Tolerable Hearing Aid Delays. V. Estimation of Limits for Open Canal Fittings

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Objectives: **Open canal fittings are a popular alternative to close-fitting earmolds for use with patients whose low-frequency hearing is near normal. Open canal fittings reduce the occlusion effect but also provide little attenuation of external air-borne sounds. The wearer therefore receives a mixture of air-borne sound and amplified but delayed sound through the hearing aid. To explore systematically the effect of the mixing, we simulated with varying degrees of complexity the effects of both a hearing loss and a high-quality hearing aid programmed to compensate for that loss, and used normal-hearing participants to assess the processing.**

Design: **The off-line processing was intended to simulate the percept of listening to the speech of a single (external) talker. The effect of introducing a delay on a subjective measure of speech quality (disturbance rating on a scale from 1 to 7, 7 being maximal disturbance) was assessed using both a constant gain and a gain that varied across frequency. In three experiments we assessed the effects of different amounts of delay, maximum aid gain and rate of change of gain with frequency. The simulated hearing aids were chosen to be appropriate for typical mild to moderate high-frequency losses starting at 1 or 2 kHz. Two of the experiments used simulations of linear hearing aids, whereas the third used fast-acting multichannel wide-dynamic-range compression and a simulation of loudness recruitment. In one experiment, a condition was included in which spectral ripples produced by comb-filtering were partially removed using a digital filter.**

Results: **For linear hearing aids, disturbance increased progressively with increasing delay and with decreasing rate of change of gain; the effect of amount of gain was small when the gain varied across frequency. The effect of reducing spectral ripples was also small. When the simulation of dynamic processes was included (experiment 3), the pattern with delay remained similar, but disturbance increased with increasing gain. It is argued that this is mainly due to disturbance increasing with increasing simulated hearing loss, probably because of the dynamic processing involved in the hearing aid and recruitment simulation.**

Conclusions: **A disturbance rating of 3 may be considered as just acceptable. This rating was reached for delays of about 5 and 6 msec, for simulated hearing losses starting at 2 and 1 kHz, respectively. The perceptual effect of reducing the spectral ripples produced by comb-filtering was small; the effect was greatest when the hearing aid gain was small and when the hearing loss started at a low frequency.**

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Introduction

When the ear canal is occluded by a hearing aid or an earmold, the level of body conducted sounds at the eardrum increases, especially for frequencies below about 500 Hz (Zwislocki, 1975). Chewing and self-voicing become more noticeable, especially if hearing levels at low frequencies are near normal. This is called the "occlusion effect." Open canal (OC) fittings deliver sound to the meatus either via a thin tube or via a receiver in the ear canal. In either case, the meatus is not blocked. This results in a much reduced occlusion effect, but also gives little attenuation of external air-borne sounds. The signal at the eardrum is therefore a mixture of a direct airborne component and a component that is amplified but delayed through the hearing aid.

Previous work (Agnew & Thornton, 2000; Stone & Moore, 1999, 2002, 2003b, 2005) has assessed the disturbance of the delay introduced by a digital hearing aid and the compromises that need to be considered when fitting a hearing aid to a particular hearing loss. That body of work was concerned primarily with closed fittings, which tend to be used when there is a risk of acoustic feedback due to the gains necessary for restoration of audibility, or when the client has a hearing loss at low as well as at high frequencies. With the advent of digital feedback cancellation, open fittings have become much more widely used, but the effect of time delays for OC fittings has received relatively little attention (Groth & Søndergaard, 2004).

In principle, the mixing of direct airborne sound and delayed sound produced by an OC hearing aid can have several undesired perceptual effects. First, there is a comb-filtering effect (ripples in the spectrum), which leads to an alteration of the timbre of the sound (coloration). This happens because com-

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ponents at frequencies that are integer multiples of the reciprocal of the delay add constructively, leading to a pattern of repeated peaks in the overall frequency response. Components at frequencies that are odd integer multiples of the reciprocal of twice the delay add destructively, leading to a pattern of valleys in the overall frequency response. When the signals in the direct and delayed paths are of equal level, the peaks and the valleys in the spectrum reach their maximum extent, of + 6 and $-\infty$ dB, respectively. The effects of comb filtering are most perceptible for relatively short delays, when the ripples are widely spaced in frequency and are therefore resolved in the auditory system (Moore & Tan, 2003, 2004). A second effect occurs for longer delays; the delayed sound may be perceived as an echo. Again, this effect tends to be strongest in a spectral region where the direct and delayed sounds are equal in level, or nearly so (Stone & Moore, 1999). Third, since the delayed sound tends to be negligible at low frequencies, but to dominate at high frequencies, the high-frequency sounds are delayed relative to the low-frequency sounds.

Across-frequency delay may typically assume one of two patterns. Figure 1 shows stylized spectrograms for an impulse (click) arriving at the cochlea via both an air path and a hearing aid. If the aid provides a delay that is constant across frequency over the frequency range for which gain is applied, the pattern of variation of delay with frequency for sounds arriving at the cochlea is similar to that

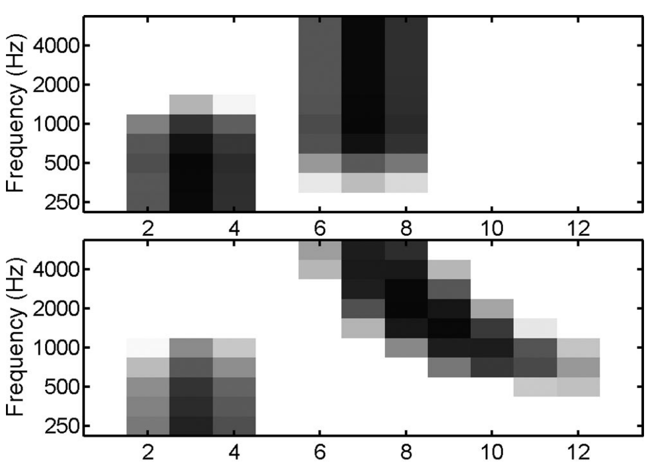

Fig. 1. Schematic spectrograms of response to a brief impulse as received at the eardrum when using an OC fitting. In both panels, the low frequencies arrive primarily via the air path, with a relatively short delay (chosen arbitrarily to be 3 msec), whereas the high frequencies arrive via the hearing aid with a delay. The top panel shows the spectrogram for an aid with a constant delay across frequency. The bottom panel shows the spectrogram for an aid with a delay that increased with decreasing frequency.

shown in the top panel of Figure 1. Low-frequency components arrive simultaneously via the air path, with little delay. High-frequency components that are amplified by the aid are more intense than high-frequency components arriving via the air path, and so the response at high frequencies is dominated by the delayed aided sound. We refer to this pattern as a monotonic variation of across-frequency delay and it is explored in experiment 2.

A different situation occurs when the aid itself produces a delay that varies across frequency, over the frequency range for which gain is applied. To keep processing delays as low as possible, some hearing aids make use of a filterbank (Hohmann, 2002) or filterbank-like approach to frequency analysis, for example, “warp” processing (Kates, 2005; Kates & Arehart, 2005). Broader analysis bands, which are used for high frequencies, give lower processing delay, whereas narrower analysis bands, which are used for low frequencies, give higher processing delay. This occurs for some currently manufactured OC hearing aids. A schematic spectrogram for this case is shown in the bottom panel of Figure 1 (for clarity, we have exaggerated the change in delay with frequency compared with that found in commercial hearing aids). Low-frequency components again arrive via the air path with little delay. Mid-frequency components, amplified by the aid, are delayed more through the aid than high-frequency components; hence the curvature of arrival time with frequency between 1000 and 4000 Hz. We refer to this overall pattern as a non-monotonic variation of across-frequency delay, and it is explored in experiment 3.

Across-frequency delay can itself have disturbing effects. For example, sounds like clicks may appear smeared in time, or may be perceived as rapid frequency glides, especially if there is a non-monotonic variation in delay. For most OC fittings, the second and third effects discussed earlier (comb filtering and the perception of echo) are most salient in the transition frequency region over which the hearing aid gain is increasing, whereas the across-frequency delays are most noticeable for broadband transient sounds, such as /t/.

Groth and Søndergaard (2004) assessed the effect of delays up to 10 msec using an OC aid that produced 10 dB of gain for frequencies above 800 Hz. Both normal hearing and mildly hearing-impaired participants gave higher disturbance ratings for the longest delay when judging the quality of their own voice. For external sounds, only the normal-hearing group rated the longest delay as being more disturbing. However, the scope of the study was limited by the use of a single hearing aid characteristic with rather low gain. Also, it is not clear which of the effects discussed earlier was mainly responsible for the reported disturbance.

