DSP LAB PROJECT

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ECHO GENERATION AND CANCELLATION USING ADAPTIVE FILTERS

Aim: To generate and remove echo signal from original signal by designing adaptive filter using LMS algorithm.

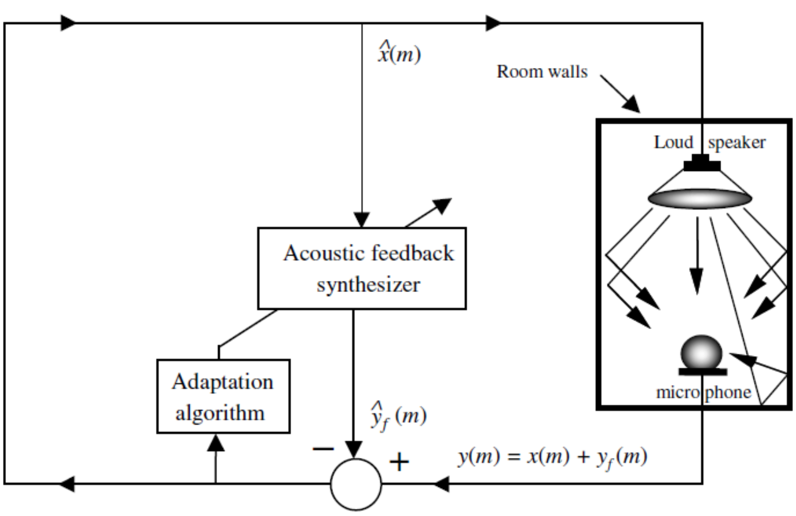
Theory :

An **Acoustic Echo** is due to Acoustic coupling between the speaker and the microphone in hands-free phones, mobile phones and teleconference systems

An **Hybrid Echo** is due to Electrical line echo due to mismatch at the hybrid circuit connecting a 2-wire subscriber line to a 4-wire truck line in the public switched telephone network.

An **adaptive filter** is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. Because of the complexity of the optimization algorithms, almost all adaptive filters are digital filter. Adaptive filters are required for some applications because some parameters of the desired processing operation (for instance, the locations of reflective surfaces in a reverberant space) are not known in advance or are changing. The closed loop adaptive filter uses feedback in the form of an error signal to refine its transfer function.

As the power of digital signal processor has increased, adaptive filters have become much more common and are now routinely used in devices such as mobile phones and other communication devices, camcorders and digital cameras, and medical monitoring equipment.



LMS:

Least mean square algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean squares of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time.

