CLARIFICATION OF SOME HEARING AID TERMINOLOGY: BANDS AND CHANNELS VERSUS HANDLES; FIRST-ORDER VERSUS SECOND-ORDER MICROPHONES

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A little history

The digital hearing aid was first introduced to the US in the late 1980s. Due, in part, to its large size and high power consumption, the first marketing attempt was not successful. In the mid-90s ear-level digital hearing aids made their way to the market and now account for nearly all hearing aids sold. In fact, many manufacturers have discontinued their analog products in favor of these higher-tech, easier-to-manufacture options.

How they work

The digital hearing aid performs its complex operations in one or more digital signal processors that are packaged in miniature integrated circuits (ICs). Sound received by a microphone is converted to analog electrical voltages and then converted to a digital format by analog-to-digital (A-to-D) converters. This process is reversed at the output when electrical voltages are directed to a transducer (usually called a receiver) to generate sound. The digital hearing aid is typically fitted by adjusting programmable microprocessors which control the hearing aid's smart internal micro-signal processors using "controls" or "handles" on the computer's graphic user interface (GUI) that is provided by each digital hearing aid manufacturer. Signal processing in a digital hearing aid is governed by *algorithms*, i.e., step-by-step series of instructions determined by sets of mathematical formulae or rules to perform a specific task. For example, the noise reduction algorithm may use a set of rules to judge if the input signal is speech or noise and react to it accordingly. Because most of the hearing aid's functions are controlled by numerical computations, the number of discrete components such as resistors and capacitors are minimal in today's digital hearing aids, thereby increasing its functionality and minimizing its size. With digital signal processing, the size of the microprocessor is much smaller than any analog circuit previously used to perform even part of the same task. In fact, many of the processing tasks we presently take for granted were not even possible with analog hearing aid circuitry. The digital hearing aid typically performs more sophisticated processing such as very steep filtering of signals, blocking loud sound impulses, and differentially controlling the dynamics of different parts of the audio spectrum. With appropriate designs, algorithms can filter digital signals, extract frequency content of those signals, obtain statistical information on the signals, and so on. This sophisticated filtering has benefited the hearing aid user in a number of ways, including significantly increasing control of acoustic feedback.

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Is it a band, a channel, or a handle?

Among the many advantages of the digital hearing aid is that of providing multiple bands, more than one channel, and the availability of handles for the clinician to use in the programming or fitting stage. Filters may be used to divide the frequency range into sections. A frequency *band* is a section, or region, of frequencies within which one may shape or adjust the gain of the hearing aid with linear processing independent of input level. However, if a signal processing algorithm is performed, such as compression or expansion, so that more than gain is adjusted within a frequency region, it is called a *channel*. A channel may include several or many bands. Such a group of contiguous bands is called a frequency channel when it is used for specific, non-linear processing, including control of gain or output sound pressure level (SPL) as a function of input level, thereby resulting in changes in compression and/or expansion. Each group of bands within a channel shares one or more digital algorithms. Within a single hearing aid there may be a different

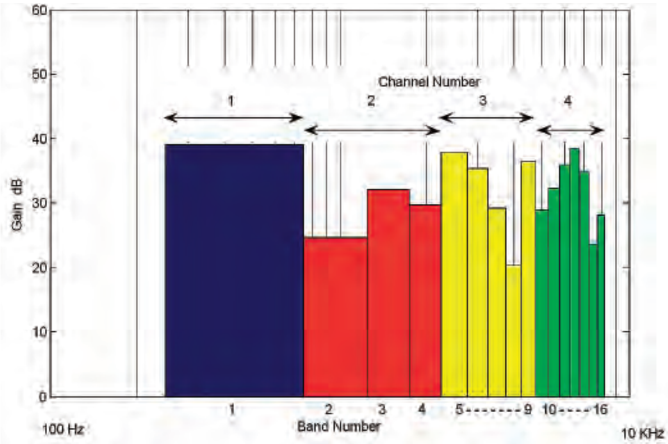

Fig. 1. A graphic representation of channels and bands within a digital hearing aid. (Courtesy of Steve Armstrong)

