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Research Paper

Real Time Implementation of Acoustic Echo Cancellation Using TMS320C6713 DSK

Geetha Jayaram*
ECE Dept., Canara Engineering College, Mangalore-575007
*Email: geethajayaram4@gmail.com

Abstract

Reverberation or Echo is a delayed and twisted adaptation of a unique sound or electrical sign which is reflected back to the source. When the reflected wave arrives a couple of several milliseconds after the main sound, it is heard as a particular reverberation. These are undesirable; thus this anticipate actualizes an Acoustic Echo Canceller System on TMS320C6713 DSK. The principle modules of this reverberation canceller are an Adaptive Filter System utilizing NLMS, VSS-NLMS.

Keywords: Acoustic echo cancellation, reverberation cancellation, NLMS algorithm, GSER algorithm, echo return loss enhancement.

1 Introduction

Echo is a phenomenon where a postponed and distorted version of an original sound or electrical signal is reflected back to the source. Echoes of our discourse are heard as they are reflected from the floors, walls, dividers and other neighboring articles. In the event that a reflected wave touches base after a brief timeframe it is considered as a contortion or resonation. Be that as it may, when the main edge of the reflected wave arrives a couple of many milliseconds after the primary sound, it is heard as a particular reverberation. Since the appearance of telephony echoes have been an issue in communication systems. Acoustic Echo Canceller is utilized on each as a part of the communication frameworks and for better clarity during video conferencing or tele-conferencing.

The quest for enhanced voice quality has prompted concentrated effort in the area of reverberation cancellation. By utilizing reverberation cancellation innovation, the nature of communication can be enhanced fundamentally. This paper describes implementation on TI's Software Development Tool: Code Composer Studio Version 5 (CCSV5) and Spectrum Digital's TMS320C6713 DSK Board which is mainly used for audio interfacing.

2 Implementation of echo cancellation using nlms and vss-nlms algorithm

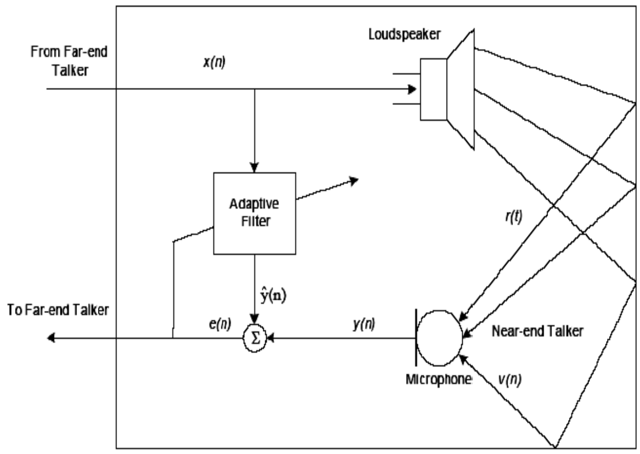

Figure 1: Block diagram of adaptive filter.

Figure 1 shows the echo canceller system using adaptive filter. The estimated echo, y^(n), is generated by passing the reference input signal, x(n), through the adaptive filter, h^(n) that is ideally matched with the transfer function of the echo path, h(n). The echo signal r(t), is produced when x(n) passes through the echo path. The echo r(r) plus the near-end talker or disturbance signal v(n), constitute the desired response, for the adaptive canceller. The two signals x(n) and r(n) are correlated since the latter is obtained by passing x(n) through the echo path. The error signal e(n) is given by equation 1:

$$e(n) = (y) - \widehat{y(n)} \tag{1}$$

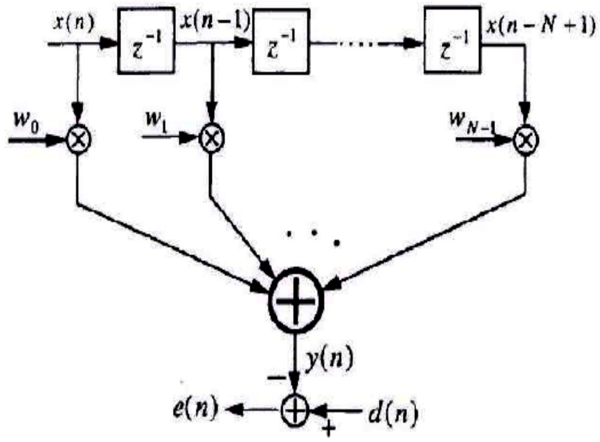

Figure 2: Block diagram of adaptive filter.