Parameters: $p =$ filter order

$\mu =$ step size

Initialisation: $\hat{\mathbf{h}}(0) = \mathbf{zeros}(p)$

Computation: For $n = 0, 1, 2, \ldots$

$$\mathbf{x}(n) = [x(n), x(n-1), \ldots, x(n-p+1)]^T$$

$$e(n) = d(n) - \hat{\mathbf{h}}^H(n)\mathbf{x}(n)$$

$$\hat{\mathbf{h}}(n+1) = \hat{\mathbf{h}}(n) + \mu\, e^*(n)\mathbf{x}(n)$$

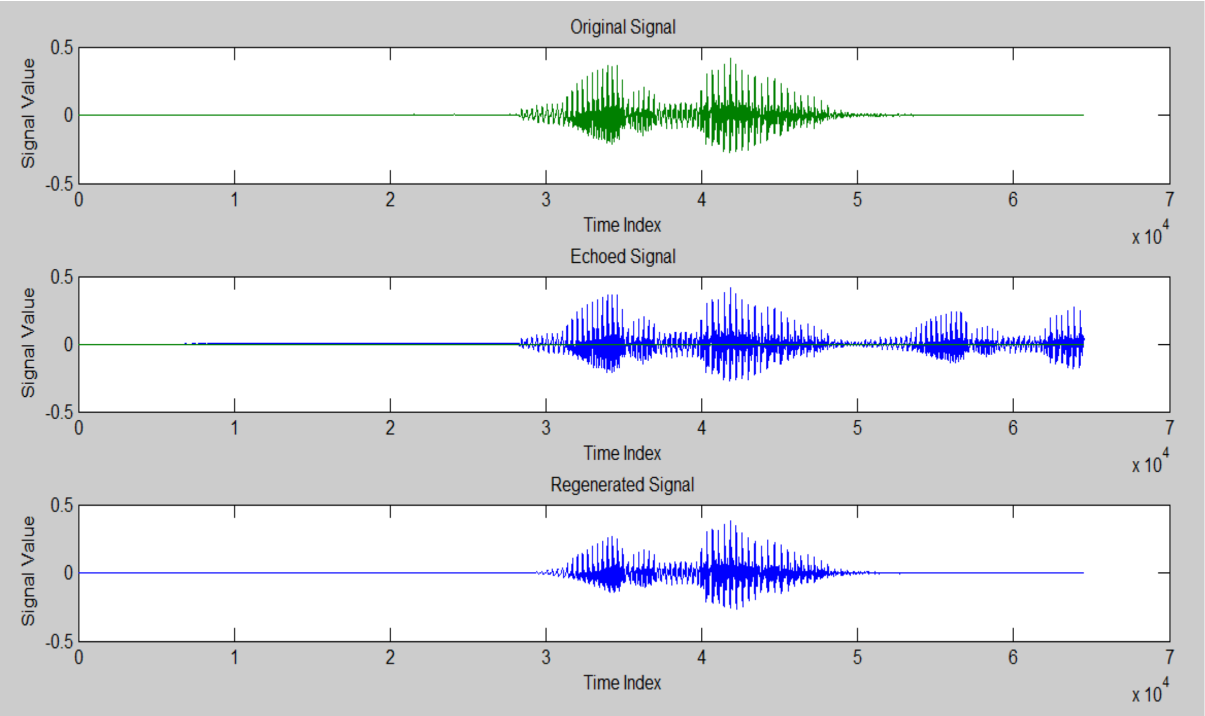
Normalised lms algorithm

$w_{l,k+1} = w_{lk} + \left(\frac{2\mu_\sigma}{\sigma^2}\right) \epsilon_k \, x_{k-l}$ in which case the convergence criteria becomes: $0 < \mu_\sigma < 1$.

Matlab code:

```
clear all;
% Audio Input
 [x,Fs] = audioread('Nirma.wav');
 sound(x,Fs)

pause(4);
 delay = 0.5; % 0.5s delay
 alpha = 0.65; % echo strength
 D = delay*Fs;

% Echoed Signal
y = zeros(size(x));
 y(1:D) = x(1:D);

 for i=D+1:length(x)
   y(i) = x(i) + alpha*x(i-D);
 end

% echoed audio signal
 sound(y,Fs);
 audiowrite('Nirma Echoed.wav',y,Fs);

%Scalling Input and Echoed Output
p=x(:);
p=p(1:length(p)/2);
p=p';

z=y(:);
z=z(1:length(z)/2);
z=z';

%Generating Original Signal from Echoed Signal by Adaptive Filter
b = fir1(31,0.5);     % FIR system to be identified
mu= 0.008;            % LMS step size.
ha= adaptfilt.lms(32,mu);
[res,e] = filter(ha,p,z);

%Regenerated Sound from Echoed Sound
sound(res,Fs);
audiowrite('Nirma Regenerated.wav',res,Fs)

%Plots
%Original Signal
subplot(3,1,1); plot(x);
title('Original Signal');
xlabel('Time Index'); ylabel('Signal Value');

%Echoed Signal
subplot(3,1,2); plot(y);
title('Echoed Signal');
xlabel('Time Index'); ylabel('Signal Value');

%Regenerated Signal
subplot(3,1,3); plot(res);
title('Regenerated Signal');
 xlabel('Time Index'); ylabel('Signal Value');
```

Results :



Applications:

- Chorus Voice Generation from one person voice
- Telephone lines echo removal

Conclusion:

- From the above design, we conclude that Echo cancellation is necessary in regions especially like conference rooms and due to internal circuits.
- In this we design the adaptive filter by using the LMS algorithm. For adaptive filter we predict the present sample with the help of previous samples.