International Journal of Electrical and Electronics Engineers ISSN- 2321-2055 (E)
http://www.arresearchpublication.com IJEEE, Vol. No.3, Issue No. 02, july-Dec, 2011

Acoustic echo cancellers for mobile devices

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Abstract:

Communication technology has advanced to the point where teleconferences that use communication lines to connect remote conference rooms are taking place every day. The use of a centralized acoustic echo canceller in the mobile switching center of a GSM network allows the operator to enhance the audio quality before transmission to all subscribers. However, the main problem is the impact of the speech coder/decoder nonlinearities along the echo path. In this paper, we propose a combined acoustic echo canceller/post-filter based on perceptual properties to reduce the coding noise. Theoretical and experimental comparisons against classical solutions are given to evaluate the performance of the proposed approach in a centralized context of echo cancellation.

Echo is the time delayed version of the original signal. Acoustic echo results from a feedback path set up between the speaker and the microphone in a mobile phone, hands-free phone .Echo can degrade the quality of service in telecommunication. Therefore echo cancellation is an important part of communication systems. A new approach based on the number of coefficients in an adaptive finite impulse response filter based acoustic echo cancellation setup is an important parameter, affecting the overall performance of the echo cancellation. Too few coefficients give under modeling and too many cause slow convergence and an additional echo due to the mismatch of the extra coefficients This paper focuses on the use of adaptive filtering techniques to cancel this unwanted echo thus increasing communication quality

Keyword: Round Trip Delay (RTD), Acoustic Echo Cancellation (AEC), Normalized Least Mean Square Algorithm (NLMS), Mean Square Error (MSE), Sub band Adaptive Filters (SAF), Adaptive Combination Normalized Sub band Adaptive Filters (NSAF), signal to noise ratio (SNR).

1. Introduction:

While the telecommunications industry is changing with new technologies, such as packet voice transport, and services, such as VoIP, one item is not changing: customers' expectations of voice quality. Engineers long ago optimized traditional network equipment for the human voice, ear, and the nature of conversation, and carriers have grown their networks with this equipment to provide a consistent level of voice quality for their customers. As such, today's customers have an established expectation of voice quality and this expectation has become a barrier to entry for new technology in a carrier's network. Even so, due to the economic advantages, voice will increasingly be transported over packet networks. Although considerable study and product development have enabled carriers to begin transitioning to packet voice networks, there is one area that has not had proper attention: hybrid and acoustic echo cancellation. The history of echo cancellation begins on 10th July 1962.In telephones and teleconferencing; a reflection can occur where there is an impedance mismatch. If the reflected signal reaches the far end subscriber with a RTD of a few milliseconds then it is perceived as reverberator. If RTD exceeds a few tens of milliseconds the reflection is known as distinct echo. Echo suppressor was used to remove echo which introduces a very large transmission loss in return path. A new technique that did not interrupt the echo return path called echo cancellation. The AEC estimates the characteristics of the echo path and generates a replica of the echo. The echo is then subtracted from the received signal. Adaptive filters are used as echo canceller. The normalized least-mean-square (NLMS) algorithm is one of the most popular adaptive filters. Speech input signal of the adaptive filter is highly correlated and the impulse response of the acoustic echo path is very long. These two characteristics will slow down the convergence rate of the acoustic echo canceller. So sub band adaptive filtering is used to solve these problems. In SAFs, each sub band uses an individual adaptive sub filters. Recently, an adaptive combination of full band adaptive filters has been proposed in [8], and its mean-square performance has been analyzed in. More recently, a combination of SAFs for AEC has been proposed [10], which is based on a conventional sub band structure. In this paper we propose a new scheme for adaptive combination of sub band adaptive filters deal with the tradeoff problem encountered in AEC which are implemented by NSAFs. The NSAF can be viewed as a sub band generalization of the NLMS based adaptive filter. In the proposed combination, mixing parameter that controls the combination is adapted by means of a stochastic gradient algorithm which employs the sum of squared sub band errors as the cost function.

2. Simulink Model Of Full band Adaptive Filter Acts As Acoustic Echo Cancellation

A block diagram of AEC is shown in Fig.1 [2].

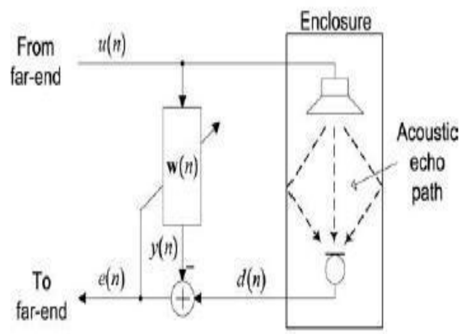

Fig.1

Speech signal originating from loudspeaker is received by microphone passing through acoustic echo path. The acoustic echo is removed by adaptive filters. The d (n) signal contains the speech signal and noise signal. The goal of the adaptive filter **w** (n) is to produce a replica of the echo signal *y* (*n*). *y*(*n*) can be used to cancel the echo by subtracting it from the microphone signal *d*(*n*) resulting in error free signal e(n)[3].Algorithm for AEC is as follows [2]: Adjustable tap weights can be expressed as:

$w(n) = [w_0\ (n), w_1\ (n), \ldots . w_{M-1}\ (n)]^T = w^T\ (n)u(n)$..(1)

Input signal can be expressed as

U (n) = $[u(n), u(n-1), \ldots u(n-M+1)]^T$ …. (2)

The output signal y(n) of adaptive filter is the multiplication of w(n) and u(n).

