A Smartphone App-Based Digital Hearing Aid with Sliding-Band Dynamic Range Compression

Nitya Tiwari [1], Prem C. Pandey [1], and Anurag Sharma [2]
[1] Dept. of Electrical Engineering, Indian Institute of Technology Bombay, Mumbai, India
[2] Quadio Devices Pvt. Ltd., Pune, India
<nitya@ee.iitb.ac.in>, <pcpandey@ee.iitb.ac.in>, <axs@quadiodevices.com>

***Abstract*—Listeners with sensorineural hearing loss have degraded speech perception due to frequency-dependent elevation of hearing thresholds, reduced dynamic range, and increased temporal and spectral masking. Signal processing in hearing aids for such listeners uses frequency-selective amplification and dynamic range compression for restoring normal loudness of low-level sounds without making the high-level sounds uncomfortably loud. Sliding-band compression has been reported earlier for reducing the temporal and spectral distortions generally associated with currently used single and multiband compression techniques. The paper presents implementation of sliding-band compression as a smartphone app for use as a hearing aid. The processing involves 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 such techniques for computational efficiency.**

Keywords—dynamic range compression; hearing aid; sliding-band; sensorineural hearing loss; smartphone app

I. INTRODUCTION

Sensorineural hearing impairment occurs due to aging, excessive exposure to noise, congenital defects, infection, or ototoxic drugs. It is caused by abnormalities in the cochlear hair cells or the auditory nerve, and is associated with frequency-dependent elevation of hearing thresholds, reduced dynamic range, and increased temporal and spectral masking. Listeners with sensorineural loss generally suffer from degraded speech perception [1]–[3]. Several signal processing techniques such as frequency-selective amplification along with dynamic range compression [4], [5], binaural dichotic presentation [6], spectral contrast enhancement [7], multiband frequency compression [8]–[10], and single-input speech enhancement [11], [12] have been reported for improving the speech perception by persons with moderate-to-severe sensorineural loss. Out of these techniques, frequency-selective amplification and dynamic range compression are considered to be the most important and they form core of the hearing aids.

Frequency-dependent elevation of hearing threshold levels without corresponding increase in the upper comfortable listening levels leads to a highly reduced dynamic range of hearing in most listeners with sensorineural loss. Increase in hearing thresholds can be attributed to loss of inner hair cells, while reduction in dynamic range and abnormal growth of loudness (known as loudness recruitment) may be attributed to the loss of outer hair cells [1], [2]. Persons with this type of loss generally do not benefit much by use of linear amplification which makes the high-level sounds intolerably loud. Signal processing in hearing aids uses dynamic range compression to reduce the level differences between the high and low level parts of the audio signal in order to present all the sounds comfortably within the limited dynamic range of the listener [4]. Currently available digital hearing aids use single-band or multiband dynamic range compression with selectable number of bands and settable attack time, release time, and compression ratios [13].

In single-band compression, the gain is calculated by estimating the dynamically varying signal level. As the estimated level of the speech signal depends mainly on the energy of the low frequency components, the level of the high-frequency components gets affected by the level of the low-frequency components. This may introduce distortions in temporal envelope and the high frequency components may become inaudible.

As a solution to the problems associated with single-band compression and for restoring near-normal loudness perception, several multiband compression schemes have been reported [14]–[17]. These schemes divide the spectral components of the input signal into multiple bands and calculate the gain for each band based on the power in that band. Use of multiple bands reduces distortions in the temporal envelope, but it decreases the spectral contrasts and modulation depths in the speech signal, which may adversely affect speech perception. Different gains in adjacent bands may distort the spectral shape of a formant (spectral peak in the speech signal, associated with acoustic resonance of the oral cavity) falling at band boundaries and may lead to perceptible discontinuities when formant transitions occur over the boundary between the adjacent bands. Furthermore, changes in the magnitude response without corresponding changes in the phase response may lead to audible distortions. This effect is more pronounced during non-speech audio. These distortions may outweigh the advantages of multiband dynamic range compression.

A sliding-band compression scheme has been reported [18] to reduce the temporal and spectral distortions associated with the single-band and multiband compression schemes currently used in hearing aids. The technique determines a frequency-

The research is supported by "National Program on Perception Engineering Phase II" sponsored by the Department of Electronics and Information Technology, MCIT, Government of India.

