Estimation of SNR based Adaptive-Feedback Equalizers for Feedback Control in Hearing Aids

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ABSTRACT

Despite the evolution of modern technology, the users of hearing aids do not realize the persistence of feedback, while wearing the device until the condition become worse. On applying the feedback cancellation algorithms, instead of cancelling the acoustic feedback, it limits the speech intelligibility. The paper presents a novel method on estimation of SNR based adaptive-feedback equalizers (SBAFE) algorithm to develop an optimized hearing aids for the feedback less sound transmission and achieving the better speech discrimination. The data gathered for the optimization is visualized and compared with the traditional technology, which provide the subjective and objective quality of the hearing aids.

KEYWORDS: Hearing aids, Acoustic feedback, Feedback control

I INTRODUCTION

The hearing loss is determined by presenting the pure tones, of different frequencies at different intensities to an individual in a sound treated room. This method of measuring the hearing threshold of a person by an audiologist is known as **Pure Tone Audiometry (PTA)** technique. An audiometry chart named the pure tone audiogram, is prepared based on the recognition of hearing, which determines the type of hearing loss. The **hearing loss** is the incapability or difficulty in sensing a certain frequency band of sounds that the person wanted to hear. Based on the type of loss identified (either Air-conduction or bone-conduction) and depending on a part of disorder in the auditory system, the hearing aids are chosen accordingly.

The **hearing aids** are the devices which assist the people during their difficulty in hearing. Among the various types, the **Behind-the-ear (BTE)** hearing aids are the commonly used listening devices. BTE can be monoaural or binaural. **Binaural** ear fittings are used to get the differences in the hearing level from both the ears. In the hearing device, the sound (includes speech) enters through the microphone, processed by a digital signal processor via ADC and DAC and reaches the receiver from which, the output signal traverses towards the ear drum.

When a part this output is picked up by the microphone again, a significant performance degradation occurs in the hearing system. That is, when the amplification signal from the microphone to the receiver is greater than, the attenuation of the amplified signals from the receiver to the microphone, the hearing system goes into oscillation. During which the whistling or ringing sound is heard. This makes the auditory system unstable and the howling is referred as **Acoustic feedback** [Fig.1]. The frequency range of acoustic feedback is around 2 to 5 kHz (Chung, 2004).

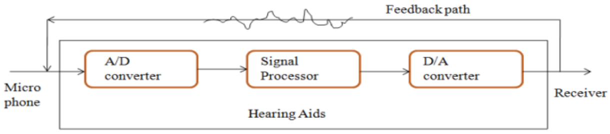

Fig.1 Acoustic Feedback in Hearing Aids

The feedback noise introduces multiple challenges to the hard-of-hearing persons, particularly reduces the maximum gain that the processor can produce and also it affects the speech intelligibility of the auditory system (Chung, 2004). Various feedback cancellation and suppression techniques are discussed to minimize the whistling or howling effect and to suppress the feedback, which are still futile.

II LITERATURE SURVEY

The feedback loop degrades the speech intelligibility of signal while using filter-windowing technique, when the noise level during quantization is high, particularly during A-D and D-A conversion (Johannes, Backstrom, 2015). The compression algorithms are applied to achieve the complete removal of redundancy. But still due to the correlation between source codec parameters, the residual redundancy is getting added.

The acoustic feedback referred as ‘Larsen-effect’, which occurs when open-loop gain of the system is greater than 1 and open-loop phase response of the system is a multiple of 2τ.

The Signal-to-Noise ratio (SNR) defines the performance of all algorithms, which is related as ratio of the ‘loudspeaker component’ to the ratio of ‘noise’ signal arriving on the microphone (Rombouts, et al, 2006). Varying the level of probe signal gives fast convergence and good SNR output for speech and strongly correlated signals; however, this is not attained for musical signals (Akhtara, Albub, Nishihara, 2019).

The gain-frequency response in the auditory system is adjusted (Kaustubh, Mahesh, 2015), to avoid the feedback oscillation. In the single channel hearing aids, the feedback cancellation can be done by reducing the overall gain in the system, while in multichannel hearing aids, there is a possibility to adjust the gain in the particular frequency where the feedback actually occurs.

