earplug (CAE) is a level-dependent device designed for military operations. The earplug is produced in both single-end and dual-end versions shown in Figure 5-23. The dual-end version can be used as either a linear (olive drab plug) or non-linear (yellow plug) HPD. A small mechanical filter with a calibrated orifice is embedded in the non-linear end of the plug. When this end is inserted into the ear canal, the CAE passes the low-intensity, low-frequency sounds with as little as 5-8 dB attenuation and allows the user to hear normal conversation, footsteps, or vehicle noise while to some degree attenuating high frequency energy of the sounds. The attenuation of the plug rapidly increases at high noise levels starting at approximately 120 dBP SPL and reaches full peak attenuation of 25 dB at approximately 190 dBP (Dancer and Hamery, 1998), providing wideband hearing protection against the dangerous high-level energy of weapons fire and explosives. The linear portion of the CAE is used for hearing protection from high level steady-state noise environments such as those created by armored vehicles or aircraft where situation awareness is not a priority. It provides approximately 35 dB insertion loss at low and middle frequencies with insertion loss gradually increasing above 1000 Hz. At very high sound intensity levels exceeding 170 dB (peak) both types of the CAE earplug provide similar attenuation for frequencies above 250 Hz.

Table 5-6.
Advantages and disadvantages of various types of passive HPDs.

HPD	Advantages	Disadvantages
Earmuffs 	-Easy to fit -Good noise reduction -Comfortable to wear if light and properly adjusted -May protect pinnae from exposure to adverse environments and burns, e.g., from improvised explosive devices (IEDs).	-Bulky and heavy -Interfere with other headgear
Foam Earplug 	-Inexpensive -Good noise attenuation -Compatible with other headgear	-May cause skin irritation -Easily become dirty -Hard to fit
Musicians Earplug 	-Provide relatively uniform attenuation across wide frequency range -Provide protection without adversely affecting sound quality -Comfortable, especially if custom fit -Compatible with other headgear	-Expensive -Hard to replace -High maintenance (require earwax cleaning from inserted filter)
Single-Flange Earplug 	-Reusable -Inexpensive -Compatible with other headgear	-Poor noise reduction -May cause skin irritation -Difficult to insert correctly -May move with normal jaw movement
Triple-Flange Earplug 	-Good noise reduction -Reusable -Inexpensive -Compatible with other headgear	-Difficult to insert correctly -May cause skin irritation -May move with normal jaw movement

The right panel of Figure 5-23 shows a new version of CAE developed by Aearo in cooperation with the US Army Research Laboratory (ARL). Its main advantage over the old version is a rotating mechanical switch to open and close the filter permitting change from one type of hearing protection to another using a single-end plug.

Active level-dependent HPDs use an external microphone and an internal loudspeaker (earphone) to bypass passive attenuation offered by the HPD when environmental noise is below a certain threshold. Such an electroacoustic system enables the wearer to hear environmental sounds and spoken messages while wearing HPDs. When the threshold level is exceeded, the bypass pathway is closed and the HPD operates as a passive HPD. Most of the amplifiers built into active level-dependent HPDs have a volume control that allows the user to adjust the amplification of the bypass system. The system may also have a built-in sound compressor or automatic gain control (AGC) circuitry that automatically decreases amplification of the external sounds as the level increases. Examples of such devices are the Peltor Tactical 7, BBP Ltd. EP171, and Bilsom 707 Impact earmuffs and the Communications Earplug (CEP). When a radio or other external audio input is added to an active level-dependent HPD it becomes a C&HPS mentioned earlier in this chapter and described more extensively below.

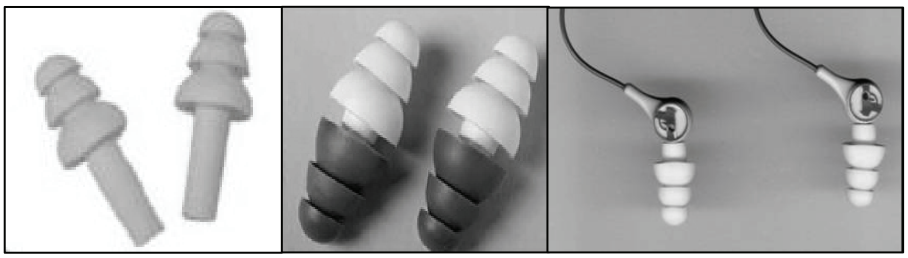


Figure 5-23. Combat Arms Earplug (CAE). Single-end version (left), dual-end version (middle), current single-end version (right) (U.S. Army Research Laboratory photos).

The advantages of a passive level-dependent HPD, such as CAE, when compared to similar active devices, are their low cost, light weight, ruggedness, and relatively easy maintenance. In addition, there is no battery requirement and no tangled wires. However, there are some issues with comfort of use of these devices if they are not available in various sizes (Scharine, Henry, and Binseel, 2005). The level-dependent passive HPD effectively complements bone conduction HMDs and together they are a viable alternative to C&HPS. Selection of a specific solution depends on the military or civilian system that is being supported and specific requirements of the missions to be performed.

Active noise reduction (ANR) devices are another class of non-linear HPDs. ANR is a method of reducing the level of environmental noise by phase cancellation. In the ANR system the surrounding noise is monitored by an environmental microphone, reversed in phase, and emitted back to the listener in an attempt to reduce the overall noise level. Such a noise-canceling scheme reduces noise only in selected locations but is a very good solution for HPDs. A general concept of an ANR system applied to an audio HMD is shown in Figure 5-24.

In the system shown in Figure 5-24, environmental noise is monitored by the external microphone (Noise Reference Mic) mounted outside of the passive HPD system. Captured noise is reversed in phase, signal processed, and emitted under the HPD by an audio transmitter (Headpone Transducer). The internal microphone (Error Mic) located close to the entrance to the ear canal monitors the overall noise level under the earmuff (earcup) and provides a differential signal that controls the amount of out-of-phase noise needed to minimize the overall noise level. The HPDs and audio HMD systems using ANR systems are frequently referred to as noise-canceling earphones or active noise-canceling earphones. Examples of audio displays using ANR technology are the Aiwa HP-CN6, AKG K-28NC, Bose QuietComfort 2, Phillips HN-110, Sennheiser PCX 250, and Sony MDR-NC10 systems. Similar ANR schemes are also incorporated into earbuds and earplugs (e.g., Etymotic ER-61C, Philips HN 060, Sennheiser CX-300, and Sony MDR-NC11).

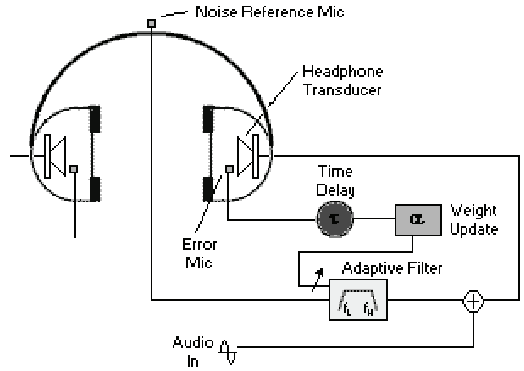


Figure 5-24. Active-noise reduction system incorporated in an audio HMD (Moy, 2001b).

The signal processing circuitry built into the ANR system usually consists of a phase inverter, filter, summer, and time delay unit. Due to the fact that environmental noise usually changes considerably both in time and in space, the filter is typically an adaptive system that self-selects optimum listening conditions. The most common form of adaptive filter used in noise-canceling earphones and hearing protectors is a finite impulse response (FIR) filter with a least mean square (LMS) algorithm. To describe such an audio HMD system the term "adaptive noise-canceling system" is frequently used.

Active Noise Reduction systems are very effective at low frequencies, providing up to 20-25 dB of noise reduction (Gowers and Casali, 1994; Nixon, McKinley, and Steuver, 1992). They are much less effective at high frequencies since reducing high frequencies requires more expensive computation. However, a combination of a passive earmuff with an ANR system provides relatively uniform high noise attenuation across a relatively wide frequency range. Such systems are becoming very common and popular. The ANR system of such devices provides the primary defense against low-frequency noises generated by engines, fans, and motors while the passive system provides the main defense against mid- and high-frequency noises generated by such devices as gas valves, pneumatic devices, saws, and power tools. It should be noted, however, that while ANR earmuffs offer a significant advantage over passive earmuffs, their total noise attenuation is similar to that offered by custom-molded earplugs (Christian, 2000). Therefore, for noise protection purposes both types of devices offer similar protection.

Traditionally, ANR HPDs have been found only in the earmuff-type of audio HMD systems. Today, however, ANR systems can also be found in supra-aural (e.g., Bose QuietComfort3) and in-the-ear devices (e.g., Panasonic RP-HC50E-A and Philips SHN 2500). However, both types of ANR systems integrated with audio HMD systems have some shortfalls. Earmuff-type ANR systems degrade both hearing protection and speech intelligibility when worn with glasses or CB (chemical/biological agent) mask (Mozo and Murphy, 1997). In some designs, ANR systems can affect the operation of the audio transducer distorting the desired signal or introducing additional high frequency noise (Wikipedia, 2007).

One of the problems with comparing various HPD systems is that noise attenuation provided by linear HPD must be measured using a standardized threshold shift method (ANSI, 1997a). However, attenuation provided by ANR systems is measured using the microphone-in-real-ear method (ANSI, 1995a) because of the low-level wideband noise normally produced by ANR systems (Mozo, 2001). Similarly, the passive nonlinear hearing protectors must be measured with the microphone method. Conversely, this method is not suitable for linear earplugs due to the difficulty with insertion of a microphone into the ear canal without compromising earplug attenuation. Unfortunately, for the devices for which both measurement methods can be used, the differences between measured attenuations can exceed 10 dB and are not uniform across the frequency range (Lancaster and Casali, 2004; Neitzel, Somers, and Seixas, 2006).