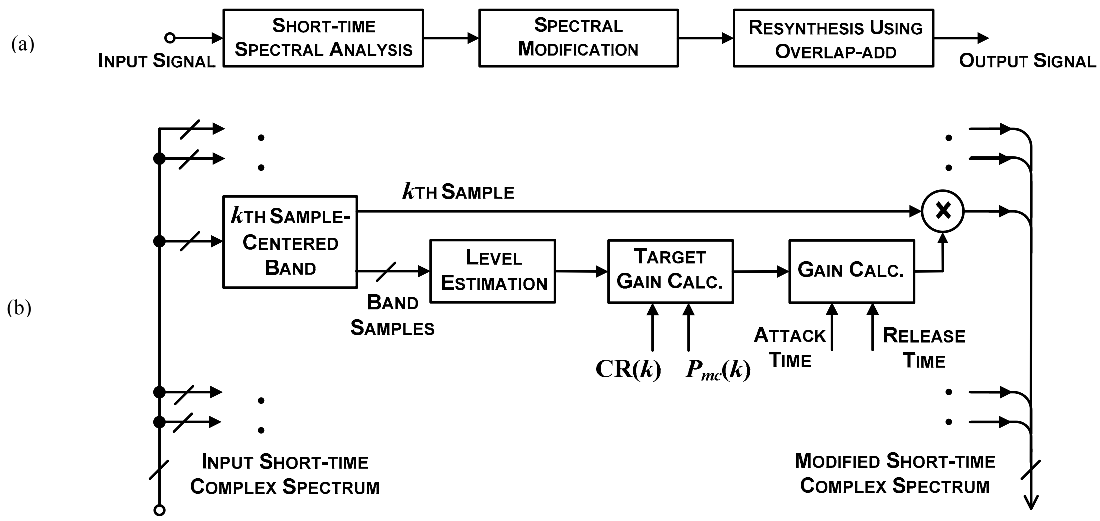

Fig. 1. Sliding-band dynamic range compression: (a) Block diagram, (b) Spectral modification, adapted from [18].

dependent gain function, wherein the gain for each frequency sample is calculated using the short-time power in the auditory critical band [19] centered at it. The technique has been implemented on a 16-bit fixed-point DSP processor and tested for real-time operation. For clinical evaluation, it needs to be incorporated as part of digital hearing aids which are currently developed using ASICs due to size and power constraints.

Development of application software (app) for smartphones (mobile telephone handsets with computing power for third-party apps) is emerging as a convenient and inexpensive platform for rapid implementation and testing of signal processing techniques for audiometry and hearing aids. Several smartphone-based audiometry apps for evaluating hearing sensitivity have been reported [20]–[22] which may be conveniently used without specialized training and may also be used by audiologists for remote adjustment of hearing aids of their patients. Hearing aid apps such as "Petralex" [23] for Android/iOS, "BioAid" [24] for iOS, and "Hearing Aid with Replay" [25] and uSound [26] for Android are available for use by hearing impaired persons. They provide frequency selective gain and dynamic range compression and also offer the flexibility of creating and storing sound profiles specific to the user's hearing loss characteristics.

To make the sliding-band dynamic range compression available for use and evaluation by hearing impaired listeners, it is implemented as a smartphone app. It is developed for iPhone, as the audio input-output latency for iOS-based handsets has been reported to be generally lower than that for handsets based on other mobile operating systems [27]. The technique is described in the second section, followed by the details of its implementation as a smartphone app in the third section. Test results are presented in the fourth section, followed by conclusions in the last section.

II. SLIDING-BAND DYNAMIC RANGE COMPRESSION

The sliding-band compression [18] realized using a DFT-based analysis-synthesis is shown in Fig. 1. The processing steps involve short-time spectral analysis, spectral modification, and resynthesis. The input segments for analysis are obtained in the spectral analysis block using L-sample window with a shift of S-samples. These segments are padded with zero-valued samples to length N and are used as input to N-point DFT to obtain short-time complex spectrum. The frequency-dependent gain function and the output spectrum are obtained in the spectral modification block. Resynthesis of output signal is done using IDFT, windowing, and overlap-add.

