1pPPb1. Effects of compression on the use of onset time differences to detect one tone in the presence of another

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It is easier to hear one of two notionally "simultaneous" tone complexes if the onset of the masker complex is delayed relative to that of the signal. However, the ability to use onset asynchrony as a cue may be reduced when using amplitude compression, due to distortion of the onset of sounds (overshoot effects). We assessed how fast- and slow-acting five-channel compression affects the ability to use onset asynchrony to detect one (signal) complex tone when another (masking) complex tone is played almost simultaneously. A 2:1 compression ratio was used with normal-hearing subjects and individual compression ratios and gains recommended by the CAM2 hearing aid fitting method were used for hearing-impaired subjects. For the normal-hearing subjects, performance improved with increasing onset asynchrony in all conditions. The improvement was greatest with fast compression and least with no compression. Preliminary results for the hearing-impaired subjects indicate smaller but similar effects of onset asynchrony and a greater benefit of compression. The benefit of compression probably occurs because compression increases the level of the part of the signal that occurs before the masker relative to the masker. Supported by Starkey (U.S.A.) and the MRC (UK).

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INTRODUCTION

When listening to or making music it is often desirable to be able to hear out individual musical lines. This is likely to be more challenging for hearing-impaired than for normal-hearing listeners, because of their reduced frequency selectivity and other deleterious effects of hearing loss. One thing that might facilitate the segregation of simultaneous musical sounds is differences in onset time; Rasch (1978) showed that onset time differences between notionally simultaneous tones make it easier to detect the tone with the earlier onset, even if it is not consciously perceived as occurring earlier in time.

Stone and Moore (2003; 2004) showed that fast-acting dynamic range compression, which is often used in hearing aids to compensate for the effects of loudness recruitment, makes it harder to hear one talker in the presence of another for speech processed to retain mainly envelope cues in different frequency bands. They attributed this to “cross-modulation”. A similar effect might occur for music, making it harder to hear out individual melodic lines. Also, compression may make it more difficult to use onset time asynchronies due to distortion of the envelope shape at onset caused by overshoot effects. On the other hand, compression might improve the ability to use onset asynchrony to detect a weak signal tone occurring just before a masker tone, by increasing the level of the signal relative to that of the masker.

The aim of this study was to explore how hearing impairment and the use of dynamic range compression affect the ability to use cues from onset time differences to detect one tone in the presence of another. Results are presented from a completed study with normal-hearing subjects and from an ongoing study with hearing-impaired subjects.

METHOD

Subjects

Five normal-hearing subjects (thresholds less than 20 dB HL over the range 0.25 to 8 kHz) between 23 and 36 years of age and six subjects with a moderate hearing loss and between 62 and 79 years of age were tested. Hearing-impaired subjects were required to have a mean audiometric threshold across 1, 2, and 4 kHz between 40 and 60 dB HL and a maximal slope of the audiogram of 60 dB/octave. The mean audiometric threshold across these frequencies for the participants included here was between 41 and 48 dB HL.

Stimuli

The two stimuli in each trial both included a “low” complex tone (the masker) with a fundamental frequency of 220 Hz, A3. One of the stimuli also contained a “high” complex tone (the signal) with a fundamental frequency of 622.25 Hz. The two stimuli are shown in musical notation below.

![Musical notation of stimuli](image)

Figure 1: The notes of the two stimuli used in this study. One stimulus contained a single complex tone (the masker) while the other also contained a second complex tone with a higher fundamental frequency (the signal).
The signal contained all harmonics up to 5000 Hz. The masker contained all harmonics up to 8000 Hz to ensure that detection of the signal did not depend on harmonics with very high frequencies. The spectral slope was \(-9\) dB/octave for the masker and \(-6\) dB/octave for the signal. The only exception was that the level of the fundamental component of the signal was reduced by 5 dB to avoid the signal being detected via its fundamental component. The duration of the signal was 200 ms. The two tones always ended together but the onset of the masker tone was delayed by 0 or 40 ms relative to that of the signal. Both tones had rise and decay times of 20 ms. The inter-stimulus interval was 400 ms.

All stimuli were processed through a hearing-aid simulator, as described below. The stimuli for the normal-hearing subjects were scaled so that the output of the hearing aid simulator in response to the 200-ms version of the masker alone was 65 dB SPL in each of the compression conditions. The stimuli for the hearing-impaired subjects were scaled so that the input level to the hearing aid simulator was 65 dB SPL for the 200-ms masker alone. The output level depended on the compression settings, which were chosen individually for each subject, as described below. The threshold for detecting the signal was expressed as the level of the signal relative to that of the masker at the input to the simulator.

