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Compressing critical bands for digital hearing aids

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We discussed an algorithm for hearing aids to compensate for the frequency selectivity of hearing impairments. We focused on the wider critical band of hearing-impaired people and proposed a method of signal processing in which a speech signal is processed to reduce interference between adjacent frequency bands. Two approaches were tested; in each case, a signal was compressed toward the center of each critical band along the frequency axis.

The first approach was based on a filter bank with a set of bandpass filters. Step 1: An input signal was passed through an FIR filter at each of 21 critical bands, and transformed into the Hilbert envelope, which contains the most essential acoustic information. Step 2: The same input signal was also passed through another FIR filter at each of 21 critical bands whose bandwidth ranged from 50 up to 90% compared to the original bandwidth. Then the carrier component, the bandpassed signal divided by its envelope, was computed. Step 3: The product of the Hilbert envelope obtained from Step 1 and the carrier component obtained from Step 2 was taken for each band. Step 4: Finally, a summation of all the outputs from the 21 bands was calculated.

The second approach was based on the fast Fourier transform (FFT). Step 1: An input signal was transformed from the time domain to the frequency domain by the FFT. Step 2: The amplitude and the phase spectra of the FFT were calculated. Step 3: The amplitude spectrum compressed toward the center of each critical band along the frequency axis was computed for each band. The compression rate ranged from 50 up to 90%. Step 4: The partially compressed amplitude spectrum was multiplied by the original phase spectrum to resynthesize a band-limited signal. Step 5: Finally, the overlap add technique was applied to the IFFT of the product from Step 4.

To evaluate the processed signals, a perceptual experiment was conducted with five severely hearing-impaired subjects. The subjects listened to three types of stimuli: the original and the processed speech by the two approaches. Then the subjects evaluated the intelligibility with a five-point scale.
The result was that subjects rated best those tokens for which the compression rates were 70% for the filter-bank approach and 60% for the FFT approach. We also discussed the applicability of our algorithm for a real-time processing system with a DSP.