Echo control in telecommunications

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This paper reviews echo control techniques for telecommunications, emphasizing the principles and applications of both circuit and acoustic echo cancellers. First, echo generating mechanisms and echo problems are described for circuit and acoustic echoes. Circuit echo is caused by impedance mismatching in a hybrid coil. Acoustic echo is caused by acoustic coupling between loudspeakers and microphones in a room. The echo problem is severe when the round-trip propagation delay is long. In this case, the echo must be removed. Next, the basic principle of the echo canceller, adaptive filter structure and adaptive algorithm are discussed. Emphasis is focused on the construction and operation of an adaptive transversal filter using the NLMS (Normalized Least Mean Square) algorithm, which is the most popular for the echo canceller. Then, applications of circuit and acoustic echo cancellers are described. Circuit echo cancellers have been well studied and implemented in LSIs for many applications. Although acoustic echo cancellers have been introduced into audio teleconference systems, they still have some problems which must be solved. Therefore, they are now being studied intensely. Finally, this paper mentions the problems of echo cancellers and the direction of future work on them. The main targets for acoustic echo cancellers are improving the convergence speed, reducing the amount of hardware and bettering the double-talk control technique.

Keywords: Telecommunication, Echo, Adaptive filter, Digital signal processing, Echo canceller

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1. INTRODUCTION

In telecommunication systems, there is a phenomenon where a sent signal (talker's voice) returns to its source after a time delay and is recognized by the talker as his signal. This phenomenon is referred to as 'echo.' Echo occurs in calls with a long propagation delay such as international telephone calls, or in hands-free telecommunications. Echo can disturb a natural conversation. When echoes occur at both ends of a communication line, a closed loop is formed. If the loop gain exceeds unity, howling occurs and the conversation can no longer be maintained.

Echo suppressors have been used to control echoes. Processing with an echo suppressor is so simple that the suppressor can be implemented by analog techniques at low cost. However, all echo suppressors cause chopping of the speech signal; hence, speech quality deteriorates.

Echo cancellers are another technique for controlling echoes. This technique was first proposed by Enomoto[1] in 1957 and its adaptive algorithm was theoretically studied in detail by Sondhi[2] in 1966. Although adaptive echo cancellers allow natural conversation, they require far more elaborate and complicated processing than echo suppressors. Therefore, the analog techniques in those days could not achieve the required ERLE (Echo Return Loss Enhancement).

In the 1970's, compact and economically feasible echo canceller hardware became possible with advances in LSI and digital signal processing techniques. These techniques are good at handling complex and

highly accurate calculations. The first echo canceller using LSIs was developed by Duttweiller[3] at AT & T Bell Laboratories. This canceller had a 128-tap transversal filter (corresponding to a 16-ms process window). Later, single-chip LSI echo cancellers were developed by AT & T, KDD, and others. These cancellers had about a 480-tap transversal filter (corresponding to about a 60-ms process window). Nowadays, echo cancellers are recommended in CCITT G.165[4] and widely used in both satellite and domestic long distance lines.

Echo canceller techniques have also been introduced into new services such as automatic call-transfer service, multiple-site telephone conference and audio teleconference systems, etc. Furthermore, echo cancellers have been studied for two-wire full duplex data modems[5] and two-way digital transmission in ISDN,[6] etc. The usage of echo cancellers is very widespread and will further expand in the future.

Taking all these points into account, this paper reviews echo control techniques for telecommunications, emphasizing the principles and applications of both circuit and acoustic echo cancellers.

2. ECHO PROBLEM

2.1 Circuit Echo

A telephone line configuration is shown in Fig. 1. 2-wire lines are used between each subscriber and their local exchange because of the cost and switching advantages. In a 2-wire local circuit, the send and receive signals are mixed. On the other hand, a 4-wire line is used between the exchanges to compensate for transmission loss and to utilize the multiple lines effectively. In the 4-wire long-distance circuit, the send and receive signals are transmitted
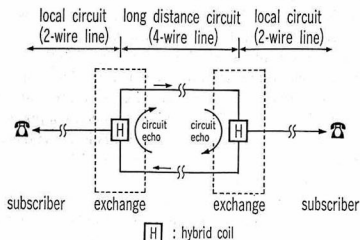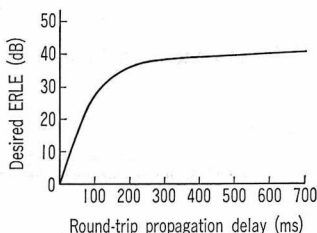
Fig. 1 Configuration of a telephone line.