We report here three experiments examining the influence of a variety of factors that may affect the disturbance produced by hearing aid delay in an OC fitting. In all three experiments, we wished to investigate what range of delays is likely to be acceptable for various values of rate of change of gain with frequency (slope), and of the maximum gain used. With a wide range of frequency-gain characteristics to test, it would have been very difficult to find enough hearing-impaired participants to test each characteristic. Hence, we adopted our previous approach of using normal-hearing participants and a simulated hearing aid and simulated hearing loss (Stone & Moore, 1999). This approach also has the advantage of reducing the influence of the high individual variability often found when using hearing-impaired participants. However, it has the disadvantage that the simulation of the hearing loss may be inaccurate or may not capture all aspects of impaired processing in the auditory system. There may also be unintended side effects of the simulation processing, as discussed in more detail later in this article. All processing was performed off-line. The intention was to simulate the effect of what a hearing-impaired person would hear when listening to an external voice.

In the first experiment, we examined the influence of the first two effects described earlier, spectral ripple and perception of an echo. To do this, the gain applied to the delayed sound was made independent of frequency, so the relative level of the direct and delayed sound was constant across frequency. The amount of delay and the gain of the delayed sound relative to the direct sound were systematically varied. All sounds were equalized in overall level, making the sounds suitable for presentation to normally hearing participants.

In the second experiment, we included the third effect, a monotonic across-frequency delay, by simulating a linear hearing aid whose gain increased with frequency. We varied the rate of change of the gain in the transition region where the gain was increasing (i.e., the slope), the maximum asymptotic gain, and the frequency at which the gain started to increase. If the gain increases rapidly, the frequency region over which there is significant spectral ripple decreases, probably reducing the disturbing effect of the ripple. However, this may be associated with some costs. First, artificial features, such as spectral edges, can be introduced into the signal. Second, the delay changes rapidly across frequency, and this may be more noticeable (and more disturbing) than a gradual change in delay with frequency. To make the sound suitable for presentation to normally hearing participants, the effect of the simulated hearing aid gain was compensated by linear digital filtering. Essentially, the filtering can be thought of as approximately simulating the attenuative component of the hearing loss (ignoring the effects of loudness recruitment). To assess the importance of the spectral ripple in the transition region, we included conditions that used additional digital filters to partially remove the ripple.

In the third experiment, we investigated the effect of delays that vary non-monotonically with frequency, as illustrated in the lower panel of Figure 1. The effect this type of delay was explored by Stone and Moore (2003b), but not for conditions simulating an OC fitting.

In experiment 3, we also took into account the fact that most modern hearing aids incorporate some form of dynamic range compression to compensate for the effects of loudness recruitment. In such aids, compression results in time-varying relative levels of the direct and delayed sounds, and this might affect subjective disturbance. In experiment 3, we simulated an OC hearing aid with multichannel, fast-acting, wide-dynamic-range compression, and followed this by a simulation of threshold elevation and loudness recruitment (Glasberg & Moore, 1992; Moore & Glasberg, 1993; Moore, et al., 1995), again making the sounds suitable for presentation to normal-hearing participants.

EXPERIMENT 1: THE PERCEPTION OF DELAY WITH FREQUENCY-INDEPENDENT GAIN

Equipment and Signal Processing

In this experiment, the gain applied to the delayed signal was independent of frequency. The signal for assessment was a 90-sec segment of continuous speech produced by a male talker, recorded in a near-anechoic environment. The recording was digitally filtered so that its spectrum matched that specified for conversational speech in the standard for calculating the speech intelligibility index (ANSI, 1997). The spectrum level was constant from 100 to 500 Hz, and decreased by 9 dB/oct above 500 Hz. The recording was resampled from 44.1 to 16 kHz, speech energy extended up to almost 8 kHz (the Nyquist frequency for the lower sampling rate). A copy of the speech file was added to itself with different combinations of delay and gain. The delay was 1, 3, 7, or 15 msec. The delay of 1 msec is at the lower end of what is currently achievable in digital hearing aids. The delay of 15 msec is slightly above the highest values currently used in digital hearing aids, but was included since more complex signal-processing algorithms, that might be implemented in the future, could lead to such a delay. The gain ranged from 0 to 20 dB in steps of 5 dB. This gave 20 combinations of delay and gain. To remove differences in overall level, all signals were scaled to have the same root-mean-square value.

Signals were replayed from precalculated files via a LynxONE™ sound card hosted inside a PC, adjusted in level by a Mackie 1202 VLZ-Pro mixing desk, and presented diotically through Sennheiser HD580 headphones. The participant was seated in a sound-isolated, double-walled chamber. The presentation level was 70 dB SPL, unweighted. All speech samples were 5-sec in duration, and were randomly selected from within the relevant 90-sec file. The speech samples were gated on and off with 0.25-sec raised-cosine ramps, to avoid clicks.

Participants

Ten participants (4 male and 6 female, aged 19–45 yrs), were selected on the basis of their having audiometric thresholds ≤15 dB HL at octave frequencies between 125 and 8000 Hz and at 3000 and 6000 Hz. All were native speakers of English, including national variants. Participants attended one session, which lasted about 1 hr after initial audiometric screening. Participants were paid for their attendance. The study was approved by the Cambridge Research Ethics Committee.

Method

Each testing session consisted of three parts. First, the experimenter played to the participant examples from the range of delays with the gain set to 0 dB. This was performed to sensitize the participant to the percept associated with the delay. Second, the participant was presented with a randomly selected example of a signal with a delay and gain, immediately followed by a signal that was the same except that the delay was set to zero. The participant was not asked to rate the signal, but was asked just to listen carefully and to notice the differences. This was repeated for several delays. The third part consisted of the testing proper.

Participants were tested using 18 blocks of 20 presentations, each block representing a randomized sequence of all conditions. Following presentation of a sample of the processed speech, the participant was asked to rate the effect of the delay, using the same seven-point scale as in our earlier studies (Stone & Moore, 1999, 2002, 2003b, 2005). For this scale, "1" corresponds to "Not at all disturbing," "4" corresponds to "disturbing," and "7" corresponds to "highly disturbing." No verbal labels were given to ratings of 2, 3, 5, or 6. For each participant, the first two blocks were treated as acclimatization or practice blocks, and results for these were discarded. The data presented here are based on 16 blocks per participant, each with a different randomization of presentation order. After eight blocks the participant was given a short break.

Statistical Analysis

Data were analyzed based on both mean values and median values across subjects. The pattern of results in terms of how ratings varied with delay and gain was essentially the same for the two cases. In what follows, we present results based on mean ratings. Also, the outcomes of analyses of variance (ANOVAs) were essentially the same whether based on each rating of each subject for each condition, or on the median rating (across repetitions) for each subject and condition. This was also true for the later experiments reported in this article. The only exceptions were for some high-order interactions. In what follows, we describe only the interactions that were significant for both types of analysis.

Results

The data for a given condition can be characterized by a triplet of values: a delay, a gain, and a disturbance rating. A graphical way of representing such triplets is by a contour plot: delay and gain form the "*x*" and "*y*" dimensions, respectively, and the disturbance ratings are represented by contours of constant disturbance, with shades of gray between contours; the darker the shading the worse the disturbance. Such a plot is shown in Figure 2. The contour lines show what combinations of level and delay produce the same mean disturbance.

As the delay increased, so did the disturbance. There was an inverse relationship between gain and disturbance: higher disturbance ratings occurred for lower values of gain. A within-subjects ANOVA was performed on the data with factors delay, gain, and repetition (block number). There were significant effects of delay [$F(3,27) = 52.7$, $p < 0.001$] and gain [$F(4,36) = 78.9$, $p < 0.001$], but no significant effect of repetition [$F(15,135) = 0.63$, $p = 0.84$]. The only significant two-way interaction was between delay and gain [$F(12,108) = 19.5$, $p < 0.001$]. Any difference greater than 0.5 scale units between two points in Figure 1 is significant at $p < 0.05$ ($t > 1.98$, df = 107.7; 2-tailed).

The decrease in disturbance with increasing gain of the delayed sound is as expected, since, for high gain values, the delayed sound dominates the direct sound, and the effects of the delay become harder to hear. For the shortest delays that are currently used in digital hearing aids, namely 2 to 3 msec, the gain needs to be 7 dB or more for the rating to be less than 3, which might be considered just acceptable.

