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Effects of Bandwidth, Compression Speed, and Gain at High Frequencies on Preferences for Amplified Music

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Abstract

This article reviews a series of studies on the factors influencing sound quality preferences, mostly for jazz and classical music stimuli. The data were obtained using ratings of individual stimuli or using the method of paired comparisons. For normal-hearing participants, the highest ratings of sound quality were obtained when the reproduction bandwidth was wide (55 to 16000 Hz) and ripples in the frequency response were small (less than ± 5 dB). For hearing-impaired participants listening via a simulated five-channel compression hearing aid with gains set using the CAM2 fitting method, preferences for upper cutoff frequency varied across participants: Some preferred a 7.5- or 10-kHz upper cutoff frequency over a 5-kHz cutoff frequency, and some showed the opposite preference. Preferences for a higher upper cutoff frequency were associated with a shallow high-frequency slope of the audiogram. A subsequent study comparing the CAM2 and NAL-NL2 fitting methods, with gains slightly reduced for participants who were not experienced hearing aid users, showed a consistent preference for CAM2. Since the two methods differ mainly in the gain applied for frequencies above 4 kHz (CAM2 recommending higher gain than NAL-NL2), these results suggest that extending the upper cutoff frequency is beneficial. A system for reducing "overshoot" effects produced by compression gave small but significant benefits for sound quality of a percussion instrument (xylophone). For a high-input level (80 dB SPL), slow compression was preferred over fast compression.

Keywords

hearing aids, bandwidth, gain, time constants, automatic gain control, compression

Introduction

One aspect of sound reproduction systems that determines the subjective quality of the sound is the frequency response. This is measured by using as input to the system a sine wave of constant amplitude but variable frequency. Ideally, the output of the system should not vary as a function of frequency. In practice there is a limit to the range of frequencies that the system reproduces, and there may also be some irregularities in the response. The degree to which the response differs from the ideal response can be specified by using as a reference level the output in response to a given frequency (often 1 kHz). The output at other frequencies can then be specified relative to this level, and the overall response can be specified in terms of the range of frequencies over which the variations in output level fall within certain limits. Thus, the frequency response for a loudspeaker might be stated as 50 to 15000 Hz ±5 dB. This would mean that the sound level did not vary over more than a 10-dB range for any frequency from 50 to 15000 Hz.

Two aspects of the response are of perceptual relevance. The first is the overall frequency range. There is little point in having a response that extends below about 30 Hz, since there is little audible energy in music below that frequency, except for some organ sounds or synthesized sounds. The desirable upper limit is the subject of some controversy. Most adults cannot hear sinusoids with frequencies above about 20 kHz (Ashihara, 2006; Ashihara, Kurakata, Mizunami, & Matsushita, 2006), and the absolute threshold for detecting sounds usually increases markedly above about 15 kHz. Consistent with this, most adults with normal hearing cannot detect the effect of low-pass-filtering recorded speech or music at 16 kHz (Muraoka, Iwahara, & Yamada, 1981; Ohgushi, 1984).

The other important aspect of the frequency response is its regularity. If the frequency response has large peaks and dips, then these affect the timbre of the reproduced sound, introducing "coloration" (Moore & Tan, 2003). Psychoacoustic studies suggest that, under ideal conditions, individuals can detect changes in spectral shape when the level in a given frequency region is increased or decreased by 1 to 2 dB

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relative to the level in other frequency regions (Bucklein, 1962; Green, 1988; Moore, Oldfield, & Dooley, 1989). Thus, a response flat within ± 1 dB will not be detectably different from a perfectly flat response.

In this article, I start by reviewing the effects of bandwidth limitations and nonflat frequency responses on the perceived quality of reproduced music for adults with normal hearing. I then describe studies of the effects of bandwidth limitations and nonflat frequency responses for hearing-impaired people, considering especially the effects of the upper cutoff frequency.

Most modern hearing aids incorporate some form of multichannel compression to compensate for the effects of loudness recruitment that are experienced by people with cochlear hearing loss (Fowler, 1936; Moore, 2007). However, the exact implementation of the compression varies markedly across manufacturers, and there is no general agreement about the form of compression that is "best," if indeed there is an optimum; for reviews, see Moore (1990, 2007, 2008), Hickson (1994), Dillon (1996), and Souza (2002). Generally, evaluations of hearing aid compression have focused on its effects on speech intelligibility and loudness. In this article, I consider the effects of some of the characteristics of compression on ratings of the quality of music. The characteristics covered include: (a) compression speed, which is related to the speed with which the gain changes in response to increases or decreases in the input sound level (Moore, 2008), and (b) "overshoot" effects, which are brief marked increases in sound level at the output of the compression system following an abrupt increase in input level. They occur because it takes the system a finite time to reduce the gain in response to the increased level. Finally, the effects of two hearing-aid-fitting methods (CAM2 and NAL-NL2) on preferences for sound quality are described.

Effects of Bandwidth Limitations and Nonflat Frequency Responses for People With Normal Hearing

It is generally accepted that "smooth" wideband frequency responses in sound reproduction systems are desirable (Bucklein, 1962; Gabrielsson, Hagerman, Bech-Kristensen, & Lundberg, 1990; Gabrielsson, Lindström, & Till, 1991; Toole, 1986a, 1986b; Toole & Olive, 1988). Reviews of early studies on the perceptual effects of irregularities in frequency response are provided in Toole (1986a, 1986b) and Toole and Olive (1988). I focus here on a study of Moore and Tan (2003). The goal of the study was to explore how the perceived naturalness of music and speech signals was affected by various forms of linear filtering. The study was designed to isolate the effects of limited bandwidth and irregularities in frequency response from other forms of distortion that can occur in transducers, such as phase distortion and nonlinear distortion. The sounds were reproduced by high-quality headphones (Sennhesier HD580) with a

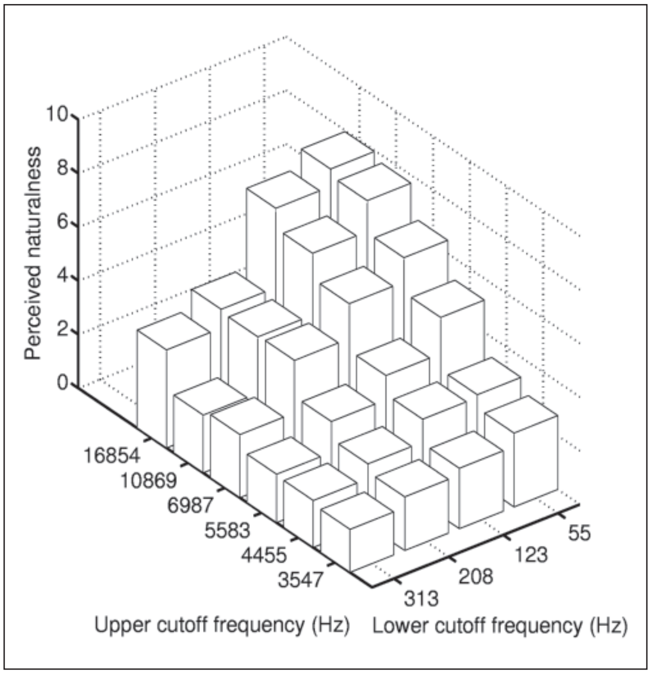

Figure 1. Effects of band-limiting music stimuli (jazz) on judged naturalness. The bottom-left axis shows the upper cutoff frequency, the bottom-right axis shows the lower cutoff frequency, and the vertical axis shows the mean naturalness rating.

good approximation to a diffuse field characteristic (i.e., the frequency response at the eardrum resembled the response that would be obtained listening with an open ear, when averaged over many different directions of sound incidence). The headphones also had low harmonic and intermodulation distortion (Tan, Moore, & Zacharov, 2003). This meant that the main factor affecting the quality ratings was the linear filtering. A large variety of forms of filtering were used, simulating different lower and upper cutoff frequencies and different types of spectral tilt and spectral ripple.