Communication and Hearing Protection Systems

Communication and Hearing Protection Systems (C&HPS) are modular audio HMD systems that can be worn with or without the helmet. As discussed above, they can also be permanently installed in helmets such as the aviation helmets (e.g., SPH-4) or tanker helmets (e.g., CVC). They typically offer approximately 20 dB of noise reduction. The level of noise reduction can be further increased if a communication helmet is combined with another kind of hearing protector (e.g. EAR earplug) or communications plug (e.g., CEP). For example, the noise level under an SPH-4B helmet earcup is approximately 91 dBA in a UH-60 helicopter flying at a speed of 120 knots and drops to approximately 69 dBA when an EAR™ foam earplug is added (DA FM 1-301, 2000). However, from the operational standpoint, the properties of C&HPS are very similar whether the system is built into the helmet or worn with the same helmet as a separate system.

There are several commercial off-the-shelf (COTS) C&HPS available offering different solutions to hearing protection and auditory awareness requirements. Available devices utilize both circumaural-earphone and insert-earphone types of design. Advantages and disadvantages of both types of hearing protection systems are similar to those listed in Table 5-5. Examples of the circumaural-earphone systems are the Bose ITH and MSA/Sordin Gen II systems. Examples of the insert-earphone systems are the CEP and Nacre QuietPro.

The Bose Improved Tactical Headset (ITH) is an earmuff-type communication system designed to protect Warfighters' hearing (up to 95-plus dB) while allowing them to communicate in the high noise of the M1114 up-armored HMMWVs (High-Mobility Multipurpose Wheeled Vehicles) and other light tactical vehicles being used by the U.S. Army. It can be secured on the head using an over-the-head headband and/or behind-the-head mounting strip. The ITH has two (left and right) forward-facing pass-through microphones and is designed to fit under the Advanced Combat Helmet (ACH). It provides hearing protection through both active and passive noise reduction of approximately 25 dB.

The MSA/Sordin Gen II is an earmuff-type headset that provides noise attenuation of approximately 25 dB. The headset has talk-through capability with a volume control and two (left and right) forward-oriented pass-through microphones. Both the Bose ITH and Sordin Gen II are shown in Figure 5-25.

Figure 5-25. Bose ITH (left) and Sordin Gen II (right) communication and hearing protection systems (Courtesy of Bose Corporation and MSA).

Another company offering earmuff-based C&HPS is Sennheiser. Its WACH 900 (Warrior Advanced Capability Headset) is an earmuff-type ANR and stereo talk-through system. It provides 15-20 dB wideband attenuation with the ANR system turned on. The system is designed for dismounted infantry and can be worn under a helmet or with a respirator. The stereo talk-through capability provides situation awareness. Another earmuff-based C&HPS system offered by Sennheiser is the SNG 100. The SNG 100 is designed for combat vehicle crewmembers and provides noise attenuation of 25-40 dB with ANR. Sennheiser also offers the SLC 110 system, which is an in-the-ear, passive, non-linear, noise reduction, militarized headset. It comes with earplug and concha tips that lock the headset in place. Situation awareness is enhanced by opening an acoustic port that bypasses the attenuation of the earplug. With the port open, impulse and gun-blast noise is attenuated using a non-linear filter that passes low intensity sound with a small loss but attenuates impulse noises by up to 30 dB. All these systems are shown in Figure 5-26.

The CEP was developed at the U.S. Army Aeromedical Research Laboratory (USAARL) for use with the aviator helmet. An expanding foam earplug is attached to a threaded hollow tube extending from the transducer. While the foam attenuates the ambient noise (Noise Reduction Rating - NRR = 29.5 dB), the tube transmits the sound from the transducer to the ear canal. When used on its own it provides noise attenuation from approximately 30 dB at low frequencies to approximately 45 dB in the 4000 Hz to 8000 Hz range. When used in combination with the aviator helmet, the earplug adds approximately 10 dB to the hearing protection provided by the helmet while improving radio communication clarity. Since the signal does not compete with the environmental noise, less audio gain is required to hear voice communication.

Figure 5-26. Sennheiser communication and hearing protection systems (C&HPS) (Courtesy of Sennheiser USA).

The Communication Enhancement and Protection System (CEPS) is an improved CEP with an additional capability of situation awareness, developed for the dismounted Warfighter or for use with the aviation helmet. The microphone providing the input signal to the radio for communication is also used to provide ambient sounds to the user. The microphone permits the Warfighter to hear environmental sounds and voice communication during dismounted operations. The system allows the user to control the level of sound from the external microphone with up to 36 dB of gain. With the lowest gain setting, the sound level to the user is limited to 95 dB (A). When impulse sound levels exceed 128 dB (peak), circuitry in the CEPS automatically disables the microphone to prevent any harmful amplified sound from reaching the ears (Mozo, 2004). The device is powered by two AAA batteries and weights only approximately 2.2 ounces (62 grams). Both the CEP and the CEPS are shown in Figure 5-27.

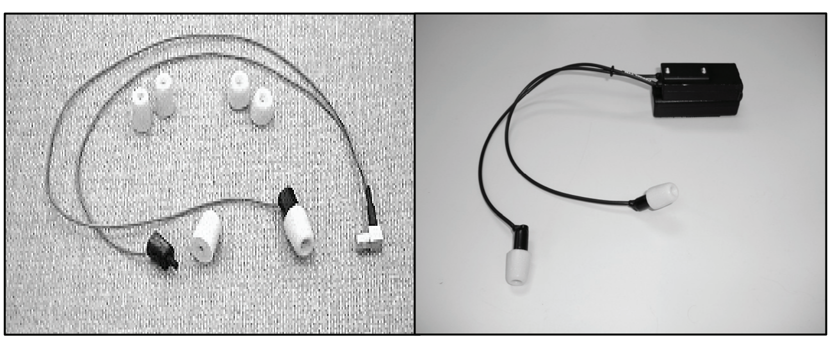


Figure 5-27. CEP (left) and CEPS (right) (Courtesy of USAARL).

The U.S. Air Force Research Laboratory (AFRL) developed a custom-molded earplug-based C&HPS called the Attenuating Custom Communication Earpiece System (ACCES). The system is shown in Figure 5-28. A small speaker is embedded in a silicon earpiece which is made from impressions of each individual user's ear canals. Its custom fit provides better comfort and higher noise attenuation compared to generic insert earphones. The ACCES can be worn alone as an earplug or plugged into the flight helmet (aircrew) or headset (ground support crew) for double protection and intercom communication.

QuietPro is an in-the-ear non-linear ANR system developed by Nacre, AS (http://www.nacre.no/). It includes one outer microphone, one inner microphone, and one miniature loudspeaker for each ear. All three elements are embedded in an earplug. The QuietPro provides binaural talk-thru, radio communication and hearing protection. Specifications for the device indicate that the active noise reduction system attenuates approximately 14 dB, targeted at frequencies between 63 to 500 Hz. Overall passive attenuation is 34 dB; and the loudspeaker is capable of reproducing a sound pressure level >125dB when combining digital ANR, talk-through audio and communication. The QuietPro system is shown in Figure 5-29.

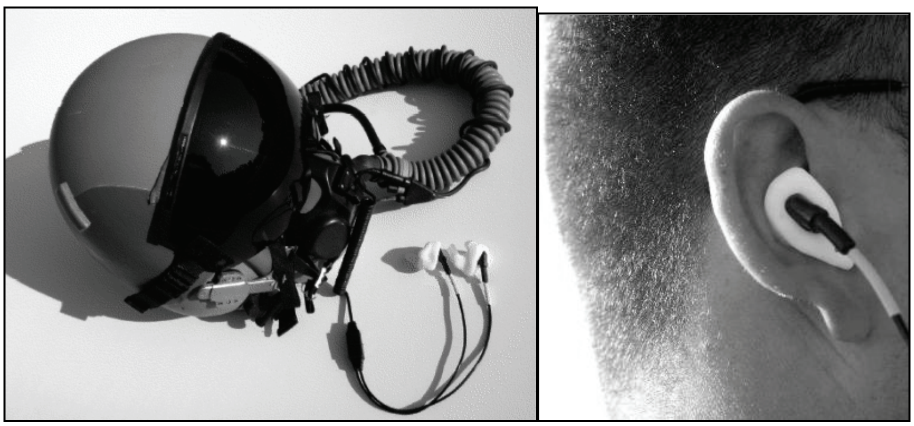


Figure 5-28. ACCES, attached to HGU-55P flight helmet (left) and in the ear (right) (Courtesy of U.S. AFRL).

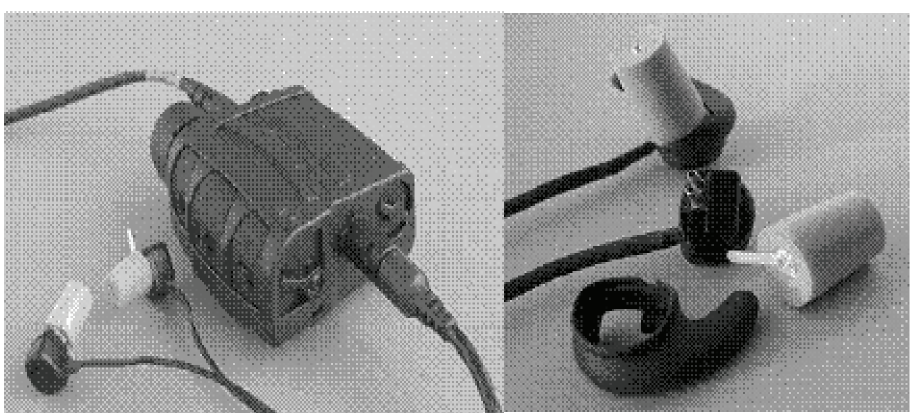


Figure 5-29. QuietPro (Courtesy of Nacre AS).

Earplug-based C&HPS have several advantages over the earphone-based systems. Proper fitting of in-the-ear style earphones provides both comfort and maximum noise isolation. These properties result in good intelligibility of transmitted speech and long-term user satisfaction. However, in-the-ear devices create hygiene-related problems in harsh environments and require careful fitting. There is also a psychological factor - not everybody wants to put something in their ears.