Figure 2 shows the block diagram of adaptive filter; here w represents the coefficients of the FIR filter tap weight vector, x(n) is the input vector samples, z-1 is a delay of one sample period, y(n) is the adaptive filter output, d(n) is the desired echoed signal and e(n) is the estimation error at time n. The aim of an adaptive filter is to calculate the difference between the desired signal and the adaptive filter output, e(n). This error signal is fed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize a function of this difference, known as the cost function. In the case of acoustic echo cancellation, the optimal output of the adaptive filter is equal in value to the unwanted echoed signal.When the adaptive filter output is equal to desired signal the error signal goes to zero. In this situation the echoed signal would be completely cancelled and the far user would not hear any of their original speech returned to them.

2.1 NLMS algorithm

The LMS calculation is a kind of versatile channel referred to as stochastic angle based calculation as it uses the inclination vector of the channel tap weights to join on the ideal wiener arrangement. With every emphasis of the LMS calculation, the channel tap weights of the versatile channel are updated. One of the essential weaknesses of the LMS calculation is a settled stride size parameter for each cycle. This requires a comprehension of the measurements of the info signal before starting the versatile separating operation.

2.2 Design Aspects of NLMS Algorithm

Step 1: Generation of echo,

$$echo(n)s = (n) \odot h_{rir} \tag{2}$$

Where s(n) is the main signal and h_{rir} is the room impulse response.

$$h_{rir} = \alpha_0 \delta(n - n_0) \tag{3}$$

α is the attenuation.

Implementation using this equation is called single delay buffer logic based.

Step 2: Generation of *hatecho* here FIR filtering is used

$$\widehat{echo}(n) = \widehat{s(n)} \odot \widehat{h_{rir}} \tag{4}$$

Here FIR filtering is used. That is generating filters using the equation

Step 2.1: Set the sampling rate $f_s = 48$ KHz and stop band frequency should be less than $f_s/2$

Step 2.2: Implementation of bank of filters using equation

$$y(n) = \Sigma N - 1k = 0b_x(n-k) \tag{5}$$

Where x(n) is the filter input y(n) is the filter output b_k filter coefficient N order of filter

Step 2.3: Here using tempx buffer in order to produce input vector. So first step is to clear the tempx buffer.

Step 2.4: To implement x(n-k) linear buffer strategy is used.

Linear buffer strategy: Keeping the present input of the sample on the top of the tempx buffer.

Step 2.5: Find the dot product between filter coefficient and tempx buffer to get y(n)

Step 2.6: Updating tempx buffer. y(n)getting by this process is the $\widehat{echo}$. That is the output of the adaptive filter.

Step 3: Echo cancellation

$$e(n) = echo(n) - \widehat{echo}(n) \tag{6}$$

Step 4: Update the filter coefficients generated in the FIR filtering.

Where

$$\widehat{h}_k + 1 = \widehat{h}_k + (\mu * errorsamp * inputvector) \tag{7}$$

Where

$$\mu = \frac{1}{signalenergy} \tag{8}$$

Step 5: update input vector

Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm and this is an acceptable increase considering the gains in stability and echo attenuation achieved.

2.3 VSS-NLMS or GSER Algorithm

Summed up Square-Error-Regularized NLMS calculation (GSER) shows great execution with quick meeting, speedy following and low relentless state MSE. The utilization of the VSS-NLMS calculation will bring about an enhanced execution. The part of epsilon is to keep the related denominator from getting excessively near zero, to keep the channel from uniqueness. Be that as it may, in utilizations of discourse flags, a too little epsilon may make the denominator near zero while a too large epsilon will back off the adjustment of the channel. Here mu is figured by utilizing diverse parameters.

$$\mu = (error_{var} * \mu_0)((error_{var} * signalenergy) + \theta) \tag{9}$$

Where,

$\mu_0 = 1, \beta = 0.1, \theta = 6$

$$error_{var} = (1 - \beta) * error * error \tag{10}$$

The execution of settled stride size and variable stride size is found by measurements. The measurements decided for the execution correlation are NSCE and ERLE. For good reverberate cancelation framework ERLE quality ought to be certain most extreme and NSCE worth ought to be negative least Echo Return Loss Enhancement (ERLE).

2.4 Echo Return Loss Enhancement (ERLE)

With a specific end goal to assess the nature of the reverberation cancelation calculation the measure of ERLE was utilized. ERLE, measured in dB is characterized as the proportion of the momentary force of the sign, d(n) and the prompt force of the leftover mistake signal, e(n) instantly after cancelation. ERLE measures the measure of misfortune presented by the versatile channel alone. Numerically it can be defined as

$$ERLE = 10log_{10}\frac{\|echo\|^2}{\|error\|^2} \tag{11}$$

Where error = echo - $\widehat{echo}$

For a good echo canceller circuit, an ERLE in the range of 30 dB – 40dB is considered to be ideal. ERLE is plotted in dB along the y-axis and the number of samples along the x-axis. The plot of ERLE implies that the ERLE for this algorithm attained the required value. ERLE value should be positive maximum.