**Hearing-aid simulator**

A hearing aid simulator with a five-channel compression system (Moore et al., 2010a) was used to process the stimuli. Three conditions were used: linear, fast compression and slow compression. The gains for the linear condition were those prescribed by the CAM2 procedure (Moore et al., 2010b) for a speech-shaped noise with a level of 65 dB SPL. The attack/release times were 10/100 ms for the fast compression condition and 50/3000 ms for the slow compression condition. Compression thresholds were 53, 36, 31, 24, and 16 dB SPL, in order of increasing channel center frequency. A compression ratio of 2:1 was used for the normal-hearing subjects and individual compression ratios and gains recommended by the CAM2 hearing aid fitting method (Moore et al., 2010b) were used for the hearing-impaired subjects.

**Experimental procedure**

A 2-AFC, 3-up 1-down experimental procedure with the level of the signal as the adaptive parameter was employed. The step size was initially 10 dB. It was reduced to 5 dB after a reversal from increasing to decreasing level, and then to 2 dB after one more such reversal. Each trial included six reversals after the minimum step size had been reached and threshold was taken as the mean signal level at the last six reversals. Each condition was tested twice for each subject in at least three sessions. The number of sessions depended on the consistency of the data. For each compression condition, a paired t-test was used to test the consistency of thresholds from same onset-time condition within each session and to test the consistency of mean thresholds from same onset time condition across sessions. More specifically, paired t-tests were used to compare pairs of thresholds from the same onset time condition within each session and pairs of mean thresholds for same condition from the first and second sessions, the second and third sessions, and the first and third sessions. The thresholds were judged to be consistent if they did not differ significantly within and across sessions. All t-tests were performed using a significance level of 1\% to provide partial correction for multiple comparisons. Only consistent thresholds from three sessions were included here.

Stimuli were played via a M Audio Delta soundcard with 16-bit resolution and 22.05 kHz sampling frequency, through Sennheiser HD 580 headphones in a sound-attenuating chamber. The stimuli were presented monaurally to the ear of the subject that had the lowest mean audiometric threshold across 500, 1000, 2000, and 4000 Hz. Stimulus presentation and response collection were controlled using the AFC software package (developed by Stephan Ewert at University of Oldenburg and Technical University of Denmark) under Matlab.
RESULTS

Figure 2: The left and right panels show the mean thresholds for the normal-hearing and hearing-impaired subjects, respectively. Error bars represent ± one standard deviation after normalizing the data such that the mean threshold was the same for all subjects.

Mean thresholds for each condition for the normal-hearing and hearing-impaired subjects, are shown in the left and right panels of figure 2, respectively. There was a clear effect of both onset asynchrony and compression for the normal-hearing subjects; thresholds were lower (better) with the 40-ms onset delay and thresholds for the 40-ms onset delay decreased when compression was applied. Also, there was a benefit of fast over slow compression for the 40-ms onset delay. The results for the hearing-impaired subjects show similar trends but no difference between fast and slow compression and a weaker effect of onset asynchrony. The latter is especially apparent for the linear condition, for which there is very little difference between the mean thresholds for the two onset times.

DISCUSSION

The improvement in performance with onset asynchrony seen for normal-hearing subjects for the linear condition is consistent with the results of Rasch (1978). Only a small effect of onset asynchrony occurred for the hearing-impaired subjects in the linear condition, indicating that they were less able than the normal-hearing subjects to detect or make use of the brief portion of the signal occurring before the masker. Compression improved the ability to use cues from the onset asynchrony for both the normal-hearing and hearing-impaired subjects. This improvement can be explained by the fact that the compression had the effect of amplifying the part of the signal that occurred before the masker relative to the level of the signal, i.e., the target-to-masker ratio was improved, hence improving detection of the signal. This effect apparently outweighed any deleterious effects of the compression produced by cross-modulation and overshoot effects. Perhaps surprisingly, compression speed did not affect the ability to use onset asynchrony for the hearing-impaired subjects, while it did for the normal-hearing subjects; the latter performed better with fast than with slow compression. We are conducting further analyses to determine why this difference across groups occurred.

It should be noted that there was a considerable difference in mean age between the hearing-impaired and the normal-hearing subjects. Hence, some of the differences between groups may be related to age rather than hearing loss. However, it was deemed reasonable to use elderly hearing-impaired subjects for the purposes of this study, since the majority of hearing aid users are elderly.
In conclusion, the results show that multi-channel dynamic range compression, as used in hearing aids, improves the ability to use cues from an onset time asynchrony when the signal precedes the masker, and this might help hearing-aid users to hear out individual melodic lines of music.

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REFERENCES