Acoustically Transparent Helmet

In modern warfare, military personnel require protection not only from kinetic threats, directed energy, and loud noises, but also from chemical and biological weapons. Therefore in early the 2000s the U.S. Army considered development of an encapsulating helmet for the Objective Force Warrior. This helmet design was intended to integrate with the biochemical protective suit and provide whole body chemical and biological protection. However, the encapsulating helmet creates a profound acoustic challenge. It would greatly attenuate sounds and distort auditory cues or even completely prevent the sounds from reaching the Warfighter's ears.

In an effort to find solutions to these negative acoustic affects of total encapsulation, various private companies, universities, and government agencies conducted research studies to develop an acoustically transparent helmet. The general concept of the acoustically transparent helmet is to place a network of microphones on the helmet shell and deliver captured spatial sound to the Warfighter's ears in order to restore natural hearing. The U.S. Army Natick Soldier Research, Development, and Engineering Center (NSRDEC) and the U.S. Air Force Research Laboratory jointly funded a project entitled *Concept and technology exploration for transparent hearing systems* that was executed by the Scorpion Audio Team, comprised of representatives of AuSIM, Inc., Fakespace Laboratories, Sensimetrics Corporation, and Boston University (Chapin et al., 2003). Figure 5-30 shows the Natick "Scorpion R2" helmet design, with potential microphone locations to capture the directional characteristics of external sounds.

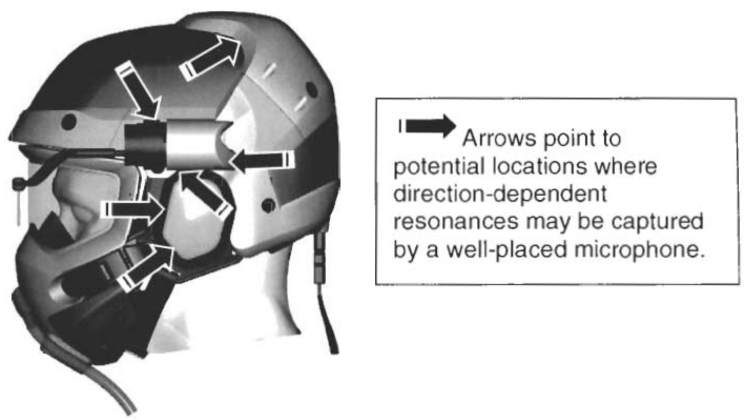


Figure 5-30. Natick "Scorpion R2" helmet design (Courtesy of Scorpion Audio Team, 2003).

Fig 5-31 shows the concept of the AuSIM 3D audio headset which is commercially available from the company. Using information captured by the microphone array, head tracker and HRTFs, AuSIM used their proprietary AuSim3D software to process the incoming sound and re-introduce its spatial cues according to head orientation. This system was used by AuSIM and Sennheiser Government Systems group in an attempt to build an encapsulating helmet providing Warfighters with restored natural situation awareness.

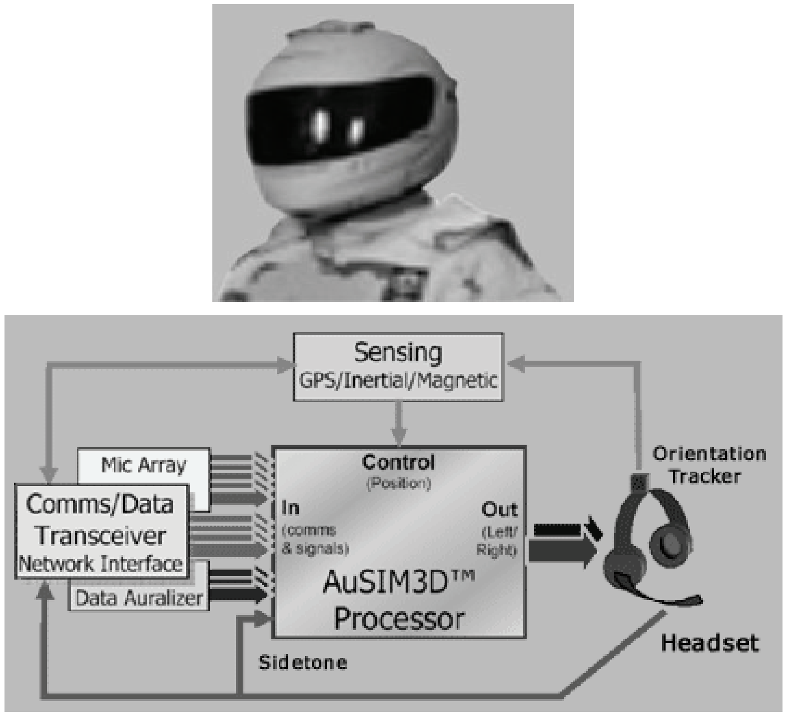


Figure 5-31. AuSIM 3D audio system for situation awareness and Sennheiser's conceptual transparent hearing helmet (Courtesy of AuSIM and Sennheiser Government Systems).

In another effort the ARL and Adaptive Technologies, Inc. (ATI) investigated natural hearing restoration solutions using a motorcycle helmet. Figure 5-32 shows prototypes of motorcycle helmets with mounted microphones built by ATI and ARL. All these efforts were terminated due to lack of funding caused by a change in the U.S. Army strategic vision.

Figure 5-32. Motorcycle helmet with microphone array built at ATI (left) and at ARL (right) for natural hearing restoration research (Courtesy of ATI and U.S. Army Research Laboratory).

Audio HMD System Design Issues

Design and selection of audio HMD systems needs to conform to general rules of human-centered design principles. Human-centered design treats the user as the final element of the HMD system rather than the screen of the monitor, membrane of the earphone, or moving plunger of the bone vibrator. Therefore, the engineering details of the display system needs to be specified not in terms of technology-based sensory stimulation parameters but in terms of perceptual and cognitive demands of the user and worked backwards toward sensory stimulation specifications.

General requirements

Human ability to respond to changing environments and to carry on required tasks requires free and effortless head movements that are minimally impeded by the additional weight of the headgear. This requirement is especially important for users that are moving on their feet and are not supported by a moving platform such as vehicle or aircraft. Therefore the mass of the audio HMD system should be made as low as possible and should not exceed 1.2 lb (545 grams) including all required batteries and not including the mass of the interface cables (Program Manager [PM] Soldier Warrior, 2007). The size and shape of the audio system should not interfere with the mission of the user including driving, crawling, parachute jumping, shooting, and the use of other headgear (e.g., video HMD or protective headgear).

An audio HMD system should be designed to fit easily in the ear or over the user's head without the need for extensive adjustment. The user should be able to don and doff the audio system in less than 10 seconds and without taking off or disengaging other personal equipment or taking off the gloves. The system should be ergonomically designed to self-set and stay stable in the operational position for the length of the user's mission. The parts of the system touching the user's skin should not create any adverse skin reaction or cause health hazards when used in operational environments. All basic mechanical, chemical, and electrical operational safety requirements for personal equipment should be met for the specified temperature, humidity, and atmospheric pressure ranges.

Discussion of various types of audio displays and hearing protection systems conducted in the previous sections clearly indicates that the design of an audio HMD system meeting all three basic operational requirements described in the first part of the chapter is a challenging effort. Further, in selecting one of many available audio systems for specific applications, there are many technical nuances and design compromises that need to be considered in order to develop a cost effective and ergonomically correct solution. Among the technical decisions that need to be made are audio transmitter technology, system interface, comfort, fit, weight, durability, mounting techniques, audio-visual integration, and compatibility with other equipment.

Audio transmitters

Both earphone-based and bone conduction audio display systems can utilize the same type of electroacoustic transducers (e.g., dynamic, piezoelectric, and electret transducers) that convert electric energy to mechanical energy and, subsequently, to acoustic energy. The main operational difference between the audio transmitters used in these two types of display is the difference in load impedance exerted upon the transmitter by air and bones of the head of the wearer. This difference, however, has a huge impact on the technical requirements for both types of transmitters. Thus, despite some physical and operational similarities between the transmitters used in both these types of systems, they differ substantially as specific technical solutions.

Magnetoelectric transducers

Most transducers used in audio HMD systems use either moving coil technology or the piezoelectric principle. Transducers using moving coil technology are called magnetoelectric or dynamic transducers. A dynamic transducer consists of a diaphragm connected to a coil of wires moving in the air gap of a permanent magnet. When an alternating electric current representing an audio signal is applied to the coil it creates an alternating magnetic flux. This magnetic flux interacts with the magnetic field of the permanent magnet pushing or pulling the coil and the attached diaphragm depending on the direction of the resulting magnetic force. The movement of the diaphragm creates changes of air pressure resulting in sound being projected to the ear. A drawing of the cross-section of simple dynamic earphone is shown in Figure 5-33.

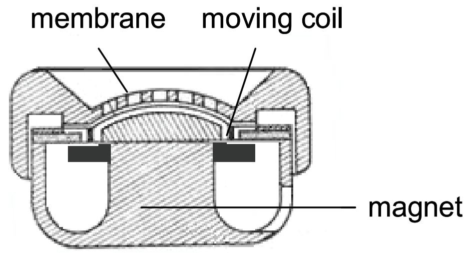


Figure 5-33. A cross-section of a simple dynamic earphone (adapted from (Kacprowski, 1956).

The diaphragm of the dynamic transmitters used in audio HMD systems is typically made of light-weight and stiff foil which requires a large radiating surface to reproduce low frequency signals. This large radiating surface also projects high frequency energy very efficiently. The magnitude of the movement of the diaphragm determines the loudness of the reproduced sound. The permanent magnets used in modern dynamic transmitters are usually neodymium and ferrite magnets. In order to improve the sound quality and the life of the transducer, some companies damp unwanted resonant frequencies and reduce the heat from the moving coil by introducing ferrofluid into the air gap of the transducer. Ferrofluids have the fluid properties of a liquid and the magnetic properties of a solid. A picture of a magnetoelectric transducer used in the earphones is shown in Figure 5-34.