A fixed null-steering beamformer is applied using multiple microphones in order to improve the acoustic feedback cancellation performance (Henning et al., 2016). Also, he minimized the min-max cost function by steering a spatial null in the beamformer in loudspeaker output. This will directly maximize the maximum stable gain of the system, which in turn results in the distortion of the signal, due to beamformer (Henning et al., 2017). A Linear-Prediction based feedback canceller is introduced with shaped probe noise, which provide less distortion and whistling problem at the loudspeaker output (Asutosh et al., 2019).

A TMS320C30 processor is used to implement a programmable noise generator to synthesis the background noise (Chassaing, Peterson, Horning, 1990). The algorithm shows efficiency in storage and computation, since he used only interpolation processing (and not decimation in prior) and never dealt with real-time signals.

The Realtime Multi-rate filtering design is modelled for the human ear in (Li, McAllister, Black, 1996), by projecting the frequency spectrum of microphone signal into the dynamic range of the hearing impaired. This provides the similar loudness perception heard by the impaired as well as normal listeners in each octave band. The TMS320C50 processor is used for the real-time implementation, which involves in different sampling rates for different processing stages. Although this method is proposed to achieve a lower order filter design and lesser computation cum storage, it leads to the generation of aliases due to the undesirable effect of sampling.

A programmable real-time multi-rate filter bank system is implemented (Li, McAllister, Black, 1996), based on frequency response characteristics of realistic audiograms using TMS320C50 processor. Although, outputs to the digital to analog converter are evenly spaced in time, the processing however cannot provide evenly spaced samples.

III IMPLEMENTATION

The hearing aid in general should facilitate the restoration of audibility, comfortable listening skills, better perception of vowel sounds, reduced dynamic range of speech signals and should discriminate the male and the female voice, for the better speech recognition. This paper dealt on applying the feedback cancellation algorithms in hearing aids for feedback control and to improve the speech intelligibility as well. A novel method on estimation of SNR based adaptive-feedback equalizers (SBAFE) algorithm is presented to develop an optimized hearing aids for the feedback less sound transmission and to achieve the better speech discrimination.

Linear-phase FIR filters are utilized due to their stability and simplicity in structure. The magnitude characteristics of FIR filters are matched with the given audiograms and parameters of synthesis of filters, related to their pass and stop bands. The yielded results are compared with standard filters and analyzed using the MATLAB tool (Ervin et al., 2015).

Here, considered the noisy-speech signals, with different levels of signal to noise ratio (SNR) from the NOIZEUS database. Initially, the speech segment mixed with Ambulance signal is chosen, with SNR of 10dB and 20 dB respectively in the basic orders of adaptive filter, whose sampling rate is 48000Hz and bit rate depth of 24. The length of the sampled data is 910080. The frequency components of the obtained signals are plotted in time domain using Fourier transformation by means of standard MATLAB tool. The plot in Fig. 2(a) represents the signal characteristics for the unfiltered signal of the ambulance sound and the plots in Fig. 2(b), (c), (d) and (e) produces the signal output, which emphasizes the noise characteristics, on the applying the adaptive-feedback algorithm using Butterworth filter of order 5, 7, 9 and 11 respectively.

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The spectrum plots (Fig. 2) clearly show that the noise of the system is equivalent to 5dB level and the second harmonic is less than -4.8dB level below the fundamental frequency. In tonal audiometry, it is important to have the sub-harmonic components below the noise level.

Since the frequency of the audio signal is from 20Hz to 20KHz, sampling frequency is considered to have 48KHz, to enable fine tuning of the frequency of generated signal. Thereby, it gives the Total Harmonic Distortion (THD) factor of about 57.54%, that requires to minimize the length of the adaptive filter of order 7 and 9.