Spectral modification using feed-forward gain compression is shown in Fig. 1(b). For each discrete frequency sample k, there are two processing paths. The first path is used for calculating the frequency-dependent gain and the second one is used for calculating the output spectral sample. The band power, $P_{ic}(k)$, is calculated as sum of the squared magnitude of the spectral samples in the band centered at k. The processing uses auditory critical bandwidth based compression with the bandwidth at the frequency sample k approximated as

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

where f is the frequency of kth spectral sample in kHz [28].

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

$$[P_{oc}(k)/P_{mc}(k)]_{\mathrm{dB}} = [P_{ic}(k)/P_{mc}(k)]_{\mathrm{dB}} / \mathrm{CR}(k) \tag{2}$$

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

$$P_{oc}(k)/P_{mc}(k) = [P_{ic}(k)/P_{mc}(k)]^{1/\mathrm{CR}(k)} \tag{3}$$

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

$$[G_t(k)]_{\mathrm{dB}} = [1 - 1/\mathrm{CR}(k)][P_{mc}(k)/P_{ic}(k)]_{\mathrm{dB}} \tag{4}$$

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

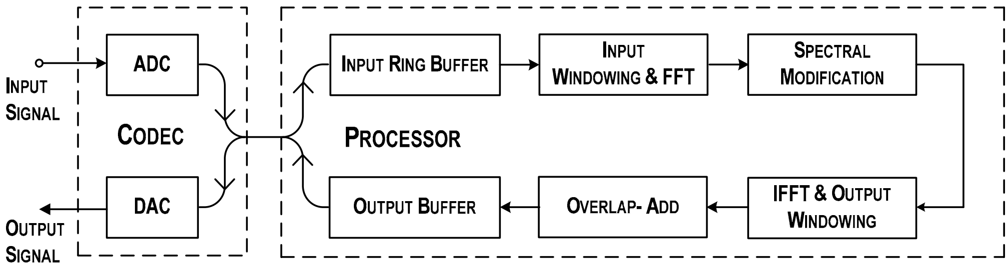

Fig. 2. Implementation of sliding-band dynamic range compression on Iphone.

The gain as calculated in (4) is applied to the input levels above the knee point $P_{kp}(k)$. Applying a high gain below $P_{kp}(k)$ may lead to over-amplification of the background noise. To avoid it, the gain is progressively reduced for input power below $P_{kp}(k)$ until it becomes unity for minimum input power $P_{min}(k)$. The expansion gain for the spectral sample k for $P_{ic}(k) < P_{kp}(k)$ is given as

$$[G_t(k)]_{\mathrm{dB}} = [\mathrm{ER}(k)-1][P_{ic}(k)/P_{\min}(k)]_{\mathrm{dB}} \tag{5}$$

where ER(k) is the expansion ratio. It is given as

$$\mathrm{ER}(k) = \frac{[P_{kp}(k)/P_{mc}(k)]_{\mathrm{dB}} + [P_{mc}(k)/P_{in}(k)]_{\mathrm{dB}}^{\mathrm{CR}(k)}}{\mathrm{CR}(k)[P_{in}(k)/P_{\min}(k)]} \tag{6}$$

Low computation time is one of the main requirements for usefulness of a signal processing algorithm in hearing aids. Taylor's series approximation for the gain expression was used by Magotra et al. [29] in a multiband compression system. It has high computational requirement as it involves calculating gain at each frequency sample, making it unsuitable for sliding-band compression. Thus a two-dimensional look-up table which relates the input power with gain as a function of frequency is used for calculating the gain as given in (4) and (5). The gain values for an intermediate input power level are obtained using linear interpolation. The look-up table requires more storage but it reduces the computational requirement significantly. It also permits the use of the most appropriate frequency-dependent compression function to compensate for the abnormal loudness growth curves of the hearing-impaired listener. This feature is currently not available in hearing aids or smartphone-based hearing aid apps.

