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A New Algorithm for Acoustic Echo Cancellation using Adaptive Filter

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Abstract

The rapid recent growth of technology has changed the whole communications. However, the presence of a large acoustic coupling between the loudspeaker and microphone would produce a loud echo that would make conversation difficult. The echo suppressor offers a simple but effective method to counter the echo problem. However, the echo suppressor can support only half-duplex communication. The full-duplex communication allows both speakers to talk at the same time. This objective of this research is to produce an improved echo cancellation algorithm, which is capable of providing convincing results. The adaptive filters are used to reduce the unwanted echo and increasing the quality of speech signal. This paper focus on different adaptive filters algorithms and characterzied by the NLMS , DTD and NLP,LMS algorithms, and the calculation of different adaptive and echo cancellation algorithms must be simulated utilizing MATLAB.

Keywords: echo cancellation, Adaptive filters, LMS, NLMS,DTD, NLP algorithm, acoustic echo cancellation

1. Introduction

1.1 . NEED FOR ECHO CANCELLATION

In this new age of global communications, wireless phones are regarded as essential communications tools and have a direct impact on people's day-to-day personal and business communications. As new network infrastructures are implemented and competition between wireless carriers increases, digital wireless subscribers are becoming ever more critical of the service and voice quality they receive from network providers. Subscriber demand for enhanced voice quality over wireless networks has driven a new and key technology termed echo cancellation, which can provide near wire line voice quality across a wireless network.

This paper discusses the overall echo problem. A definition of echo precedes the discussion of the fundamentals of echo cancellation and the voice quality challenges encountered in today's network.

1.2. TYPES OF ECHO

In telecommunications networks there are two types of echo. One source for an echo is electrical and the other echo source is acoustic, the electrical echo is due to the impedance mismatch at the hybrids of a Public Switched Telephony Network, (PSTN), exchange where the subscriber two-wire lines are connected to four-wire lines. If a communication is simply between two fixed telephones, then only the electrical echo occurs. However, the development of hands-free teleconferencing systems gave rise to another kind of echo known as an acoustic echo. The acoustic echo is due to the coupling between the loudspeaker and microphone. These electrical and acoustic echoes are discussed in greater detail in.

1.3 THE PROCESS OF ECHO CANCELLATION

An echo canceller is basically a device that detects and removes the echo of the signal from the far end after it has echoed on the local end's equipment. In the case of circuit switched long distance networks, echo cancellers reside in the metropolitan Central Offices that connect to the long distance network. These echo cancellers remove electrical echoes made noticeable by delay in the long distance network. An echo canceller consists of three main functional components:

- Adaptive filter
- Doubletalk detector
- Non-linear processor

1.4. ADAPTIVE FILTER

The adaptive filter is made up of an echo estimator and a subtractor. The echo estimator monitors the received path and dynamically builds a mathematical model of the line that creates the returning echo. The model of the line is convolved with the voice stream on the receive path. This yields an estimate of the echo, which is applied to the subtractor. The subtractor eliminates the linear part of the echo from the line in the send path. The echo canceller is said to converge on the echo as an estimate of the line is built through the adaptive filter.

1.5. DOUBLETALK DETECTOR

A doubletalk detector is used with an echo canceller to sense when far-end speech is corrupted by near-end speech. The role of this important function is to freeze adaptation of the model filter when near-end speech is present. This action prevents divergence of the adaptive algorithm.

1.6. NONLINEAR PROCESS

The non-linear processor evaluates the residual echo, which is nothing but the amount of echo left over after the signal has passed through the adaptive filter. The nonlinear processor removes all signals below a certain threshold and replaces them with simulated background noise which sounds like the original background noise without the echo.

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1.11 AVOIDING DIVERGENCE

The process of divergence is an adaptive filter problem that arises when a suitable solution for the line model is not found through the use of a mathematical algorithm. Under specific conditions, certain algorithms are bound to diverge and corrupt the signal or even add echo to the line. Good echo cancellers are tuned to avoid divergence situations in nearly all conditions.