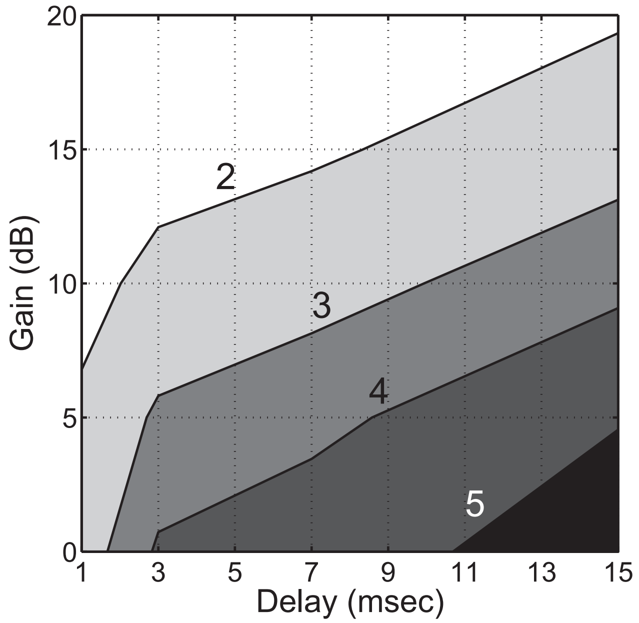

Fig. 2. Contour plot of mean ratings from experiment 1 showing contours of equal disturbance as a function of delay and hearing-aid gain.

For a delay of 10 msec, the gain needs to exceed 10 dB for the rating to be less than 3.

EXPERIMENT 2: THE PERCEPTION OF DELAY WITH GAIN INCREASING ACROSS FREQUENCY

Equipment and Signal Processing

The equipment and presentation methods were the same as for experiment 1. This experiment was intended to simulate the situation where a linear hearing aid is fitted to a person with normal low-frequency hearing but a hearing loss at high frequencies. Above an "edge" frequency, the gain of the simulated aid increased with a slope of 6, 12, or 18 dB/oct. The slope was constant until the net gain resulting from the combination of the direct sound and delayed sound (assuming power summation of the direct and delayed sound) reached an asymptotic value of 12, 18, or 24 dB. The edge frequency, defined as the frequency at which the gain of the aid was the same as for the direct path, namely 0 dB, was either 1 or 2 kHz. The edge frequency was intended to correspond to the frequency below which hearing was near normal. Below the edge frequency, the gain was reduced as rapidly as possible so as to reduce the audibility of the delayed sound in the region of "good" hearing. The frequency-dependent gains were implemented using finite-impulse-response (FIR) digital filters. Delay values of 2, 4, 7, or 12 msec were used. This range is similar to that used for experiment 1, except that the 1-msec delay was omitted, as it was anticipated that such a delay would have only a small perceptual effect.

The insertion gains were chosen from a range likely to be suitable for presentation of speech with a level of 65 dB SPL to a hearing-impaired person with a mild to moderate hearing loss. Generally, above 1 kHz, prescribed gains range from about one-third to one-half of the hearing loss (Byrne & Dillon, 1986; McCandless & Lyregard, 1983; Moore & Glasberg, 1998). The gain slopes were therefore appropriate for audiogram slopes up to about 40 dB/oct, and the gain asymptotes were appropriate for asymptotic high-frequency losses up to about 50 dB. It should be noted that the desired asymptotic gain at the Nyquist frequency of 8 kHz could not be achieved for three combinations of edge frequency and gain slope. These combinations can be expressed as a trio of edge frequency, slope, and asymptotic gain, and were 1 kHz, 6 dB/oct, 24 dB; 2 kHz, 6 dB/oct, 24 dB; and 2 kHz, 6 dB/oct, 18 dB, respectively. In practice, the first of these conditions repeated the condition (1 kHz, +6 dB/oct, +18 dB) and the remaining two repeated condition (2 kHz, +6 dB/oct, +12 dB). To balance the order of presentation and the subsequent statistical analysis, these repeated conditions were actually tested.

Because we wished to present the sounds to normal-hearing participants, the stimuli resulting from the addition of the direct and amplified/delayed sound were subjected to additional linear-phase (constant delay) digital filtering. The filters were designed using the FIR2 function in MATLAB. One form of filter, called a "gain-compensation" (GC) filter, was designed to flatten the overall frequency response, so that the overall spectral shape of the processed signal was similar to that of the original unprocessed signal. This filtering can be thought of as approximately simulating the attenuative component of the hearing loss (ignoring the effects of loudness recruitment), and its effect was to offset the frequency-dependent gain produced by the hearing aid. The GC filter had a flat response at low frequencies. The response decreased above the edge frequency and then remained constant at higher frequencies. The effect of the GC filter is illustrated in the left-hand panels of Figure 3. The dashed line in the upper panel shows the gain of the simulated hearing aid (direct plus amplified/delayed sound) before application of the GC filter, and the solid line shows the characteristic of the GC filter. The lower panel shows the response of the system after application of the GC filter. The response is globally flat, but there are distinct spectral ripples (comb filtering) that are most prominent in the spectral region of the edge frequency, but persist at higher frequencies.

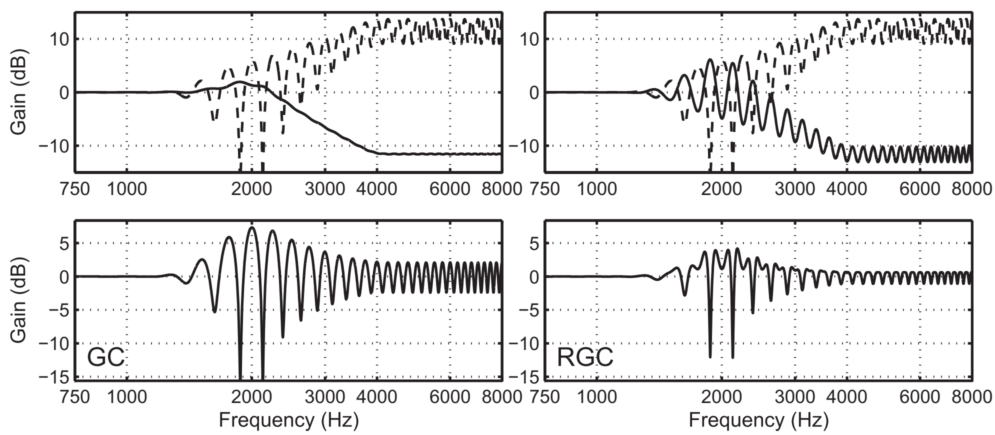

Fig. 3. Examples of the effect of the two filter types used in experiment 2. The processing conditions are edge frequency = 2 kHz, delay = 4 msec, gain slope = 12 dB/oct, and asymptotic gain = 12 dB. The left- and right-hand panels are for the GC and RGC filters, respectively. The top two panels show gain responses (resulting from combination of the direct unamplified sound and the amplified delayed sound) before compensation (dashed line) and the response of the compensation filter (solid line). The lower two panels show gain responses after application of the compensation filter.

To assess the importance of the spectral ripples, we used a second form of filter that compensated partially for the ripples in the spectrum as well as for the gain changes with frequency. We call this the "ripple plus gain compensation" (RGC) filter. This filter was similar in overall shape to the GC filter, but it had the additional effects of decreasing the gain at the peaks and increasing the gain in the valleys of the comb response. Because the valleys can be very deep, the necessary compensation can be very high. The maximum compensation for frequencies in the valleys was limited to the value of the asymptotic gain. The effect of the RGC filter is illustrated in the right-hand panels of Figure 3. The dashed line in the upper panel shows the gain of the simulated hearing aid (direct plus amplified/delayed sound) before application of the RGC filter, and the solid line shows the characteristic of the RGC filter. The lower panel shows the response of the system after application of the RGC filter. The magnitude of the ripples was reduced relative to those left by the GC filter (bottom-left panel), but some narrow ripples remained, especially in the spectral region of the edge frequency.

The response of the RGC filter varied more rapidly with frequency than that of the GC filter, and this required a filter with a longer impulse response. The impulse response of the RGC filter had a duration equal to five times the simulated aid delay, whereas the corresponding duration for the GC filter was only twice the delay. Both filters introduced some smearing in the temporal domain, and this smearing was greater for the RGC filter than for the GC filter. This should be borne in mind when interpreting the results, because the possible beneficial effects of the RGC filter in the spectral domain might be partially offset by deleterious effects in the temporal domain.

To summarize, the steps in the processing were as follows:

1. Application of an FIR filter to implement high-pass filtering and gain, simulating an open-fit linear hearing aid. The characteristics of this filter determined the edge frequency, the slope, and the asymptotic gain.
2. Application of a delay to the output of the FIR filter described in step (1).
3. Summation of the original signal and the signal produced in step (2).
4. Application of the GC or RGC filter.