There are two basic types of dynamic transducers: orthodynamic and isodynamic transducers. In an isodynamic transducer, the coil is embedded in the diaphragm in such a way that the resulting magnetic force applied to the diaphragm is equally distributed on the entire diaphragm surface. In an orthodynamic transducer, the force is applied to the diaphragm at only one point. The advantage of the isodynamic transducer is that it reproduces sound more accurately when compared to the orthodynamic transducer. However, the orthodynamic transducer is more efficient in the sense that it can produce louder sound with a given input voltage.

Figure 5-34. A frontal view of magnetoelectric (dynamic) earphone transducer (http://en.wikipedia.org/ wiki/Headphones).

Electromagnetic transducers

Electromagnetic transducers are similar to magnetoelectric transducers except that the coil is stationary and wound around an electromagnet and a metal membrane (or a dielectric membrane with an attached piece of magnetic material) is the moving element. Therefore, the electromagnetic transducers are sometimes called moving magnet transducers. In comparison to the magnetoelectric transmitters, the electromagnetic transmitters are generally smaller and more efficient but have a narrower frequency response. When they are incorporated in the insert earphones they require a good seal to the ear canal to provide reasonably wide frequency response. A view of electromagnetic earphone transducer is shown in Figure 5-35.

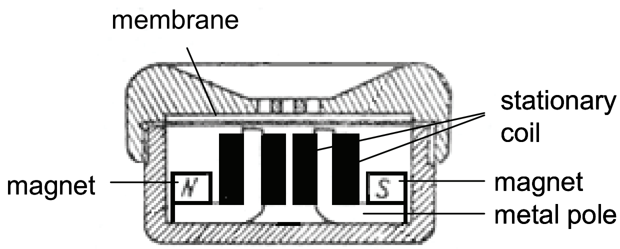


Figure 5-35. A cross-section of an electromagnetic earphone (adapted from Kacprowski, 1956).

Electromagnetic transducers are the transmitters of choice for in-the-ear devices because of their high efficiency. The most common type of electromagnetic transducer used in hearing aids and insert earphones is the magnetic balanced armature transducer in which armature is symmetrically balanced to minimize non-linear distortions of the system. The armature is ferrous material attached to the magnet and excited by an alternating magnetic field created by an audio current passing through a stationary coil surrounding the armature. The armature is attached to a plate or a membrane that vibrates and produces the sound. Typically the armature, coil, and magnetic structure are centered on the axis of the cylindrical construction, and motion of the armature in the axial direction is transmitted to the diaphragm by a pin coinciding with the axis of the cylinder. The whole assembly is supported by a shock-absorbing system. Examples of insert earphones that use balanced armature technology are the Etymotic ER6, and Westone UM1. Another type of electromagnetic transducer is the rocking armature transducer shown in Figure 5-36.

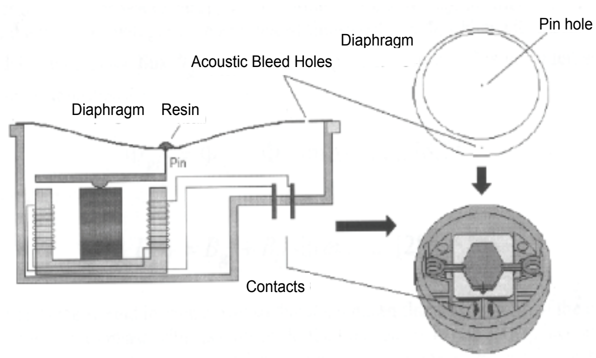


Figure 5-36. Electromagnetic transducer with rocking armature element (Moulton, 2004).

In some insert earphones the electromagnetic transmitter works together with a dynamic transducer to provide both efficiency and wide bandwidth with a wide frequency range. An example of hybrid canal earphones is the Ultimate Ear UE-10Pro that features two loaded armature electromagnetic transducers and one magnetoelectric transducer.

The construction of an electromagnetic transducer used in bone conduction vibrators from RadioEar is shown in figure 5-37. The magnet and coil assembly is attached to a lead block to increase its mass. The spring with spacers maintains the air gap, a critical separation between the permanent magnet poles and the armature. Consider the mass of the enclosure and armature as one part and the mass of the magnet, coils, and the lead block as another part of the system. Both parts of the system are connected together by the spring. When an ac signal is applied to the coil, the varying magnetic force in the air gap causes the metal armature to vibrate vertically. The mass of the magnet assembly is large enough make it appear fixed so that the armature and the enclosure move away and toward the magnet assembly creating mechanical vibrations of the transducer.

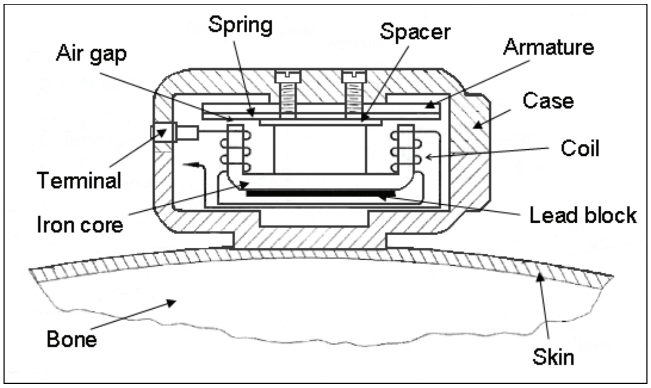


Figure 5-37. Schematic diagram of the RadioEar B-71 vibrator (Courtesy of RadioEar, Inc.).

Electrostatic transducers

A basic electrostatic (condenser) transducer consists of a thin diaphragm suspended at the center of two perforated flat metal plates. The plates and diaphragm form a capacitor. A high voltage bias is applied to the diaphragm polarizing it against both stationary electrodes. Because the suspended diaphragm is located at the center of the gap between the outer plates, the resulting attractive force to the outer plates is cancelled holding the diaphragm in a fixed position when no audio signal is applied to the electrodes. When an audio signal is applied to the outside plates, it creates an alternating electrostatic field between the plates pushing or pulling the suspended diaphragm. The movement of the diaphragm pushes air through the holes in the metal plates generating auditory signals. In some cases only one metal plate is used and the audio signal is delivered between the metal plate and the diaphragm. A schematic view of the electrostatic transducers is shown in Figure 5-38.

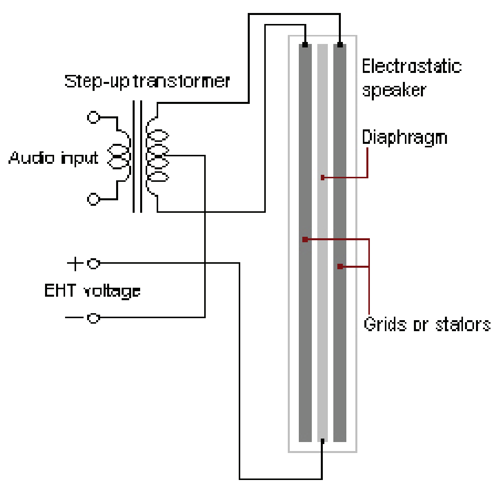


Figure 5-38. Schematic diagram of electrostatic transmitter (http://en.wikipedia.org).

Electrostatic transmitters are usually large, heavy, and expensive and require a high bias voltage. A step-up transformer is needed for the audio signal and is usually built into an adaptor box powered by commercial or generator power. Therefore they are seldom used in audio HMD systems. The primary advantages of electrostatic transducers are fast response, low distortion, and high fidelity sound reproduction. High quality electrostatic earphones include the Koss ESP950, Sennheiser HE60, Stax SR-1, and Stax 4070.

An *electret* transducer is an electrostatic device with the suspended dielectric diaphragm permanently polarized or with dielectric material filling the gap between the metal plate and the diaphragm. These transducers do nor require an external bias voltage and, thus, are much smaller, less expensive, and more rugged. However, they are very inefficient and thus only used in microphone assemblies.

Piezoelectric transducers

Piezoelectric transducers utilize the ability of crystals and some ceramic materials to generate a voltage in response to applied mechanical stress. When a voltage is applied across a piezoelectric material, the material is deformed. Conversely, if mechanical pressure is applied to the material, a potential difference is created on the opposite sides of the crystal. This unique property has many applications in electronic devices, especially in the audio industry. Because of this two-way effect, piezoelectric materials can be used as transmitters (earphone, vibrator) or receivers (microphone). An example of a piezoelectric earphone is shown in Figure 5-39.

Construction of piezoelectric transmitters is simple; they require fewer parts than magnetic and electrostatic transducers, and are very efficient. However, the frequency range of a piezoelectric transducer is limited. Therefore, in general, piezoelectric transducers are not suitable for applications where a wide, flat frequency response is required. Typical sizes of piezoelectric transducers used as microphones or buzzers are shown in Figure 5-40.

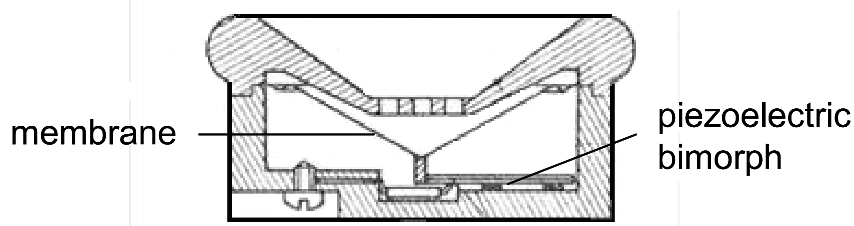


Figure 5-39. A simplifier view of a piezoelectric earphone (adapted from Kacprowski, 1956).

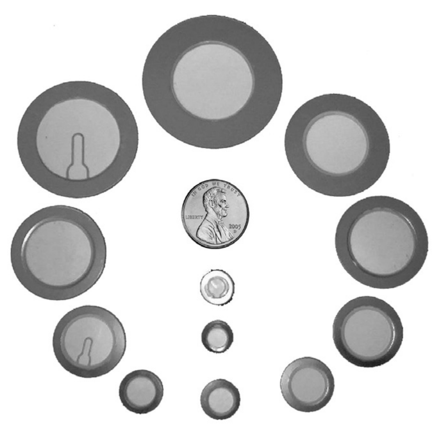


Figure 5-40. Piezoelectric transducers (Courtesy of Piezo Solutions).