number of channels for each feature activated, such as noise reduction, adaptive directional microphones, and feedback management. A *handle* is a graphic representation of a control on the computer screen. It is provided in the manufacturer's software to alter a parameter of the hearing aid, e.g., by sliding the control handle up or down. This GUI is functional in a Windows environment. The use of handles is common among different manufacturers to permit the clinician to change the magnitudes and/or ranges of more than one band or channel at a time in a consistent manner. The interface allows the clinician to choose to make either more global or finer, more detailed, adjustments. The hearing aid depicted in Fig. 1, for example, has 16 bands (for gain shaping across frequency) and four channels for specific non-linear signal processing. The manufacturer may also choose to use four handles so that the clinician can adjust the response for the contiguous bands in one channel at a time.

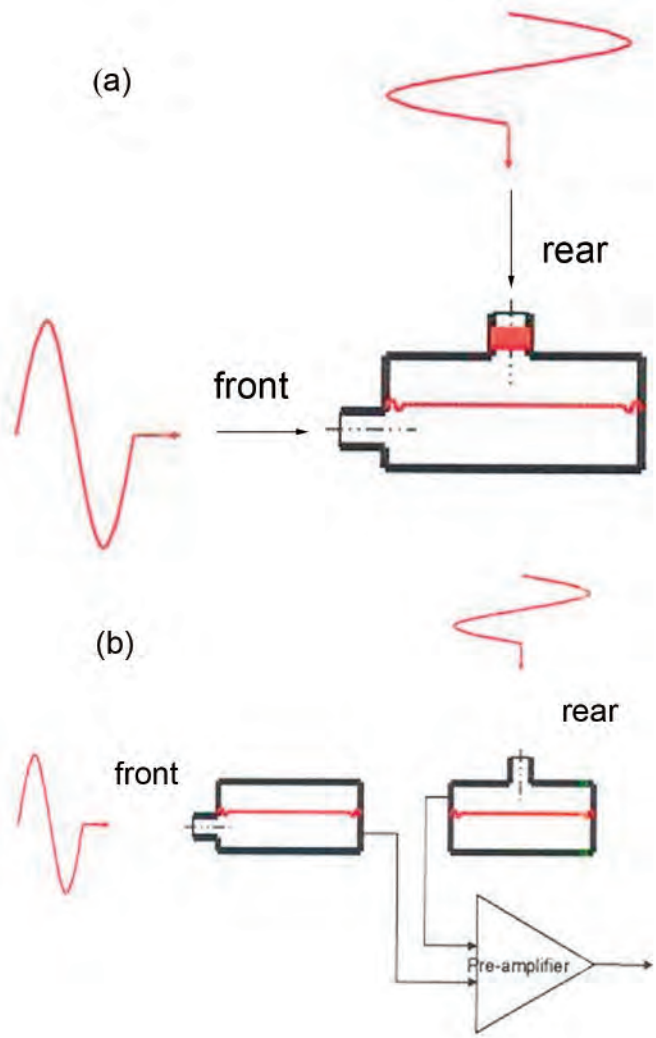

Fig. 2. (a) Shows a single-element directional microphone scheme. (b) Shows a two-element scheme.

Re-introduction of directional microphones

Although directional microphones have been used for many years in the electronic news gathering and recording industries, they first became popular in hearing aids in the mid-1970s. For a number of reasons, including the introduction of custom in-the-ear styles of hearing aids, their popularity waned until the late 1990s. The rationale for using directional microphone systems in hearing aid design is based on different spatial properties of speech and noise. Typically, speech comes from a listener's front while competing noises come from the front and other directions other than the front. Directional technology has the potential to improve speech intelligibility in noise because it can maintain sensitivity to sounds originating in the front of the listener, while reducing sensitivity to sounds at other azimuths.