2.5 Normalized Squared Coefficient Error (NSCE)

NSCE is used to evaluate the performance of the algorithms. The NSCE is defined as

$$NSCE = 10log_{10}\frac{\|h\|^2}{\|h\|^2} \tag{12}$$

Where h is the room impulse response and impulse response of FIR filter. NSCE is plotted in dB along the y-axis and the number of samples along the x-axis. The plot of NSCE implies that the NSCE for this algorithm attained the required value. NSCE value should be negative minimum.The ERLE and NSCE values are found by processing an additive white gaussian signal. Additive white gaussian signal is required because highly uncorrelated samples are needed to be differentiated.

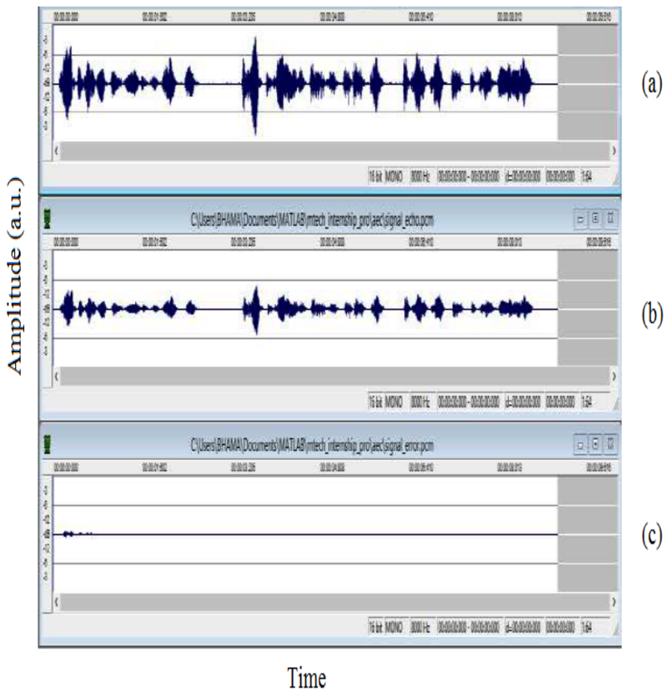

Figure 3: a) Main signal, b) echo signal and c) error signal in Wavosaur.

3 Results and discussion

MATLAB code can be written according to the algorithm. The output is analysed using WAVOSAUR software. The signal created in MATLAB has to import to WAVOSAUR. The signals like main signal, echo signal and residual error signal in WAVOSAUR are shown in Figure 3. In MATLAB residual error will be less. But while executing in CCSV5 residual error will be zero. The main signal, echo signal and error signal in CCSV5 are shown in Figure 4. ERLE and NSCE graph for fixed step size NLMS is shown in Figure 5 and Figure 6. Similarly, ERLE and NSCE graph for GSER NLMS is shown in Figure 7 and Figure 8.

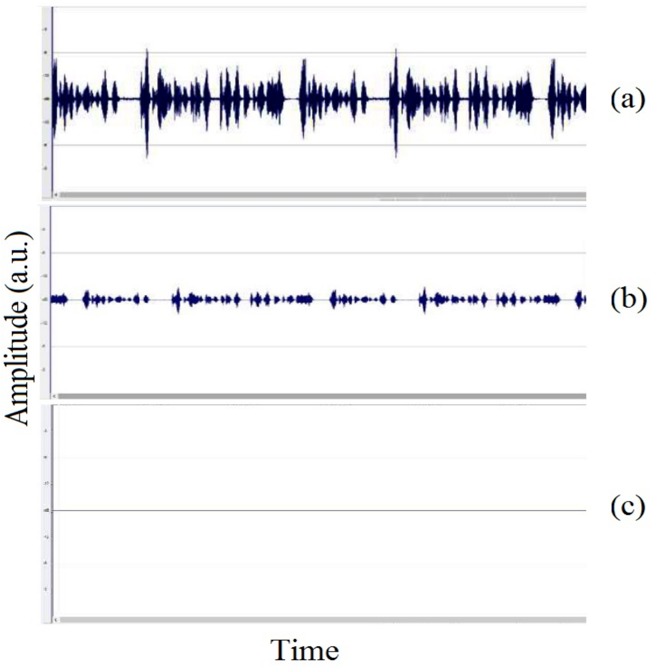

Figure 4: a) Main signal, b) echo signal and c) error signal in CCSV5.