Multiple-transducer designs

Audio HMD systems usually have one audio transmitter delivering a signal to the ear. However, some systems consist of two or more transducers delivering signals to the ear, built like large multi-loudspeaker systems. Such systems were mentioned above during the discussion of electromagnetic transducers.

In some high quality audio earphones there can be as many as three transducers; one for each frequency range - low-, mid-, and high-frequency. In some others, multiple transducers are used to create spatial displays. Recall that the spatial audio displays projected by proximal transmitters are normally created by means of HRTFs (discussed in Section 5-4). However, another method to create a spatial display is to use multiple transducers spatially separated slightly in an earcup and delivering multi-channel signal. For example, Konig (1996; 1997) described 4-transducer and 6-transducer earphones producing spatial sound without using HRTFs. The optimal arrangements of dynamic transducers inside the earcup for 4- and 6-channel headphones are shown in Figure 5-41. Figure 5-42 shows an actual arrangement inside the ear cup of a commercial surround sound headphone.

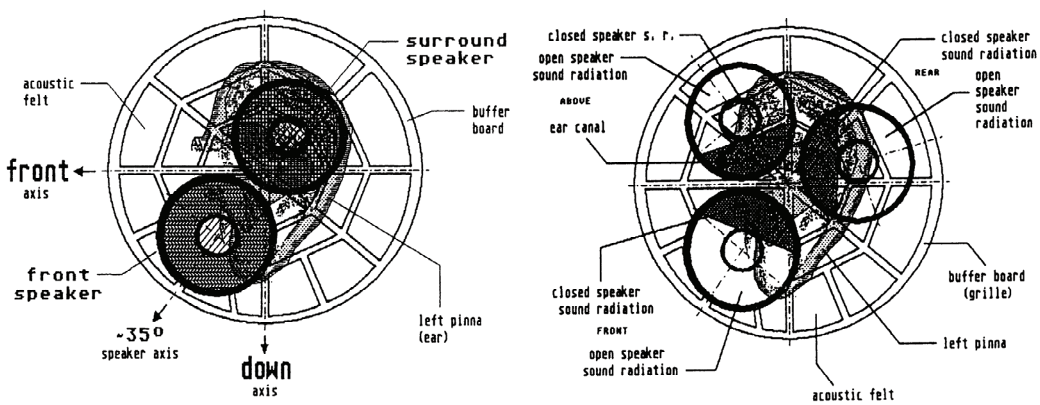


Figure 5-41. Transmitter arrangements inside left ear cup for 4-channel and 6-channel headphone (Konig, 1997).

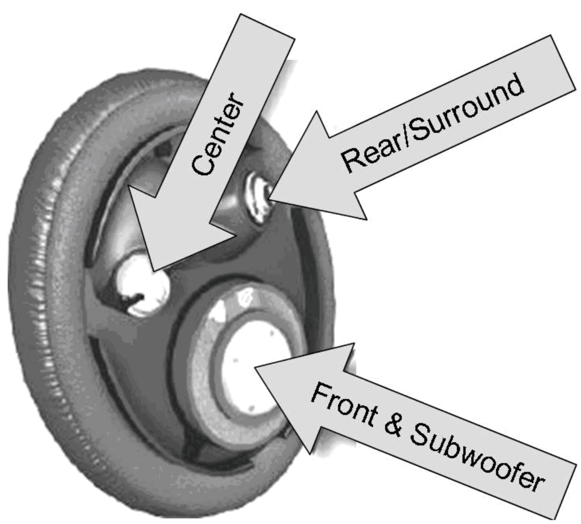


Figure 5-42. View of the earcup of LTB Magnum 5 surround audio display LTB-MG51-USB) (Courtesy of LTB Audio Systems, Inc.).

Few research papers describing this auditory spatial display technique have been published (e.g., Makowski and Letowski [1975] and Letowski and Makowski ([980]). The advantage of this technique is that there is little signal processing involved. However, today the use of HRTF in spatial displays is a more common practice due to increased microprocessor speed, reduced power consumption, and the low cost of hardware.

Audio transmitter calibration

The frequency response of an audio HMD system is typically measured as a pressure response using a standardized acoustic coupler. An acoustic coupler is an interface device that represents a standardized load to the acoustic transmitter used in the display. In the case of earphone-type audio systems it is a small chamber of specific shape and volume with an opening for coupling the audio transmitter (an earphone) to the chamber and with a measuring microphone terminating the chamber. Calibration procedure requires a specific static force pressing the transmitter against the coupler and specific environmental conditions to operate properly. In the case of bone conduction audio systems an accelerometer (motion transducer) is attached to a mechanical device providing a standardized load for the audio transmitter (a vibrator). The role of acoustic and mechanical couplers is to provide standardized and repeatable load conditions similar to the load conditions of the ear or the skull bones.

Standardized couplers provide repeatable data but such data are not necessarily a good representation of the signal delivered to the human listener. To know the actual frequency response of the audio transmitter seen by the ear or the bones of the head, the acoustic coupler used for such measurements must exactly represent the actual load provided by the human ear or the human head. The acoustic couplers that intend to represent the exact load provided by an average human ear are referred to as artificial ears or ear simulators (e.g., B&K 4153 and Larson-Davis AE100). The couplers that intend to simulate the load provided by a mastoid bone of the human head are called artificial mastoids (e.g., B&K 5090 and Larson-Davis AMC93). They have a specified range of frequencies within which such simulation can be assumed. Outside this range, the devices should be treated just as regular acoustic and mechanical couplers which do not necessarily match human characteristics.

Another method of measuring frequency response of earphone-based systems is to mount such displays on a manikin with artificial ears built in the head of the manikin. Such manikins are called artificial heads, binaural heads, or dummy heads by their developers. Examples of such heads include the B&K head and torso simulator (HATS), Aachen Head (HEAD Acoustics), and Knowles Electronic Manikin for Auditory Research (KEMAR). An artificial head provides a more natural coupling between the transmitter and the measuring system and simultaneous assessment of two transmitters (left and right) in their natural positions on the head. However the artificial ear terminates the ear canal rather then being mounted flush with the head and the collected data must be compensated for the additional travel of the acoustic wave along the canal.

Still another possibility is to put the display on a real person with miniature microphones mounted at the entrance to the ear canal. Such a real (human) load is not standardized and repeatable but provides the users with information regarding the effects of their own head on the auditory stimulus emitted by the transmitter.

The frequency response of a typical audio HMD is not flat and has several resonance and anti-resonance regions. In many applications not attempting to simulate the recording space or specific virtual environment such a frequency response is fully acceptable. However, any faithful reproduction of the recording environment requires a flat frequency response. This flat response is especially important if the signals are convoluted with a specific HRTF to create realistic immersion in the virtual environment.

There are three basic methods of earphone calibration/equalization: pressure equalization, free-field equalization, and diffuse field equalization. Each of these methods attempts to flatten the frequency response of an earphone with respect to a specific reference point. Pressure equalization flattens the frequency response in reference to the sound pressure measured using an artificial ear (acoustic coupler) or artificial head methods. Free-field equalization intends to recreate the conditions of sound field listening in which the listener is in front of a sound source in a non-reflective environment. Diffuse-field equalization assumes the sound source is not necessarily in front of the listener and the sound arrives at the ear with the same intensity and the same probability from any direction (Killion, Berger, and Neuss, 1987; Thiele, 1986; Larcher, Jot, and Vandernoot, 1998). Diffuse-field equalization is the most appropriate for simulating listening to distant sound sources in an enclosed space or in a free field environment when the sound sources surround the listener. Diffuse-field equalized earphones provide better spatial impression of the sound and make it easier to differentiate between sounds coming from the front and back of the listener. The compensated frequency response for diffuse-field listening is called a diffuse-field frequency response. Diffuse-field equalization is commonly built into high-quality earphones intended for music listening or virtual reality listening. Such products are provided by AKG (e.g., K 240D), Etymotic (e.g., ER 4P), and Sennheiser (e.g., HD 250, HD 580, HD 600, HD 650), Stax (e.g., Lambda Pro), and other manufacturers. An example of the relation between the pressure response and the diffuse-field response for the Telephonics TDH-39 earphones is shown in Figure 5-43.

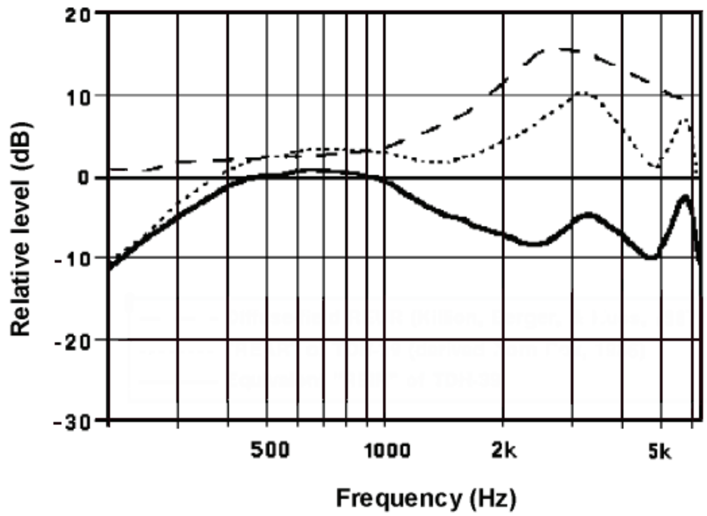


Figure 5-43. Relation between the pressure response (dotted line) and diffuse-field response (solid line) of the TDH-39 earphones. The dashed line shows the transformation function between pressure and diffuse-field environments (Cox, 1986; Killion, Berger and Nuss, 1986).

