# A Sliding-band Dynamic Range Compression for Use in Hearing Aids

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***Abstract*— Sensorineural hearing loss is associated with elevated hearing thresholds, reduced dynamic range, and loudness recruitment. Dynamic range compression in the hearing aids is provided for restoring normal loudness of low level sounds without making the high level sounds uncomfortably loud. A sliding-band compression is presented for significantly reducing the temporal and spectral distortions generally associated with the currently used single and multiband compression techniques. It uses a frequency-dependent gain function calculated on the basis of critical bandwidth based short-time power spectrum and the specified hearing thresholds, compression ratios, and attack and release times. It is realized using FFT-based analysis-synthesis and can be integrated with other FFT-based signal processing in hearing aids to save computation. The technique is implemented and tested for satisfactory real-time operation, with sampling frequency of 10 kHz, window length of 25.6 ms with 75% overlap on a 16-bit fixed-point DSP processor with on-chip FFT hardware.**

***Keywords*— auditory critical bandwidth; dynamic range compression; hearing aid; sensorineural hearing loss**

## I. Introduction

Sensorineural hearing impairment is caused by abnormalities in the cochlear hair cells or the auditory nerve. It occurs due to aging, excessive exposure to noise, infection, or abnormalities at the time of birth. It is generally associated with elevated hearing thresholds, reduced dynamic range, and increased temporal and spectral masking, leading to degraded perception of speech [1] – [3]. Several signal processing techniques have been reported for reducing the effect of increased intraspeech spectral masking caused by widening of auditory filters. For persons with moderate bilateral loss and using binaural hearing aids, dichotic presentation with a pair of comb filters having complementary magnitude responses has been reported to be useful [4]. Spectral contrast enhancement [5] and multiband frequency compression [6] – [8] have been reported for use in monaural hearing aids. For improving speech perception in noisy environment, single-input speech enhancement technique such as spectral subtraction [9], [10] may be used for noise reduction.

Most of the listeners with sensorineural loss have a highly reduced dynamic range of hearing, with a significant frequency-dependent elevation of hearing threshold levels without corresponding increase in the upper comfortable listening levels. The sensory mechanism in the cochlea consists of inner and outer hair cells. The loss of inner hair cells results in elevated thresholds while that of outer hair cells leads to abnormal growth of loudness known as loudness recruitment and a significantly reduced dynamic range [1], [2]. Dynamic range compression in hearing aids is provided with the objective of presenting all the sounds comfortably within the limited dynamic range of the listener. It reduces the level differences between the high and low level parts of the audio signal in order to amplify the low level sounds without making the high level sounds uncomfortably loud [11].

Signal processing in digital hearing aids generally involves an adjustable frequency-dependent gain function and a dynamic range compression which can be either a single-band compression or a multiband compression with selectable number of bands and settable attack time, release time, and compression ratios [11] – [13]. The most commonly used compression systems employ single-band compression with the gain dependent on the dynamically varying signal level. As the power is mostly contributed by the low-frequency components, the amplification of the high-frequency components depends on the energy in the low-frequency components. Thus the high frequency components may become inaudible and distortions in temporal envelope may get introduced. As a solution to these problems, several multiband compression systems have been reported. In these systems, the spectral components of the input signal are divided in multiple bands and the gain for each band is calculated on the basis of signal power in that band.

Fig. 1 shows a schematic representation of multiband compression using a feed-forward gain control in each band. A delay equivalent to the processing time required for gain estimation is introduced in the signal path. Delayed signal is multiplied by the gain estimated in accordance with the selected compression threshold (Th), and compression ratio (CR). The resultant signals from each band are summed to produce the output signal. Lippmann et al. [14] compared four linear systems and two 16-channel amplitude compression systems. Use of compression resulted in an average increase of 9% in recognition scores for persons with severe loss and with speech material having significant level variation. Asano et al. [15] realized a compression system as a single time-varying FIR filter and implemented it on a DSP board with 32-bit