Order of directionality

Another descriptor used in directional microphone design is the *order* of directionality. There are two commonly used schemes in hearing aids today as shown in Fig. 2: a single microphone element (capsule) with two ports and two microphone elements (capsules–both omnidirectional microphones) that are connected electrically. In each case, the designs are *first-order* directional systems; that is, two ports are used for sound entry to the diaphragm and the sounds from different directions are either reinforced or reduced based upon the slight differences in arrival time at each of the two sound entries (ports). A *second-order* directional system can be achieved by adding one more omnidirectional microphone, in effect one more port, at a carefully defined spacing from the first port, sometimes mathematically calculating the necessary time delay. In effect, any *higher order* directional system can be labeled by the "number of ports minus 1" rule. An omnidirectional microphone is technically a zeroth-order microphone. A first-order directional microphone system with two microphone ports typically has a free field directivity index of 6 dB or less. A second-order directional microphone system with three microphone ports typically has a free field directivity index of 6 to 9.5 dB. The recently proposed revision of Annex B of ANSI S3.35 (*Method of Measurement of Performance Characteristics of Hearing Aids Under Simulated Real Ear Working Conditions*) contains definitions for first- and second-order directional systems. The technical definition of

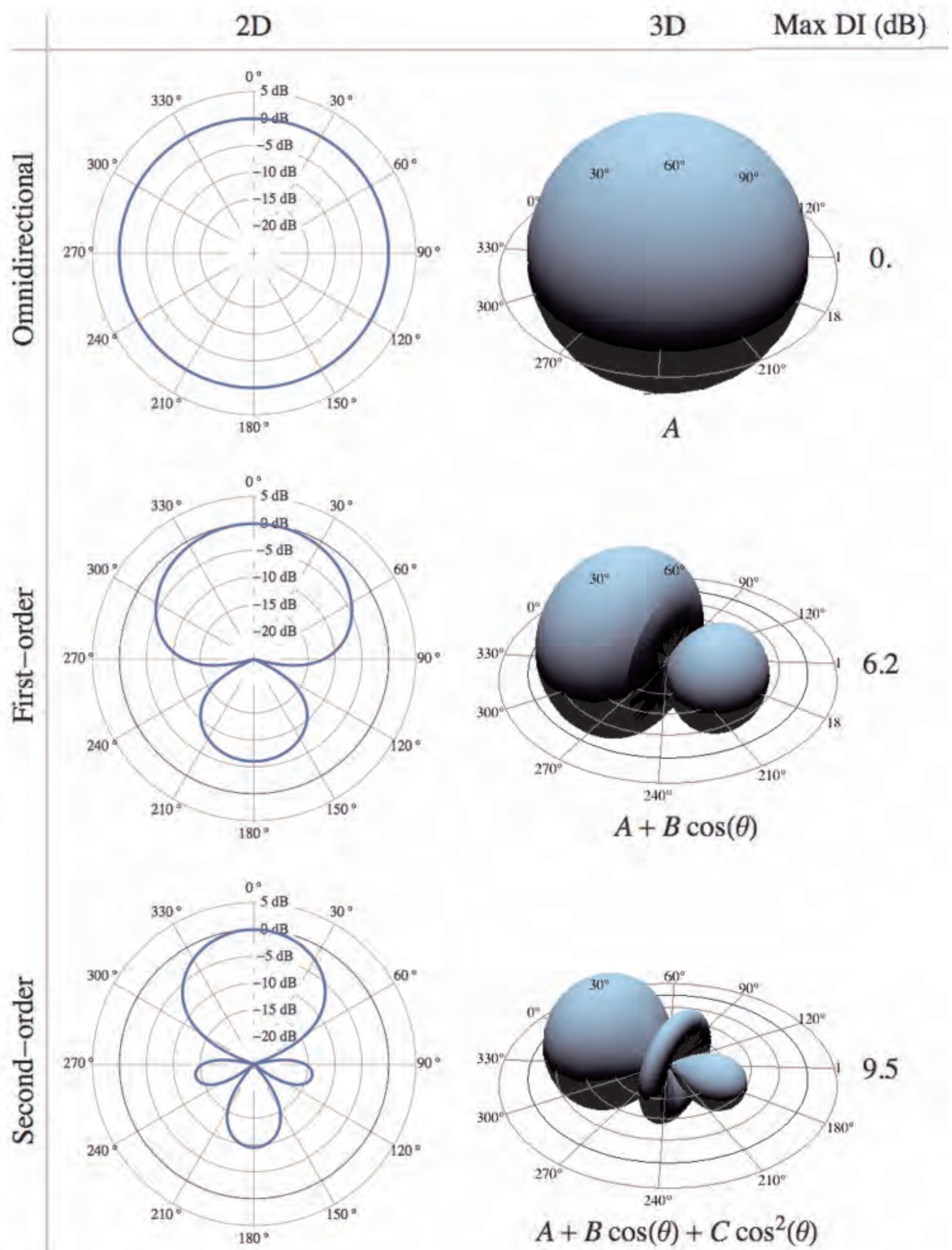

Fig. 3. Graphic representation of microphone order. Polar patterns are shown in two and three dimensions with accompanying limacon equations and directivity indices (reprinted with permission from Daniel Warren).

the *order* of the microphone is the power of the cosine term in the limacon equation for directionality, as shown in Fig. 3. While a higher order directional microphone may be capable of more directivity (narrower beam), the problems associated with microphone matching, low-frequency roll-off, internal noise are all magnified with increasing microphone (or port) numbers.

The directional systems on the market can be one of the following or some combination thereof: *fixed, automatic,* and *adaptive*. In a fixed system, the directional pattern (polar response) has a fixed internal delay and does not change. In the automatic system, an algorithm detects and analyzes acoustic characteristics of environments, and then switches automatically to the appropriate microphone mode (directional or omnidirectional) according to predefined decision rules. Adaptive refers to the ability of the directional system to change the directional pattern in different noise configurations based on measurements made (sampled) in the changing environments and adjusted based on rules programmed in its signal processor. Specifically, the system can "steer" the nulls and/or *look direction* of the directional pattern to the appropriate azimuths (e.g., the azimuth of the noise source) and optimize signal-to-noise (SNR) improvement. This adaptivity can be realized independently in different frequency channels in a multi-channel adaptive directional hearing aid.