Audio receivers

Audio HMD systems are designed to provide information to the user via the auditory path, but the technologies used to generate acoustic stimuli may also function as collectors of mechanical (acoustic) energy and can be used to convert this energy into electric signals. The transducers that convert acoustic signals into electric signals are called audio receivers or microphones. Microphones used in conjunction with audio HMD systems are typically dynamic (moving coil microphones or condenser (electret) microphones. They can be used as air conduction microphones, bone conduction microphones, or throat microphones, which are mounted on the neck and receive signals directly from vibration of the vocal folds. Air conduction microphones are typically the noise-canceling type designed in such a way that the unwanted ambient noise is presented to two out-of-phase microphone elements while the desired speech communications is presented to only one microphone element. Using this technique, the unwanted noise may be reduced through phase cancellation. Bone conduction or throat microphones are much less susceptible to air conducted noise energy than air microphones and provide a good signal-to-noise ratio without additional signal processing.

Audio receivers should be designed and selected for specific applications. Frequency response, sensitivity, impedance, etc. of a microphone must be matched to the equipment to which it is attached and to the input conditions in which it operates. In addition, environmental conditions in which the microphone operates must be considered and they include dust, shock, vibration, rain, salt spray, temperature, and humidity extremes.

System interfaces and connections

The input to audio HMD systems comes from microphones, computers, intercom systems and/or radio communication systems through wired or wireless connections. To operate properly within the required communication regime, the audio system wiring and input circuitry should be compatible with technical specifications of the whole communication system. These requirements include type of connectors and pin assignments, signal level and impedance matching, and common ground requirements. Switching from the send to receive mode of operation and vice versa should be accomplished by a voice operating switch or by a push-to-talk (PTT) switch which is easily accessible and sufficiently large to be operated without removing gloves. All cables and fixed connections shall withstand 20 lbs tension to operate securely and reliably (PM Soldier Warrior, 2007).

For wired connections, audio HMD systems are normally connected to sound sources by using plugs and jacks to facilitate easy detachability. Different sizes of plugs are available to mate with different form factors of jacks. A mini-plug known as the 1/8 inch (3.5 mm) plug is the most common for portable devices; a smaller plug (2.5 mm) is common for cellular phones; and the full size 1/4 inch phone plug is often used in professional audio or laboratory applications. The universal serial bus (USB) connector is another new type of audio connector used to interface digital audio signals to/from personal computers or game consoles. As discussed in the initial part of this chapter, there are many methods to display audio signals to the listener, so these plugs can be mono or stereo plugs (2 or 3 electric contacts), or a connector with multiple pins to accommodate combinations of different signals as the application may require.

Cables connecting the audio source to the audio HMD system also contribute to the quality of the reproduced sounds. Unshielded cables and connectors are susceptible to electric interference from other sources. High conductivity cable provides improved signal transmission and results in less signal distortion. In some high quality sound systems, optical fiber is used for optimal signal transmission. When fiber optic cable is used the signal must be converted from a light signal to a mechanical signal using electronic converters located within the HMD system. For military applications, the standard electric interface to radios and intercommunications systems is the U-329/U connector shown in Figure 5-44. This connector provides single-channel audio to the headset (handset). Since military radios provide monaural audio, the military has not yet adopted a standard multi-channel connector configuration. Typically the connector is wired as follows:

- Pin A Common Ground
- Pin B Transmitter
- Pin C Push-to-Talk Switch
- Pin D Receiver (microphone)
- Pin E DC Power (not standardized)

Figure 5-44. U-329/U Audio accessory connector used on U.S. military radios. This 6 pin connector is common to many military radios including the AN/VRC-111 and SINCGARS (Courtesy of Tactical Engineering).

Sound signals can be also delivered to audio HMD systems through wireless networks. Although cordless telephones have been available for many years, wireless audio HMD systems are only recently becoming common. The developments of digital radio frequency communication and of low cost transceiver microprocessors make wireless systems more attractive and affordable although they produce an electro-magnetic signature that is not desirable in some cases.

The most popular communication protocol used with wireless audio HMD systems is Bluetooth, also known as the Institute of Electrical and Electronic Engineers (IEEE) standard 802.15.1. It allows two devices to communicate with each other via unlicensed short range radio frequency (RF) signals. Bluetooth, developed in 1994 at Ericsson Radio Systems, Netherlands, was designed for low power short range communication (Institute of Electrical and Electronic Engineers, 2005; McDermott-Well, 2005). With Bluetooth technology, the audio HMD can receive signals at a maximum range of 1 m, 10m or 100m, depending upon the RF power of the system. A picture of an audio display utilizing Bluetooth technology is shown in Figure 5-45. For greater ranges, Wireless Fidelity -WiFi - a spread-spectrum system operating on several channels in the 2.4 GHz band (also known as IEEE 802.11, ANSI/IEEE Standard 802.11, 1999 edition (R2003)) is used. Other wireless signal transmission methods used with audio HMD systems are analog radio frequencies (very high frequency [VHF] or ultra high frequency [UHF]) or infrared light (Moy, 2001a).

Figure 5-45. Bluetooth mobile phone headset (http://en.wikipedia.org/wiki/Bluetooth).

Although wireless networking provides great convenience (no tangled wire, no tether), each wireless technology has its own advantages and limitations. In general, the wireless network is susceptible to interference, introduces noise, and drops connections occasionally. Infrared technology uses infrared light to transmit audio signals, thus requiring line of sight to the base system. Radio frequency signals can transmit through walls and often interfere with other surrounding radio frequency systems.

Mounting and hearing protection considerations

Mounting of the audio HMD on the head is very important consideration because it affects both effectiveness of the interface and user's comfort. In considering a mounting solution for an audio HMD three basic factors need to be taken into account: technical quality of coupling the audio HMD to the user, user's comfort, and system durability.

In order to provide proper interface to the ears or the head of the user an audio HMD can be built in the headgear, worn with a headband or other supporting structure, or inserted in the ear canal. There are several types of headbands used with different audio HMD systems that fit over the head, behind the head, behind the neck, or under the chin. The over-the-head mounting technique supports the weight of heavy ear cups and provides stability of the headphones on the head. The headband can be a hard bow conforming to the shape of the head or a soft harness around the head with optional top head support. The former type of headband is commonly used with circumaural and supra-aural audio systems whereas the latter one is used with bone conduction systems and lighter earphone-type audio HMD systems such as ear buds, and insert earphones.

In addition to soft harness design another headband style that is especially designed to be used with headgear is the behind-the-neck headband. The behind-the-neck headband is curved under the hair line along the neck and up around the ears. The headband hangs relatively loose behind the neck providing spring-like action holding audio transducers in place. This design interferes less with hair or helmet than over-the head or behind the head designs and is usually used with lighter audio display systems as supra-aural or ear bud systems.

Due to the variety of head sizes, the length of a headband needs to be adjustable for comfort and fit. The width of a headband also contributes to its pressure on the head. Padding with cushions provides better fit and makes long-term wear more comfortable. Some headbands are also foldable for better storage and portability. The over-the-head headband is probably the most reliable of all the on-the-head mounting system; however, it can interfere with other headgear.

Another mounting method is the clip-on-the-ear design, which includes a clip attached to the miniature audio transmitter by a hinge in such a way that it can be easily opened and closed to remove or keep it in place. This design is usually seen in light and inexpensive audio display systems using ear buds, supra-aural earphones, or insert earphones. The limitation of the clip-on-the-ear design is lack of comfort during the prolonged use due to the pressure of the clip on the ear.

Some small and lightweight audio display systems such as ear buds or insert earphones can be worn without any additional support by fitting snugly on the conchae or inside the ear canals. However, with this mounting technique, the earphones have a tendency to be pulled out of the ears by the weight of the connecting cables, especially during physical activities that require a variety of movements. Examples of mounting techniques used with helmet independent (add-on) audio display systems are shown in Figure 5-46.

Figure 5-46. Examples of various light-weight headsets with over-the-head (left), behind-the-neck (center), and clip-on-the-ear (right) mounting (http://www.amazon.com; http://www.thanko.jp; http://www.boscovs.com).

For military and firefighter applications it is desirable to integrate an audio HMD into the helmet. Transducers can be integrated into the impact resistant shell or embedded into the padding of the helmet support system. This mounting method gives users the convenience of fewer cables and only one piece of equipment to care for. For example, the Bose CVC helmet has its audio display and communication system integrated into the helmet shell. This helmet is shown in Figure 5-47 (left panel). Similarly, the Gentex aviation helmet HGU-56/P is equipped with audio HMD in a form of a pair of earphones mounted under the helmet. The CEP and CEPS in-the-ear systems can also be successfully used with the HGU-56/P and SPH-4B aviation helmets. These implementations are shown in Figure 5-47 (middle and right panels).

Audio displays integrated in the helmet are state-of-the art solutions for the aviators, tankers, and other users that need protection from noise and reliable communication within a moving platform. However, when the audio HMD system used for communication purposes is integrated into the helmet, the Warfighter does not have radio communication capability without donning the helmet. This creates the need for modular audio HMD systems for dismounted Warfighters, firefighters, security personnel, and others who may or may not wear helmets. Such systems employ C&HPS. The C&HPS may be worn using one of the mounting techniques described above or can be embedded into a fabric cup or harness worn under the helmet.

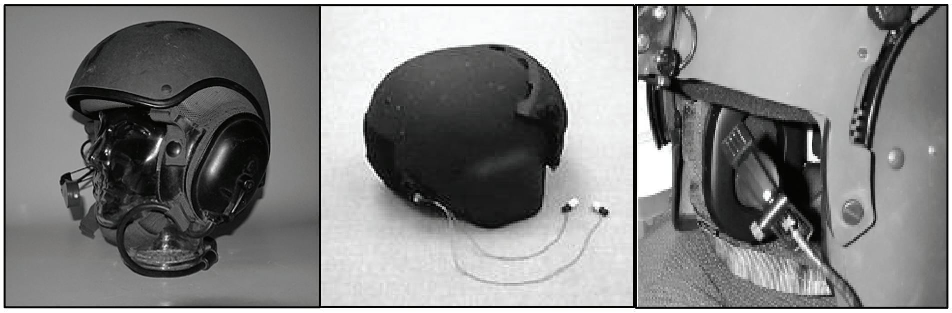


Figure 5-47. CVC helmet (left panel), HGU-56/P with CEP (middle panel), and HGU-56/P with CEPS (right panel) (Courtesy of USAARL).