Participants

Fourteen participants (8 men, 6 women, aged 19–24 yrs) were selected on the same basis as for experiment 1. Participants attended two sessions, each of which lasted about 1.5 hr. Participants were paid for their attendance.

Method

Each testing session was similar to that for the first experiment, including training. However, during the stage of training where participants compared signals with delay and gain and signals without delay and gain, they were asked to give a rating of the disturbance of the delay. A series of 36 trials was then given where single signals with delay and gain were presented and the participant was asked to rate the effect of the delay using the same seven-point scale as before. In the testing proper, participants were tested using blocks of 72 presentations, each block representing a randomized sequence of all the aid processing conditions (2 edge frequencies, 3 slopes, 3 asymptotic gains, and 4 delays). In the first session, half of the participants were tested with the GC filter and the remainder with the RGC filter. In the second session, each participant was tested with the alternative filter. The data presented are based on seven blocks per participant, each with a different randomization of presentation order. After three blocks, the participant was given a short break.

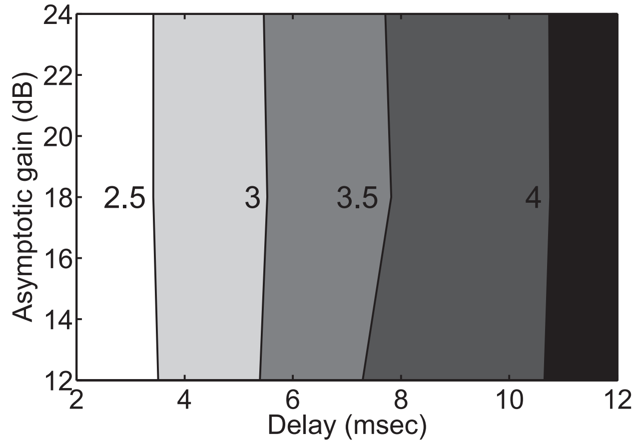

Fig. 4. Contour plot of mean ratings from experiment 2, showing contours of equal disturbance as a function of delay and asymptotic gain.

Results

The ratings were again subjected to a within-subjects ANOVA with factors delay, asymptotic gain, filter type (GC or RGC), edge frequency, slope, and repetition. Data for conditions where the asymptotic target gains were not achieved at 8 kHz were treated as "missing values." The levels of significance and the grand means for each factor are shown in Table 1.

As expected, disturbance increased significantly with increasing delay: the main increase occurred between 4 and 12 msec. In contrast to the effect of gain in experiment 1, asymptotic gain had very little effect on the disturbance ratings, as illustrated in Figure 4. A disturbance rating of 3 was reached for a delay of about 5.3 msec, regardless of asymptotic gain. The lack of an effect of gain suggests that disturbance ratings were dominated by the across-frequency delay rather than by the spectral ripples associated with comb-filtering, because increasing gain led to reduced ripple depth at higher frequencies. There was no significant effect of filter type, although we later describe a significant interaction of filter type with delay (illustrated in Fig. 5). This indicates that the compensation for the spectral ripples produced by comb-filtering did not have a large effect, consistent with the finding of Moore and Tan (2003). They showed that spectral ripples of moderate depth produced only a mild degradation in the perceived naturalness of speech signals, especially when the ripples occurred over a limited frequency range. However, the small difference between ratings for the GC and RGC filters could have occurred because the beneficial effects of the RGC filter in the spectral domain were partially offset by deleterious effects in the temporal domain.

TABLE 1. Significance levels and means for all one-way effects in experiment 2

(a) Delay: $F(3,39) = 120.8, p < 0.001$							
Delay (msec)	2	4	7	12			
Rating	2.18	2.62	3.40	4.21			
(b) Asymptotic gain: $F(2,26) = 0.2, p > 0.05$							
Gain (dB)	12	18	24				
Rating	3.11	3.10	3.11				
(c) Filter type: $F(1,13) = 2.3, p > 0.05$							
Filter type							
GC	3.03						
RGC	3.17						
(d) Edge frequency: $F(1,13) = 6.84, p = 0.021$							
Edge (kHz)	1	2					
Rating	3.04	3.17					
(e) Slope: $F(2,26) = 48.2, p < 0.001$							
Slope (dB/oct)	6	12	18				
Rating	3.47	3.00	2.84				
(f) Repetition: $F(6,78) = 3.90, p = 0.002$							
Rep	1	2	3	4	5	6	7
Rating	3.04	3.00	3.11	3.17	3.13	3.10	3.17

The ratings were significantly higher (worse), but only by about 0.1 scale units, for the 2-kHz than for the 1-kHz edge frequency. This may reflect the fact that the detection of onset or offset asynchrony in the components of complex sounds is better when the asynchronous components are centered around 2 kHz than when they are centered around 1 kHz (Zera & Green, 1993). If the across-frequency asynchrony is more detectable, it is likely to be more disturbing. Significant two-way interactions with edge frequency were associated with larger effects and will be discussed later.

Disturbance decreased significantly as the slope of the gain characteristic increased, the main

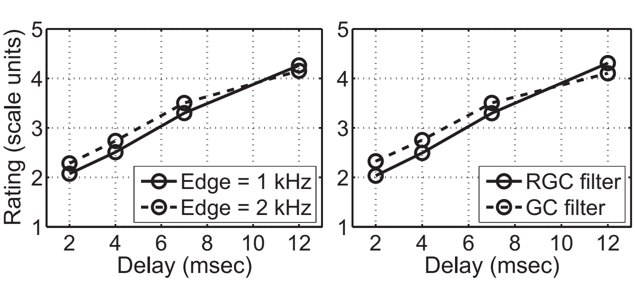

Fig. 5. Mean ratings as a function of delay for the two edge frequencies (left) and the two filter types (right).

TABLE 2. Statistics for the significant two-way interactions in experiment 2

(a) Filter type × edge frequency: $F(1,13) = 67.7$, $p < 0.001$

Filter type	Edge frequency (kHz) 1	2
GC	3.22	3.13
RGC	2.85	3.21

(b) Edge frequency × delay: $F(3,39) = 11.7$, $p < 0.001$, plotted as left-hand panel in Fig. 5

(c) Filter type × delay: $F(3,39) = 11.7$, $p < 0.001$, plotted as right-hand panel in Fig. 5

(d) Slope × delay: $F(6,78) = 3.02$, $p = 0.01$, plotted in Fig. 6

Mean ratings are given for significant interactions which are not shown in a figure.

change occurring between 6 and 12 dB/oct. This probably happened because increasing the slope decreased the frequency range over which marked comb-filtering and echo perception would have occurred.

The results of the ANOVA for significant two-way interactions are shown in Table 2 (three-way interactions were not significant in the ANOVA based on medians, and accounted for only a small proportion of the variance in the ANOVA reported here). Mean ratings are given either in the table, or in the figures that are presented below. The interaction of filter type and edge frequency reflects the fact that disturbance was lower for the RGC than for the GC filter when the edge frequency was 1 kHz, but disturbance was similar for the two filters when the edge frequency was 2 kHz. This probably occurred because spectral ripple is more disturbing when it occurs over a large frequency range than when it occurs over a limited frequency range (Moore & Tan, 2003). Hence, reducing the ripple with the RGC filter had a greater effect for the lower edge frequency. The significant interaction of edge frequency and delay is shown in the left panel in Figure 5. For the three smallest delays, disturbance was slightly higher for the 2-kHz than for the 1-kHz edge frequency, but for the longest delay, ratings were almost the same for the two edge frequencies. The significant interaction of filter type and delay is shown in the right panel of Figure 5. For the three smallest delays, disturbance was slightly higher for the GC filter, but for the longest delay this pattern reversed.

The data for the two-way interaction of slope and delay are shown in the contour plot of Figure 6. For a given delay, disturbance decreased with increasing slope, and the rate of decrease in disturbance was greater for the longer delays. A relatively high disturbance rating of about 4 occurred when a long delay (8 msec) was combined with a shallow slope (6 dB/oct), and, for this delay, there was a reduction in disturbance of about 0.75 scale units as the slope was increased to 18 dB/oct.

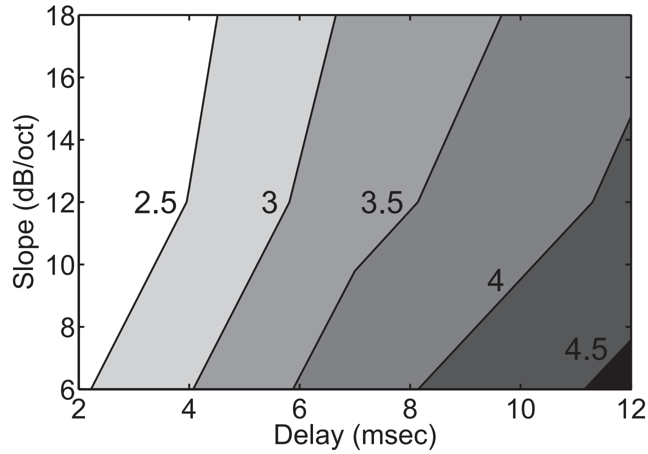

Fig. 6. Contour plot of mean ratings from experiment 2, showing contours of equal disturbance as a function of delay and gain slope.