1.12 HANDLING DOUBLETALK

In an active conversation, both talkers often speak at the same time or interrupt each other. Those situations are called “doubletalk”. Doubletalk presents a special processing challenge to echo cancellers. Taken step-by-step, doubletalk proceeds as follows:

1. A speaks. The echo canceller must compare the received speech from Speaker A to what would be transmitted back to A in order to approximate an echo point.
2. B speaks over the echo signal. B speaking constitutes doubletalk. The echo canceller must detect the doubletalk and cancel the echo without affecting what is heard locally, which is speaker B’s words.
3. The echo canceller must send B’s speech, as well as the echo-cancelled version of A’s own speech, back to A. Handling doubletalk so that it sounds natural is technically challenging. A good echo canceller must be able to do the following:

1.13 COMPONENTS OF AN ACOUSTIC ECHO CANCELLER (AEC)

The previous section attempted to give some valuable first hand knowledge on the functioning of a basic echo canceller. The following sections offer a detailed theoretical and mathematical account of the three fundamental components of echo cancellers. The three fundamental components that combine to form an echo canceller are

1. Adaptive Filter
2. Doubletalk Detector
3. Nonlinear Processor

1.14 ADAPTIVE FILTERING

As previously demonstrated, the best solution for reducing the echo is to use some form of adaptive algorithm. The theory behind such an algorithm and the reasons for choosing that algorithm will be described in this section. Basically filtering is a signal processing technique whose objective is to process a signal in order to manipulate the information contained in the signal. In other words, a filter is a device that maps its input signal into another output signal by extracting only the desired information contained in the input signal. An adaptive filter is necessary when either the fixed specifications are unknown or time-invariant filters cannot satisfy the specifications. Strictly speaking an adaptive filter is a nonlinear filter since its characteristics are dependent on the input signal and consequently the homogeneity and additivity conditions are not satisfied.

Additionally, adaptive filters are time varying since their parameters are continually changing in order to meet a performance requirement. In a sense, an adaptive filter is a filter that performs the approximation step on line.

2. LEAST MEAN SQUARE (LMS) ALGORITHM

The least mean square, (LMS), is a search algorithm that is widely used in various application of adaptive filtering. The main features that attracted the use of the LMS algorithm are low computational complexity, proof of convergence in stationary environments and stable behavior when implemented with finite precision arithmetic. Figure illustrates how such an algorithm works. A path that changes the signal x is called h. Transfer function of this filter is not known in the beginning. The task of the LMS algorithm is to estimate the transfer function of the filter. The result of the signal distortion is calculated by convolution and is denoted by r. In this case r is the echo and h is the transfer function of the hybrid. The near-end speech signal v is added to the echo. The adaptive algorithm tries to create a filter w. The transfer function of the filter is an estimate of the transfer function for the hybrid. This transfer function in turn is used for calculating an estimate of the echo. The echo estimate is denoted by $\hat{r}$.

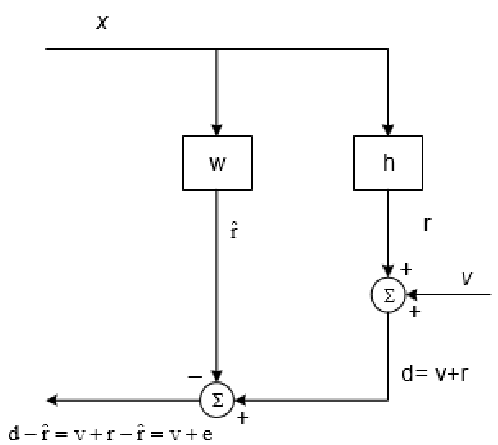

Fig 1. LMS Algorithm

3. NORMAILISED LMS ALGORITHM (NLMS)

There are a number of algorithms for adaptive filters, which are derived from the conventional LMS algorithm. The objective of the alternative LMS-based algorithms is either to reduce computational complexity or convergence time. The normalized LMS, (NLMS), algorithm utilizes a variable convergence factor that minimizes the instantaneous error. Such a convergence factor usually reduces the convergence time but increases the misadjustment.