In the case of the bone conduction audio HMD systems the selection of an appropriate mounting technique is challenging due to contradictory requirements of a minimum static force on the contact area needed for comfortable use of the display and the need for some minimum static pressure to provide good contact to the head and efficient sound transmission. These requirements favor large low-profile curved transmitters or a distributed network of miniature transmitters. In commercial bone conduction headsets, vibrators are secured with over-the-head, around-the-forehead, behind-the-head or behind-the-neck headbands (e.g., Sensory Devices and Vonia systems), or are incorporated into a web cap as in the case of the Temco HG-17 headset. Temco also produces an integrated bone conduction audio display and communication system intended to be mounted on a gas mask (Temco FM-1) or attached by an adhesive to the skin over the temporal bone behind the ear (Temco SK-1). For military Special Forces, security personnel, police, or intelligence agents the bone conduction transmitters can be secured on the head under the hair or mounted in inconspicuous head covers such as a baseball cap or hat.

Similar to air conduction transducers, over-the-head headbands are found in most commercial bone conduction headsets with the vibrators pressed securely to the face bones. Typically, the headband of a bone conduction audio HMD is stiff and flexible enough to maintain adequate static force on the vibrators. However, when worn with a helmet, the pressure of the helmet on the stiff headband can cause the vibrators to lose contact with the skull. Therefore such modular use requires soft harness mounting rather than hard headband mounting of the transmitters. Figure 5-48 shows typical mounting techniques used in commercial bone conduction audio HMD systems.

Figure 5-48. Examples of mounting techniques used with Temco bone conduction audio systems. The pictures show the over-the-head (left panel), behind-the-neck (center panel), and on-the-gas-mask (right panel) systems (Courtesy of Temco Communications, Inc.).

User's comfort is the most critical element of mounting considerations for audio HMD systems. Sound quality is usually considered the most important factor of audio HMD systems with comfort usually considered as a secondary requirement when the systems are used for short periods of time, that is, only when needed. However, comfort may actually be equally important as sound quality when an audio display system must be worn for long periods of time. Long-term discomfort results not only from an uncomfortable fit of the audio HMD but also from the degree of psychological isolation caused by the headgear and fatigue caused by system unbalance, weight, and a large number of controls that need to be operated when the system is used. A user may mildly complain about an audio HMD system with less than optimum sound quality but that same user will typically refuse to wear uncomfortable equipment for long periods of time – tens of hours, not minutes. Many factors (weight, compatibility with other equipment, mounting technique, fit) contribute to quality and comfort. If an audio HMD system is uncomfortable, it will not be used regardless of how well it performs and protects.

Earmuff-type HMD systems are typically built in the CVC and aviator's helmets providing hearing protection and housing audio communication transmitters. They perform a significant role in providing stability of the helmet on the user's head and overall comfort of the helmet (Mozo, 2001). They also isolate the ears from potential contact with the helmet liner, which increases the overall comfort of the helmet. Their main drawback in CVC and aviator's helmet applications is that they do not provide good ballistic and lateral impact protection (Shanahan, 1985). However, there are some design considerations (e.g., lower weight, modified structural strength) that may increase the lateral impact protection of earmuff-type HMD systems (Mozo, 2001).

The amount of hearing protection needed for the audio HMD system is a function of frequency and depends on the type of application and specific use of the system (with or without the helmet). Typical earmuff and earplug protectors provide some limited protection at low frequencies and the amount of protection increases with frequency. The minimum noise attenuation values by hearing protection tactical headsets recommended in a recent draft of the U.S. Army document are listed in Table 5-7.

The values listed in Table 5-7 reflect attenuation curves of typical HPDs. However, this curve is just the opposite to what may be required in most continuous vehicle, industrial, and environmental noises that typically have energy density distributions inversely proportional to frequency (1/f). This means that they are predominantly low frequency noises. In addition, good speech recognition requires good audibility of speech energy from 1000 Hz to 4000 Hz, which is usually significantly attenuated by most hearing protectors. Thus, in the applications that require live speech communication it makes sense to use audio HMD systems that offer noise attenuation that do not increase much with frequency. This philosophy is reflected, for example in the US. Air Force document MIL-PRF-89819/4 (DAF, 1997b) that specifies minimum attenuation for the in-flight headset-microphone to be approximately 20 dB for any frequency above 800 Hz and gradually lower attenuation with decreasing frequency due to the technical reasons.

It must be stressed that comfort and weight of an HMD (audio or otherwise) system are inversely related. In general, heavy HMD systems place more strain on the user's neck during movement or prolonged activity in any position. Thus, from a purely weight consideration, any lightweight in-the-ear or bone conduction devices may be favored over heavier and more bulky earmuff-type systems assuming they provide the same amount of hearing protection, speech intelligibility and situation awareness. However, the discomfort of having a device inserted in the ear canal or pressing against the head may be greater than the discomfort caused by the earmuff-type solutions. This stresses the need for considering the comfort of in-the-ear and bone conduction devices as a priority issue in designing such systems. Long-term comfort must be a primary consideration when designing any audio HMD, whether in-the-ear, bone conduction, or conventional systems. This calls for comfort evaluation of the devices by the users and response instruments (questionnaires, scales) that can provide a thorough feedback to the designers. An example of a scale that is used for comfort rating is the Wong-Baker FACES pain scale (Wong and Baker, 1988). This is a six step scale (from 0 to 5) illustrated by six faces expressing gradually increasing degree of pain from happy (no pain) to very unhappy (a lot of pain). This scale adapted for comfort rating is shown in Figure 5-49.

Table 5-7.
Noise attenuation criteria for hearing protection tactical headset.
(Modified from PM Soldier Warrior [2007]).

Octave band attenuation requirements [1] (dB)	**Frequency (Hz)**						
	125	250	500	1000	2000	4000	8000
Moderate noise level exposure [2]	15	16	21	27	28	35	35
High noise level exposure [3]	28	34	34	34	34	40	40

[1] *Levels should be determined following ANSI S12.42-2002 standard (ANSI, 2002) measurement procedure and subtracting 1 standard deviation*

[2] *Infantry Soldiers in wheeled tactical vehicles*

[3] *Infantry Soldiers in tracked tactical vehicles*

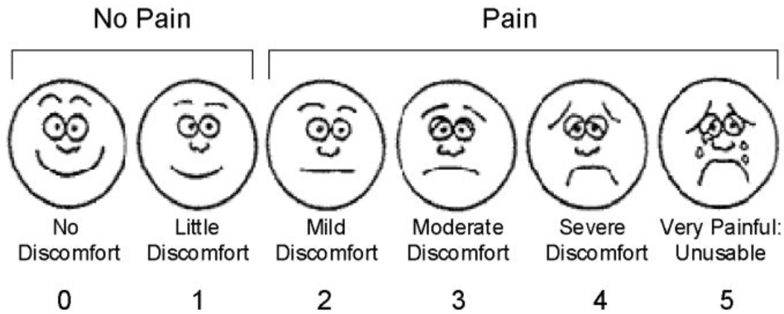


Figure 5-49. The Wong-Baker FACES pain scale adopted for comfort rating (Modified from: Hockenberry M, Wilson D, and Winkelstein ML: Wong's Essentials of Pediatric Nursing, ed. 7, St. Louis, 2005, p. 1259 (Copyright, Mosby) (used with permission).

The third important factor that needs to be considered in designing or selection of audio HMD systems is the durability of the system. Most COTS audio systems are not suitable for the harsh environments of military or firefighter operations. Audio HMD systems for the military must be sustainable in high impact, high temperature, and dusty environments. In some cases waterproof devices are required. For military applications under combat conditions, equipment should meet the requirements of MIL-STD-810F (DOD, 2001). This standard requires materiel to meet certain environmental design criteria and specifies tests and methods which replicate field conditions to verify compliance. This standard addresses and specifies minimum performance requirements for the following categories of environmental conditions: low pressure, high temperature, low temperature, temperature shock, contamination by fluids, solar radiation (sunshine), rain , humidity, fungus, salt, fog, sand dust, explosive atmosphere, immersion, acceleration, vibration, acoustic noise, shock, pyroshock, acidic atmosphere, gunfire vibration, temperature, humidity, vibration, and altitude , icing/freezing rain, ballistic shock, and vibro-acoustic and temperature conditions (DOD, 2001).

Speech intelligibility

Audio HMD systems are required to provide audio signals that result in auditory stimuli that are heard, recognized, and localized by the listener. The primary stimuli to consider are speech stimuli and its intelligibility. Speech intelligibility is defined as the percentage of speech units that can be correctly identified by an ideal listener over a given communication system in a given acoustic environment. If the properties of the listener, such as hearing loss or divided attention, are taken into consideration, it is more appropriate to refer to speech recognition rather than speech intelligibility.

Poor speech intelligibility increases task difficulty, compromises human performance, and may lead to loss of life (Peters and Garinther, 1990). The criteria for minimum required speech intelligibility in voice communication systems are stated in MIL-STD-1472F (DOD, 1999; Table VI) and are listed in Table 5-8.

The Modified Rhyme Test (MRT) criterion scores listed in Table 5-8 are the adjusted for guessing word recognition scores for the six-alternative MRT (House et al., 1965). The MRT is one of the three speech tests recommended for testing speech intelligibility in communication systems (ANSI, 1989).

The values listed in Table 5-8 are desirable goals and criteria for fielding live voice communication equipment and for natural person-to-person communication. However, the referenced standard does not specify the test conditions leaving some room for interpretation. More specific test conditions are included in the Communication Clarity Criteria being developed by the Program Manager Soldier Warrior office (PM Soldier Warrior, 2007) and shown in Table 5-9. Specified criteria that need to be met require performing the MRT as described in ANSI S3.2-1989 standard (ANSI, 1989), the talker to be in ambient noise environment of 75 dB SPL or more, and the listener in a pink noise environment with the overall sound pressure level as specified in Table 5-9. These criteria are based on the U.S. Air Force criteria for headset-microphone (DAF, 1997a). The difference between these two documents is that the Air Force document requires 85 % speech intelligibility at 105 dB SPL and 80% intelligibility at 115 dB SPL.