Overall, the interaction of filter type with other factors can be understood in the following way. Longer delays are associated with more closely spaced spectral ripples, which are less well-resolved by the auditory system (Moore & Tan, 2003, 2004). Also, the depth of the ripples at high frequencies decreases with increasing asymptotic gain. Together, these two factors would lead to the ripples being less perceptually important for long delays and large asymptotic gains, which is why the reduction of the ripples using the RGC filter had little effect for this combination of factors.

In summary, the lack of an effect of gain suggests that disturbance ratings were dominated by the subjective effect of the across-frequency delay rather than by the timbre change produced by the spectral ripples associated with comb-filtering. Compensation for the spectral ripples produced by comb-filtering did have a small influence on disturbance ratings, but this occurred mainly for the 1-kHz edge frequency and the shorter delays.

EXPERIMENT 3: INCLUDING THE EFFECTS OF MULTICHANNEL FAST-ACTING COMPRESSION AND A RECRUITMENT SIMULATION

Linear hearing aids do not compensate for the loudness recruitment and reduced dynamic range that is commonly found for people with cochlear hearing loss (Moore, 2007). Multichannel, fast-acting wide-dynamic-range compression (WDRC) is one solution to this problem. In experiment 3, we examined the effect of this nonlinear aspect of hearing aid processing on the perceptual effects of delay in OC fittings. With WDRC, the gain of the aided/delayed signal relative to the direct signal is time varying. To make the processed signal suitable for presentation to normal-hearing listeners, we followed the hearing aid processing with a simulation of threshold elevation combined with loudness recruitment (Moore & Glasberg, 1993), as was done in some of our earlier work (Stone & Moore, 1999).

Hearing Aid Processing

The simulated hearing aid comprised

a. A high-pass filter at the relevant edge frequency to remove low-frequency components that did not require amplification.
b. An allpass FIR filter to produce variable delay in the hearing aid.
c. Fast-acting multichannel WDRC.
d. An FIR filter to produce the required gain as a function of frequency.

The high-pass filter (stage a) was designed as a three-pole elliptic "Infinite-Impulse-Response" filter with passband ripple of 0.25 dB, and stop-band attenuation of at least 50 dB. The response was −0.5 dB at the edge frequency, declining to −18 dB and −45 dB at 1 and 2 octaves, respectively, below the edge frequency, with a steadily increasing slope. For the filter with 1-kHz edge-frequency, the maximum group delay was 0.66 msec at 0.85 kHz, falling to 0.54 msec at 1 kHz, and less than 0.08 msec for frequencies above 2 kHz. This delay was not corrected and was included as part of the delay parameter to be investigated in the experiment. The group delays of the infinite-impulse-response filter for the 2-kHz edge frequency were half these values at the corresponding relative frequencies (i.e., 1.7, 2, and 4 kHz).

The allpass filter (stage b) was intended to approximate all the subsequent delays that would occur in a "real" hearing aid, except for a further small delay introduced by the dynamic range compression (see below). The total delay at a given frequency is the sum of two components, one fixed and one varying with frequency. The former arises from fast Fourier transformation methods and from signal conversion between the analog and digital domains. The latter may arise from frequency analysis methods that attempt to mimic the across-frequency variation of resolution present in the normal auditory system. This can be expressed mathematically as:

$$\tau_{\text{total}}(f) = (\tau_{\text{conversion}} + \tau_{\text{fixed}}) + (\tau_{\text{high-pass}}(f) + \tau_{\text{processing}}(f)) \quad (1)$$

where (f) denotes a delay that is frequency dependent. $\tau_{\text{total}}(f)$ is the total delay produced by the hearing aid between microphone and receiver, $\tau_{\text{conversion}}$ is the delay produced by the domain conversion process, τ_{fixed} is a fixed processing delay, $\tau_{\text{high-pass}}(f)$ is the delay produced by the high-pass filter at the relevant edge frequency, and $\tau_{\text{processing}}(f)$is a frequency-dependent processing delay. The value of τ_{fixed} was 0.3, 1.1, 2, 3.5, 5.5, or 8 msec.

For a typical signal-processing technique used to implement an array of filters simulating the auditory filters, the bandwidths are chosen to be proportional to the values of ERB_N, where ERB_N is the mean value of the Equivalent Rectangular Bandwidth of the auditory filter, as determined using young participants tested at moderate sound levels (Glasberg & Moore, 1990). For such a filter bank, the group delay is

$$\tau_{\text{varying}} = \boldsymbol{k}/\text{ERB}_\text{N} \quad (2)$$

where k is a constant. We chose k to range between 0.1 and 0.80 in steps of 0.14. This sort of range occurs for the two techniques mentioned in the Introduction section (Hohmann, 2002; Kates, 2005; Kates & Arehart, 2005). The range implies an analysis bandwidth ranging from very broad to nearly as narrow as would be found in the healthy auditory system. An across-frequency variation of delay is commonly used as a method of reducing the audibility of delay while providing reasonable frequency resolution at lower frequencies.

Each value of k was paired with a value of τ_{fixed}, the value of k increasing monotonically with the value of τ_{fixed}. The pairs are labeled 1 to 6, in ascending order of delay. Below 707 Hz, the delay was held constant, since there was little information at these frequencies from the aided signal after the high-pass filtering stage. Figure 7 shows the delay

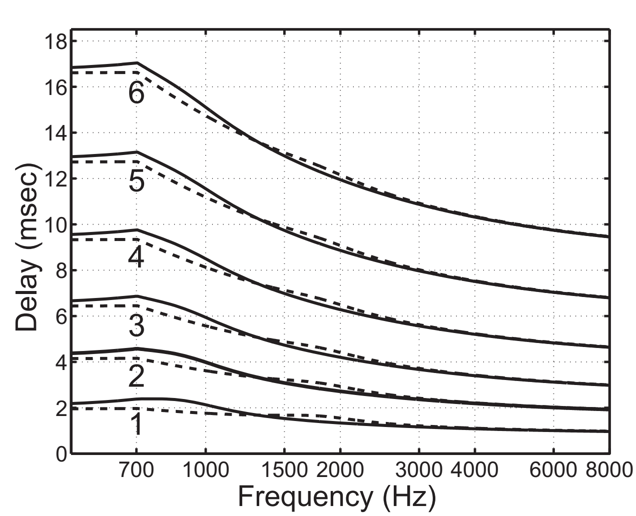

Fig. 7. Across-frequency variation of delay implemented in the simulated hearing aid used in experiment 3, for an edge frequency of 1 kHz (solid lines) or 2 kHz (dashed lines). The numbers 1–6 are used as labels for the delay settings.

as a function of frequency for each of the six delay settings, including the effect of the high-pass filter at the relevant edge frequency. The range of delays covered those typical of commercial hearing aids, and included some larger values that might occur with more complex signal processing. Over the frequency range where the simulated aid provided gain, the total across-frequency variation was less than about 5 msec, a value that Stone and Moore (2003b) found not to produce a significant reduction in consonant intelligibility. The delay produced by the filtering in stage (d) above was effectively included in the delay produced by the allpass filter in stage (b). For allpass filter i, the delay was

$$\tau_{\text{allpass}(i)}(f) = \tau_{\text{conversion}} + \tau_{\text{fixed}(i)} + \boldsymbol{k_i}/\text{ERB}_\text{N}, \quad i = 1 \text{ to } 6 \quad (3)$$

The average delay over the frequency range where the aid produced gain was about 1.5, 3, 4.5, 6, 8, and 11 msec for delay settings 1 to 6, respectively. These averages depended somewhat on whether the 1 or 2 kHz edge frequency was used.

For both edge frequencies, the compression channels were 0.5-octave wide and were centered at 1, 1.4, 2, 2.8, 4, and 5.6 kHz. The compression thresholds were 40, 35, 35, 35, 30, and 30 dB SPL, respectively. For the 2-kHz edge frequency, only the upper five of the channels had compression ratios above one. The bandpass filters used to create the channel signals were implemented as linear-phase FIR filters, and the delay introduced by the 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 −20 dB point. The attack and release times for each channel, as defined by ANSI (1996), were 5 and 35 msec, respectively. The audio signal was delayed by 0.5 msec relative to the gain control signal to reduce the amount of overshoot at sudden increases in signal level (Robinson & Huntington, 1973). This delay was included in the delay parameter shown in Figure 7.