In a given session, a participant was tested using either speech or music stimuli. The music was a fragment of jazz (piano, bass, and drums) with a relatively constant overall level, taken from a commercial CD (digital recording). The same fragment was used throughout. Its duration was 7.3 s. Filtered stimuli were presented in a randomized order. After each stimulus presentation, there was a pause, during which the listener was required to rate the perceived quality on a 10-point scale where 10 = *very natural—uncolored* and 1 = *very unnatural—highly colored*. The response categories were displayed on the computer screen, and participants responded using the mouse to "click" on their category of choice.

To illustrate the meaning of the descriptors for the categories, before the experiment proper started, samples were presented of wideband signals (55 to 16854 Hz) without any spectral ripple or tilt; these were described as examples of Category 10. Similarly, samples were presented with large amounts of ripple and spectral tilt and described as examples of Category 1. Ten participants with normal hearing were tested. The pattern of results was very consistent across participants and across test sessions.

Figure 1 illustrates the effects of band-limiting the music stimuli. The bottom-left axis shows the upper cutoff frequency, the bottom-right axis shows the lower cutoff frequency, and the vertical axis shows the mean naturalness rating. The highest naturalness was obtained for the broadband signal (55 to 16854 Hz). Increasing the lower cutoff frequency from 55 to 313 Hz resulted in a progressive decrease of naturalness. There was also a progressive decrease in naturalness as the upper cutoff frequency was decreased from 16854 to 3547 Hz. Typical telephone bandwidth (313 to 3547 Hz) gave very poor naturalness, and typical hearing-aid bandwidth (about 200 to 4500 Hz) also gave poor naturalness.

The effect of ripples in the frequency response was investigated using ripples that were uniformly spaced on the ERB^{N}-number (Cam) scale, which is a scale related to the equivalent rectangular bandwidth (ERB) of the auditory filters for young listeners with normal hearing (for details, see Glasberg & Moore, 1990; Moore, 2012). For medium and high center frequencies this scale is similar to a logarithmic frequency scale. The ripples either extended over a wide frequency range (87 to 6981 Hz) or were restricted to low frequencies (87 to 606 Hz), mid frequencies (701 to 2224 Hz), or high frequencies (2503 to 6981 Hz). The ripple density in ripples/Cam varied from 0.05 (coarsely spaced ripples) to 0.5 (finely spaced ripples).

Figure 2 shows the mean ratings for music when processed through filters giving spectral ripples with a depth of 10 dB (peak-valley ratio). Ripples of this magnitude occur quite commonly in the frequency response of hearing aids and cheap headphones. The bottom-left axis shows the ripple rate (ripples/Cam), the bottom-right axis shows the ripple range, and the vertical axis shows the mean rating. The degradation in naturalness produced by the ripples is clearly evident. The ripples degrade naturalness to a greater extent when they extend over a wide frequency range than when they extend over subranges, and perceived naturalness tends to decrease with increasing ripple rate up to 0.2 ripples/Cam.

Overall, these results indicate that, for normal-hearing listeners, natural reproduction of music requires a wide bandwidth, from 55 Hz or less up to about 16000 Hz, and minimal ripples in the frequency response. Ripples with a depth of 5 dB had only small deleterious effects on naturalness (not shown), but ripples with a depth of 10 dB had a substantial effect.

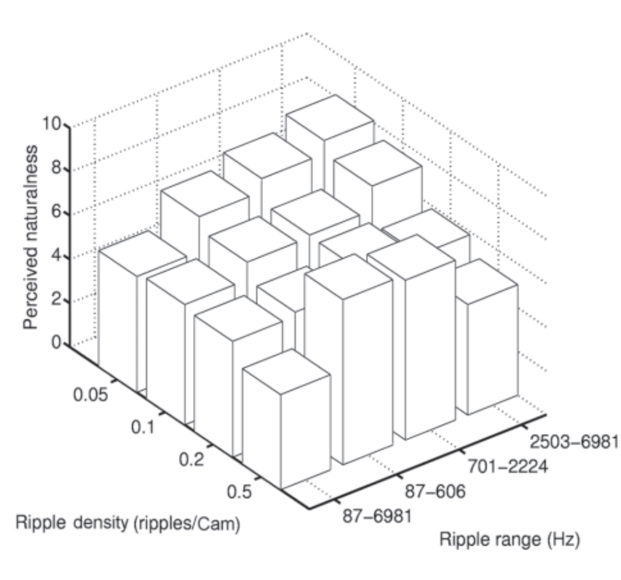

Figure 2. Effect of spectral ripples with a depth of 10 dB (peak-valley ratio) on judged naturalness. The bottom-left axis shows the ripple rate (ripples/cam), the bottom-right axis shows the ripple range, and the vertical axis shows the mean rating.

Effects of Bandwidth Limitations for People With Impaired Hearing

There have been several studies examining the effects of bandwidth limitations on evaluations of sound quality for music by hearing-impaired people. Franks (1982) obtained paired-comparison judgments of preference for hearing-aid processed music, varying both the lower and upper cutoff frequencies. Upper cutoff frequencies were either 4 or 10 kHz. Listeners with normal hearing preferred the higher cutoff frequency, but preferences were not consistent across listeners with impaired hearing. Ricketts, Dittberner, and Johnson (2008) pointed out that the absolute thresholds of the hearing-impaired listeners used by Franks varied over a wide range and suggested that the inconsistency across listeners might have occurred because some listeners consistently preferred the higher cutoff frequency and some consistently preferred the lower cutoff frequency.