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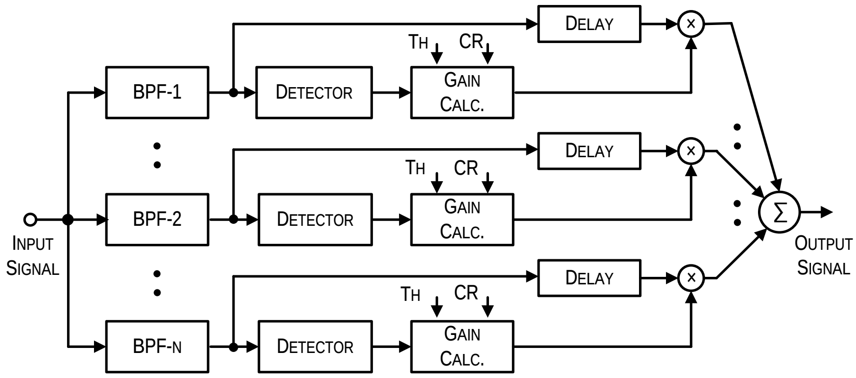


Fig. 1. A multiband amplitude compression using feed-forward gain control, adapted from [11].

floating-point processor. They have reported that the compression realized using a single FIR filter with smoothened frequency response resulted in much less spectral distortion than the compressions realized using band-pass filters. Stone et al. [16] compared the effect of compression ratio, compression threshold, and attack and release times using single and four-channel compression schemes implemented on a wearable digital hearing aid. No specific preference for the schemes was observed in tests for speech quality and intelligibility conducted on eight hearing-impaired subjects. Li et al. [17] described a compression scheme in which the speech signal was divided into 7 octave sub-bands using wavelet filter bank and the output was resynthesized after applying a logarithmic compression on the wavelet coefficients. On the basis of informal listening test, they have reported that the compression scheme increased the intelligibility without introducing noticeable distortions.

It has been reported in [16] that a fast-acting multiband compression can help in restoring near-normal loudness perception, restoring the audibility of weak sounds following intense sounds, and giving good results when two voices alternate with markedly different levels. Its main disadvantages are that it can introduce spurious spectral distortions. If many bands are used, it reduces spectral contrasts and the modulation depth of speech, which may have an adverse effect on the perception of certain speech cues. Spectral shape of a formant falling at the boundary between two adjacent bands may get distorted due to different gains applied in these bands. Formant transitions over the boundary between two adjacent bands may lead to perceptible discontinuities. Further, the frequency response of a multiband compression system has a time-varying magnitude response without corresponding variation in the phase response, which can cause audible distortions, particularly for non-speech audio. These distortions may partly offset the advantages of dynamic range compression for the hearing-impaired listener.

In order to significantly reduce the temporal and spectral distortions associated with the currently used single-band and multiband compressions in hearing aids, a "sliding-band compression" is presented. It involves calculating a frequency-dependent gain function, in which the gain for each spectral sample is determined by the short-time power in a band centered at it and the bandwidth selected to be equal to the auditory critical bandwidth [18]. The gain calculation takes into account the specified hearing thresholds, compression ratios, and attack and release times. Unlike single-band compression, it does not result in any significant temporal distortions because the effect of short-time energy of a spectral component on other spectral components is limited to those located within a critical bandwidth. Due to use of sliding critical bands for calculating the power spectrum, formant transitions do not result in discontinuities in the processed output. The technique is realized using an FFT-based analysis-synthesis method which masks phase related discontinuities. It can be integrated with other FFT-based signal processing in hearing aids. The technique is implemented and tested for satisfactory real-time operation on a 16-bit fixed-point DSP processor.

## II. SLIDING-BAND DYNAMIC RANGE COMPRESSION

A block diagram of the sliding-band compression realized using a DFT-based analysis-synthesis platform is shown in Figure 2. The processing consists of short-time spectral analysis, spectral modification, and resynthesis. In the spectral analysis block, the segments are obtained using $L$-sample window and window shift of $S$-samples. These segments are padded with zero-valued samples to length $N$ and short-time complex spectrum is obtained using $N$-point DFT. Spectral modification for dynamic range compression involves frequency-dependent gain calculation and output spectrum calculation. The output signal is resynthesized using IDFT, windowing, and overlap-add.