Recruitment Simulation

The simulation had the following stages. Initially, the signal was filtered to account for the frequency-response shaping that occurs as the sound passes through the outer and middle ear to the cochlea (Moore, et al., 1997). The signal was then filtered into multiple bandpass channels with passbands broader than ERB_N. The broadening is intended to simulate the loss of independence between adjacent frequencies found in an impaired ear. Moore and Glasberg (1993) used 13 analysis filters, each three times broader than ERB_N, which is typical of the degree of broadening found in cases of severe hearing loss (Glasberg & Moore, 1986; Moore, 2007). We wanted to simulate losses less severe than those simulated by Moore and Glasberg, for which the filters would be less broad (Moore, 2007). The simulation software was rewritten to use 22 channels, each with bandwidth 2 ERB_N, but with the same relative spacing of center frequencies as used by Moore and Glasberg, namely 0.75 times the channel bandwidth. The filter center frequencies covered the range 80 to 6000 Hz, and the overall response of the filter bank was "flat" over that range.

The short-term signal level in each channel was estimated by calculating the Hilbert envelope of the signal (Bracewell, 1986; Hilbert, 1912) and low-pass filtering it with a sliding rectangular 10-msec window. The short-term level was used to compute a gain that was applied to the channel signal so as to expand the dynamic range. The amount of expansion was intended to simulate the degree of recruitment that would be expected from the hearing loss at the channel center frequency. If the short-term level in the channel was 100 dB HL or more, the gain was set to 0 dB. The applied gain was progressively reduced as the short-term level decreased below 100 dB HL. If the short-term level was equal to the absolute threshold (AT) of the simulated hearing loss in dB HL at the center frequency of the channel, the gain was set to −AT, so the signal level was transformed to 0 dB HL. Channel signals with short-term levels below AT were mapped to levels below 0 dB HL, and were therefore inaudible.

Following the application of the time-varying gains in each channel, the channel signals were added. Then, a further stage of frequency-response shaping was applied, which was the inverse of the frequency-response shaping applied at the start of the processing. This was done to allow for the fact that the signals would pass through the outer and middle ear of the participant. The resultant signal was suitable for presentation over headphones with a diffuse-field response, as used here.

The narrow-bore tubing used for OC fittings typically leads to little or no insertion gain for frequencies above about 5000 Hz. Although the direct sound is not affected by the tubing, as frequency increases, the eartip in an OC fitting starts to behave more like an earplug and to produce attenuation of high frequencies (as seen in manufacturers' data sheets). Although these effects were not explicitly taken into account in our simulations, the roll-off in the response of the filter bank above 6000 Hz, as described earlier, was intended to mimic the reduced audibility of high frequencies typically associated with OC fittings.

Conditions

For a hearing aid with WDRC, the gain at a given frequency depends on the input level. The gains discussed in this section are the gains that would occur for a signal with the same long-term average spectrum as speech at a typical conversational level of 65 dB SPL (ANSI, 1997). We initially wanted to explore the same range of gains as used in experiment 2, but this was not possible in all cases, as the effective gain rolled off above 6000 Hz. The low slope condition of +6 dB/oct would have led to maximum gains of +15 and +9 dB at 6000 Hz, for the edge frequencies of 1000 and 2000 Hz, respectively. Consequently, several of the tested conditions would have been duplicates of conditions with lower asymptotic gain. To reduce the number of such duplicates, the rates of change of gain were increased to 8, 13, and 18 dB/oct. The repeated combinations of slope and asymptotic gain expressed as a trio of edge frequency, slope, and asymptotic gain were 1 kHz, +8 dB/oct, +24 dB; 2 kHz, +8 dB/oct, +18 dB; 2 kHz, +8 dB/oct, +24 dB; and 2 kHz, + 13 dB/oct, +24 dB, respectively. The first of these was close to condition 1 kHz, +8 dB/oct, +18 dB, the next two were close to condition 2 kHz, +8 dB/oct, +12 dB, and the last one was close to 2 kHz, +13 dB/oct, +18 dB. The edge frequencies of 1000 and 2000 Hz and the asymptotic gain values of 12, 18, or 24 dB were kept as before.

In the first two experiments, only speech produced by a male talker was used. In this experiment, we used the same male-produced speech as previously [with a fundamental frequency (f_0) between 100 and 200 Hz], but also used speech from a female talker with an f_0 range of 200 to 280 Hz. Each speaker was assessed in one test session.

Compression and Expansion Ratios

Usually, the gains and compression ratios of a hearing aid would be selected based on the audiogram of the individual being fitted. Here, we started with sets of gain values, appropriate for a signal with the same long-term average spectrum as speech at a typical conversational level of 65 dB SPL (ANSI, 1997), and we used the CAMEQ fitting rationale (Moore, 2005; Moore, et al., 1999) as implemented in the "Camfit" software, to determine what hearing loss would require such a set of gain values. The CAMEQ rationale is based on the goal of giving speech at 65 dB SPL the same overall loudness as normal, and making the specific loudness pattern evoked by the speech "flat" over the frequency range from about 500 to 5000 Hz.

TABLE 3. Parts (a) and (b) show audiograms that require specific combinations of slope and asymptotic gain for speech with a level of 65 dB SPL, as determined using the CAMEQ procedure

Slope, Gain	Threshold (dB HL)							Channel compression ratio					
Center freq. (kHz)	1	1.5	2	3	4	6	8	1.4	2	2.8	4	5.6	7.3
(a) Edge frequency of 1 kHz													
8, 12	0	15	15	30	25	25	25	1.23	1.41	1.53	1.42	1.61	1.67
8, 18	0	15	15	30	35	35	35	1.23	1.41	1.53	1.55	1.81	1.90
8, 24*	0	15	15	30	35	40	45	1.23	1.41	1.53	1.55	1.91	2.03
13, 12	0	20	25	30	25	25	25	1.34	1.55	1.55	1.42	1.61	1.67
13, 18	0	20	25	40	35	35	35	1.34	1.55	1.68	1.55	1.81	1.90
13, 24	0	20	25	40	50	45	45	1.34	1.55	1.68	1.80	2.09	2.19
18, 12	0	25	25	30	25	25	25	1.42	1.55	1.55	1.42	1.61	1.67
18, 18	0	25	35	40	35	35	40	1.42	1.70	1.70	1.55	1.81	1.90
18, 24	0	25	35	50	50	50	50	1.42	1.70	1.86	1.80	2.22	2.38
(b) Edge frequency of 2 kHz													
8, 12			0	10	15	25	25		1.04	1.14	1.31	1.58	1.67
8, 18*			0	15	20	30	35		1.04	1.24	1.36	1.59	1.67
8, 24*			0	15	20	30	35		1.04	1.24	1.36	1.59	1.67
13, 12			0	20	25	25	25		1.04	1.34	1.42	1.61	1.67
13, 18			0	20	25	35	35		1.04	1.34	1.42	1.77	1.90
13, 24*			0	20	25	40	45		1.04	1.34	1.42	1.87	2.03
18, 12			0	25	25	25	25		1.04	1.38	1.42	1.61	1.67
18, 18			0	25	35	35	35		1.04	1.38	1.55	1.81	1.90
18, 24			0	25	40	50	50		1.04	1.38	1.63	2.17	2.38
(c) Threshold (dB HL)	5	10	15	20	25	30	35	40	45	50			
Expansion ratio	1.05	1.11	1.18	1.25	1.33	1.43	1.54	1.67	1.82	2.00			

The right half of the table shows compression ratios required for each hearing loss, as determined using the CAMEQ procedure, for the 0.5-octave wide channels of the WDRC hearing aid for each channel center frequency. Asterisks (*) denote combinations of slope and gain where the full gain was not achieved by 6 kHz. Part (c) shows the expansion ratios used in the recruitment simulation as a function of simulated hearing loss.

For each combination of edge frequency, gain slope, and asymptotic gain, the gain as a function of frequency was determined. We refer to the resulting gain values as "target" gains. A starting audiogram was entered into Camfit, and the audiogram value at each frequency was adjusted in 5-dB steps until the gains prescribed for 65-dB SPL speech using the CAMEQ rationale matched the target gains. The audiograms obtained in this way are shown in the left-hand part of Table 3 for each combination of edge frequency, gain slope, and asymptotic gain. The characteristics of the simulated hearing aid (channel edge frequencies and compression thresholds) were also specified in the Camfit software. The software then gave a recommended compression ratio for each channel. These ratios were implemented in the simulated hearing aid, and are shown in the right-hand part of Table 3.