Ricketts et al. (2008) obtained paired-comparison judgments of preference for (simulated) hearing-aid processed sounds using upper cutoff frequencies of 5.5 and 9 kHz. The sounds were a piece of music and a movie soundtrack, both chosen because they contained a relatively large amount of high-frequency energy. Both normal-hearing and hearing-impaired participants were tested. The latter were selected to have mild hearing losses; average absolute thresholds were 38, 43, and 53 dB HL at 8, 10, and 12 kHz, respectively. The gains were adjusted for each listener using the NAL-NL1 fitting method (Byrne, Dillon, Ching, Katsch, & Keidser, 2001). Since this method does not give recommended gains for frequencies above 6 kHz, gains at high frequencies were based on a form of extrapolation. Results were similar for these two signals. Both normal-hearing and hearing-impaired participants showed, on average, a preference for the higher cutoff frequency, but the strength of preference was greater for the former than for the latter. Preferences within the hearing-impaired group did not appear to be related to average absolute thresholds, but a steep slope of the audiogram was associated with a preference for the lower cutoff frequency.

It is not obvious why some hearing-impaired listeners appeared to prefer the lower cutoff frequency. One possibility is that these listeners were unused to hearing frequencies above about 6 kHz because of their long-standing hearing loss and the limited frequency range of their hearing aids (if any). When these high frequencies were amplified and presented in a laboratory setting, the sound quality may have appeared somewhat harsh or tinny. If this were the case, such listeners might come to prefer a higher cutoff frequency after an acclimatization period (Gatehouse, 1992). Another possibility is that the fitting rule used for the high frequencies, which was acknowledged by Ricketts et al. (2008) to be somewhat arbitrary, may have led to excessive loudness of the high frequencies for some listeners.

Until recently, hearing-aid-fitting methods, such as NAL-NL1 (Byrne et al., 2001), DSL[i/o] (Cornelisse, Seewald, & Jamieson, 1995), and CAMEQ (Moore, Alcántara, & Marriage, 2001; Moore, Glasberg, & Stone, 1999) did not give gain recommendations for frequencies above 6 kHz. Recently, two procedures have been introduced that give recommendations for gain at higher frequencies: NAL-NL2 (Keidser, Dillon, Flax, Ching, & Brewer, 2011) gives gain recommendation for frequencies up to 8 kHz, and CAMEQ2-HF (Moore & Füllgrabe, 2010), now called CAM2 (Moore et al., 2011), gives gain recommendations for frequencies up to 10 kHz. This allows evaluation of the effects of varying the upper cutoff frequency using gains and compression ratios (CRs) that are selected in a consistent way rather than arbitrarily.

Moore et al. (2011) examined the influence of upper cutoff frequency on preferences for music using a simulated five-channel compression hearing aid and the method of paired comparisons. The participants had mild hearing losses for frequencies up to 2 kHz, but the losses increased at higher frequencies, reaching an average of 60 dB at 8 kHz. All participants had normal middle-ear function. The TEN(HL) test (Moore et al., 2004) was used to check for the presence of cochlear dead regions over the frequency range 0.5 to 4 kHz. None were found. The gains and CRs of the simulated hearing aid were set individually for each of the 14 hearing-impaired participants, using the CAM2 method. The effects of several other aspects of the simulated hearing aid were also investigated, including compression speed and the use of a system for reducing overshoot effects; these are discussed in more detail later on.

Moore et al. (2011) used three types of music signals. One was a segment of a jazz trio (piano, bass, and drums), the same as used by Moore and Tan (2003), as described earlier. The second was an excerpt from an orchestra playing Bizet's Carmen, lasting 5.6 s. It included the sound of brass instruments and cymbals. The third, which was used only for evaluation of the effects of overshoot reduction, was extracted from Track 27 of the compact disc produced by Bang and Olufsen called "Music for Archimedes" (CD B&O 101). It was a recording of a solo percussion instrument, the xylophone, playing the "Sabre Dance" by Khachaturian (anechoic recording). The excerpt lasted 3.5 s. The long-term-average spectra of the three music signals, expressed as the relative level in 1/3-octave bands, are shown in Figure 3. The classical music and the percussion signals had spectra with relatively more high-frequency energy than the jazz signal. This should be kept in mind when interpreting the results.

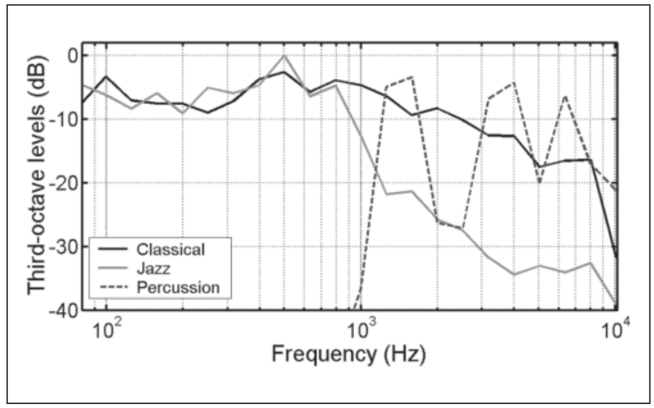

Figure 3. Spectra of the jazz, classical music, and percussion sounds, as used by Moore, Füllgrabe, and Stone (2011), specified as the relative levels in 1/3 octave bands.

The simulated hearing aid had the same general structure as described by Moore, Füllgrabe, and Stone (2010). Stimuli were processed to simulate listening through a hearing aid with five-channel wide dynamic range compression. The aid simulator included a facility for overall shaping of the frequency response; the gain for a speech-spectrum noise with a level of 65 dB SPL could be set for center frequencies of 0.25, 0.5, 1, 2, 3, 4, 6, 8, and 10 kHz. This noise had the spectrum described by Moore, Stone, Füllgrabe, Glasberg, and Puria (2008). The shaping was done using a single linear-phase finite impulse response (FIR) filter that was applied before compression. The four higher compression channels were each 1-octave wide and were centered on frequencies of 1, 2, 4, and 8 kHz. The lowest channel included all frequencies up to 0.71 kHz. FIR filters were used to create the channel signals, and the delay introduced by these filters was removed. The frequency response of each filter overlapped that of its neighbor at the −6-dB point and overlapped that of its next-but-one neighbor at the −65-dB point or lower. The compression thresholds were set to 49, 41, 40, 39, and 38 dB SPL, in order of increasing channel center frequency. The values were chosen to make soft speech audible, in keeping with the philosophy behind CAM2 (Moore, Glasberg, & Stone, 2010). The CR could be set independently for each channel. The attack and release times could be set to any desired value for each channel. The values actually used are specified for each experiment described below.

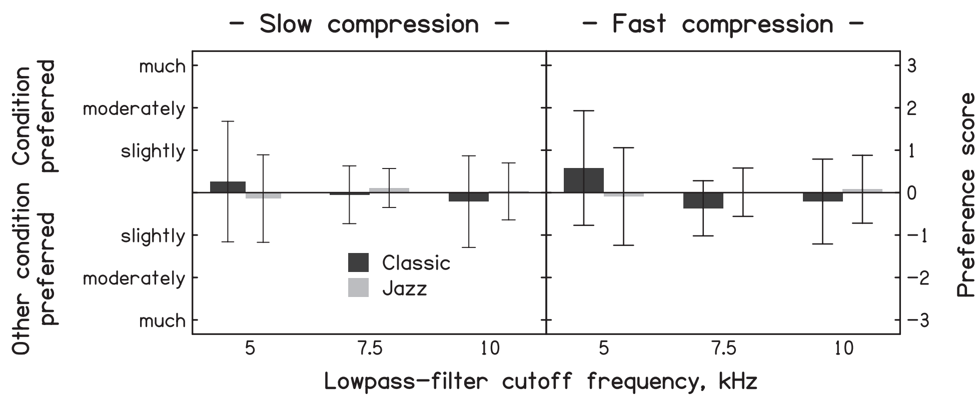

Figure 4. Mean preference judgments for each upper cutoff frequency for the slow compression system (left) and the fast compression system (right). In this and subsequent similar figures, error bars indicate ±1 standard deviation.