The updating equation of the LMS algorithm can employ a variable convergence factor μk in order to improve the convergence rate. In this case, the updating formula is expressed as

$$w(k+1)=w(k)+2\mu_k e(k)x(k)=w(k)+\Delta\hat{w}(k), \qquad (1)$$

where μ_k must be chosen with the objective of achieving faster convergence, Using the variable convergence factor the updating equation for the NLMS algorithm. Usually a fixed convergence factor μn is introduced in the updating formula in order to control the misadjustment since all the derivations are based on instantaneous values of the squared errors and not on the MSE.
Also a parameter γ should be included in order to avoid large steps when $x^T(k)x(k)$ becomes small.

4. EXISTING SYSTEM

4.1 ECHOES IN TELECOMMUNICATION NETWORK

This chapter deals with echoes that are generated in telecommunication systems. As discussed in chapter one, there are two main types of echo, which are termed electrical, or hybrid, and acoustic.

4.2. HYBRID/ELECTRICAL ECHO

Hybrid echoes have been inherent within the telecommunications networks since the advent of the telephone. This echo is the result of impedance mismatches in the analog local loop. For example, this happens when mixed gauges of wires are used, or where there are unused taps and loading coils. In the Public Switched Telephone Network, (PSTN), by far the main source of electrical echo is the hybrid. This hybrid is a transformer located at a juncture that connects the two-wire local loop coming from a subscriber's premise to the four-wire trunk at the local telephone exchange. The four-wire trunks connect the local exchange to the long distance exchange.

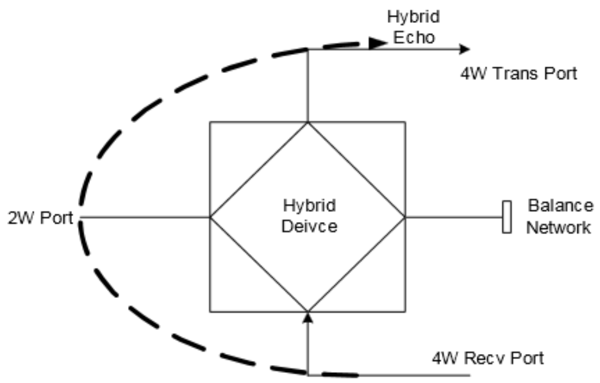

Fig 2. Hybrid Echo

The hybrid splits the two-wire local loop into two separate pairs of wires. One pair is used for the transmission path and the other for the receiver path. The hybrid passes on most of the signal.

However, the impedance mismatch between the two-wire loop and the four-wire facility causes a small part of the received signal to "leak" back onto the transmission path. The speaker hears an echo because the far-end receives the signal and sends part of it back again. Electrical echo is definitely not a problem on local calls since the relatively short distances do not produce significant delays. However, the electrical echo must be controlled on long distance calls echoes are produced.

4.3 ACOUSTIC ECHO

The acoustic echo, which is also known as a "multipath echo", is produced by poor voice coupling between the earpiece and microphone in handsets and hands-free devices. Further voice degradation is caused as voice-compressing and encoding/decoding devices process the voice paths within the handsets and in wireless networks. This results in returned echo signals with highly variable properties. When compounded with inherent digital transmission delays, call quality is greatly diminished for the wire line caller.

Acoustic coupling is due to the reflection of the loudspeaker's sound waves from walls, door, ceiling, windows and other objects back to the microphone. The result of the reflections is the creation of a multipath echo and multiple harmonics of echoes, which are transmitted back to the far-end and are heard by the talker as an echo unless eliminated. Adaptive cancellation of such acoustic echoes has become very important in hands-free communication systems such as teleconference or videoconference systems.

5. PROPOSED SYSTEM

The solution to these problems is the elimination of the echo with an echo suppression or echo cancellation algorithm. The echo suppressor offers a simple but effective method to counter the echo problem. However, the echo suppressor possesses a main disadvantage since it supports only half-duplex communication. Half-duplex communication permits only one speaker to talk at a time. This drawback led to the invention of echo cancellers. An important aspect of echo cancellers is that full-duplex communication can be maintained, which allows both speakers to talk at the same time.