In actuality, it is difficult to specify the real-life directional response of hearing aids with simulated real-ear conditions and especially difficult to define with measurements performed in one plane. With the digital hearing aid in position on or in the ear, the directional response, which is often described by its polar pattern, is changed by the head and by the objects nearby and the degree of reverberation in the environment. Hearing aid manufacturers, through the standardization process, develop and confirm methods to describe the spatial properties of the sound field with different types of directional microphone systems (ANSI S3.35) and report the performance to the clinicians.

Conclusion

The intent of this brief article is to clarify the terminology that is often used by both industry and clinicians to describe some of the current hearing aid technologies. Many of these terms are also incorporated into standards. As the technology evolves, so will new descriptors, and it is important that both entities understand and use the terminology in an unambiguous manner.

Ruth A. Bentler obtained her Ph.D. in Speech and Hearing Sciences from the University of Iowa in 1987, where she is now a Professor of Audiology. Dr. Bentler is a Fellow of the American Speech-Language-Hearing Association and the American Academy of Audiology, as well as a member of the Acoustical Society of America and the International Society of Audiology. She is Director of the Doctor of Audiology (Au.D.) Studies at the University of Iowa where she teaches graduate students in courses related to hearing aids and adult auditory rehabilitation. As the Director of the Hearing Aid Laboratory for Basic and Applied Research within the Department of Communication Sciences and Disorders, Dr. Bentler has been involved in numerous research endeavors involving directional microphones, digital noise reduction, and, most currently, frequency-lowering algorithms. She has authored over 85 articles and chapters related to hearing aid technology and fitting practices. Outside of the academic setting, she is the Global Director of the Hearing Aid Program for Special Olympics International, and the Co-Director of the Iowa China Project. Her current involvement with Acoustical Society of America is as a member of the working groups on hearing aid standards (S3/WG48) and probe-tube measures of hearing aid performance (S3/WG80).

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