The audio signal could be slightly delayed relative to the gain-control signal, to reduce overshoot and undershoot effects (Robinson & Huntington, 1973). This delay is referred to as the "alignment delay." Its effects are described in more detail below.

All processing was performed offline, using at least 24-bit precision. During each experiment, stimuli were generated via an Echo Indigo 24-bit sound card, using a sampling rate of 22.05 kHz, and presented via Sennheiser HDA200 headphones. These are often used to measure audiometric thresholds for frequencies above 8 kHz. The response of the headphones was measured with a Knowles Electronic Manikin for Acoustic Research (KEMAR) (Burkhard & Sachs, 1975), using the "large" pinnae. Digital filtering was used to "correct" the response of the headphones so that it corresponded closely to the diffuse-field response of the ear as specified in ANSI S3.4-2007 (American National Standards Institute [ANSI], 2007). This ensured that the insertion gains prescribed by CAM2 were implemented accurately.

In one experiment, the upper cutoff frequency of the stimuli was set to 5, 7.5, or 10 kHz. Two compression speeds were used. For the fast system, the attack and release times were 10 ms and 100 ms, respectively, both defined as specified in ANSI (2003). For the slow system, the attack and release times were 50 and 3000 ms, respectively. The alignment delay was set to 2.5 ms.

The overall stimulus level at the input to the simulated hearing aid was 65 dB SPL. Within a run, the compression speed and type of stimulus were fixed and all possible pairs of cutoff frequencies were compared, in both possible orders. For each pair, the participant had to indicate which of the two stimuli was preferred (in terms of pleasantness) and by how much. This was done using a slider on a screen, which could be moved, using a mouse, along a continuum from "stimulus 1 much preferred" to "stimulus 2 much preferred," with "no preference" in the middle. The participant moved the slider to the appropriate point to indicate his or her preference. Before a trial commenced, the stimuli to be compared in that trial were presented in alternation in random order to help the participant to attend to the way in which the sounds differed perceptually.

Preference scores for each participant and each condition were computed in a manner similar to that described by Keidser, Dillon, and Byrne (1995). For a given pair of stimuli, the preferred one was assigned a score from 0 to 3 (depending on the slider setting), where 3 corresponded to *much preferred*, and the nonpreferred one was assigned a score of the same absolute magnitude but the opposite sign. The overall score for a given condition and stimulus type (e.g., classical music) was obtained by averaging all of the subscores obtained for that condition and stimulus type. If a given condition was always much preferred, it would have received a score of 3, whereas if the other conditions were always much preferred, they would have received a score of −3. Note that the preference score for a given condition expresses degree of preference relative to the other conditions tested in that series of paired comparisons.

The mean results are shown in Figure 4. The error bars indicate ±1 standard deviation across participants. The effects of cutoff frequency were generally small for both slow and fast compression. Separate one-way ANOVAs (analyses of variance) for each type of stimulus and each compression speed showed no significant effect of cutoff frequency. However, there were substantial individual differences, with some participants consistently preferring the 7.5- and 10-kHz cutoff frequencies and some consistently preferring the 5-kHz cutoff frequency. To assess whether the preferences were associated with audiometric thresholds or the slope of the audiogram at high frequencies, correlational analyses were conducted. The measure of bandwidth preference was taken as the difference between the preference

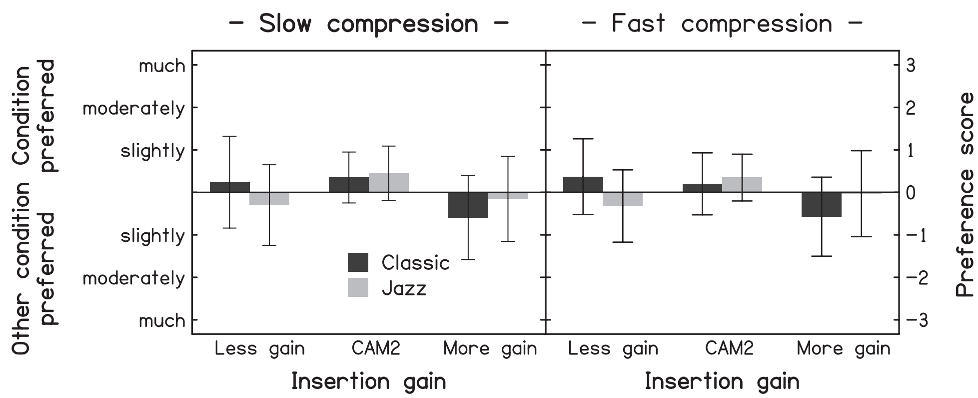

Figure 5. Mean preference judgments for the high-frequency gains prescribed by CAM2, and either less gain or more gain. Results are shown for the slow compression system (left) and the fast compression system (right).

scores for the 5- and the 7.5-kHz bandwidths; the 10-kHz bandwidth was not considered as there was no clear difference in preferences between the 7.5- and 10-kHz bandwidths. The audiogram measures were the audiometric threshold at 6 and 8 kHz (each considered separately), and the difference between audiometric thresholds at 4 and 8 kHz, which is a measure of audiogram slope at high frequencies. There were no significant correlations between the bandwidth preference measures and the audiometric thresholds at 6 and 8 kHz. However, the slope measure was correlated with the pleasantness preference for jazz ($r = .538$, $p = .047$). A steep audiogram slope was associated with a preference for the narrower bandwidth, and a shallow slope was associated with a preference for the wider bandwidth, as found by Ricketts et al. (2008).

The individual variability in terms of preferences for cutoff frequency may have been related to the amount of high-frequency gain prescribed by CAM2; the gain might have been higher than preferred for some participants, leading them to prefer a lower cutoff frequency. To assess this possibility, preference judgments were obtained with the high-frequency gains of the simulated hearing aid set both lower and higher than recommended by CAM2. To obtain reduced gains, the gains for a speech-shaped noise with a level of 65 dB SPL were multiplied by 0.9, 0.85, 0.8, and 0.8 for center frequencies of 4, 6, 8, and 10 kHz, respectively. To obtain increased gains, the gains for a speech-shaped noise with a level of 65 dB SPL were multiplied by 1.11, 1.18, 1.25, and 1.25, for center frequencies of 4, 6, 8, and 10 kHz, respectively. The modified gains were implemented in the linear FIR filter prior to filtering into compression channels. Thus, the CRs in each channel were not altered. Gains for frequencies below 4000 Hz were not modified.

