INTRODUCTION

Signal processing, as it applies to hearing aids, is a term that has many connotations. In the broadest sense, any change made to an input signal by a circuit can be considered signal processing. Thus, even the simplest hearing aid performs some degree of processing by amplifying and frequency-shaping an input signal. Generally however, hearing aid signal processing usually refers to somewhat more complex signal modification intended to enhance, in a defined, predictable manner, the output signal relative to the input signal.

Signal processing can be beneficial or it can be detrimental, depending upon its intended function and the resulting outcome. Peak clipping is an example of a simple form of signal processing. A peak-clipping circuit clearly modifies the input signal in a well-defined, predictable manner by limiting the amplitude of signals that exceed a certain level and thereby maintaining the output at a specified level. This processing can be viewed as beneficial if looked at from the design goal of preventing the hearing aid output from reaching sound pressure levels that exceed a user’s comfort level (UCL). On the other hand, peak clipping can be considered detrimental in view of its effect upon the frequency content and sound quality of the output signal. Peak clipping results in severe distortion and significantly degrades sound quality. This example illustrates that signal processing can be viewed in different ways, and that in and of itself, signal processing is neither inherently good nor bad, but must be examined in the context of its intended application, how well it achieves the desired goal, and the consequences of the processing on various aspects of the output signal.

GOALS OF SIGNAL PROCESSING

Hearing aid signal processing can generally be categorized into two very broad areas of intended application. The first includes those circuits designed to alter the input signal in some manner to better fit the hearing impairment of the user. In other words, their intent is to compensate for the hearing loss. Linear amplification with a prescriptive frequency response would represent a very simple example in this category. A more complex example is a compression circuit designed to transform the wide range of sounds in the environment into a narrower, restricted range of output levels to fit within a reduced dynamic range.
The second category includes circuits designed to enhance the signal in some manner, based upon the properties of the input signal itself. In this case, the processing is intended to somehow improve the signal, thereby making it more intelligible or more pleasant. Almost all forms of noise reduction fall into this category, since their primary design goal is to modify the input signal, based upon its frequency, intensity, or temporal characteristics, in such a way that the output contains less noise and more signal. Although some processing in this category may yield an improved fit to the hearing impairment, this is typically a secondary effect. An example of this would be traditional ASP (automatic signal processing) hearing aids that use an adaptive filter to reduce low frequency output when input levels exceed a defined threshold. The original design intent of such circuits was to reduce the amplification of background noise dominated by low frequency, and thereby reduce the incidence of the upward spread of masking (i.e., the loss of audibility of high frequency sounds caused by louder low frequency sounds).

Within both application categories exists a range of signal processing complexity, from the simple examples listed, to very complex implementations under development, such as systems using multiple microphones, and digital signal processors incorporating sophisticated noise reduction algorithms.

**CHARACTERIZING SIGNAL PROCESSING**

Signal processing circuits are usually characterized in terms of their effects on three primary signal domains: the frequency domain, the intensity domain, and the time domain. Although each is identified individually, the three represent inseparable attributes of any analog signal, such as speech. For example, speech can be characterized in the time-intensity domain by a waveform like that shown in Figure 1, where the top graph represents a plot of sound intensity versus time. The various individual speech components in this utterance of the word “shoot” are indicated below the corresponding section of the waveform. In this example, the vowel portion (“oo”) has a greater intensity than the consonants (“sh” and “t”). In the frequency domain, the speech signal can be described by the relationship between frequency and intensity of the long-term average of speech, known as the speech spectrum (Figure 1, bottom). As in the short speech sample, the long-term speech spectrum has the highest intensity in the low frequency vowel region and less energy in the high frequency consonant region. To assess the effects of hearing aid signal processing on speech or any other signal, the input and output signals can be compared in these different domains. The differences between the output and the input are the result of the signal processing.

**CLASSIFYING SIGNAL PROCESSING**

Recently, several authors have summarized and created a hierarchy of the different types of signal processing currently available, particularly in programmable hearing aids (1,2). For the most part, these categorizations are in general agreement. The classification scheme presented below will ignore for the moment the aspect of programmability and the potential for multiple-response capabilities, which represents a further point of differentiation, and focus specifically on analog signal processing.