The processing for spectral modification using feed-forward gain compression is shown as a block diagram in Figure 3. For each discrete frequency sample $k$, there is a processing path for calculating the frequency-dependent gain and there is another path for calculating the output spectral sample. For the band centered at $k$, the band power $P_{ic}(k)$ is calculated as sum of the squared magnitude of its spectral samples. For auditory critical bandwidth based compression, the bandwidth at the frequency sample $k$ can be approximated as the following

$$\mathrm{BW}(k) = 25 + 75(1 + 1.4 f^{2})^{0.69} \qquad (1)$$

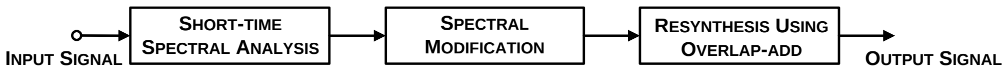


Figure 2. Block diagram of sliding-band compression system using spectral modification.

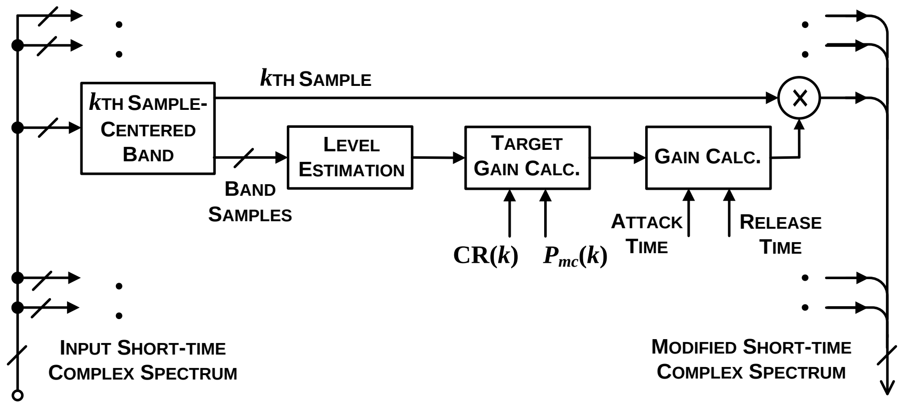


Figure 3. Block diagram of spectral modification for sliding-band compression system.

where $f$ is the frequency of $k$th spectral sample in kHz [19]. For dynamic range compression, the input power $P_{ic}(k)$ and the output power $P_{oc}(k)$ are linearly related on a dB scale and the relationship is given as

$$\left[P_{oc}(k)/P_{mc}(k)\right]_{\text{dB}} = \left[P_{ic}(k)/P_{mc}(k)\right]_{\text{dB}} /\text{CR}(k) \quad (2)$$

where $P_{mc}(k)$ is the maximum power representing the upper comfortable listening level and CR($k$) is the compression ratio. The relationship can also be written as

$$P_{oc}(k)/P_{mc}(k) = \left[P_{ic}(k)/P_{mc}(k)\right]^{1/\text{CR}(k)} \quad (3)$$

This relation results in a gain for the spectral sample $k$ as

$$\left[G_t(k)\right]_{\text{dB}} = \left[1-1/\text{CR}(k)\right]\left[P_{mc}(k)/P_{ic}(k)\right]_{\text{dB}} \quad (4)$$

The input complex spectrum is multiplied with the gain function to obtain the output spectrum which is used for resynthesizing the output signal.

Reduction in computation time is one of the requirements for real-time implementation of the proposed algorithm. Magotra et al. [20] used a Taylor's series approximation for gain calculation in a multiband compression system. This method is not suitable for sliding-band compression as it involves gain calculation at each of the frequency samples. Therefore, the gain as given in (4) is calculated using a two-dimensional look-up table relating the input power with gain as a function of frequency. It requires a large storage space but it significantly reduces the computational requirement. Further, it permits use of a frequency-dependent compression function most suited to compensate for the abnormal loudness growth curve of the hearing-impaired listener.