Both the slow and fast compression speeds were used. Within a run, the compression speed and type of stimulus were fixed and all possible pairs of gains (less gain, CAM2, and more gain) were compared, in both possible orders. The overall stimulus level at the input to the simulated hearing aid was 65 dB SPL. The upper cutoff frequency for all stimuli was 10 kHz.

The mean results and standard deviations are shown in Figure 5. The results were very similar for the two compression speeds. For the classical music signal, the CAM2 gains and the reduced gains were approximately equally preferred, whereas the increased gains were not preferred. Preferences did not differ significantly for reduced gains and CAM2 gains. For the jazz signal, which had relatively less high-frequency energy, CAM2 gains tended to be preferred over either reduced or increased gains. However, the effects were small. Of the three participants who regularly used hearing aids in their everyday life, two preferred CAM2 gains over either reduced or increased gains for both signals. This suggested that experienced participants may prefer slightly greater gains than inexperienced participants (Marriage, Moore, & Alcántara, 2004). The version of CAM2 currently in use recommends slightly lower gains for inexperienced than for experienced users, more so at high frequencies.

Overall, it seems clear that whereas normal-hearing listeners clearly prefer upper cutoff frequencies greater than 5 kHz when listening to music, preferences among hearing-impaired listeners are less clear, and vary markedly across listeners. Preference for an upper cutoff frequency above 5 kHz seems to be associated with audiograms that do not have a steep slope (Moore et al., 2011; Ricketts et al., 2008). It may also be the case that listeners who are not experienced hearing-aid users prefer lower gains at high frequencies than experienced users. A study presented later in this article, comparing the NAL-NL2 and CAM2 hearing-aid-fitting procedures, supports the idea that amplification at frequencies above 5 kHz is preferred, provided that gains are chosen appropriately.

There have been relatively few studies of preferences for the lower cutoff frequency in hearing aids. Most hearing aids have frequency responses that roll off below about 200 Hz. This is often considered desirable when listening to speech, as the roll off reduces masking from intense low-frequency environmental sounds, such as car noise or noise from air-conditioning. This can be important, as hearing-impaired people are often very susceptible to the "upward spread of masking" (Glasberg & Moore, 1986). However, for listening to music, Franks (1982) showed that hearing-impaired participants clearly preferred cutoff frequencies below 200 Hz.

Many hearing-impaired people have normal or near-normal hearing at low frequencies. For such people, open-fit hearing aids are often used, and low-frequency sounds are heard via leakage past the open dome. In such cases, the low-frequency roll off of the hearing-aid response is largely irrelevant. However, if a more closed fit is used, either because the client has a hearing loss at low frequencies or because a closed fit is required to reduce acoustic feedback, then the low-frequency response of the hearing aid becomes much more important. Tests using sealed-dome receiver-in-the-canal hearing aids suggest that a lower cutoff frequency of about 50 Hz is required for good sound quality when listening to music (Füllgrabe, Moore, van Tasell, & Stone, 2007), consistent with the results obtained by Moore and Tan (2003) for normal-hearing listeners.

Effects of Irregularities in Frequency Response for People With Impaired Hearing

The frequency response of a hearing aid at or close to the eardrum often shows distinct ripples. For "closed-fit" hearing aids, these can be caused by resonances in the acoustical delivery system, for example, the tubing leading from a behind-the-ear hearing aid to the earmold. It is possible to reduce these peaks, smoothing the overall frequency response, by suitable modifications to the tubing and/or by the use of acoustic resistors (Killion, 1982; Libby, 1981). For "open-fit" hearing aids, ripples in the frequency response can be caused by the interference of amplified sound from the hearing aid with sound leaking past the open dome (Stone, Moore, Meisenbacher, & Derleth, 2008). These ripples can be reduced to some extent by adjusting the gain in individual frequency channels, provided that the aid has many such channels. However, the ripples are difficult to eliminate completely, and, for a hearing aid with multichannel compression, the pattern of the ripples may change with input sound level.

A single broad peak in the frequency response around 3 kHz is desirable, since this mimics the normal response of the outer ear. However, additional peaks and dips are not desirable and can have adverse effects on sound quality. It is now widely accepted that the frequency response should be as smooth as possible. Although one study showed no significant differences in judged clarity and pleasantness between hearing aids with unsmoothed and smoothed responses (Cox & Gilmore, 1986), most other studies have shown that hearing aids with smoothed frequency responses are preferred (Libby, 1981; Mertz, 1982).

To study the effects of frequency response irregularities in a better controlled manner, van Buuren, Festen, and Houtgast (1996) artificially imposed peaks in the frequency response of a sound reproduction system via digital filtering prior to delivery via headphones. The peaks were centered at 1.3, 2.8, or 5.5 kHz and had heights of 10, 20, or 30 dB. The peaks were presented either singly or all three together. A reference condition without any such peaks was included. A total of 26 participants with sensorineural hearing loss were tested, most with hearing loss increasing with increasing frequency. Frequency-dependent amplification was applied to ensure that the signals fell within the dynamic range of each participant (the range between the detection threshold and the uncomfortable loudness level). Several music signals were used, including (a) flute, piano, and voice; (b) trumpet and orchestra; (c) drums, synthesizer, and voice; and (d) piano. Participants were asked to rate each sound sample on a scale with response options ranging from *very unpleasant* to *very pleasant*. Pleasantness ratings decreased systematically with increasing peak height and also tended to decrease with increasing center frequency of the peak. Multiple peaks led to lower pleasantness than a single peak. Even the smallest peaks used (10 dB) led to noticeable reductions in pleasantness for some of the music signals.

It can be concluded that the quality of music perceived by hearing-impaired people is reduced by frequency-response irregularities when the peak-to-valley ratio in the response is 10 dB or more, as is common in hearing aids. In addition, there are at least three benefits of smoothing the frequency response other than effects on sound quality: (a) It can reduce acoustic feedback; (b) it can reduce the distortion (including temporal distortion produced by rapid phase changes) that often occurs at frequencies around peaks in the response; (c) it can allow a greater proportion of the spectrum of the sound to be above threshold before the uncomfortable loudness level is reached.