**Number of Channels**

A main point of differentiation between circuits is the number of channels. Instruments with multiple chan-
nels contain filters that separate the input signal into frequency bands that can then be processed independently. These instruments generally offer the ability to more precisely shape the response to match the hearing loss.

**Number of Channels of Compression**

Generally viewed as more important than the number of channels is the number of channels of compression within an instrument. Single- and multiple-channel hearing instruments may contain zero, one, or several compression circuits. For either instrument, linear amplification combined with peak-clipping circuits to limit output represents the simplest form of signal processing.

Single-channel compression instruments can generally be classified into one of three categories based upon the frequency region that the compression circuit controls (Figure 2). Compression systems that operate primarily in the low frequency region produce bass increase at low levels (BILL) processing (3). These systems reduce low frequency gain in response to higher input levels in that region (note: some BILL devices do not use compression circuits but accomplish a similar effect through the use of active filters that change the response slope in the low frequencies). High frequency compression systems provide treble increase at low levels (TILL), reducing high frequency gain in response to louder inputs and increase high frequency gain as input levels decrease. Single-channel compression systems that control overall gain are generally known as automatic gain control (AGC) circuits; these vary the overall gain across the full frequency range of the instrument, much like a volume control, rather than having a frequency-specific effect.

Multichannel instruments may or may not contain compression circuits; some use simple linear amplifiers and peak-clipping circuits. Those with compression may have: (a) one compression circuit that controls the overall gain of the instrument; (b) a compression circuit in one channel and peak clipping in others; (c) individual compression circuits within each channel; or (d) a combination of the above. In the signal processing hierarchy, instruments that utilize independent compression circuits within each channel are generally considered the most advanced. This design, known as multichannel compression, allows the gain of each channel to be controlled by inputs within each channel’s bandwidth. Multichannel compression instruments provide the most flexibility, and can adapt their signal processing in response to changes in input level across the frequency spectrum. Unlike single-channel systems, limited to one type of processing, some multichannel compression in-

![Figure 2](image-url)
Instruments can produce BILL, TILL, or overall AGC effects in response to different types of input signals (low frequency, high frequency, or broadband), providing variations of the three different response patterns illustrated in Figure 2, depending upon the intensity and frequency content of the input signal.

**TYPE OF COMPRESSION CIRCUIT**

**Compression Limiting**

The compression circuit within an instrument also represents a point of differentiation. Circuits with high compression ratios (5:1 or greater) and high thresholds of compression (65 dB SPL or greater) are known as compression limiters, and are designed to prevent output from exceeding a predetermined level to avoid circuit overload and user discomfort (Figure 3). Compression limiters perform essentially the same function as peak clippers: they limit the hearing aid output at a specified level, but generally do so with significantly lower distortion levels.

**Wide Range Compression**

Other circuits designed to compensate for the reduced dynamic range associated with hearing impairment typically use lower compression ratios and thresholds (Figure 3). These circuits operate over a wider range of input levels and preserve relative intensity information of speech and other inputs, compared to compression limiting systems that severely degrade intensity information above the compression threshold. Some instruments also combine different compression circuits within the same hearing aid. Typically, these consist of a combination of one or more wide-range compression circuits paired with a compression limiting circuit that activates at a much higher level and functions to limit output.

**Variable Release Times**

Some circuits also employ a variable release time feature. Associated primarily with single-channel compression systems, variable release time circuits adapt the compressor release time, based upon the duration of the input that triggers the compression circuit: longer release times are used for longer duration inputs and shorter release times are used for more transient inputs. This system is typically not required with multichannel compression systems where the release time for each channel is set for the appropriate duration of signals within the channel’s bandwidth.