In order to avoid distortions due to sudden gain changes and to realize the desired attack and release times, the gain as calculated in (4) is taken as the target gain and the gain changed from previous value to the target value in steps. In the DFT based implementation, the gain applied to $k$th spectral sample in $i$th frame is given as

$$G(i,k) = \begin{cases} \max\left[G(i-1,k)/\gamma_a\ , G_t(i,k)\ \right], G_t(i,k) < G(i-1,k) \\ \min\left[G(i-1,k)\,\gamma_r\ , G_t(i,k)\ \right], G_t(i,k) > G(i-1,k) \end{cases} \quad (5)$$

Here $\gamma_a$ and $\gamma_r$ are the gain ratios for the attack phase and the release phase, respectively. These are given as

$$\gamma_a = \left(G_{\max}/G_{\min}\right)^{1/s_a} \quad (6)$$

$$\gamma_r = \left(G_{\max}/G_{\min}\right)^{1/s_r} \quad (7)$$

where $G_{\max}$ is the maximum target gain corresponding to the minimum input level, and $G_{\min}$ is the minimum target gain corresponding to the maximum input level. The parameters $s_a$ and $s_r$ are the number of steps during attack and release, respectively and are selected to set the attack time as $T_a = s_a S/f_s$ and release time as $T_r = s_r S/f_s$ where $f_s$ is the sampling frequency and $S$ is the number of samples for window shift. A fast attack may be used to avoid the input level to exceed the uncomfortable level during transients, and a slow release may be used to avoid the pumping effect or amplification of breathing.

Modifications in the short-time magnitude spectrum without corresponding changes in the phase spectrum can result in audible distortions, particularly for non-speech audio. To avoid these distortions, the sliding-band compression technique is realized using the analysis-synthesis method based on the least squared error estimation (LSEE) as proposed by Griffin and Lim [21]. The method masks the effect of phase discontinuities in the modified short-time complex spectrum and has been used earlier in real-time implementation of multiband frequency compression for reducing the effects of increased spectral masking [8]. For analysis, the speech signal is segmented using $L$-sample frames with 75% overlap. The segmented frames are multiplied with modified-Hamming window [21]. The samples are zero padded and $N$-point FFT is calculated to obtain the short-time complex spectrum. After spectral modification, the output signal is re-synthesized by using $N$-point IFFT and overlap-add after multiplying the output segment with the analysis window.

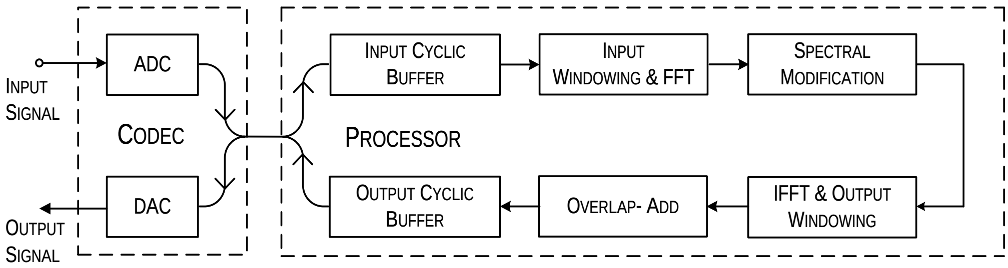

Fig. 4. Implementation of sliding-band dynamic range compression on the DSP board.

## III. OFFLINE AND REAL-TIME IMPLEMENTATIONS

In order to evaluate the performance of the proposed technique and the effect of processing parameters, it was implemented for offline processing in Matlab 7.10. The processing was carried out using a sampling frequency of 10 kHz and window length $L$ = 256 (25.6 ms). A 75% overlap-add was used corresponding to a window shift $S$ = 64. Analysis-synthesis was carried out using 512-point FFT. For look-up table based gain calculation, use of 20 logarithmic intervals to divide the input power range resulted in an acceptable trade-off between discontinuous gain changes due to quantization effect and size of the look-up table. Thus with 512-point FFT, there are 256×20 entries in the look-up table. The input critical band power and the corresponding target gain values were obtained and stored as a look-up table. Changing the maximum value of input power corresponds to a change in the threshold values, which can be adjusted according to hearing loss characteristics. The parameters $s_a$ and $s_r$ were set equal to one and 30 respectively. This corresponds to attack and release times of 6.4 ms and 192 ms, respectively.