5.1 THE ECHO CANCELLATION ALGORITHM

This Concept discusses the echo cancellation algorithm environment. The basic idea behind the algorithm, its terminology, modes of operation and the problems addressed by the algorithm are discussed in detail.

5.2 .BASIC ECHO CANCELLER

A basic echo canceller used to remove echo in telecommunication networks is presented in Figure, The echo canceller the transfer function of the echo path in order to a replica of the echo. Then the echo canceller subtracts the synthesized replica from the combined echo and near-end speech or disturbance signal to obtain the near-end signal. However, the transfer function is unknown in practice. Therefore, it must be identified. This problem can be solved by using an adaptive filter that gradually matches its estimated impulse response, $\hat{h}$, to that of the impulse response of the actual echo path, h.

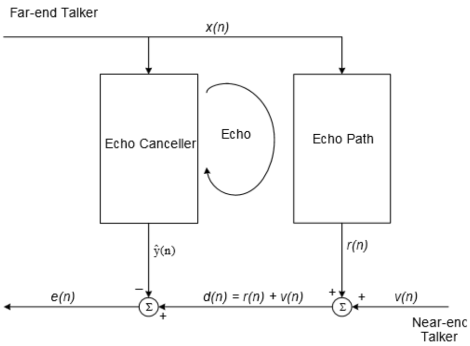

Fig 3. Basic echo canceller

This process is illustrated in Figure. The echo path is highly variable and can even depend on such things as the movement of people in the room as well as other things. These variations are accounted for by the adaptive control loop, which is built ,

The estimated echo , $\hat{y}(n)$, is generated by passing the reference input signal , x(n), through the adaptive filter, $\hat{h}$(y n), that will ideally match the transfer function of the echo path, h(n). The echo signal, r(n), is produced when x(n) passes through the echo path. The echo r(n) plus the near-end talker or disturbance signal, v(n), constitute the desired response ,

$$d(n) = r(n)+v(n), \quad (2)$$

for the adaptive canceller. The two signals x(n) and r(n) are correlated since the later is obtained by passing x(n) through the echo path. The error signal e(n) is given by

$$e(n) = d(n) - \hat{y}(n) . \quad (3)$$

In the ideal case, e(n) = v(n), which represents the case when the adaptive echo canceller is perfect.

Similar to the echo suppressors, adaptive echo cancellers also face the problem of double talking when both near and far end speakers talk simultaneously. If double talk occurs, the system may try to adjust the adaptive filter parameters to imperfectly cancel the near-end talker signal. This will result in making large corrections to the estimated echo path, $\hat{h}$, in an attempt to mimic h. In order to avoid this possibility the coefficients in the adaptive filter must not be updated as soon as double talking is detected as illustrated in Figure. The design of a good double talking detector is difficult. Even with the assumption of a fast-acting detector, there is still a possibility of changes occurring in the echo channel during the time that the echo canceller is not updated, which leads to increasing amount of un cancelled echoes. Fortunately, the duration of double talking is usually short. In addition to these problems, it sometimes occurs that a well-working echo canceller leaves some residual un cancelled echo. In such a case, a nonlinear processor is used to remove the residual echo. The goal of the nonlinear processor is to block this small unwanted signal if the signal magnitude is lower than An certain small threshold value during single talking. The nonlinear processor will only distort and not block the near-end signal during double talking. The distortion is generally unnoticeable and the process or does not have to be removed during double talking.

5.3 ECHO RETURN LOSS ENCHANCEMENT

Since you have access to both the near-end and far-end speech signals, you can compute the echo return loss enhancement (ERLE), which is a smoothed measure of the amount (in dB) that the echo has been attenuated. From the plot, observe that you achieved about a 35 dB ERLE at the end of the convergence period