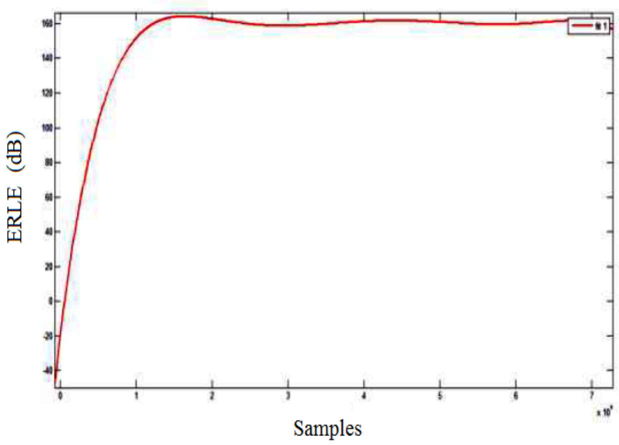

Figure 5: ERLE curve of NLMS based on fixed step size.

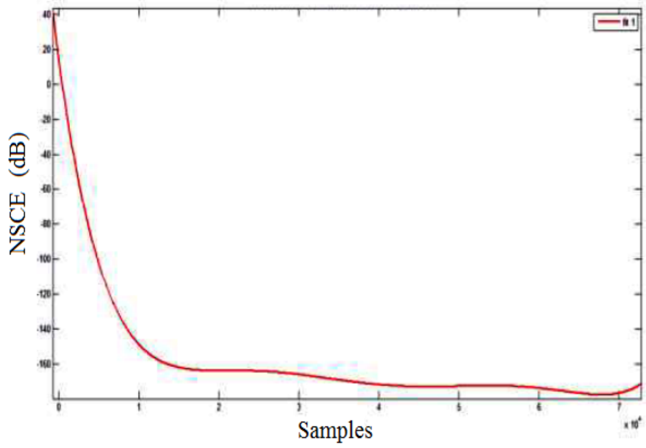

Figure 6: NSCE curve of NLMS based on fixed step size.

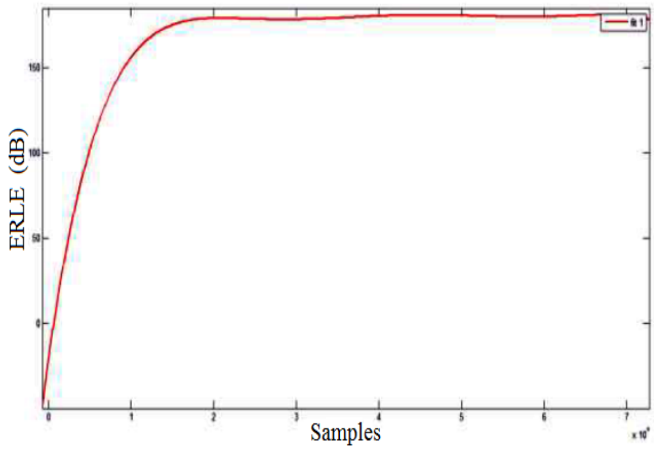

Figure 7: ERLE curve of GSER based VSS NLMS.

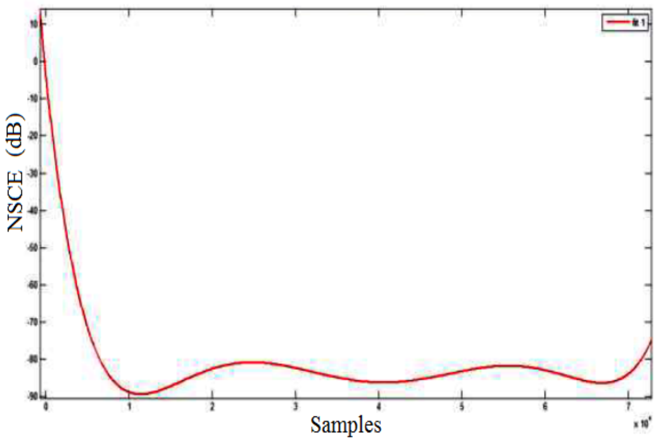

Figure 8: NSCE curve of GSER based VSS NLMS.

4 Conclusion

Rapid innovations in the quest for enhanced voice quality has prompted serious investigations into the zone of reverberation cancellation. Such research is led with the aim of generation of techniques that can decrease foundation commotion and evacuate half breed and acoustic echoes before any transcoder. By utilizing reverberation cancellation innovation, the nature of discourse can be enhanced altogether.

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