The technique was implemented for real-time processing on a low-power DSP chip, using input-output operations similar to that described in [8] and [10]. A block diagram of the implementation is shown in Fig. 4. It uses a DSP board based on the 16-bit fixed point processor TI/TMS320C5515 [22], with a maximum clock rate of 120 MHz and 16 MB address space with 320 KB on-chip RAM (including 64 KB dual access RAM), and 128 KB on-chip ROM. It features three 32-bit programmable timers, four DMA controllers each with four channels, and a tightly coupled FFT hardware accelerator supporting 8 to 1024-point FFT. The DSP board "eZdsp" [23], with 4 MB on-board NOR flash for user program and codec TLV320AIC3204 [24] with stereo ADC and DAC supporting 16/20/24/32-bit quantization and sampling frequency of 8 – 192 kHz, was used for the implementation. The implementation uses the left channel of the codec, with 16-bit quantization and 10 kHz sampling. The input samples from ADC are acquired by DMA channel-2 and output to DAC by channel-0. The program was written in C, using TI's "CCStudio, ver. 4.0" as the development environment.

Fig. 5 shows the data transfer and buffering operations involved. To reduce the conversion overheads, the input samples, spectral values, and the processed samples are all stored as 4-byte words with 16-bit real and 16-bit imaginary parts as used in [10]. A 5-block DMA input cyclic buffer, with $S$-word blocks is used for signal acquisition. An input data

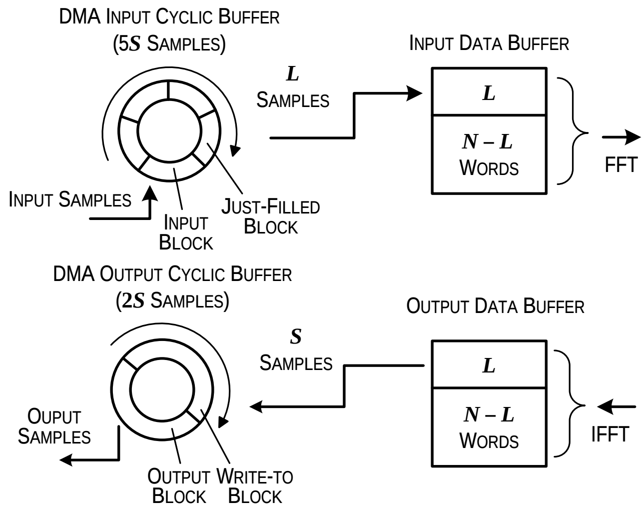

Fig. 5. Data transfer and buffering on the DSP board ($S$ = $L$/4) [8].

buffer of $N$ words is initialized with zero values. To keep a track of the current input, just-filled input, current output, and write-to output blocks, cyclic pointers are used. The pointers are initialized to 0, 4, 0, and 1, respectively and are incremented at each DMA interrupt which is generated when an input block gets filled. The DMA-mediated reading from ADC and writing to DAC are continued. Input window with $L$ samples is formed using the samples of the just-filled and the previous three blocks. These $L$ samples multiplied by modified Hamming window of length $L$ are copied to the input data buffer. They are padded with $N-L$ zero-valued samples to serve as input to $N$-point FFT. This method of data handling, results in an efficient realization of 75% overlap and zero padding. Input power in an auditory critical band centered at each of the first $N$/2 complex spectral components is calculated.

A look-up table for input power and target gain is used to calculate the target gain for each spectral sample. The gain is calculated using (5), with attack and release times as given in (6) and (7), respectively. Modified spectrum is obtained by multiplying first $N$/2 complex spectral component with the corresponding gain, with the other samples remaining zero valued. The first $L$ samples of $N$-point IFFT of the complex spectrum are multiplied by twice the modified Hamming window to get time domain signal. The output signal is synthesized using overlap-add operation. The processing has to get completed in $S$ sampling intervals for real-time operation.