The expansion ratio (E_i) in the ith channel of the recruitment simulation was based on the audiometric threshold, T_i in dB HL, at the center frequency of that channel, and was given by:

$$E_i = 100/(100 - T_i) \tag{4}$$

The gain for each channel was calibrated so that, for an input level of 100 dB HL, the output level was 100 dB HL. Effectively, this was based on the assumption that loudness recruitment was complete (loudness "caught up" with normal) when the level reached 100 dB HL, which is generally true for mild to moderate hearing losses (Miskolczy-Fodor, 1960). Values of the expansion ratio as a function of hearing level are given in section (c) of Table 3.

Participants

Sixteen participants (10 men, 6 women, aged 19–24 yrs) were selected on the same basis as for experiment 1. Participants attended two sessions, each of which lasted about 1.5 hr. Participants were paid for their attendance.

Method

Each testing session was similar to that of the first experiment, including the training. In the testing proper, participants were tested with blocks of 108 presentations, each block representing a randomized sequence of all the processing conditions. Half of the participants started with the male speaker and half with the female. In their second test session, each participant was swapped to the alternative speaker. After presentation of a 5-sec sample of the processed speech, the participant was asked to rate the disturbance produced by the delay using the same 7-point scale as described above. For each participant, the first block was treated as an acclimatization or practice block, and results for this block were discarded. The data presented are based on six blocks per participant, each with a different randomization of presentation order. After three blocks the participant was given a short break.

The equipment and presentation methods were the same as for experiments 1 and 2.

TABLE 4. Significance levels of all main effects for the data of experiment 3

(a) Delay setting: $F(5,75) = 83.9$, $p < 0.001$						
Delay setting	1	2	3	4	5	6
Rating	1.97	2.13	2.42	2.85	3.21	3.60
(b) Speaker gender: $F(1,15) = 0.62$, $p > 0.05$						
Gender	Male	Female				
Rating	2.65	2.74				
(c) Edge frequency: $F(1,15) = 83.7$, $p < 0.001$						
1 kHz	2.95					
2 kHz	2.44					
(d) Slope: $F(2,30) = 11.6$, $p < 0.001$						
Slope (dB/oct)	8	13	18			
Rating	2.71	2.62	2.77			
(e) Gain: $F(2,30) = 17.2$, $p < 0.001$						
Gain (dB)	12	18	24			
Rating	2.44	2.61	3.14			
(f) Repetition: $F(4,60) = 0.74$, $p > 0.05$						

Mean ratings are given for significant effects.

Results

The ratings were subjected to a within-subjects ANOVA with factors delay setting, speaker gender, edge frequency, slope, gain, and repetition. Again, data for conditions where the asymptotic target gains were not achieved were treated as missing values. All three- and four-way interactions were insignificant for the ANOVA based on medians, so these will not be discussed further. The levels of significance and the means for each level of each factor are shown in Table 4. As expected, the disturbance ratings increased with increasing delay setting, and ratings exceeded 3 for delay settings 5 and 6. The effects of edge frequency and gain were opposite to those found in experiments 1 and 2, disturbance being greater for the lower edge frequency and the higher gains. Although the effect of slope was significant, the effect was very small, amounting to only 0.1 scale units.

The data for the significant two-way interactions are shown in Table 5. Where one of the factors was assessed only at two levels, the means are tabulated. Some of the significant interactions involved a change in the mean rating less than 0.5 scale units and will not be discussed further.

The interaction of edge frequency and delay setting reflects the fact that edge frequency had a greater effect for the longer delays. A disturbance

TABLE 5. Statistics for all significant two-way interactions for the data of experiment 3, together with selected means

(a) Speaker gender × edge frequency: $F(1,15) = 13.8$, $p = 0.002$						
Edge frequency (kHz)	1	2				
Male	2.83	2.48				
Female	3.08	2.40				
(b) Edge frequency × delay setting: $F(5,75) = 24.7$, $p < 0.001$, plotted in Fig. 8						
Delay setting	1	2	3	4	5	6
Edge frequency (kHz)						
1	2.08	2.29	2.60	3.10	3.56	4.08
2	1.87	1.97	2.24	2.60	2.87	3.11
(c) Speaker gender × slope: $F(2,30) = 5.17$, $p = 0.012$						
Slope (dB/oct)	8	13	18			
Male	2.71	2.56	2.69			
Female	2.71	2.67	2.85			
(d) Edge frequency × slope: $F(2,30) = 10.8$, $p < 0.001$						
Slope (dB/oct)	8	13	18			
Edge frequency (kHz)						
1	2.90	2.89	3.07			
2	2.52	2.35	2.46			
(e) Slope × gain: $F(3,45) = 16.1$, $p < 0.001$						
(f) Edge frequency × gain: $F(2,30) = 6.01$, $p = 0.006$						

rating of 3 was exceeded for delay settings 4, 5, and 6 for the 1-kHz edge frequency, but this happened only for delay setting 6 for the 2-kHz edge frequency. The nature of this interaction is illustrated in Figure 8.

In summary, the disturbance ratings increased progressively with delay setting and exceeded 3 for setting 5. Disturbance was greater for the lower edge frequency and the effect of edge frequency increased with increasing delay. In contrast to experiment 1, disturbance increased with increasing gain. Possible reasons for this are discussed below.

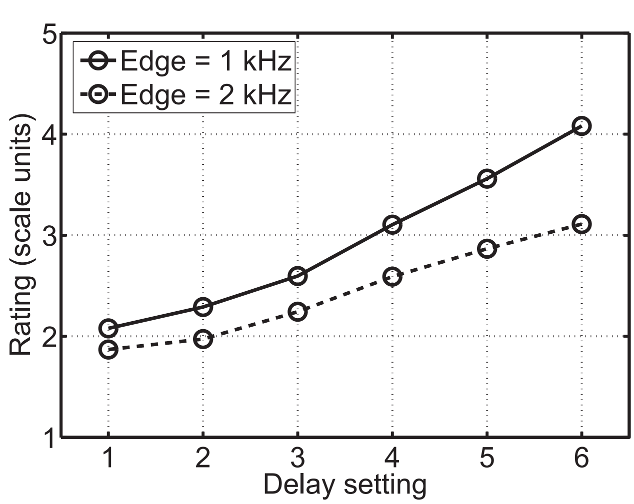

Fig. 8. Mean ratings from experiment 3 plotted as a function of delay for edge frequencies of 1 kHz (solid line) and 2 kHz (dashed line).

Discussion

Reasons for the Discrepancies Across Experiments: Effects of Gain and Edge Frequency

In all three experiments, disturbance increased monotonically with increasing delay. However, in experiments 1 and 2, disturbance decreased with increasing asymptotic gain, whereas in experiment 3 disturbance increased with increasing asymptotic gain. Also, in experiment 2 disturbance increased with increasing edge frequency and with decreasing slope, whereas in experiment 3 disturbance decreased with increasing edge frequency and hardly changed with slope. Thus, the pattern of results was very different for experiments 2 and 3.

The main difference between experiments 2 and 3 is that the signals in experiment 3 were subjected to dynamic processing, both in the simulated WDRC hearing aid and in the recruitment simulation. Ideally, the two dynamic processes would cancel each other out. In practice complete cancellation did not occur for several reasons:

a. The compression channels in the simulated hearing aid were different from, and fewer in number than, the channels used in the recruitment simulation.
b. The compression ratios and gains in the simulated hearing aid were different from the expansion ratios and attenuations used in the recruitment simulation. CAMEQ, like most

other fitting procedures, does not aim to compensate completely for the hearing loss.

c. The time constants of the WDRC were longer than those used in the recruitment simulation. We did not wish to use even shorter time constants in the simulated hearing aid, as these can lead to intermodulation distortion and poor sound quality.

For similar reasons, WDRC processing in a real hearing aid does not (and cannot) completely compensate for the dynamic effects produced by loudness recruitment. The latter acts like a very fast-acting expander, so that amplitude-modulated sounds seem to be more modulated for an impaired than for a normal ear; this effect occurs for modulation rates up to at least 32 Hz (Moore, et al., 1996). On the other hand, most WDRC hearing aids do not reduce amplitude modulation depth for rates exceeding a few Hertz (Braida, et al., 1982; Moore, et al., 2001; Stone & Moore, 1992, 2003a). It seems likely that the deleterious effects of increasing gain found in experiment 3 occurred because greater gains were associated with larger simulated hearing losses (see Table 3). These larger simulated losses were associated with greater side effects of the dynamic processing, resulting from the imperfect match between the dynamic aspects of the processing used to simulate the hearing aid and the processing used to simulate threshold elevation and loudness recruitment. Similarly, one would expect that the perceived quality of speech subjected to WDRC would worsen with increasing real hearing loss, partly as a result of the increasing mismatch between the dynamic aspects of the WDRC and the dynamic aspects of recruitment. This effect would occur independently of any disturbing effects of hearing-aid delay. Hence, the finding of experiment 3 that disturbance increased with increasing gain should not be taken as indicating that the effects of delay per se increase with increasing gain. Rather, the disturbing effects of delay probably decrease with increasing gain, as shown in experiments 1 and 2.