Effects of Overshoot Reduction on Music Preferences

A small time delay of the audio signal relative to the gain control signal, called here the alignment delay, has been proposed as a method of reducing overshoot effects in compression systems (Robinson & Huntington, 1973; Verschuure & Dreschler, 1996; Verschuure et al., 1993). The concept is illustrated in Figure 6, which shows the output of the simulated five-channel hearing aid described above in response to a 2000-Hz tone with an abrupt 25-dB increase in level, lasting 0.03 s. The CR was 2 for all channels, and both the low and high signal level were above the compression threshold. For the left panels, the attack and release times were 10 ms and 100 ms, respectively, both defined as specified in ANSI (2003). This corresponds to the fast compression system described earlier (Moore et al., 2011). For the right panels, the attack and release times were 50 and 3000 ms, respectively, corresponding to the slow compression system described earlier. Without the alignment delay (0 ms, top panels), when the input level of the signal is suddenly

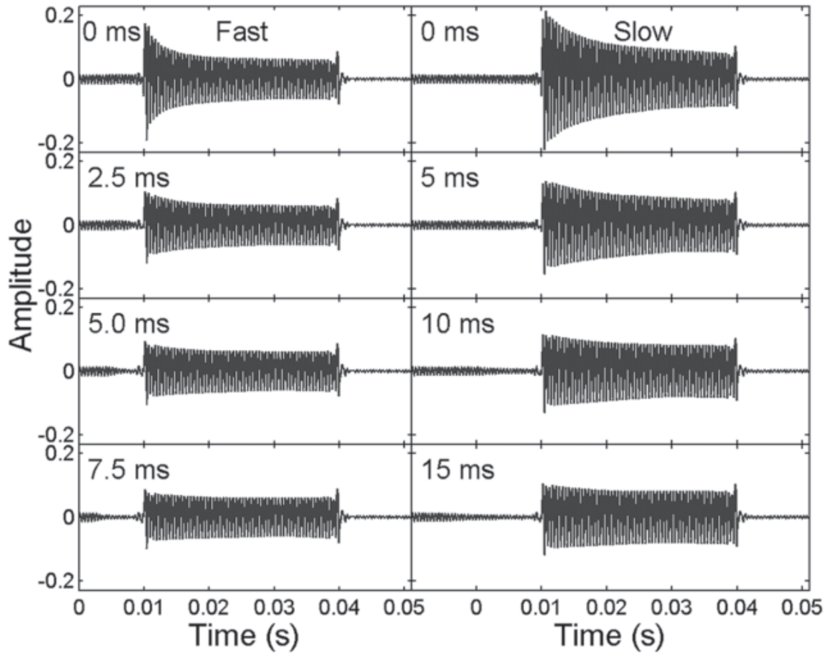

Figure 6. Illustration of the effect of alignment delay on the overshoot effect. The input signal was a 2000-Hz tone with an abrupt 25-dB 0.03-s increase in sound level. The output of the simulated compression hearing aid is shown for the fast compression system (left: alignment delays of 0, 2.5, 5, and 7.5 ms) and the slow compression system (right: alignment delays of 0, 5, 10, and 15 ms).

increased, it takes a certain time for the gain to be reduced; this time is determined by the attack time. As a result, the output level becomes high for a short time (the overshoot effect) and then decreases smoothly to a steady value. This may mean that the signal momentarily becomes uncomfortably loud. Also, the envelope shape of the signal is distorted.

The alignment delay has an effect similar to that of looking into the future. The gain is reduced before the sudden increase in sound level occurs, and the overshoot effect is much reduced. This effect is most apparent for the alignment delays of 5 and 7.5 ms for the fast compression system and the alignment delays of 10 and 15 ms for the slow compression system. Note that increase of gain following the end of the high-level section of the signal (at 0.04 s) is not apparent because the release times for both compression speeds are much longer than the attack times; the gain does not recover significantly over the 10-ms duration shown.

A problem with the use of alignment delay is that it necessarily involves a delay in the overall signal at the output of the hearing aid. This delay is added to other delays that are inherent in the operation of the hearing aid, such as those produced by spectral analysis. The overall throughput delay can have disturbing effects on speech production and speech perception when the delay exceeds 10 to 20 ms (Stone & Moore, 2002, 2003a, 2005; Stone et al., 2008). These effects place a limit on the alignment delay that can be used in practice.

Moore et al. (2011) assessed the effect of alignment delay for the slow and fast compression systems. For the slow system, alignment delays of 0, 5, 10, and 15 ms were used. For the fast system, alignment delays of 0, 2.5, 5, and 7.5 ms were used.

The percussion instrument (xylophone) was used as the test stimulus. This was felt to be appropriate for exploring perceptual effects of overshoot, since the abrupt onset of each note would tend to elicit strong overshoot effects. The overall stimulus level at the input to the simulated hearing aid was 65 dB SPL. Within a run, the compression speed was fixed and all possible pairs of alignment delays were compared, in both possible orders.

A preference score for each alignment delay was determined as described earlier. The preference score for a given alignment delay expresses degree of preference relative to the other alignment delays used with that compression speed. The mean results are shown in Figure 7. Note that the scale ranges only from −1 to +1. The effects of alignment delay were subtle. However, there was a trend for pleasantness to increase with increasing alignment delay, especially for the fast compression system.

Analysis of variance showed that the effect of alignment delay was significant for the slow system: $F(3, 24) = 3.22$, $p = .04$. Post hoc tests, based on Fisher's protected least significant difference (LSD) test, showed that the 15-ms delay was preferred over the delays of 5 and 10 ms (both $p < .05$).

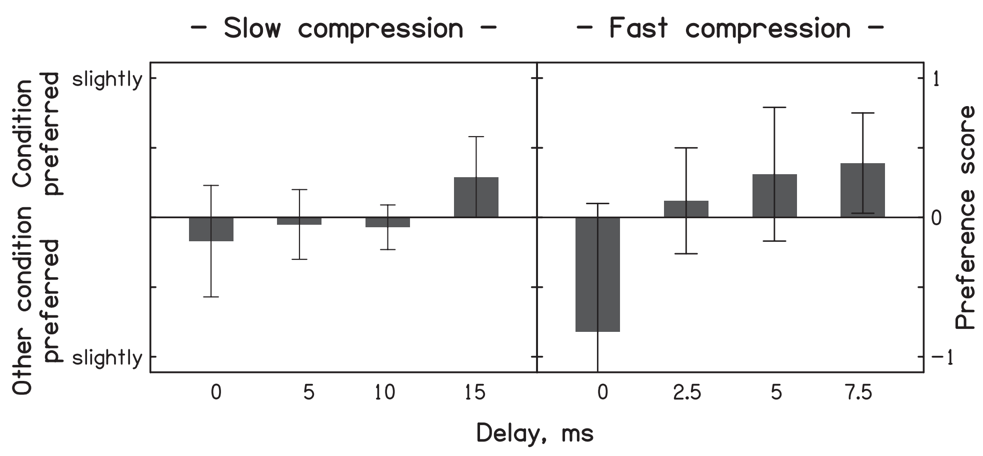

Figure 7. Mean preference judgments for each alignment delay for the slow compression system (left) and the fast compression system (right).

The effect of alignment delay was also significant for the fast system: $F(1.347, 10.77) = 6.33$, $p = .022$. LSD tests showed that the delays of 2.5, 5, and 7.5 ms were preferred over the delay of 0 ms, but preferences did not differ for delays of 2.5, 5, and 7.5 ms.

In summary, for the percussion sound, pleasantness did increase significantly with increasing delay, but the effect was small. The most marked effect was that, for the fast compression system, the delay of 0 ms received the lowest rating.

