MULTIBAND COMPRESSION AND CONTRAST-ENHANCING FREQUENCY SHAPING IN HEARING AIDS

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1. INTRODUCTION

Contrast-enhancing frequency shaping (CEFS) produces a better representation of formants in the auditory nerve (AN) response of an impaired ear than conventional amplification schemes (Miller et al., 1999; Sachs et al., 2002). CEFS compensates for the broadened tuning curves and elevated thresholds of an impaired ear by adjusting the relative amplitudes of the formants without distorting the spectral envelope between the formants. Multiband compression, on the other hand, has been utilized in hearing aids to compensate for the reduced dynamic range of the impaired ear.

We have previously shown that multiband compression and CEFS amplification can work together when used in series without counteracting one another (Bruce, 2004), in contrast to previous spectral enhancement schemes that are not compatible with multiband compression (Franck et al., 1999). In this paper we describe the combination of CEFS amplification and multiband compression in a single frequency-domain filterbank implementation, thus reducing the computational complexity and the signal delay. The CEFS gain-frequency response has also been improved to give a better neural representation of F2 and F3. This new multiband-CEFS (M-CEFS) algorithm is evaluated with models of the normal and impaired ears (Bruce et. al., 2003) and compared to linear amplification, multiband compression and CEFS without compression.

2. METHOD

As illustrated in Fig. 1, the M-CEFS algorithm utilizes a formant tracker (Mustafa and Bruce, 2004) to direct the gain-frequency response of a time-varying filter. The gain in each frequency band also depends on the short-term input signal energy in that band (to apply multiband compression) and the hearing-loss profile (to tailor the algorithm to the audiogram of a hearing aid user).

Compression was realized using a filterbank of 15 filters spaced at 1/3-octave, starting at 250 Hz. Filter bandwidths were approximately 2 equivalent rectangular bandwidths. A sampling frequency of 16 kHz was used. Details of the FFT-based implementation of the filterbank can be found in Bruce (2004).

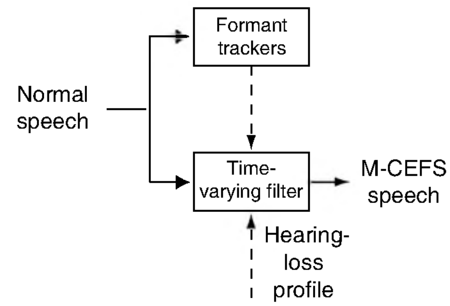

Fig. 1. Schematic of the M-CEFS amplification system.

Compression was applied independently in each frequency band of the filterbank based on the short-term input power in that band. The compression knee point was 40 dB SPL, above which a compression ratio of 2:1 was applied. The gain in each frequency band was adjusted dynamically to give a near-instantaneous attack and a release time of approximately 60 ms.

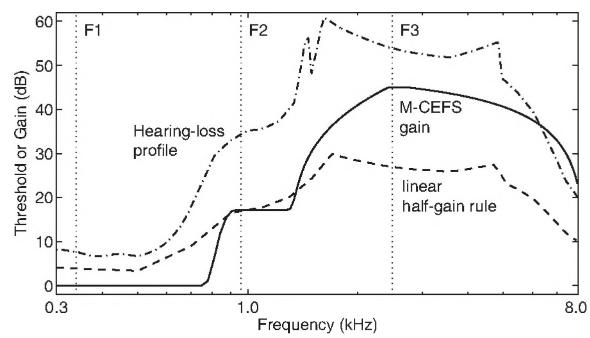

Fig. 2. Example gain-frequency response. The dot-dashed line gives the hearing-loss versus frequency profile used in the impaired auditory-periphery model. The dashed-line shows a linear gain-frequency response following the "half-gain rule." The solid line gives the M-CEFS gain-frequency response for the formant values F1, F2 and F3 indicated by the vertical dotted lines.