### Classifying Hearing Aid Signal Processing

<table>
<thead>
<tr>
<th>Number of channels</th>
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<tbody>
<tr>
<td>Single channel</td>
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<tr>
<td>Multiple channel</td>
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<table>
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<tr>
<th>Number of channels of compression</th>
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<tr>
<td>0 — peak clipping</td>
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<tr>
<td>1 — Single Channel: BILL, TILL, AGC, variable release time</td>
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<td>2 or more — Multichannel compression</td>
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<th>Type of compression circuit</th>
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<tr>
<td>Input/Output</td>
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<tr>
<td>Compression limiting</td>
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<td>Wide range compression</td>
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### Input/Output Compression

For hearing aids with a volume control, the compression system can also be categorized as either input or output compression. Very simply, with an input compression system, the volume control affects both the gain and the maximum output of the hearing aid, providing the user with control over the level of both soft and loud sounds. With an output compression circuit, the maximum output of the aid is fixed and independent of the volume control, which affects the gain and the compression threshold. Because the user does not have control of maximum output of the aid with this system, an appropriate output setting is critical to successful use. Typically, output compression systems tend to be associated

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*Figure 3.* Compression input-output characteristics. Representative illustrations of input-output curves for different compression circuits.
with compression limiting circuits. Wide range compression circuits are more often configured as input compression systems.

**CHARACTERIZING ALTERED SIGNAL CONDITIONS**

Which one of the various types currently available in hearing aids is the most appropriate under “altered” signal conditions? Assuming that an appropriate fitting exists for “average” conditions (conversational speech in quiet), one approach to answer this question is to determine how the altered conditions differ from the average condition, and attempt to create signal processing to compensate for the differences. For some conditions, a simple change in frequency response of a linear instrument may be sufficient; for others, the use of more complex multichannel compression may be required to best optimize performance.

Fittings designed for altered listening conditions, or specific listening environments, are generally associated with the use of multiple-response instruments (i.e., multiple-memory programmables). With single-response instruments, it is unlikely that the fitting will be optimized for the characteristics of one specific listening situation, but more likely to be fit for average conditions. In attempting to determine the most appropriate signal processing for specific conditions there are a number of characteristics of listening situations that should be defined including: 1) the signal of interest; 2) the background noise conditions; and 3) the acoustic nature of the listening environment.

**Signal Of Interest**

The primary consideration in creating a fitting for any specific listening situation is the signal of interest and how it differs from the conversational speech signal for which the baseline fitting was optimized. The altered signal of interest can then be compared in the different domains outlined earlier to determine how it differs from the average speech signal. For example, are the differences primarily in the frequency content or in the intensity domain? By defining these basic differences, the appropriate signal processing modifications may become more evident.

**Background Noise**

Similarly, the level and composition of any background noise in the altered conditions that will compete

with the signal of interest need to be defined. Here again, the characteristics of the background noise: its frequency content (low frequency versus broadband); its intensity range, and its temporal nature (constant versus intermittent) will all factor into the signal processing modifications that may be appropriate.

**Listening Environment**

Finally, the acoustic nature and specific conditions of the listening environment also require consideration. For example, situations with multiple speakers located at various distances from the listener, as in a large meeting, can be problematic for many users due to large differences in the vocal levels of the various speakers. Is the signal of interest presented live or via a sound reproduction system, the characteristics of which may alter the signal? Are the surroundings reverberant? These factors will also influence the signal processing required to optimize performance.

**Characterizing “Altered” Listening Conditions**

- **Signal of Interest**
  - Frequency composition
  - Intensity levels

- **Background Noise**
  - Frequency composition
  - Intensity levels
  - Temporal characteristics

- **Listening Environment**
  - Signal source(s)
  - Acoustics (reverberance)

**ALTERED SPEECH SIGNALS**

Everyday speech covers a wide range of frequencies and intensity levels. Figure 4 illustrates a series of long-term average spectra produced at different vocal efforts (4). In addition to changes in overall intensity, there are also changes in the shape of the spectrum, and in the relative levels of the low and high frequency components.

**Whispered Speech**

The waveform and spectrum of whispered speech is considerably different from that of typical voiced speech, as shown in Figure 5. The top panel shows two waveforms of the word “shoot,” one spoken normally (upper) and the second whispered (lower). In whispered speech, all components are essentially unvoiced. Therefore, the
dominant low frequency portion of the spectrum, made up of harmonics of vocal cord vibrations and vocal tract resonances associated with vowels, is considerably reduced. This is illustrated in the bottom panel, which shows a comparison of the spectrum of voiced speech versus whispered speech.