To reduce distortions due to sudden changes in gain and to realize the desired attack and release times, the gain as obtained from the look-up table using interpolation is taken as the target gain. The gain applied to the kth spectral sample in the ith frame is changed from its previous value to the target value in steps and is given as

$$G(i,k) = \begin{cases} \max[G(i-1,k)/\gamma_a, G_t(i,k)], G_t(i,k) < G(i-1,k) \\ \min[G(i-1,k)\,\gamma_r, G_t(i,k)], G_t(i,k) > G(i-1,k) \end{cases} \tag{7}$$

Here γ_a and γ_r are the gain ratios for the attack phase and the release phase, respectively. These are given as

$$\gamma_a = (G_{\max}/G_{\min})^{1/s_a} \tag{8}$$

$$\gamma_r = (G_{\max}/G_{\min})^{1/s_r} \tag{9}$$

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 $T_a = s_aS/f_s$ and release time as $T_r = s_rS/f_s$ where f_s = sampling frequency and S = 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.

A change in magnitude response without corresponding changes in the phase response may lead to audible distortions. To mask the effect of phase discontinuities in the modified short-time complex spectrum, 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 [30]. The analysis uses L-sample frames with 75% overlap to segment the speech signal. Modified-Hamming window [30] is applied on the segmented frames which are zero padded to length N. Short-time complex spectrum is obtained using N-point FFT. After spectral modification, an N-point IFFT is taken, following which the output segment is multiplied with the analysis window and the output signal is re-synthesized using overlap-add.

III. App Implementation for Real-Time Processing

Block diagram of the implementation is shown in Fig. 2. The implementation has been carried out and tested using iPhone-4s with iOS-7.0. The processing uses a sampling frequency of 11.025 kHz and window length L = 256 (23.2 ms). A 75% overlap-add is used corresponding to a window shift S = 64.

Input/output handling is carried out using functions from Novocaine [31], an open source library for audio management. It provides a ring buffer of sufficiently large size (32K word buffer) for storing the audio samples from ADC for real-time operation. During input callback, the sample from ADC is written to the ring buffer. During output callback, S samples are read from the ring buffer and are used for processing. Write and read pointers are maintained to track the number of samples written to the ring buffer and the number of samples read from it for processing. A 2S-word output buffer is used for outputting the processed samples through DAC. While the samples from one half of the buffer are being output through the DAC, the other half of the buffer is used for storing the processed samples.

The input samples, real and imaginary parts of spectral samples, and the output samples are stored as 32-bit floating point arrays. The input samples from ring buffer are acquired in *S*-word blocks. For each *S*-word just-filled block, a 3*S*-word buffer stores the previous input samples. Input window with *L* samples (4*S* words) is formed using the samples of the just-filled block and the samples from previous three blocks stored in 3*S*-word buffer. These *L* samples multiplied by modified Hamming window of length *L* and *L*-point FFT is calculated. Open source Kiss FFT library [32] has been used for floating point FFT computation. Input power in an auditory critical band centered at each of the first *L*/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 (4) and (5), with attack and release times as given in (8) and (9), respectively. Modified spectrum is obtained by multiplying first *L*/2 complex spectral samples with the corresponding gains, with the other samples remaining zero valued. The *L* samples of *L*-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 of a frame has to get completed before the processing for the next frame. Thus the processing has an algorithmic delay of *L* samples and the computational delay has to be less than *L*/4 samples.

The program was written using a combination of C++ and objective C, with Apple's "Xcode, ver. 7.0.1" as the development environment. Analysis-synthesis was carried out using 256-point FFT. For look-up table based gain calculation, use of 15 logarithmic intervals to divide the input power range, and a linear interpolation resulted in an acceptable trade-off between discontinuous gain changes due to quantization effect and size of the look-up table. Thus with 256-point FFT, there are 128×15 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 5.8 ms and 174 ms, respectively.

IV. TEST RESULTS

The app was tested on an iPhone 4S with iOS 7.0. For qualitative evaluation, the headset of the handset was used for acquiring speech input through its microphone and listening to audio output through its earphone. For objective evaluation of the processing, an audio input-output interface to the headset port of the handset was devised with a 4-pin TRRS (tip-ring-ring-sleeve) connector. The interface used a resistive attenuator for attenuating the input audio signal to a level compatible with the microphone signal level and had output resistance of 1.8 kΩ for it to be recognized as external microphone. The audio input was applied using the audio output from a PC sound card.