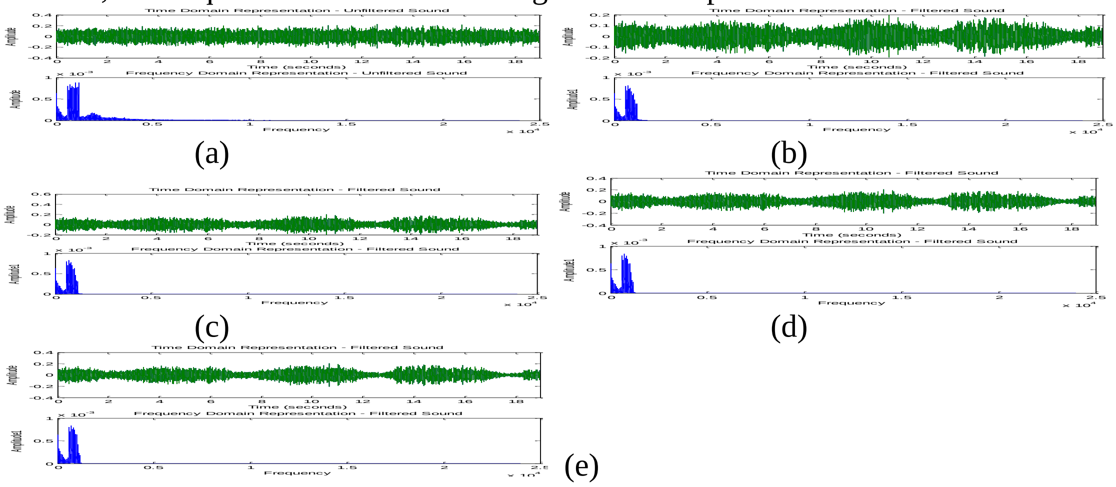

Fig. 2 (a) Measurement of Fundamental Frequency of unfiltered sound (b) Measurement of Fundamental Frequency of filtered sound of order 5 (c) filtered sound of order 7 (d) filtered sound of order 9 (e) filtered sound of order 11.

The total harmonic distortion is measured by, calculating the distortion in the power caused by harmonics, using SNR-Gain ratio, that gives the non-linearity level of the system. The speech enhancement algorithms in the frequency domain, provides the multiplication of noisy speech magnitude spectrum by gain $G(\tau)$, the function of SNR.

The gain function, $$G(\tau) = \frac{\tau}{\tau+1} \quad (1)$$

and the priori SNR, $$\tau > \frac{E[X^2]}{E[N^2]} \quad (2)$$

where, $E[X^2]$ and $E[N^2]$ are the estimation of magnitude spectra of the speech and noise signals respectively. The uncertainty of any ADC bit is $\pm\frac{1}{2}$, where q is the size of LSB and M is the number of bits.

$$\text{SNR (dB)} = 20\, log \frac{2^{M-1} * \frac{q}{\sqrt{12}}}{\frac{q}{\sqrt{12}}} \quad (3)$$

$$= 6.02\, M \;+\; 1.76\, dB \quad (4)$$

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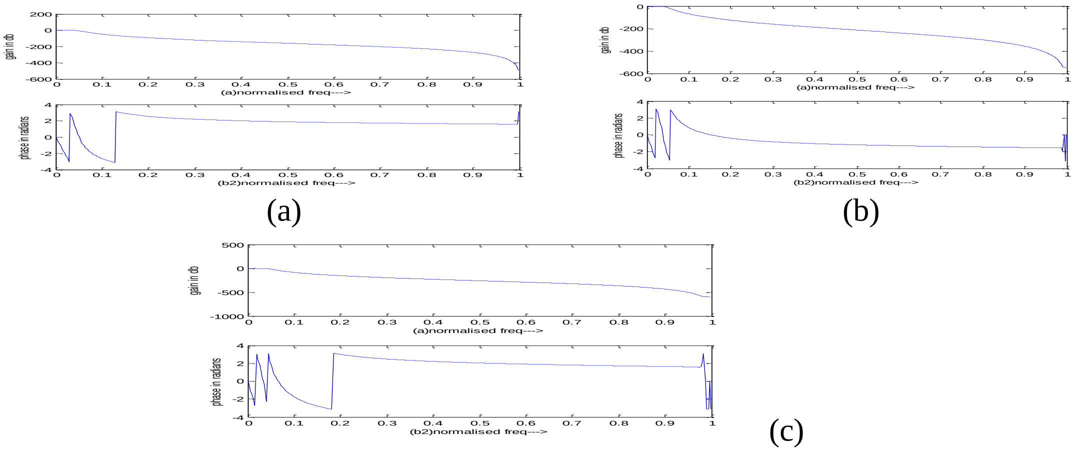

Fig.3 (a) Estimation of SNR of filtered sound of order 7 (b) Estimation of SNR of filtered sound of order 9 (c) Estimation of SNR of filtered sound of order 11

Using the above SNR estimation, a set of 10 sampled data values (SDV) from the of filtered sound of order 7, 9 and 11 are calulated using SBAFE adaptive algorithm which are processed through MATLAB and obtained their corresponding filtered sound values (FSV) and their corresponding plots are shown in Fig. 3(a), (b) and (c).