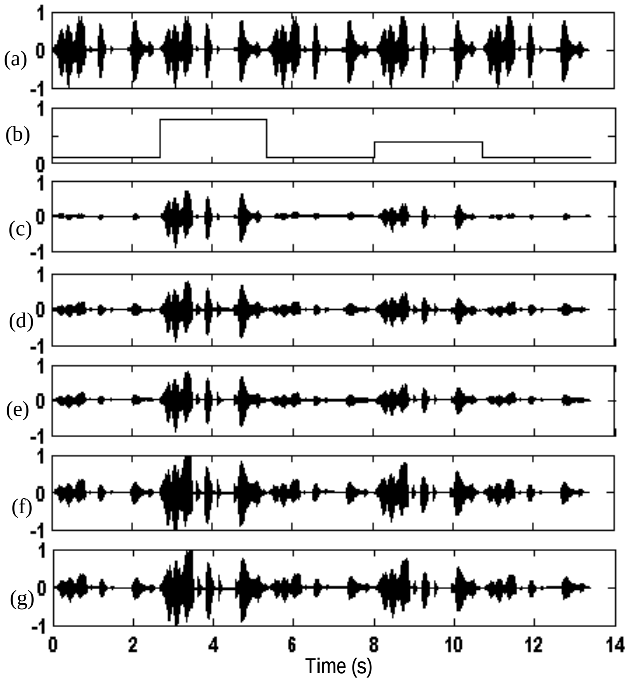


Fig. 6. Example of offline processing of sentence "you will mark ut please" concatenated with scaling factors of 0.1, 0.8, 0.1, 0.4, 0.1: (a) input waveform (b) scaling factor, (c) unprocessed waveform, (d) processed with $s_R$ = 0 and low $P_{mc}$, (e) processed with $s_R$ = 30 and low $P_{mc}$, (f) processed with $s_R$ = 0 and high $P_{mc}$, (g) processed with $s_R$ = 30 and high $P_{mc}$. CR = 2, for all spectral samples [26].

Thus the processing has an algorithmic delay of $L$ samples and a computational delay of less than $L$/4 samples.

## IV. Results

An example of the results from offline processing is shown in Fig. 6, for compression ratio CR = 2 (for all spectral samples) and two values of $P_{mc}$. The input speech material consists of an English sentence "*you will mark ut please*" concatenated with different scaling factors to observe the effect of variation in the input level on the output waveform. It can be observed from the figure that as input level increases the gain applied decreases. The processing gives highest gain when input level is lowest. The smallest gain is applied when the input level is highest. The results obtained with $s_r$ = 30 at two values of $P_{mc}$ are shown in Fig. 6(c) and 6(e) respectively. It can be observed that higher $P_{mc}$ leads to a higher initial gain and a larger compression.

To examine the difference in the processed outputs of single-band, multiband, and sliding-band compressions during spectral transitions, the compressions were applied on an input consisting of a sine wave with constant amplitude and changing frequency. Fig. 7 shows the results for frequency linearly swept from 125 Hz to 250 Hz over 200 ms. The multiband compression was implemented with band frequencies corresponding to 18 critical bands. The amplitude of the output of the multiband compression changes as the tone frequency changes over the band boundaries. The sliding-band compression output does not show these amplitude changes. Similar results were obtained for different swept tones and