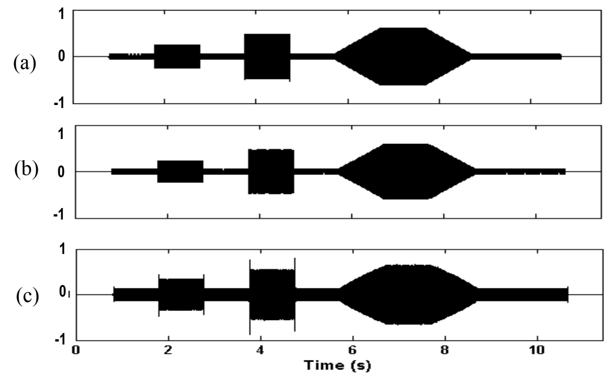

Fig. 3. Processing of amplitude-modulated sinusoidal tone of 840 Hz: (a) input, (b) input looped back from iPhone, and (c) processed output. CR = 2 for all spectral samples.

The processed audio output was acquired using the audio input of the PC sound card and also displayed on a DSO.

Examples of processing by the app for different types of inputs consisting of amplitude-modulated and frequency modulated tones and broad-band noises as well as amplitude modulated speech are available at [33].

The processing was tested using amplitude-modulated tones of different frequency and amplitude envelopes as inputs. An example of the processing is shown in Fig. 3, for compression ratio CR = 2 for all spectral samples. The input is an amplitude-modulated tone of 840 Hz. The processing gives highest gain when the input level is lowest and the gain is smallest for the highest input level. Spikes in the amplitude envelope of the output signal in response to step changes in the amplitude envelope of the input signal, as seen in the figure, are typical of the dynamic range compression with a finite attack time and window shift.

To examine the advantage of sliding-band compression over the conventional multiband compression, the later was implemented with band frequencies corresponding to 18 critical bands. As the differences in the two forms of compression primarily occur during transitions of spectral components across bands, testing was carried out with a sinusoidal input of swept-frequency and constant amplitude. An example of the processing is shown in Fig. 4 for frequency linearly swept from 100 Hz to 5 kHz over 8 s. In the output of multiband compression, the amplitude envelope changes as the tone frequency moves across the band boundaries. These changes may cause audible distortions, particularly for listeners with reduced dynamic range and correspondingly increased sensitivity to amplitude changes. The sliding-band compression output does not show these amplitude variations. Similar results were observed for different swept tones and narrowband noises with swept center frequencies.

The processing was tested for speech and music modulated with different types of amplitude envelopes. An example of the processing is shown in Fig. 5 for CR = 2 for all spectral samples and two release times. 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. As the