Modifying a hearing aid to listen specifically to whispered speech would primarily involve altering the shape of the frequency response to accommodate the altered speech spectrum, and providing adequate gain to ensure audibility of the reduced input levels. A flatter response, with more mid- and low-frequency gain may help to emphasize the reduced mid-range spectrum in whispered speech. Appropriate signal processing for this type of listening situation would include TILL, which provides maximum gain for soft, high frequency sounds, or wide range, multichannel compression, which would boost softer inputs. Even a linear, peak-clipping circuit, with an appropriate frequency response, would likely suffice under this listening condition, as long as input levels remain low, and the user has access to a volume control.

Loud Speech

At the other end of the speech intensity range is shouting, or loud speech, which has a slightly different spectrum from casual speech (Figure 4). For loud speech, the peak of the spectrum shifts upward, above 1,000 Hz, and the roll-off between the peak and the lower intensity, high frequency portion of the spectrum is steeper. During these intense vocal efforts, the vocal cords open and close more rapidly and remain open for a shorter period of time, which increases the amplitude of the higher harmonics. For most individuals, less gain would be needed for this signal, since the level is significantly elevated compared to average conditions. Also, because the low and mid-frequencies contribute the major portion of the loudness of speech, a frequency response with reduced low and mid-frequency gain may be more appropriate. BILL processing, single-channel AGC, or multichannel compression would be appropriate processing to deal with the elevated low and mid-frequency input levels to maintain user comfort. Peak-clipping circuits may exhibit severe distortion under these conditions and, therefore, compression limiting would be the preferable form of output limiting.

For a multichannel compression instrument, where the crossover frequency between channels may be near 1,000 Hz, the change in the loud speech spectrum may
actually shift the peak energy from one channel into another. Because the loudest signal within a channel’s bandwidth controls the gain of a compression amplifier, this shift of spectral peak may actually result in a large reduction in gain in the high frequency channel. One possible modification for a multichannel compression aid for this type of input signal would be to shift the crossover frequency upward, to maintain the peak of the spectrum within the low frequency channel, thereby minimizing its effect on gain for softer high frequency components.

COMMUNICATION SYSTEMS

Speech that has been preprocessed through another communication device, such as a telephone or intercom, will have a different frequency spectrum from that shown in Figure 1, owing to the transmission characteristics of the device. In addition to the system’s frequency response, some of these devices contain built-in AGC circuits, which further alter the normal speech signal.

Telephone

The telephone, for example, has a bandpass characteristic that is reasonably flat from approximately 300 to 3,000 Hz. Below and above those frequencies, however, the spectrum rolls off quite dramatically. There are two main considerations for acoustically coupling a hearing aid to a telephone handset: (a) minimize feedback, and (b) compensate for the altered speech spectrum. The first goal can usually be accomplished by significantly reducing the gain above 3 kHz through frequency-shaping modifications, such as a high cut trimmer or programmable gain. Because of the limited bandwidth of the telephone system, reducing the gain in this frequency region does not alter the speech signal, yet does significantly reduce the likelihood of feedback. More emphasis can be provided to the mid-frequency region for this type of fitting, to maximize audibility of the speech information in that range. Typically, most telephone speech occurs over a fairly restricted intensity range, and signal-processing considerations are, therefore, of secondary importance compared to providing an appropriate frequency response with adequate gain and minimal feedback.

For magnetic telecoil systems, response modifications are fairly limited in most instruments; however, some programmables do allow the frequency response of the telecoil system to be modified independently of that of the microphone (5). There is recent evidence that shaping the telecoil frequency response to provide real ear gain that matches a speech-based prescriptive target, such as NAL, may improve performance (6). Use of a telecoil has become more difficult in many of today’s office environments, where high levels of electrical interference from fluorescent lighting, computer terminals, or other electronic equipment exist. These devices radiate energy at harmonics of the frequency of the power supply circuit (60 Hz). By reducing the gain of the telecoil in the frequency region of the higher level harmonics (below 300 Hz), this electrical background noise can be minimized. Since the telephone signal rolls off below 300 Hz, there is little information lost through the modification.

Intercoms

For intercoms and other speaker systems, the frequency response, as well as its overall level, will dictate the type of fitting modification required. Most intercom systems do not use particularly high quality components; therefore, the output signal may be significantly degraded to begin with. In general, the frequency response of the system is the major consideration, as opposed to intensity concerns, since here again, the output range for most of these systems is fairly restricted. Typically, they contain relatively small speakers, which reproduce mid- and high frequencies more efficiently than low frequencies. The spectrum of the signal of interest then, will likely be more mid- and high-frequency biased than live speech. To compensate for this variation in input signal, more low frequency gain can be provided in the hearing aid response to improve the sound quality. In many cases, intelligibility may be determined more by the characteristics of the output device, rather than by the processing of the hearing aid.