A similar argument can be made about the differing effects of edge frequency across experiments 2 and 3. In experiment 3, the mean disturbance was greater for the edge frequency of 1 kHz than for the edge frequency of 2 kHz. This probably happened because the side effects of the dynamic processing, as discussed earlier, extended over a greater frequency range for the lower edge frequency, and were therefore more disturbing. When there was no dynamic processing, as in experiment 2, disturbance was slightly greater for the edge frequency of 2 kHz, probably because the discriminability of across-frequency delays is better around 2 kHz than around 1 kHz (Zera & Green, 1993). It seems likely that the disturbing effects of delay per se are greater for an edge frequency of 2 kHz than for an edge frequency of 1 kHz.

Because of the problems associated with the dynamic aspects of the processing to simulate the hearing aid and the effects of loudness recruitment, it would be desirable to assess the effects of delay in OC fittings using "real" hearing-impaired participants with a range of hearing losses and using WDRC hearing aids. Unfortunately, our previous work on the perception of across-frequency delay by hearing-impaired participants (Stone & Moore, 2003b) does not allow any insight into how the effects of delay are influenced by degree of hearing loss or hearing aid gain, because all participants were selected to have moderate gently sloping hearing losses.

Effects of Fixed Delay and Across-Frequency Delay

Insight into the relative importance of overall delay and across-frequency delay can be gained by comparing the results of experiments 1 and 2 (see Figs. 2 and 4). The conditions in the two experiments were similar, except that in experiment 1 the delay did not vary across frequency (because the gain did not vary across frequency), whereas in experiment 2 it did; below the edge frequency, the undelayed direct sound was more audible, whereas above the edge frequency the delayed aided component was more audible. Some caution is needed here, because Figure 4 shows data collapsed across different slopes of the transition band. However, it seems clear that, for a given delay and for gains greater than about 10 dB, disturbance ratings are higher for experiment 2 than for experiment 1. For example, for a gain of 12 dB, a delay of about 13 msec led to a disturbance rating of 3 in experiment 1, wheras for an asymptotic gain of 12 dB that same disturbance rating was reached in experiment 2 when the delay was only about 5.3 msec. Comparing for an equal delay of 5 msec and an equal gain of 13 dB across the two experiments, the mean disturbance rating was about 2 for experiment 1 and about 2.8 when there was an across-frequency delay. These comparisons indicate that the across-frequency delay clearly had a disturbing effect.

Effect of Gain Slope

The results of experiment 2 showed that, for a given delay, disturbance increased with decreasing slope of the gain (Fig. 6). This effect may be partly a consequence of the width of the transition band; as the slope increased, the width of the transition band decreased, reducing the frequency range over which comb-filtering and echo effects would be audible. The results suggest that the slope in the transition band should be at least 10 dB/oct.

Effects of Spectral Ripple

Disturbance was lower for the RGC filter, which partially removed spectral ripples, than for the GC filter when the edge frequency was 1 kHz, but disturbance was similar for the two filters when the edge frequency was 2 kHz. This may reflect the fact that spectral ripple is more disturbing when it occurs over a large frequency range than when it occurs over a limited frequency range (Moore & Tan, 2003). Hence, reducing the ripple with the RGC filter had a greater effect for the lower edge frequency. However, even for low delays and the 1-kHz edge frequency, the effect of filter type was only about 0.5 scale units. Again, it should be noted that the potential benefit of the RGC filter might have been partly offset by the (unavoidable) deleterious side effects of temporal smearing introduced by that filter.

Comparison With Earlier Work

The effects of gain, gain slope, and frequency differed across experiments 2 and 3, and this was attributed to the mismatch between the dynamic processing in the simulated WDRC hearing aid and in the recruitment simulation used in experiment 3. A hearing aid simulation combined with a simulation of recruitment was also used by Stone and Moore (1999). Their Figure 4 shows a contour plot representation of data gathered in a similar way to experiment 3 here. In contrast to the results of experiment 3, their results showed decreasing disturbance with increasing simulated hearing loss. The difference may be due to several factors:

1. They simulated the disturbing effects when a user of a hearing aid talks, and hears their own voice through several paths. This meant that the disturbing effects mainly originated from low frequencies in the signals. In our experiment 3, the disturbing effects originated mainly from frequencies above the edge frequency of 1 or 2 kHz.
2. Their simulated hearing aid used a faster compression system (but with fewer channels) than the one used here. This aid might have more effectively compensated for the dynamic effects introduced by the recruitment simulation.
3. Their recruitment simulation used channels that were 3-ERB_N wide, as opposed to the 2-ERB_N wide channels used here. This would have reduced the independence between different frequency regions, perhaps reducing the dynamic effects introduced by the recruitment simulation.

It remains unclear which of these factors is most important.

Tolerable Delays

If we choose scale point "3" as indicating the maximum acceptable delay (since 4 corresponds to "disturbing"), then

a. When the hearing aid gain is independent of frequency (experiment 1), delays longer than 10 msec are not acceptable unless the aid provides at least 10 dB of gain. However, OC fittings would not normally be used for cases that required uniform amplification across frequency.
b. When the gain varies monotonically across frequency (experiment 2), delays greater than about 5 msec are not likely to be acceptable when the edge frequency at which the gain starts to increase is 2 kHz. When the edge frequency is 1 kHz, the largest acceptable delay is about 6 msec. The maximum acceptable delay increases slightly when the slope of the gain function is moderate to high.

Certain combinations of parameters led to high disturbance ratings, which are "hidden" when the data are averaged across one of the parameters. These parameter combinations can be regarded as "no-go" areas. Parameter combinations for which mean ratings reached or exceeded 4 (disturbing) were as follows:

i. Experiment 2: All gains when the delay was about 12 msec or greater.
ii. Experiment 2: All delays exceeding 8 msec when using a gain slope of 6 dB/oct. However, the disturbance decreased as the gain slope increased. Delays above 12 msec received ratings below 4 if the gain slope exceeded 15 dB/oct.
iii. Experiment 3: All gains when using the male speaker, 1-kHz edge frequency, delay setting 6, and slopes of 13 and 18 dB/oct.
iv. Experiment 3: All slopes when using the female speaker, 1-kHz edge frequency, delay setting 5, and gain of 24 dB.
v. Experiment 3: All slopes and all gains when using the female speaker, 1-kHz edge frequency, and delay setting 6.

The no-go areas found in experiment 3 may reflect side effects of the dynamic processing rather than effects of the delay per se. However, the no-go areas found in experiment 2 probably do reflect an effect of the delay, and indicate that a shallow slope of the gain should be avoided unless the hearing aid delay is very short.

CONCLUSIONS

Hearing aid delays up to about 5 to 6 msec are likely to be acceptable when the gain (and hence the delay) is constant across frequency and when the gain is 10 dB or more. However, in a typical OC fitting, the gain varies across frequency, being 0 dB at low frequencies and increasing above an edge frequency. Under these conditions, the maximum acceptable delay is about 6 msec for an edge frequency of 1 kHz and 5 msec for an edge frequency of 2 kHz. The across-frequency variation in delay does seem to have a disturbing effect as, for comparable gains, disturbance ratings were higher when the gain varied across frequency than when it was fixed.

A problem when using a simulated hearing aid with WDRC, followed by a simulation of recruitment (as in experiment 3), is that the dynamic effects of the time-varying gain produced by the WDRC are not (and cannot be) exactly compensated by the expansion implemented in the recruitment simulation. As a result, there are disturbing dynamic side effects that increase as the simulated hearing loss increases. This makes the interpretation of the results ambiguous, as the disturbing effects of the delay are confounded with the disturbing side effects of the simulation. It seems likely that most real hearing aids, with their limited compression speed, do not compensate fully for the dynamic effects of loudness recruitment (Moore, et al., 1996), and that the disturbing effects of this increase with increasing hearing loss. This may make the hearing aid user more tolerant of the disturbing effects produced by delay in the hearing aid.

The results of experiment 2 suggest that the spectral ripples introduced by the addition of the delayed/amplified sound to the direct sound have a relatively small perceptual effect. The ripples are more salient when the edge frequency is low, but are less perceptually important for long delays and large asymptotic gains.

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