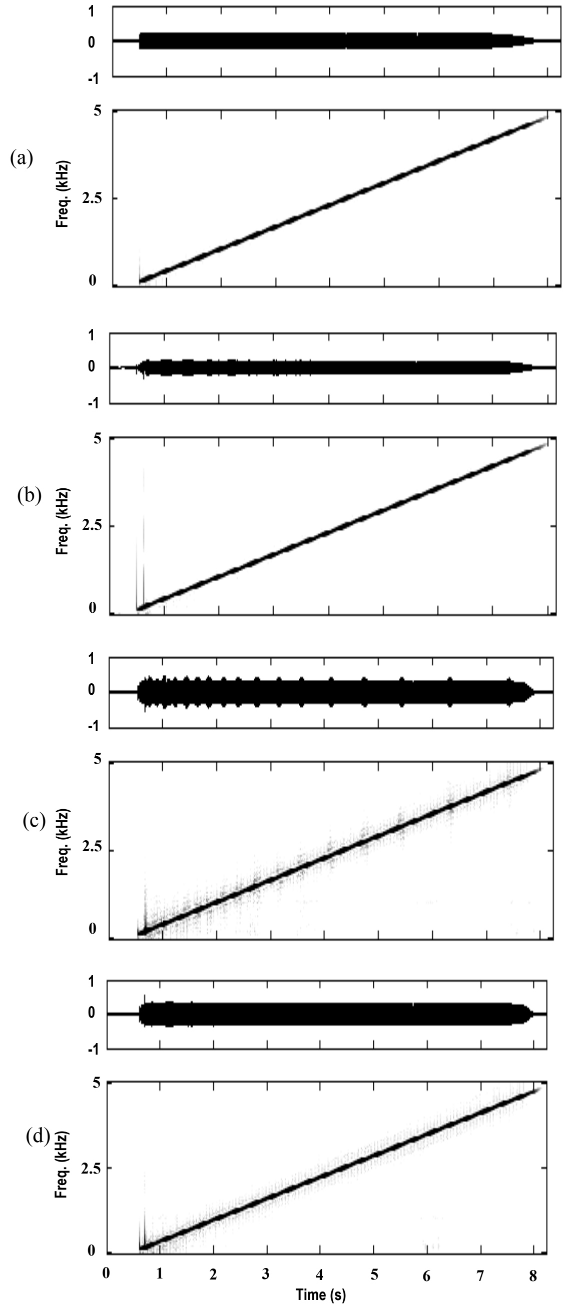

Fig. 4. Processing of sinusoidal tone with constant amplitude and frequency linearly swept from 100 Hz to 5 kHz over 8 s: (a) input, (b) input looped back from iPhone, (c) processed output using multiband compression, and (d) processed output using sliding-band compression. CR = 10 for all spectral samples. In each case, the top panel shows the time-domain waveform and the bottom panel shows the corresponding spectrogram.

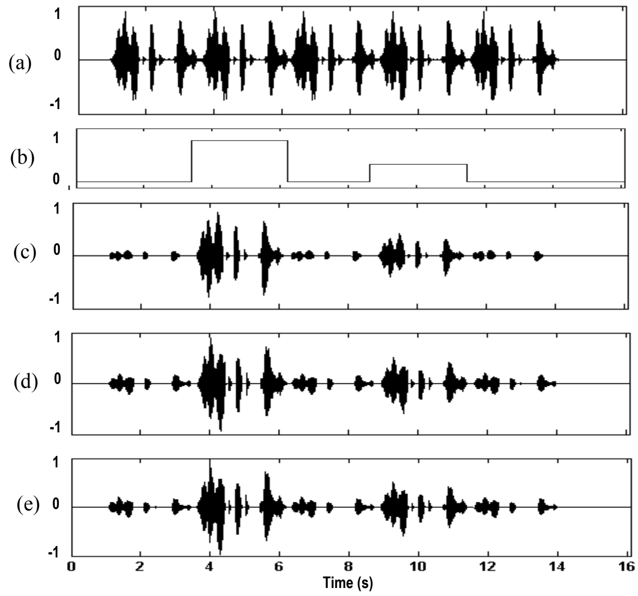

Fig. 5. Processing of amplitude-modulated speech, with the sentence "you will mark ut please" concatenated with scaling factors of 0.1, 0.8, 0.1, 0.4, 0.1: (a) speech signal, (b) amplitude scaling factor, (c) amplitude-modulated speech input, (d) processed output with $s_R = 0$, (e) processed output with $s_R = 30$. CR = 2 for all spectral samples.

input level increases the gain applied decreases, without any perceptible distortion.

Informal listening test was carried out with different speech materials, music, and environmental sounds with large variation in the sound level as inputs. The outputs exhibited the desired compression without introducing perceptible distortions. There were no perceptible differences between the processed outputs from the app and the offline implementation.

To measure the real-time latency (signal delay, consisting of the processing delay and input-output delay), a 1 kHz tone burst of 100 ms was given as input. The delay was measured using a digital oscilloscope and was found to be approximately 36 ms. The processing delay (sum of the algorithmic and computational delays) is approximately 29 ms (1.25 times the window length). The additional delay is due to audio input-output latency of the handset. The latency is well within the acceptable delay for audio-visual perception. The profiler of Xcode was used to monitor the processing capacity used by the application. It was found that the technique needed approximately 39% of the maximum available processing capacity of the device.

V. CONCLUSION

A smartphone app for sliding-band dynamic range compression has been developed for use as a hearing aid application. It aims 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. Compensation for frequency responses of the microphone and earphone of the handset can be included as part of a look-up table with additional adjustment through the graphical user interface of the app, without any run-time processing overhead.

The app has been developed for iPhone and it has been tested for satisfactory operation on iPhone 4S with iOS 7.0. The real-time latency is acceptable for audio-visual perception and hence the app is useable as a hearing aid by hearing impaired persons during face-to-face conversation. It uses only a part of the processing capacity of the device. As the processing capacity of newer handsets is expected to be much higher, there is scope for further developing the app to incorporate speech enhancement for extending its usability in noisy environments [11], [34]. The app needs to be developed for other popular mobile platforms.