SPEECH COMBINED WITH OTHER SIGNALS

Because of the importance of communication in everyday life, speech remains the primary signal of interest in the majority of typical listening situations. There are times, however, when nonspeech signals are the point of interest, such as listening to music, or other situations, often entertainment-related, where speech may be combined with other signals, such as music or sound effects.
Television

Listening to television is often one of the first situations where hearing loss becomes a noticeable problem. Because television programming contains speech at a variety of levels, as well as music, audience sound tracks, and any number of other sounds, the viewer is exposed to dynamically changing intensity levels. From the softest speech sounds of a quiet conversation to a loud trumpet fanfare introducing the newest variety of breakfast cereal, the individual with hearing loss, aided or unaided, is faced with a challenging listening situation. Unaided, most raise the television volume to the level needed to hear the softest speech segments. The much louder commercials and music may then be too loud, due to the reduced dynamic range associated with many types of hearing loss. For a hearing aid wearer, particularly if using a linear circuit, a similar outcome may result (the television volume control is simply a linear amplifier). By turning up the gain of the instrument to hear the soft speech, too much gain may be provided for louder sounds, or distortion from peak clipping may occur, either one an uncomfortable listening experience.

On television there may be multiple signals of interest, although speech most likely remains the primary one. The speakers used in most sets are often of fairly low quality, although that aspect has improved recently with the advent of stereo broadcasts. In general, however, television speakers are smaller in size, which limits their ability to reproduce low frequency sound efficiently. From a frequency response point of view, a hearing aid fitting with additional low frequency gain will produce a more pleasant sound quality for listening to television, but the primary concern of a fitting specific to this situation is the wide range of intensities to which the listener is exposed. This artificially manipulated range may, in fact, be greater over a relatively short period of time than one might otherwise experience during a typical day away from the TV set. Because of this, the use of some type of compression circuit will generally provide the most comfortable listening experience for most users. The more adaptive multichannel compression processing, which can respond appropriately to the broad spectral and intensity changes, may provide the best performance in this situation. Single-channel devices with compression-limiting circuits would also provide an improvement over peak-clipping devices, in terms of sound quality. For this situation, the issue of intensity dynamics generally outweighs the consideration of frequency response.

Theaters and Auditoriums

Much of the above discussion related to television is germane to considerations for movie theaters, live theaters, and other large listening environments. In these situations, consideration of intensity dynamics remains the primary focus over frequency content issues. In fact, most theaters have relatively high quality sound reproduction systems (capable of producing high output levels); therefore, the frequency response of the output source is generally not a limiting factor. However, in these larger venues room acoustics may become more of a factor for consideration. Places of worship, auditoriums, and the like, with bright, solid, reflective surfaces often tend to create reverberant listening conditions. Reverberation of low frequencies degrades the temporal envelope of speech, masking many lower intensity speech cues. In such a situation, a multichannel compression system, where low frequency speech components are processed separately from high frequency components, may prove advantageous. By separating speech information into multiple bands, appropriate gain can be provided for softer high frequency components, independent of reflected, low frequency energy. Single-channel BILL processing would also serve to minimize low frequency gain in these types of conditions, and would likely provide an improvement over linear processing.

Music

Listening to live or recorded music and to many of the situations discussed above where music may be part of the overall content, is a different listening experience from most others. Certainly, the frequency content of music can differ quite significantly from that of speech. Similarly, the intensity range of music is also quite likely to exceed that of speech. In addition, the receptive attitude of the listener may also constitute a major factor in fitting for this listening situation. Many individuals listen to music at relatively loud levels (some to the extent of being dangerously loud) compared to other types of materials. In terms of absolute intensity, these levels may be far beyond what the same individual would tolerate for any period of time for listening to speech, or to a baby crying, or to nails on a chalkboard. These examples illustrate that the nature of the material, rather than simply the absolute levels alone, exerts a major influence on the acceptability of intensity levels for specific stimuli. Bentler observed exactly this phenomena in a study that examined the relationship of stimulus material to the reported UCL (7). Her data showed a clear trend toward lower re-
ported UCLs for stimuli typically considered to be aversi-
ve, compared to those considered pleasant.