TABLE1

Comparison chart for Sampled data values (SDV) and Filtered sound values (FSV):

SDV	FSV	SDV1	FSV1	SDV2	FSV2
0.0105	4.07	4.0414	3.6765	3.8541	3.5062
0.0138	5.99	-2.8809	-3.1271	2.8145	3.0033
0.0217	4.44	-2.3133	-2.2805	0.0000028	0.00000258
0.0308	2.22	-3.5092	-3.2186	3.6293	3.2939
0.0371	8.55	0.6086	0.9716	4.4077	4.5255
0.0383	2.71	1.0483	1.2404	3.3119	3.0993
0.0402	7.38	-0.2639	-0.3587	2.9799	6.0237
0.0425	1.79	2.2557	1.9282	3.2085	2.8854
0.0423	3.94	0.5569	0.2133	8.8621	8.5456
0.0392	8	-2.7296	-2.8647	3.2115	3.1478

The sampled data values and its corresponding filtered sound values of the order 7 are illustrated as SDV1 and FSV1 and for order 9 is SDV2and FSV2 respectively, are tabulated and shown in table 1. It is clearly noted that for the filter of order 9 shows the minimized values for the sampled and filtered values, compared to seventh order adaptive filter. The prediction of SNR estimation contributes a real alternative for the real-time process, in which the prior knowledge is not required.

IV SIMULATION

The simulation was carried out on the DSP evaluation board **TMS320C55X™**, the high-performance, low-power processor including Serial-Port Interface (SPI) with four chip-cycle time selects of 150-MHz clock rate. It consists of four inter-IC sound (I2S bus™) for data as dual multipliers transport. It has tightly-coupled FFT hardware accelerator and software-compatible with C55X devices ("TMDX5505EZDSP", n.d). On-chip memory options of up to 320KB that reduce the need for external memory in several applications, providing the cost-efficient way to boost performance and for saving the system power.

The C5505 DSPs are supported by **Code Composer Studio™** (CCS) Integrated Development Environment (IDE). The CCS-IDE is a fully functioning high level C-code used to write, download, run and debug the programs, on to the hardware, for each application. It gives emphasis on real-time and low-power operations. The Texas Instruments (TI) uses the Code Support Library (CSL) for simplifying the design process.

The DSP processor which uses the CCS, should have **CODEC** (Coder - Decoder) in which the Sampling rate, Input ADC gain, Output DAC gain can be adjusted. TMS320C55X™ DSP processor kit comprising the audio codec TLV320AIC3204 is utilized here, to obtain the simulation results for system analysis.

To maximize the performance of audio codecs in processor, external clock values with frequencies 256 times the sampling rate is selected. For common clock values of 12.288 MHz and 11.2896 MHz, sampling rate of 48kHz or 44.1kHz is required respectively. The frequency dividers in the audio codecs are used to provide the required sampling rate, through an input clock. By using these valued clocks, the power consumption is reduced and avoids the usage of internal PLL.

The ADC gain considers the value between 0 and 48, i.e., 0 dB for line input, 30 dB for microphone input and 48 dB for low output microphone input.

Equalization that alters the frequency response of an audio system using linear filters is applied here, to find the reversal of distortion incurred by a signal transmitted through the input channel. The SNR-based adaptive-feedback equalizers (SBAFE) algorithm is applied in adaptive filter, to adjust the frequency components of audio signal, to perform the equalization of signals by using an 'equalize' function. The structure of the equalizer (i.e., number of taps), the adaptive algorithm (i.e., step size) and the signal constellation used by modulator are based on SBAFE algorithm. The Fig. 4 (a), (b) and (c) show the equalization of corresponding received signals for the SBAFE filter, of order 7 and 9 respectively.