As the audio output levels of the smartphones are not adequate for persons with severe loss, headsets with higher levels will be needed by such listeners. With use of such headsets, effectiveness of the app in improving the speech intelligibility for the hearing-impaired listeners needs to be evaluated.

Acknowledgment

Authors are grateful to Mr. Paresh Patel (Quadio Devices, Pune) for supporting the investigation and his colleagues Mr. Ketan Pandya and Mr. Naveen Francis for technical assistance.

References

[1] H. Levitt, J. M. Pickett, and R. A. Houde, Eds., *Senosry Aids for the Hearing Impaired*, New York: IEEE Press, 1980.

[2] B. C. J. Moore, *An Introduction to the Psychology of Hearing*, London, UK: Academic, 1997, pp 66–107.

[3] S. A. Gelfand, *Hearing*: *An Introduction to Psychological and Physiological Acoustics*, 3rd ed., New York: Marcel Dekker, 1998, pp. 314–318.

[4] H. Dillon, *Hearing Aids*, New York: Thieme Medical, 2001.

[5] Robert E. Sandlin, *Textbook of Hearing Aid Amplification*, San Diego, Cal.: Singular, 2000, pp. 210–220.

[6] P. N. Kulkarni, P. C. Pandey, and D. S. Jangamashetti, "Binaural dichotic presentation to reduce the effects of spectral masking in moderate bilateral sensorineural hearing loss," *Int. J. Audiol*., vol. 51, no. 4, pp. 334–344, 2012.

[7] J. Yang, F. Luo, and A. Nehorai, "Spectral contrast enhancement: Algorithms and comparisons," *Speech Commun*., vol. 39, no. 1–2, pp. 33–46, 2003.

[8] T. Arai, K. Yasu, and N. Hodoshima, "Effective speech processing for various impaired listeners," in *Proc. 18th Int. Congr. Acoust. (ICA 2004)*, Kyoto, Japan, 2004, pp. 1389–1392.

[9] P. N. Kulkarni, P. C. Pandey, and D. S. Jangamashetti, "Multiband frequency compression for improving speech perception by listeners with moderate sensorineural hearing loss," *Speech Commun.*, vol. 54, no. 3, pp. 341–350, 2012.

[10] N. Tiwari, P. C. Pandey, and P. N. Kulkarni, "Real-time implementation of multi-band frequency compression for listeners with moderate sensorineural impairment," in *Proc. 13th Annu. Conf. Int. Speech Commun. Assoc. (INTERSPEECH 2012)*, Portland, Oregon, 2012, pp. 1860–1863.

[11] P. C. Loizou, *Speech Enhancement: Theory and Practice*, New York: CRC, 2007.

[12] S. K. Waddi, P. C. Pandey, and N. Tiwari, "Speech enhancement using spectral subtraction and cascaded-median based noise estimation for hearing impaired listeners," in *Proc. 19th Nat. Conf. Commun. (NCC 2013)*, New Delhi, India, 2013, doi: 10.1109/NCC.2013. 6487989.

[13] L. D. Braida, N. I. Durlach, R. P. Lippmann, B. L. Hicks, W. M. Rabinowitz, and C. M. Reed, *Hearing aids–a review of past research on linear amplification, amplitude compression, and frequency lowering*, vol. 19, J. C. Hardy, Ed., Rockville, Maryland, USA: ASHA Monographs, 1979, pp. 1–114. [online] Available: http://www.asha.org/ uploadedFiles/publications/archive/Monographs19.pdf

[14] R. P. Lippmann, L. D. Braida, and N. I. Durlach, "Study of multichannel amplitude compression and linear amplification for persons with sensorineural hearing loss," *J. Acoust. Soc. Am*., vol. 69, no. 2, pp. 524–534, 1981.

[15] F. Asano, Y. Suzuki, T. Sone, S. Kakehata, M. Satake, K. Ohyama, T. Kobayashi, and T. Takasaka, "A digital hearing aid that compensates for sensorineural impaired listeners," in *Proc. IEEE Int. Conf. Acoust. Speech Signal Process. (ICASSP 1991)*, Toronto, Ontario, Canada, 1991, pp. 3625–3628.