Thus, in addition to frequency content and dynamic con-
sideRations, the nature of the signal of interest, in the
case of music, must also be accounted for. In general,
most listeners will tolerate, and may in fact prefer, louder
levels for music than for other materials. Hearing aid fit-
tings for listening to music should address all of these is-
ues. Compared to listening to speech, the preferred
frequency response for music will generally have more
low frequency gain, and a somewhat flatter response (8).
This setting will provide a fuller, richer sound quality and
provide emphasis to the bass components, compared to
the typical high frequency emphasis desired for speech,
where priority is placed on intelligibility. In a study of
user preferences of different frequency responses for vari-
ous listening materials, Fabry and Stypulkowski found
that most listeners tended to select the greatest amount of
low frequency gain for listening to music, compared to
listening to speech in quiet or noise (9).

The intensity range of music is also often greater
than that of everyday speech, suggesting that compres-
sion circuits may be appropriate, certainly over the use of
peak-clipping circuits, since sound quality is a major fac-
tor when listening to music. Multichannel, wide-range
compression may improve sound quality and naturalness
in perceived intensity levels, particularly if the dynamic
range is significantly different in the low and high fre-
quency regions. Because of the nature of the material,
output levels may be set higher for this specific listening
condition than for others, as described above. Input com-
pression circuits, which provide the user with control of
both gain and output levels, would be more appropriate
here than would output compression circuits, where the
user would be unable to control the loudness of higher in-
tensity sounds.

In general, as the relative spectral differences be-
tween the low and high frequencies of an input signal be-
come greater, multichannel compression offers a greater
potential to improve signal processing performance. Sim-
ilarly, as differences in the user’s dynamic range become
greater in different frequency regions, multichannel pro-
cessing may also prove more beneficial.

MULTIPLE-RESPONSE INSTRUMENTS

As outlined earlier, the use of very specific fittings
or signal processing for individual listening situations is
likely to involve the use of multiple-response instru-
ments. Creating different fittings for specific listening
conditions is a luxury that is really only available with
such programmable instruments. The decision as to
whether to prescribe one can be based to a great extent
upon user lifestyle and exposure to distinctly different
listening situations. Many of the situations described in
this and earlier chapters represent listening environments
where multiple-response instruments can be beneficial
when appropriately fit to optimize performance for the
specific conditions. A number of recent studies have
shown that individuals will consistently and effectively
utilize a number of different programs of a multiple-
memory instrument in different listening situations
(10–13). For example, Ringdahl reported that most sub-
jects used 2–5 different programs of a multiple memory,
multichannel compression instrument in their daily use,
and consistently selected specific programs for specific
listening conditions (14).

It should be noted that just as with single-response
hearing aids, multiple-response instruments span a com-
plete range of signal-processing capabilities. Some multi-
ple-response instruments contain single-channel, linear,
peak-clipping circuits. This system primarily allows dif-
fences in frequency response to be created between the
different programs. At the other end of the spectrum are
multiple-response, multiple-channel compression instru-
ments that not only allow changes in the frequency re-
sponse of different programs, but also in the signal
processing that can be created for different listening sit-
uations (15). Instruments with this capability offer the
greatest flexibility, and the greatest ability to create dra-
matic differences in program responses within the same
hearing instrument.

Successful use of multiple-response instruments is
not dictated by audiogram type, prior hearing aid experi-
ence, or age of the user. Studies have reported successful
use of them across a wide population of users with di-
verse audiometric profiles and history of use, including a
large population of children (10,14). In general, individ-
uals with hearing losses on the extremes of the distribu-
tion (i.e., very mild or very severe) may not benefit as
much from the use of multiple memories due to limita-
tions in frequency response or signal processing differ-
ence that can be provided between programs for these
types of losses. Most users, however, can benefit from
the availability of one or two additional programs de-
dsigned for specific listening circumstances in their life.
Experienced hearing aid users, familiar with the situa-
tions that present listening difficulties for them, are often
the most successful users of multiple-response instru-
ments.
REFERENCES


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