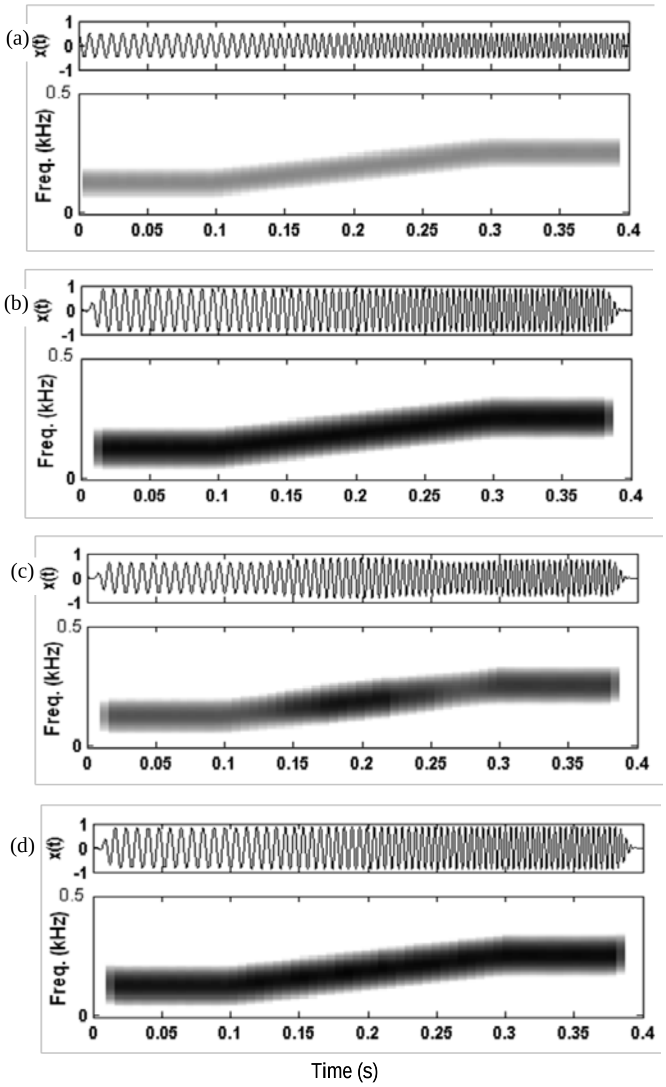


Fig. 7. Example of offline processing of sine wave with constant amplitude and frequency linearly swept from 125 Hz to 250 Hz over 200 ms: (a) unprocessed waveform, (b) processed using single-band compression, (c) processed using multiband compression, and (d) processed using sliding-band compression. CR = 30, for all spectral samples [26].

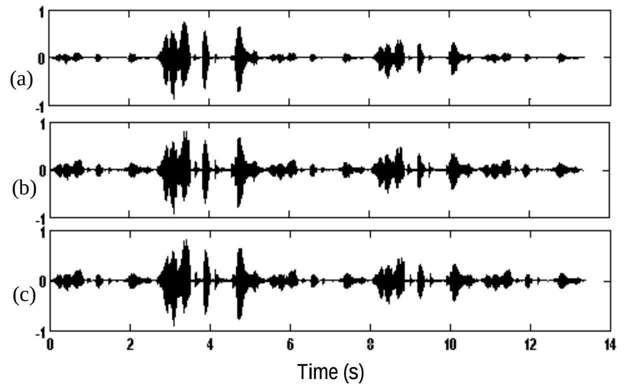


Fig. 8. Example of real-time processing (for the input waveform same as in Fig. 6(c)): (a) unprocessed waveform, (b) offline processed waveform, (c) real-time processed waveform. $s_R$ = 30, low $P_{mc}$ [26].

narrowband noises with swept center frequencies [26]. These results confirm that the sliding-band compression is successful in avoiding the distortions which occur in multiband compression during spectral transitions.

The outputs of the real-time processing exhibited a very close match to the corresponding outputs of offline processing. An example of the results from real-time processing for the input speech material as used in the example of offline processing, with CR = 2, $s_a = 1$ and $s_r = 30$, is shown in Fig. 8. The implementation was evaluated using informal listening and objective evaluation using perceptual evaluation of speech quality (PESQ) measure [25]. Informal listening showed that the processed output from the DSP board was perceptually similar to the corresponding output from the offline implementation for speech as well as other audio signals. The processed output signal from DSP board was acquired through a PC sound card. PESQ-MOS for speech outputs from the real-time processing with those from the offline processing was 3.50. The technique was also tested by informal listening of different speech materials, music, and environmental sounds with large variation in the sound level. No perceptible distortions were noticed in the processed outputs.

To estimate the computation load on the processor, the system clock was progressively decreased from 120 MHz. It was found that the processing requires a clock frequency of 50 MHz indicating that the technique needed approximately 41% of the maximum available processing capacity. Total signal delay (algorithmic delay, computation delay, and input-output delay) was found to be approximately 36 ms.

## V. Conclusion

A sliding-band dynamic range compression has been presented for use in hearing aids to compensate for frequency-dependent loudness recruitment associated with sensorineural hearing loss without introducing the distortions generally associated with single-band and multiband compression techniques. The technique has settable attack time, release time, and compression ratios. It has been implemented on a 16-bit fixed-point processor and tested for satisfactory operation. Its benefits in improving the speech intelligibility for the hearing impaired listeners need to be evaluated. Further, a combination of the proposed technique with spectral subtraction and multiband frequency compression needs to be implemented for real-time processing and evaluated.

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