[16] M. A. Stone, B. C. Moore, J. I. Alcántara, and B. R. Glasberg, "Comparison of different forms of compression using wearable digital hearing aids," *J. Acoust. Soc. Am*., vol. 106, no. 6, pp. 3603–3619, 1999.

[17] M. Li, H. G. McAllister, N. D. Black, and T. A. Perez, "Wavelet based non-linear AGC method for hearing aid loudness compensation," *IEE Proc. Vision Image Signal Process.,* vol. 147, no. 6, pp. 502-507, 2000.

[18] N. Tiwari and P. C. Pandey, "A sliding-band dynamic range compression for use in hearing aids," in *Proc. 20th Nat. Conf. Commun. (NCC 2014)*, Kanpur, India, doi: 10.1109/NCC.2014. 6811300.

[19] E. Zwicker, "Subdivision of the audible frequency range into critical bands (Freqenzgruppen)," *J. Acoust. Soc. Am.*, vol. 33, no. 2, pp. 248, 1961.

[20] F. Larrosa, J. Rama-Lopez, J. Benitez, J. M. Morales, A. Martinez, M. A. Alañon, D. Arancibia-Tagle, A. Batuecas-Caletrio, M. Martinez-Lopez, N. Perez-Fernandez, C. Gimeno, A. Ispizua, A. Urrutikoetxea, and J. Rey-Martinez, **"**Development and evaluation of an audiology app for iPhone/iPad mobile devices**,"** *Acta Oto-Laryngologica*, vol. 135, no. 11, pp. 1119–1127, 2015.

[21] S. Abu-Ghanem, O. Handzel, L. Ness, M. Ben-Artzi-Blima, K. Fait-Ghelbendorf, and M. Himmelfarb "Smartphone-based audiometric test for screening hearing loss in the elderly**,"** *Eur Arch Otorhinolaryngol*, Feb. 2015, doi: 10.1007/s00405-015-3533-9.

[22] Mimi Hearing Technologies. (2015). "Mimi hearing test v3.0." [online] Available: itunes.apple.com/in/app/mimi-hearing-test/id932496645 ?mt=8

[23] IT ForYou. (2014). "Petralex hearing aid v1.4.3." [online] Available: play.google.com/store/apps/details?id=com.it4you.petralex, itunes. apple.com/us/app/petralex-hearing-id

[24] N. Clark. (2015). "Bio aid v1.0.2." [online] Available: itunes. apple. com /gb/app/bioaid

[25] Lamberg Solutions. (2014). "Hearing aid with replay v2.0.0." [online] Available: play.google.com/store/apps/details?id=com.ls.sound amplifier

[26] S. A. Newbrick. (2015). "uSound (hearing assistant) v.1.6." [online] Available: play.google.com/store/apps/details?id=com.newbrick. usound

[27] Superpowered Inc. (2015). "Android audio latency and iOS audio latency test app" [online] Available: superpowered.com/latency /#axzz3hutxeg2s

[28] J. W. Picone, "Signal modeling techniques in speech recognition," *Proc. IEEE*, vol. 81, no. 9, pp. 1215 – 1247, 1993.

[29] N. Magotra, S. Kamath, F. Livingston, and M. Ho, "Development and fixed-point implementation of a multiband dynamic range compression (MDRC) algorithm, in *Proc. 34th Asilomar Conf. Signals Syst. Comput.*, vol. 1, Pacific Grove, CA, USA, 2000, pp. 428-432.

[30] D. W. Griffin and J. S. Lim, "Signal estimation from modified short-time Fourier transform," *IEEE Trans. Acoust. Speech Signal Process.*, vol. 32, no. 2, pp. 236 – 243, 1984.

[31] A. Wiltschko, "Novocaine," [online] Available: github.com/alexbw/ novocaine

[32] M. Borgerging, "Kiss FFT," [online] Available: sourceforge.net/ projects/kissfft

[33] N. Tiwari, "Dynamic range compression results from iPhone app implementation", 2016, [online] Available: www.ee.iitb.ac.in/~spilab/ material/nitya/ncc2016.

[34] N. Tiwari and P. C. Pandey, "Speech enhancement using noise estimation based on dynamic quantile tracking for hearing impaired listeners," in *Proc. 21st Nat. Conf. Commun. (NCC 2015)*, Mumbai, India, 2015, paper no. 1570056299.