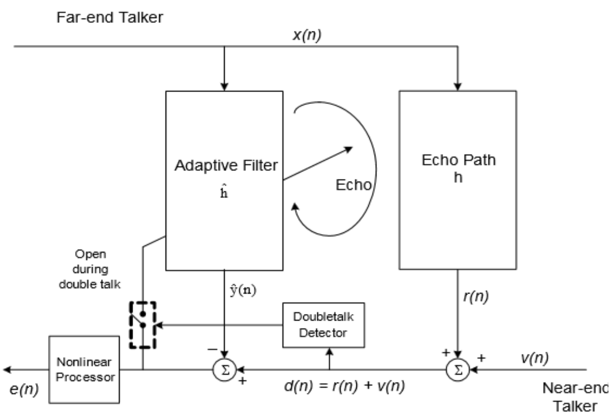

Fig 4. Echo Canceller With doubletalk Detector And Nonlinear Processor

5.4 ECHO RETURN LOSS ENHANCEMENT COMPARISION

With a larger step size, the ERLE performance is not as good due to the misadjustment introduced by the near-end speech. To deal with this performance difficulty, acoustic echo cancellers include a detection scheme to tell when near-end speech is present and lower the step size value over these periods. Without such detection schemes, the performance of the system with the larger step size is not as good as the former, as can be seen from the ERLE plots

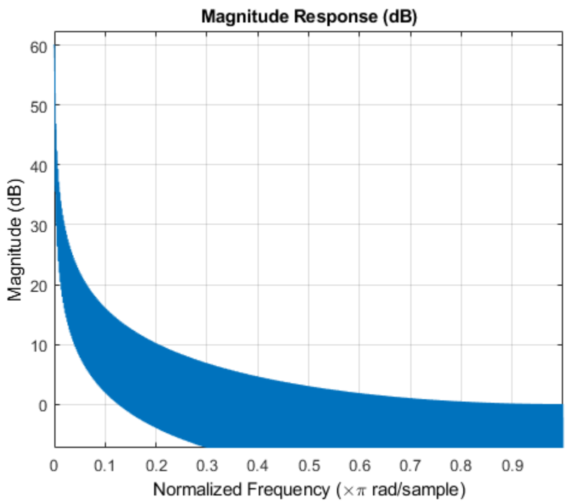

Fig 5. Magnitude response of ERLE

5.5 NEAR END SPEECH SIGNAL

The teleconferencing system's user is typically located near the system's microphone. Here is what a male speech sounds like at the microphone

5.6 FAR END SPEECH SIGNAL

In a teleconferencing system, a voice travels out the loudspeaker, bounces around in the room, and then is picked up by the system's microphone. Listen to what the speech sounds like if it is picked up at the microphone without the near-end speech present