Effects of Compression Speed and Input Level on Preferences

There is no general consensus about whether slow- or fast-acting compression is better for listening to music. In principle, fast-acting compression might be better for dealing with sudden changes in sound level, as can occur in the transition from a *Forte* passage to a *Piano* passage, or vice versa. This is illustrated on a compact disk containing simulations of the effects of hearing loss (Moore, 1997). On the other hand, fast compression can introduce cross-modulation effects between the signals from different sound sources (Stone & Moore, 2003b, 2004, 2008) and this might make it more difficult to hear out the different instruments or groups of instruments in an ensemble performance.

Moore et al. (2011) determined relative preferences for music for three compressor speeds. Two of these were the slow and fast systems described earlier. The third, designated "medium," used an attack time of 20 ms and a release time of 300 ms in all channels. The stimulus bandwidth was 10 kHz in all cases.

The stimuli were the jazz and classical music described earlier. Since one purpose of compression systems is to make sounds both audible and comfortable over a wide range of input levels, three input levels were used: 50-, 65-, and 80 dB SPL. Within a run, the type of stimulus and input level were fixed and all possible pairs of compression speeds were compared, in both possible orders.

The mean results are shown in Figure 8. For the input levels of 50- and 65 dB SPL, compression speed had almost no effect. For the input level of 80 dB SPL, preferences tended to decrease with increasing compression speed. A separate one-way ANOVA was conducted for each stimulus type and each input level. For the 50-dB level, there was no significant effect of compression speed for any stimulus. For the 65-dB level, there was a significant effect only for the classical music: $F(2, 26) = 4.07$, $p = .029$. LSD tests showed that the slow system was preferred over the fast system ($p = .038$). For the 80-dB level, there was a significant effect for classical music, $F(2, 26) = 18.2$, $p < .001$, and for jazz, $F(2, 26) = 11.94$, $p < .001$. LSD tests showed that the slow system was consistently preferred over the medium and fast systems (all $p < .034$). The strength and consistency of the preferences varied across participants. For the 80-dB input level, 4 participants (HI3, HI8, HI12, HI13) showed clear and consistent preferences, and the remaining 10 showed patterns of preference that were small and varied across stimuli.

Overall, the results indicate that the effects of compression speed on pleasantness and clarity were greater for high than for low input levels and that, for the 80 dB SPL input level, slow compression was generally preferred over medium or fast compression. The strength and consistency of the preferences varied across participants, but none showed a clear preference for fast compression.

Effects of Fitting Method on Preferences: CAM2 Versus NAL-NL2

As mentioned earlier, two new fitting methods have been introduced, the NAL-NL2 method (Keidser et al., 2011) and the CAM2 method (Moore, Glasberg et al., 2010). NAL-NL2

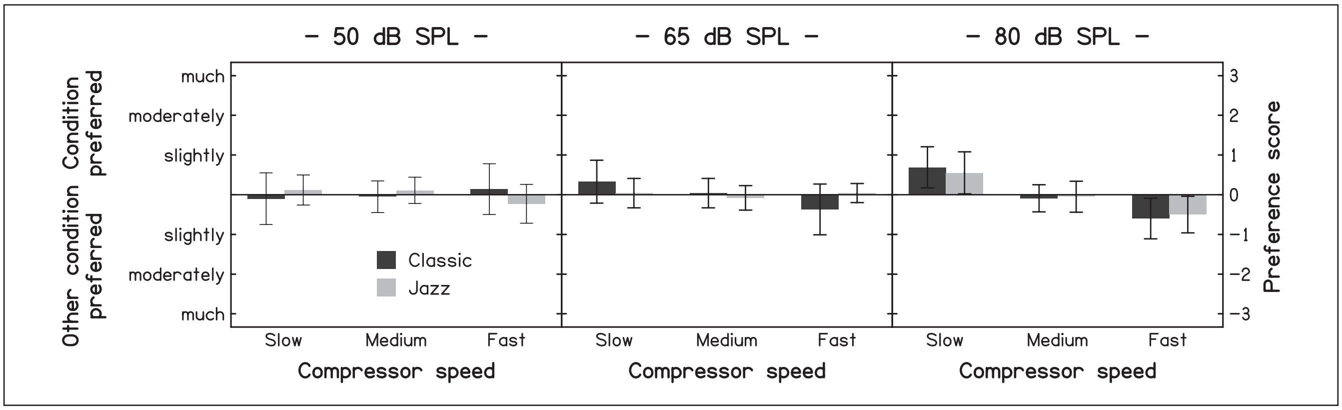

Figure 8. Mean preference scores as a function of compressor speed for stimuli with input levels of 50 (left), 65 (middle), and 80 (right) dB SPL.

recommends less mid-frequency gain and more low- and high-frequency gain than NAL-NL1 (Johnson & Dillon, 2011). NAL-NL2 recommends gain for frequencies up to 8 kHz, whereas the limit for NAL-NL1 is 6 kHz. CAM2 is conceptually similar to the CAMEQ method (Moore, 2005; Moore et al., 1999), which was developed using a loudness model (Moore & Glasberg, 1997). The gains recommended by CAM2 are close to, but typically 1 to 3 dB lower, than those recommended by CAMEQ for frequencies from 1 to 4 kHz. CAM2 gives recommended gains for center frequencies up to 10 kHz, whereas the upper limit for CAMEQ is 6 kHz.

Moore and Sek (2012) compared the relative preferences for sound quality of hearing-aid fittings based on CAM2 and on NAL-NL2, using the same simulated five-channel compression hearing aid as described previously. Both fast and slow compression were used, as described earlier. The music stimuli were the same as described previously (jazz, classical, and percussion), plus a man singing (a counter-tenor accompanied by guitar and recorder). For all four music signals, the input level to the simulated hearing aid was 50, 65, or 80 dB SPL. The level was always the same for the two sounds within a pair to be compared. In the paired-comparison procedure, the same segment of sound was presented twice in succession, once with the gains and CRs prescribed by NAL-NL2 and once with the gains and CRs prescribed by CAM2. The possible orders were used equally often and the order was randomized across trials. Within a given pair of sounds (one trial), the only difference between the signals was in the fitting method; the input level and compression speed were always the same. For each pair of sounds, the participant was asked to indicate which of the two was preferred and by how much. Participants responded using a slider on the screen, which could be moved, using the mouse, along a continuum labeled *1 = much better, 1 = moderately better, 1 = slightly better, equal, 2 = slightly better, 2 = moderately better*, and *2 = much better*.

Within each block of trials, six types of signals were presented (classical music, jazz, male singing, percussion, female speech, and male speech). The two compression speeds were also used within a single block, but the input level was kept constant. Within a block, the 24 pairs of sounds (6 signal types × 2 compression speeds × 2 presentation orders [CAM2 first or NAL-NL2 first]) were presented in random order. Two blocks of trial were used for each participant and each input level. The order of presentation of input levels across blocks was random.