Y (n) = $\sum_{m=0}^{M-1} w_m\ (n)u(n-m) = w^T\ (n)u(n)$…. (3)

The error signals difference between the desired responses d (n) and filter response y (n) is expressed as:

e (n)=d(n)- $w^T\ (n)$u(n)……..(4)

3. Simulink Model of Nsaf and Its Adaptive Combination

3.1 Simulink Model of NSAF

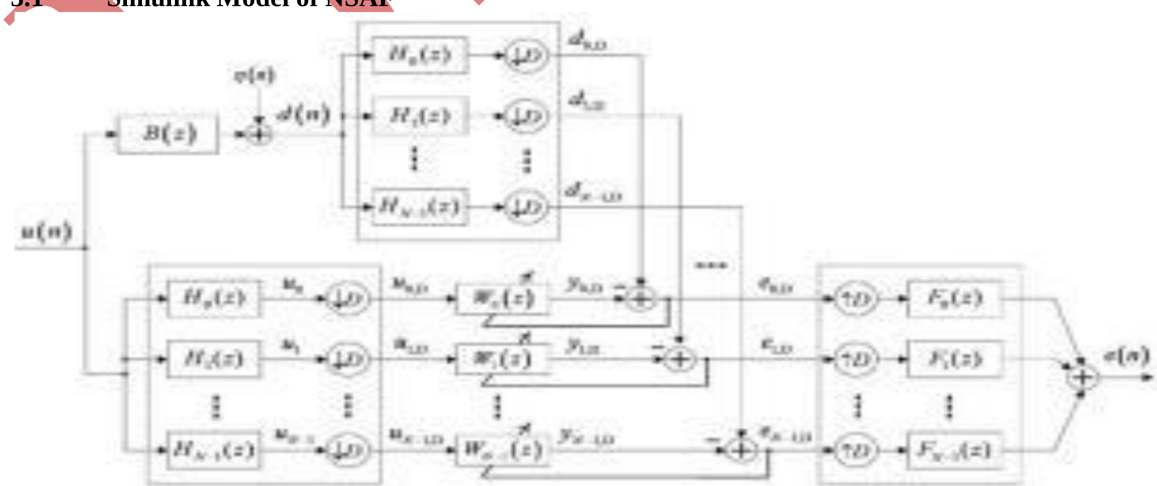

Fig.2

The input speech signal u (n) and desired output d(n) are decomposed into N spectral bands using analysis filters. Analysis filtering is then performed in these sub-bands by a set of independent filters (h0 (n), h1 (n),…hM-1(n))[2].The sub band signals are further processed by individual adaptive sub filters Wi(z).Each sub band is computing error signal e(n). By updating the tap weights, minimizes the sub band error signal. The full band error signal *e(n)* is finally obtained by interpolating and recombining all the sub band error signals using a synthesis filter bank. The updating equation of NSAFs is written as follows [3]:

$w_i(n+1) = w_i(n) + \mu_i \sum_{i=0}^{N-1} \frac{u_{i(n)}}{\delta + \|u_{i(k)}\|^2} e_i(n)$…..(5)

Where i=1,2,…N-1. Whe

$u_i(k) = [u_i(kN), u_i(kN-1) \ldots, u_i(kN-M+1)]T$,M is the length of the adaptive filter **w**ˆ (*k*) ,μ is the step-size, and δ is the
regularization parameter [3].

3.2 Simulink Model of Adaptive Combination of NSAFs

The block diagram of adaptive combination of normalized sub band adaptive filters is shown in fig.3[9].

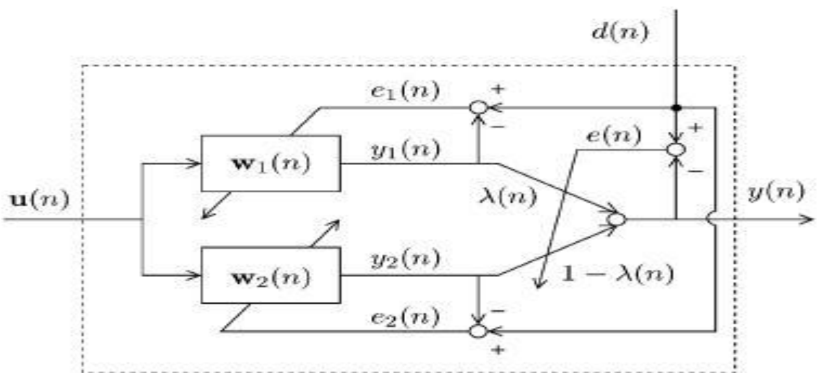

Fig. 3

A large step size yields a fast convergence rate but also a large steady state MSE [7].To achieve fast convergence rate and small steady state MSE, adaptive combination of sub band adaptive filters is done .So that large step sizes adaptive filters give fast convergence rate and small step sizes adaptive filters give small steady state MSE. So idea becomes to adapt different step sizes filters independently and combination is carried out by using a mixing parameter lambda.