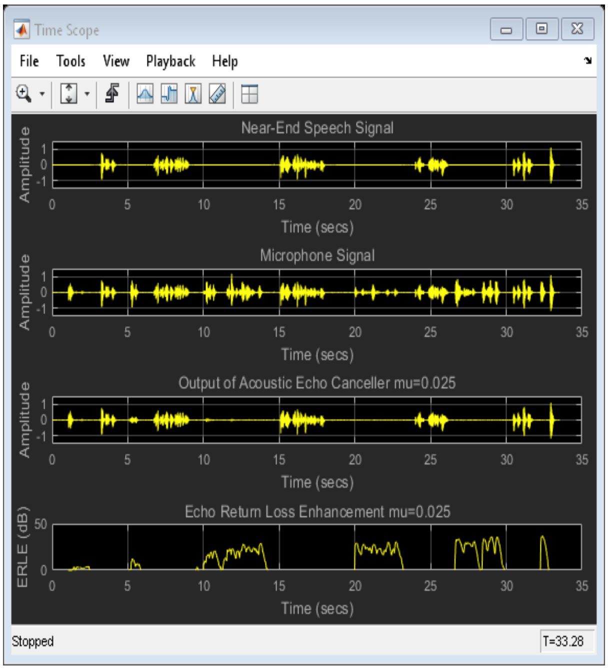

Fig 6. Echo return loss enhancement

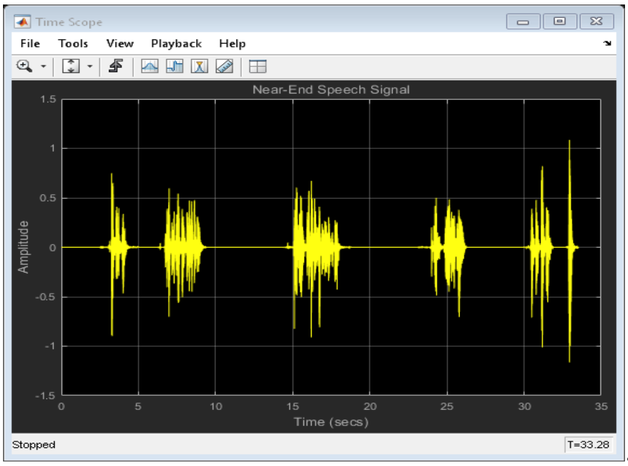

Fig 7.Plot the near end signal is v(n)

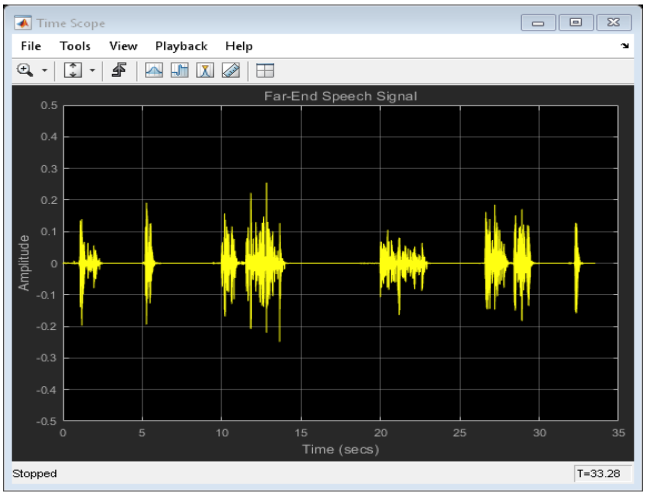

Fig 8. Plot the far end signal

6. CONCLUSION

With the world shrinking into a global village because of superior communications, telephones, both conventional and hands-free sets occupy a prominent position in solving people's communication needs. One of the major problems in a telecommunication application over a telephone system is echo. The proposed algorithm has minimized echo return loss significantly. There is a scope for framing new algorithm which can find a software solution for the problem of echoes in the telecommunications environment. The algorithm can run in any PC with MATLAB software installed.

REFERENCES

1. Andrzej Borys, "Nonlinear Aspects of Telecommunications", CRC Press, 2001[10] A.V. Oppenheim and R.W. Schafer, "Discrete Time Signal Processing", Prentice Hall, 1996
2. A.V. Oppenheim and R.W. Schafer, "Discrete Time Signal Processing", Prentice Hall, 1996
3. J. Benesty, T. Gansler, D.R. Morgan, M.M. Sondhi and S.L. Gay, "Advances in Network and Acoustic Echo Cancellation", Springer-Verlag, 2001
4. J. Benesty, D.R. Morgan and J.H. Cho, "A New Class of Doubletalk Detectors Based on Cross-correlation", IEEE Trans. Speech Audio Processing, vol. 8, pp. 168-172, March 2000
5. Jerry . Whitaker, "Master Handbook of Audio Production", McGraw-Hill, 2003
6. J.G. Proakis and D.G. Manolakis, "Digital Signal Processing - Principles, Algorithms and Applications", Prentice Hall, 1996
7. Lester S.H Ngia, "System Modeling using Basis Functions and Application to Echo Cancellation", Ph. D. Dissertation, Chalmers University of Technology
8. Mu Tian, P.K. Gupta, Marcus Harte and Danny Yip, "Improved Echo Canceller and Implementation", Dialogic Corporation
9. Paulo S.R. Diniz, "Adaptive Filtering Algorithms and Practical Implementation", Kluwer Academic Publishers, 1997
10. S.Haykin, "Adaptive Filter Theory", Prentice Hall, 1996

Author's Biography

Senthil Murugan Boopalan received his B.E from Madras University and M.E from Anna University. He is working as Assistant Professor in the Department of Electronics and Communication Engineering at Thanthai Periyar Government Institute of Technology, Vellore. He has more than 18 years of teaching experience. He is currently pursuing his research in the field of Signal Processing and communication.