In addition to the gain adjustments made in each band to produce compression, M-CEFS amplification was applied using gain adjustments as a function of the current formant estimates and the hearing loss profile. An example gain-frequency profile is given in Fig. 2, where a comparison is made to a linear amplification scheme based on the "half-gain rule" (Dillon, 2001). The M-CEFS gain-frequency response has the same gain as the linear scheme at the F2 frequency and is similar on average to the linear gain profile up to just below the F3 frequency. However, the M-CEFS gain is shaped to increase the contrast of the formants, and at F3 and above the M-CEFS gain is substantially higher than the linear gain. In contrast, the original CEFS gain-frequency response only applied contrast enhancement between F1 and F2 (Miller et al., 1999; Bruce, 2004).

The test speech signal used in this paper was the synthesized sentence 'Five women played basketball' (courtesy of R McGowan of Sensimetrics Corp, Somerville, MA). Using the auditory-periphery model of Bruce et. al. (2003), the neural representation of this sentence was evaluated via the short-term average discharge rate and the short-term synchronized rate versus time for a population of AN fibers.

3. RESULTS

Spectrograms of the original sentence, the linear-amplified sentence and the M-CEFS amplified sentence are show in Fig. 3. The linear amplification scheme applies the most gain in the F2 and F3 frequency region (~ 1–3 kHz), which helps prevent impaired model AN fibers (with elevated and broadened tuning curves) in this region from responding erroneously to F1. However, the contrast between the formants is not enhanced, and consequently the tonotopic representation of F2 and F3 are not correctly restored in the short-term average discharge rates and the synchronized rates of these impaired fibers.

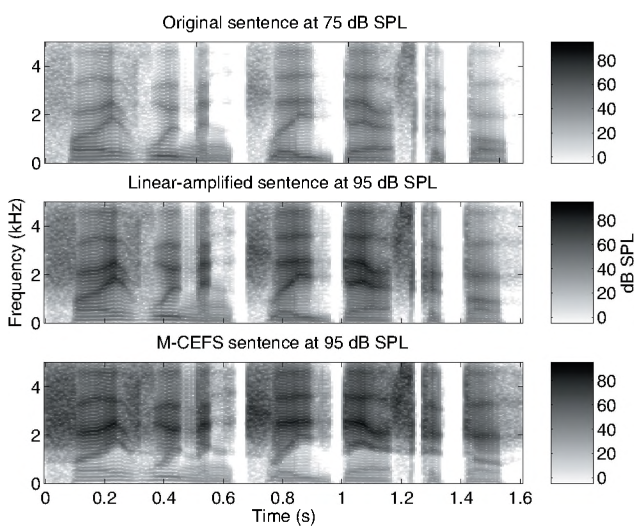

Fig. 3. Spectrograms of the original sentence (top panel), the linear-amplified sentence (middle panel) and the M-CEFS amplified sentence (bottom panel).

In comparison, the M-CEFS sentence (processed at an input level of 75 dB SPL) exhibits increased contrast between the formants. Particularly evident is the contrast at F2 as the formant tracker successfully adjusts the M-CEFS gain-frequency profile as the F2 frequency changes over time. Increased enhancement between F2 and F3 is also achieved. The resulting impaired model AN fiber responses have a more normal tonotopic representation of F2 and F3 than is produced by the linear-amplification gain-frequency response or the original CEFS scheme.

4. DISCUSSION

In this paper we have presented a hearing-aid amplification scheme that uses a formant-tracking algorithm and time-varying filter to produce improved contrast enhancement between formants. This contrast enhancement is not degraded by the application of multiband compression. The amplification scheme was incorporated into the same FFT-based filterbank implementation of the compression algorithm, thereby reducing the signal delay. The currently version has an average signal delay of 16 ms, which is somewhat larger than is desirable for a hearing aid speech processing scheme, so we will investigate if the delay can be reduced further without degrading performance. The modeling results indicate that M-CEFS should produce a better representation of formants in hearing aid users; human testing will be conducted to evaluate the actual performance.

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AUTHOR NOTES

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