Preference scores for each participant and each condition were computed in the following way. Regardless of the order of presentation in a given trial (CAM2 first or NAL-NL2 first), if CAM2 was preferred the slider position was coded as a *negative number* and if NAL-NL2 was preferred the slider position was coded as a *positive number*. For example, if the order on a given trial was NAL-NL2 first and CAM2 second, and the participant set the slider position midway between *2 = slightly better* and *2 = moderately better*, the score for that trial was assigned a value of −1.5. The overall score for a given fitting method and stimulus type (e.g., classical music) was obtained by averaging all of the subscores obtained for that fitting method and stimulus type. Hence, a score of −3 would indicate a very strong and perfectly consistent preference for CAM2 whereas a score of +3 would indicate a very strong and perfectly consistent preference for NAL-NL2. A score of 0 would indicate no preference. In practice, because of variability in the responses and a general reluctance to use the extremes of the rating scale, most preference scores fell in the range -1 to +1.

Fifteen participants with mild-to-moderate sensorineural hearing loss were tested. Their ages ranged from 54 to 76 years. There were 11 male and 4 female participants. Six had experience with multichannel compression hearing aids and nine were inexperienced. Gains were reduced for the inexperienced participants according to the CAM2 and NAL-NL2 recommendations. Recommended insertion gains at low

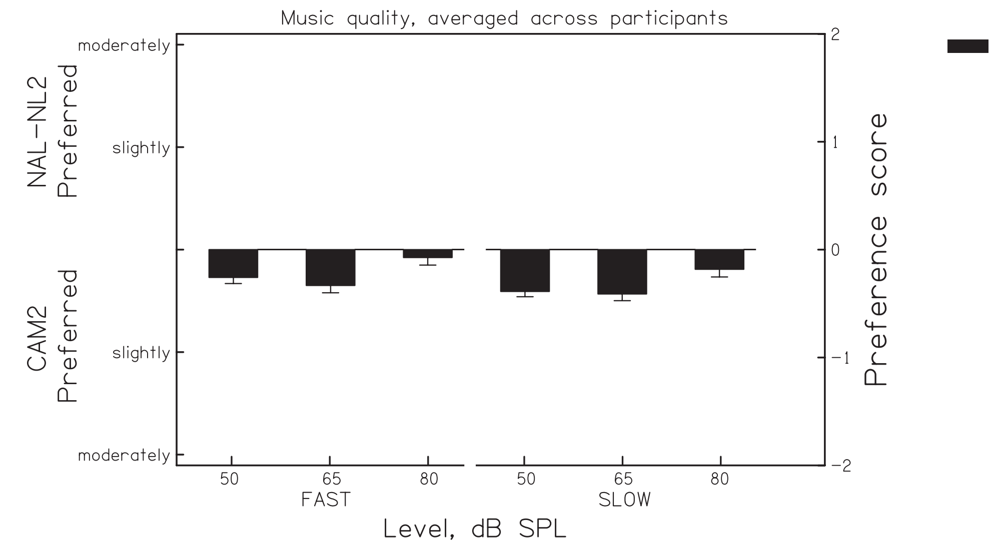

Figure 9. Preference scores averaged across participants for the overall quality of classical music and jazz using a simulated hearing aid fitted using CAM2 or NAL-NL2. Results are shown separately for the two compression speeds and three input levels.

frequencies were small for both fitting methods, which is as expected given that most participants had near-normal hearing at low frequencies. Over the frequency range 1 to 4 kHz, CAM2 insertion gains were 0 to 5 dB higher than NAL-NL2 insertion gains. For frequencies above 4 kHz, CAM2 insertion gains were markedly higher than NAL-NL2 insertion gains; the insertion gains recommended by CAM2 increased with increasing frequency above 4 kHz, whereas the insertion gains for NAL-NL2 remained nearly constant. The CRs for the two fitting methods were similar.

Figure 9 shows preference scores averaged across participants for the overall quality of classical and jazz music (results were similar for the other two music signals). The results are shown separately for the two compression speeds and three input levels. CAM2 was preferred relative to NAL-NL2 for both types of music, both compression speeds and all three levels, although the effect was very small for the input level of 80 dB SPL. The mean score of −0.29 was significantly different from zero, based on a t test, $t(1439) = -10.6$, $p < .001$.

As described earlier, the differences between the insertion gains prescribed by CAM2 and by NAL-NL2 were small (usually less than 5 dB) for frequencies up to 4 kHz, but CAM2 prescribed considerably more insertion gain than NAL-NL2 for frequencies above 4 kHz. Thus, the preference for CAM2 over NAL-NL2 can mainly be ascribed to the greater high-frequency gain prescribed by CAM2. In the work described earlier in this article, increasing the upper cutoff frequency from 5 to 7.5 kHz or from 5 to 10 kHz, using simulated hearing aids fitted using the original version of CAM2 (Moore, Glasberg et al., 2010), resulted in small but significant reductions in judged overall sound quality (Füllgrabe, Baer, Stone, & Moore, 2010; Moore et al., 2011); the sound quality was described as too shrill or too harsh for the sounds with the wider bandwidth. This effect was ascribed to the fact that most of the participants in those studies were not experienced users of hearing aids. CAM2 now incorporates reduced gains for inexperienced users, and those reduced gains were employed in the study of Moore and Sek (2012) for the nine participants who did not have experience with hearing aids. This gain reduction seems to have been successful, in that most of the participants tested preferred the greater high-frequency gains prescribed by CAM2 over the lower gains prescribed by NAL-NL2.

Conclusions

For normal-hearing participants, the highest ratings of quality for sounds reproduced via headphones were obtained when the reproduction bandwidth was wide (55 to 16000 Hz) and ripples in the frequency response were small (less than ± 5 dB). For hearing-impaired participants listening via a simulated five-channel compression hearing aid with gains set using the CAM2 fitting method, preferences for upper cutoff frequency varied across participants: Some preferred a 7.5- or 10-kHz upper cutoff frequency over a 5-kHz cutoff frequency and some showed the opposite preference. There were no clear differences in preference for the 7.5- and 10-kHz cutoff frequencies. Preferences for a higher upper cutoff frequency were associated with a shallow high-frequency slope of the audiogram. A subsequent study comparing the CAM2 and NAL-NL2 fitting methods, with gains slightly reduced for participants who were not experienced hearing-aid users, showed a consistent preference for CAM2. Since the CAM2 and NAL-NL2 methods differ mainly in the gain applied for frequencies above 4 kHz (CAM2 recommending higher gain than NAL-NL2), these results suggest that extending the upper cutoff frequency is beneficial. For high-quality reproduction of music, the lower cutoff frequency of a hearing aid should be well below 200 Hz, and preferably about 50 Hz, and ripples in the frequency response should be less than ±5 dB. A system for reducing overshoot effects produced by compression gave small but significant benefits for sound quality of a percussion instrument (xylophone). For a high input level (80 dB SPL), slow compression was preferred over fast compression.

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