Table 5-8.
Intelligibility criteria for voice communication systems.

Communication Requirement	MRT* Score
Exceptionally high intelligibility; separate syllables understood	97 %
Normal acceptable intelligibility; approximately 98% of sentences correctly heard; single digits understood	91%
Minimally acceptable intelligibility; limited standardized phrases understood; approximately 90% sentences correctly heard (not acceptable for operational equipment)	75%

* Modified Rhyme Test

Table 5-9.
Communication clarity criteria for hearing protection tactical headsets.
(Program Manager - PM Soldier Warrior, 2007)

Sound pressure level of pink noise (dB SPL)	75	95	105
Minimum score percent correct (adjusted for guessing) in %	90	85	80

Relations between speech intelligibility scores (in %) and speech level (in dB A) for the communication system of the SPH-4B aviation helmet operating without and with the addition of CEP or Bose ANR system are shown in Figure 5-50. The data were obtained in UH-60 helicopter cabin noise of approximately 110 dB produced during flight at a forward speed of 120 knots (Mozo, 2001; Mozo and Murphy, 1997).

In practical situations, when worn equipment is being used in adverse listening conditions, the actual speech intelligibility is much worse than was expected. Therefore, it is critical to test speech intelligibility under the worst expected operation conditions (the worst case scenario) as well as under normal operational conditions. In addition, worn equipment does not perform as well as new equipment and should be tested periodically.

One important consideration in selecting an audio HMD system is the bandwidth of the radio communications channel that will be used to provide the data. Typical telecommunications and military radio systems are frequency band-limited to a pass-band from approximately 300 Hz to 3.4 kHz. With the introduction of digital telephony, based on the International Telecommunication Union standard G.711 (the standard for encoding telephone audio on a 64 kbps channel), the upper frequency limit of the telephone network is now commonly accepted to be approximately 3.3 kHz at best. The last Bell public switched telephone network tests in 1984 showed significant high-frequency roll-off at 3.2 kHz for short and medium distance connections, dropping to 2.7 kHz in long distance connections. The telephone network carries frequencies no lower than 220Hz, and most commonly the lower limit is 280 or 300 Hz. (Rodman, 2003). For this application an audio HMD with frequency response tailored to human voice data transmitted over a bandwidth limited channel will outperform an audio HMD with a frequency response which covers the full range of human hearing due to both noise and power constraints. Conversely, for applications where sound source localization cues are required and sufficient bandwidth is available to present needed high-frequency sound energy, the auditory performance of the listener would be hampered by a frequency-limited HMD (i.e., an HMD lacking a flat frequency response over the entire range of frequencies perceived by a human).

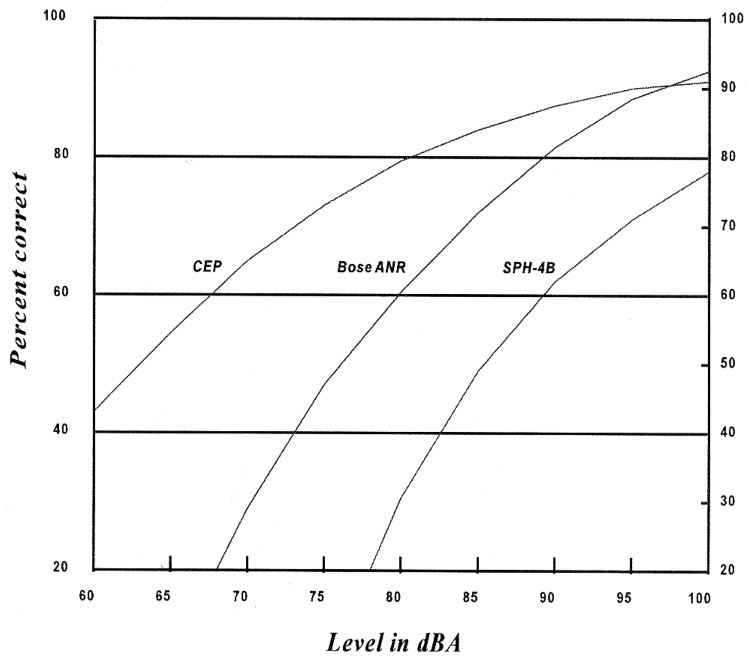


Figure 5-50. Radio communication speech intelligibility scores for SPH-4B aviation helmet without (SPH-4B) and with CEP (CAP) or Bose ANR (Bose ANR) systems as a function of speech level in UH-60 helicopter noise (Mozo, 2001) (Courtesy of USAARL).

Other primary factors affecting speech intelligibility are poor speech articulation by the talker and loss of signal intensity during speech transmission. The sound attenuation provided by the aircrew and tanker helmets and by other ear-encapsulating headgear greatly affects intelligibility of natural live speech. For example, Garinther and Hodge (1987) observed that the presence of the M25 respiratory mask and the NBC (nuclear-biological-chemical) protective hood restricted effective speech communication range to less than 12 meters. Conversely, the typical infantry helmets provide only minimal speech attenuation in the frequency range below 4.0 kHz (Randall and Holland, 1972), that is within the range that is responsible for providing more than 80% of speech intelligibility (ANSI, 1997b). The potential detrimental effect of an infantry helmet on speech communication is in providing false cues regarding the direction of incoming speech.

Audio and radio communication systems that provide good speech intelligibility have been reported to improve combat performance and decrease Warfighter's fatigue. Garinther and colleagues (Garinther, Whitaker and Peters 1994; Whitaker, Peters and Garinther 1989) reported that a specified percent of improvement in speech intelligibility provides an almost equal improvement in crew performance. The functional relationship between mission success and speech intelligibility is shown graphically in Figure 5-51.

Audio HMD Systems: Closing Remarks

The audio HMD system is a sub-system of a larger multi-functional display system of the helmet assembly. It is a challenging sub-system since it must provide hearing protection and auditory situation awareness in addition to an audio display. In addition, to operate properly as a part of the large system, the audio HMD must be physically and electrically compatible with the remainder of the HMD system and not interfere with the functioning of the non-audio system components of the HMD. This requires appropriate design considerations so as to provide an engineered solution suitable for the desired application. Issues such as comfort, power requirements, size, weight, location, desired sound pressure level, fidelity (bandwidth and dynamic range), number of audio channels, wiring, and connectivity must be considered together and from the perspective of optimizing auditory performance of the user awhile considering total system requirements and functionality. For example, high power and wide bandwidth audio signals may require larger and heavier transmitters that may not be feasible to be incorporated in the overall design.

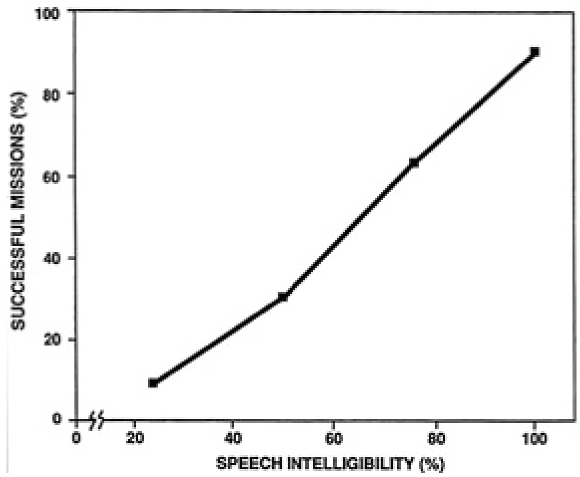


Figure 5-51. Percent of mission success as a function of speech intelligibility (Garinther, Whitaker and Peters, 1994).

Human hearing range extends from approximately 0 dB SPL to 120 dB SPL; therefore high quality output of an audio HMD may theoretically require a 120 dB dynamic range. This would be, for example, the ideal transmission range for high fidelity symphonic orchestra listening in ideal listening conditions. However, in most applications this wide intensity range is not necessary and may be dangerous. Prolonged listening to sounds (signal and/or noise) with intensity exceeding 85 dB SPL can be a source of hearing loss. For listening in quiet to normal verbal messages, the dynamic range of speech communications can be drastically limited since the effective dynamic range of speech is only approximately 50 dB. In the audio HMD systems operating in varying environmental conditions or used for environmental listening this range needs to be extended to accommodate various voice intensities from whisper to shouting and must allow hearing faint environmental sounds. Note that limited dynamic and frequency range of the transmitting channel also removes some contextual and environmental information and adversely affects transmission of emotions and physical state of the talker. However, for environmental listening it is also necessary to have an intensity limiter built into the system to protect the listener from dangerous high intensity sounds.

To protect the user from the harmful effects of high level environmental and military noises, the amount of hearing protection provided to the user by the audio HMD system must be carefully considered and integrated into the overall design from the beginning; it cannot be added as an afterthought. Natural speech communication and auditory awareness of the environment must be considered in parallel with the hearing protection system. Overprotection is actually worse than under protection since the user most-likely will defeat the protection or fail to use the system as it was intended. As discussed previously in this chapter, hearing protection can be provided in two primary forms, in-the-ear, or over-the-ear (circumaural earmuff). Adding in-the-ear hearing protection will reduce the efficiency of any external earphone-based audio systems and may make certain types of audio HMD systems unusable. Conversely, in-the-ear protection systems work well with bone conduction audio HMD systems. Circumaural hearing protection may be acceptable when used with both in-the-ear and bone conduction systems, but maximum efficiency is achieved when the audio HMD system and hearing protection are implemented as one fully integrated C&HPS. Circumaural audio HPDs are difficult to integrate into the overall helmet system because there is limited available space under the helmet in the vicinity of the ears. The decision as to which approach to take when designing audio HMDs - earmuff- or insert-type earphone, linear or non-linear, active or passive or active hearing protection, must be dictated by the mission which must be accomplished. There is no one-size fits all solution. In summary, audio HMD systems selected for a specific platform and operations must be tailored to their intended use, both operationally and environmentally. In addition, regardless of operational requirements of the system, it has to provide a long-term comfort for the user. An uncomfortable system will never be worn properly and used all the time when needed, thereby affecting users' mission effectiveness and safety.

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