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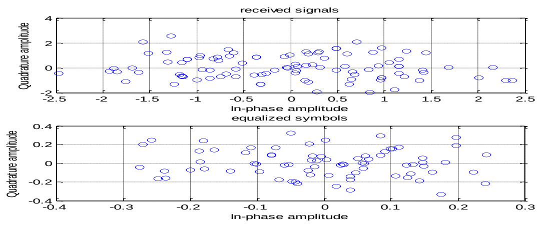
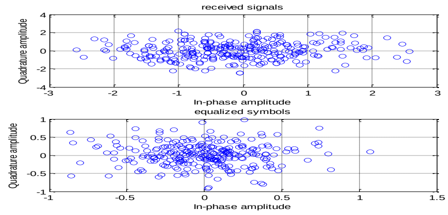
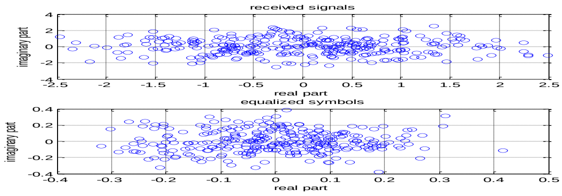


(a)



(b)

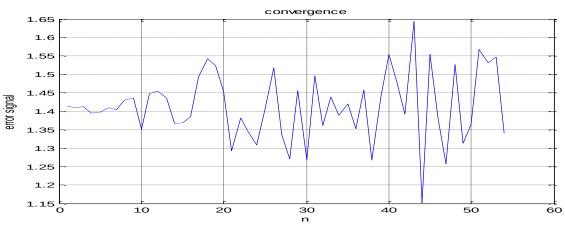


(c)

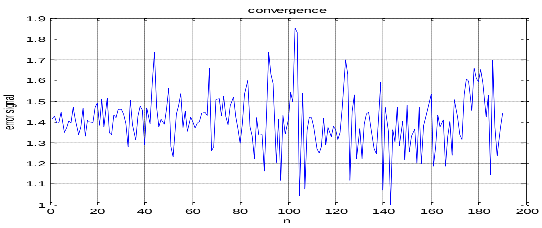
Fig. 4 (a) Equalization of unfiltered sound (b) Equalization of filtered sound of order 7 (c) Equalization of filtered sound of order 9

The equalization process is most commonly used to correct the signals that sound unnatural, to adjust the balance between frequency components, within an electronic signal and modifies the frequency of sound. The above plots in Fig. 4, show the variation in equalization of corresponding received signals for the respective filters. Also, it is clear that, a proper equalization is carried out in ninth order SBAFE filter, shown in Fig. 4 (c).

Communication in signal processor is carried out through RS232 serial interface access, between DSP kit and system terminal to execute quick data transfer of the program (Roberto, Stanko, Vladimir, 1997).



(a)



(b)

Fig. 5 (a) Convergence of signal using SBAFE filter of order 7 (b) SBAFE filter of order 9

The convergence of signal using SBAFE adaptive algorithm of order 7 and 9, shown in Fig. 5 (a) and (b). DSP processor enables the convergence of signal of SBAFE filter for fine tuning of frequency of order 7 and 9, for the audio signal frequency. And thereby gives, the Total Harmonic Distortion (THD) factor reduced to 28.33% with respect to total squared error and convinced for any length of filter.

V CONCLUSION

The developed estimation of SNR based adaptive-feedback equalizers (SBAFE) algorithm is shown with the estimation of unknown system parameters and it has the capacity to estimate as many parameters as long as the necessary data are available. This indicates that the algorithm developed is suitable for the simulation of feedback control process, when the SNR estimation is quite good. Thus, the estimation-based approach used to eradicate redundancy, through which the process becomes a real and simple alternative step for real-time signal processing, where the prior knowledge is not required. Though the prediction accuracy produced by the developed estimation is not perfect, the error indices obtained are quite acceptable.

Also, the audio codecs in processor, are sensitive to noise issues generated from multiple sources. The distortion in A-D converter of codec, affects the spectral purity of generated signal at its output, which had been evaluated by the using network analyzer. It is suggested to optimize the codec, design layout, internal power supplies and registers, before evaluation.

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