The input signal is U (n)=$[u(n), u(n-1), \ldots u(n-M+1)]^T$, weight vectors are $w(n) = [w_0(n), w_1(n), \ldots . w_{M-1}(n)]^T = w^T(n)u(n)$. So output becomes

y(n)=$w^T(n)u(n)$.

The update eq. of sub band adaptive filter is given in eq. 5.

$w_i(n+1) = w_i(n) + \mu_i \sum_{i=0}^{N-1} \frac{u_{i(n)}}{\delta + \|u_{i(k)}\|^2} e_i(n)$ Where

$e_1(n)$= d (n)-y_1(n) and $e_2(n)$= d(n)-y_2 (n) and

$y_1(n) = w_1^T(n)u_n$, $y_2(n) = w_2^T(n)u_n$.

Consider $\mu_1 > \mu_2$, then $w_1(n)$ adaptive filter has faster convergence rate and large steady state MSE whereas $w_2(n)$ has slower faster convergence rate but small steady state MSE.So our purpose is to get large convergence rate and small steady state MSE ,So combine both adaptive filters. The output of overall filter is: Y(n)=$\lambda(n)\, y_1$(n)+[1- $\lambda(n)$] y_2(n)……….(6) where is mixing parameter. The overall filter with tap weight factor of the form is: w (n) = $\lambda(n) w_1$(n)+ [1- $\lambda(n)$] w_2(n).For adaptation of mixing parameter $\lambda(n)$,use stochastic gradient method to minimize error of

overall filter $e^2(n) = [d(n) - y(n)]^2$.

However instead of directly adjusting (n), we will adapt a variable $\alpha(n)$ that defines (n) as a sigmoid function. (n)=sigma

[α (n)]=$\frac{1}{1+e^{-\alpha(n)}}$…….(7)

The update eq. for α(n) is given as:

α(n+1)=α(n)+μ e(n) [y_1(n)- y_2(n)] (n)[1- (n)]……….(8)

where μ is the step size for adapting α(n).since the mixing parameter is defined by the sigmoidal function[15],which insists the mixing parameter to lie exactly inside the interval (0,1), this combination is convex combination.[14][3].

4. Improved Adaptive Combination of Normalized Sub band Adaptive Filters

In improved adaptive combination of normalized sub band adaptive filters, we take the following assumption.

1) d(n) and u(n) are related by a linear regression model d(n)= w_o^T (n)u(n)+ $\eta(n)$

for some unknown weight vector w_0 of length M and where $\eta(n)$ is an independent distributed noise.

2) The initial condition w_1 (0), w_2 (0) and a (0) are independent of u (n), d (n) or all n.

5. Experimental Results

The full-band and sub-band systems, adaptive combination of sub band adaptive filters and its improvement were modeled in Mat lab Simulink and many simulations for different inputs and number of sub-bands was performed. For the adaptive algorithm several different algorithms can be used, but the most common one is the normalized least mean squares (NLMS). The order of the NLMS filters was chosen from N=64 to N=2. The designs were made in Mat lab-Simulink environment and the simulations were run for 5000 samples for Gaussian noise and sine wave input, respective 12*104 samples in the case of speech input. A reverberating effect was added to the input by an artificial Schroeder I reverberate which contained four comb filters in parallel and two all-passes filters series connected. The first estimation of a system capability is represented by the (output error-voice input), but in order to measure its potential, Echo Return Loss Enhancement (ERLE) should be computed; it is defined as the ratio of the power of the desired signal over the power of the residual signal. Comparison between full band, sub bands, adaptive combination, and their improvement is done based on SNR and MSE and by using output error- voice input and ERLE

6. Conclusion

The NSAF is a good candidate for implementing acoustic echo cancellers because of its fast convergence rate. However, it requires a tradeoff between fast convergence rate and small steady-state MSE. This paper presented an adaptive convex combination of two NSAFs to solve this problem. In addition to the conventional coupling update method for component filters, we also proposed a coupling update mechanism which requires less number of adaptive filters as than used in conventional method. To verify the effectiveness of the proposed scheme, simulations using different input signals as well as system noises with different SNRs were performed. The experimental results demonstrated that the proposed scheme can obtain improved performance as compared to the conventional NSAF.

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