N. Tiwari / Supervisor: Prof. P. C. Pandey, "Dynamic range compression and noise suppression for use in hearing aids," *Ph.D. Thesis*, Department of Electrical Engineering, Indian Institute of Technology Bombay, June 2019.

Abstract

Sensorineural hearing loss is associated with elevated hearing thresholds, reduced dynamic range, and increased temporal and spectral masking, resulting in degraded speech perception, particularly in noisy environments. The research objective is to develop signal-processing techniques for dynamic range compression and background noise suppression to enhance the performance of hearing aids for the listeners with sensorineural loss, with considerations for low computational requirements, low audio latency, and low perceptible distortions.

A dynamic range compression technique, named as 'sliding-band compression' (SLBC), is proposed to overcome the shortcomings of the single-band and multiband compressions. The gain for each spectral sample is based on the short-time power in an auditory critical band centered at its frequency. The technique avoids the attenuation of high-frequency components in the presence of strong low-frequency components, which may occur in single-band compression. Further, it avoids distortions in the shape of spectral resonances and discontinuities during the resonance transitions, which may occur in multiband compression.

Two techniques for quantile-based noise estimation are developed for single-input speech enhancement. A technique is proposed for dynamic tracking of quantiles of a data stream, without storage and sorting of the past samples and without prior knowledge of the distribution. The quantile is estimated recursively by applying an increment, calculated as a fraction of the dynamically estimated range, such that the estimate converges to the sample quantile. This technique is applied for the tracking of the quantiles of the spectral samples of the noisy speech spectrum for noise spectrum estimation without voice activity detection, resulting in the technique named as 'dynamic quantile tracking based noise estimation' (DQTNE). For improving the tracking of nonstationary noises, the technique named as 'adaptive dynamic quantile tracking based noise estimation' (ADQTNE) and using an adaptive quantile and an adaptive convergence factor is proposed. The two techniques are evaluated and compared with some of the existing techniques in terms of computational requirement and noise tracking and in a speech enhancement framework using spectral subtraction based on the geometric approach. Considering residual noise and speech attenuation together, ADQTNE provides the highest quality output. DQTNE, having the lowest computational requirement, provides a similar output, except that the output has a higher residual noise in case of low SNRs. Considering the increase in PESQ scores for the different noises, ADQTNE and DQTNE provide SNR advantages of 4−11 and 3−10 dB, respectively.

The proposed techniques are implemented using a fixed-point processor for real-time processing with audio latency acceptable for face-to-face communication. They are also implemented as a smartphone app with a graphical touch interface for setting the processing parameters in an interactive and real-time mode.