Fig. 2 Desired ERLE for circuit echo.

via separate 2-wire lines. The 2-wire/4-wire conversion device is called a hybrid coil.

This hybrid coil is designed so that the impedance of the 2-wire line matches that of the balancing network. If the hybrid coil works perfectly, the received signal of the 4-wire line does not leak into the send line. However, differences in the length and diameter of the 2-wire line result in improper impedance-matching. Consequently, some of the received signals leak into the send line. This is how circuit echo arises.

Experimental results[7] regarding the desired ERLE (Echo Return Loss Enhancement) versus round-trip propagation delay are shown in Fig. 2. The longer the round-trip propagation delay, the larger the desired ERLE value. This means that as the delay increases, the telephone user begins to hear a disturbing hollow effect due to echo.

The round-trip propagation delay on the international lines between Japan and the USA is 500 ms via satellite and 300 ms via submarine cable. Therefore, circuit echo cancellers are used in these types of international lines. Circuit echo cancellers are also used in domestic lines in Japan in areas where round-trip propagation delay is over 60 ms.

2.2 Acoustic Echo

A hands-free telecommunication configuration with loudspeakers and microphones is shown in Fig. 3. The voice of the talker is broadcast by the loudspeaker in the opposite room and received by the microphone in that room. Then, it is rebroadcast by the loudspeaker in the talker's room. This is how the acoustic echo arises.

Experimental results[8] regarding the desired ERLE

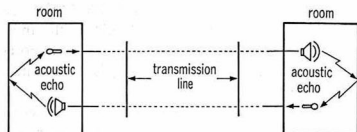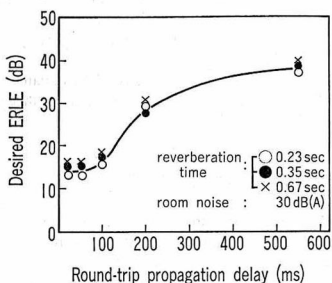versus round-trip propagation delay are shown in Fig. 4. The difference between these results and those for circuit echoes (Fig. 2) indicates that the acoustic echo must be removed even for short round-trip propagation delays of less than 10 ms.

3. CONFIGURATION AND OPERATION OF ECHO CONTROL SYSTEMS

3.1 Echo Suppressor

Echo suppressors control the echo by inserting a loss into the send path when one listens, or into the receive path when one talks. Echo suppressors are simple enough to make economically. However, they do have basic design limitations, e.g., they introduce speech distortions such as speech chopping. In addition, they do not allow simultaneous conversation.

3.2 Echo Canceller
3.2.1 Basic principle

The basic principle of the echo canceller is shown in Fig. 5. The echo canceller estimates the impulse response $h(j)$ of an echo path. An echo replica $\hat{y}(j)$ is computed by convoluting the estimated impulse response $\hat{h}(j)$ with the received input $x(j)$. The echo replica $\hat{y}(j)$ is then subtracted from the real echo $y(j)$ to produce the residual echo $e(j)$. The near-end talker signal $s(j)$ is transmitted without being affected by the echo canceller.

Since the impulse response $h(j)$ varies with time, an adaptive filter $\hat{h}(j)$ is used to estimate $h(j)$. An adaptive algorithm should provide real time opera-
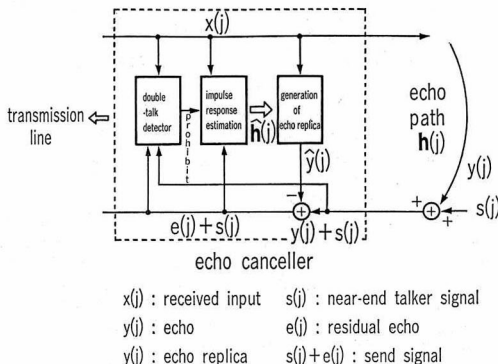

Fig. 3 Configuration of a hands-free tele-communication design.

Fig. 4 Desired ERLE for acoustic echo.

x(j) : received input	s(j) : near-end talker signal
y(j) : echo	e(j) : residual echo
ŷ(j) : echo replica	s(j)+e(j) : send signal

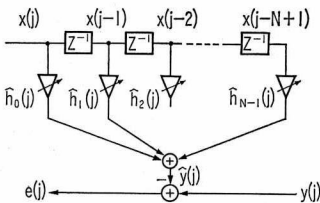**Fig. 5** Basic principle of an echo canceller.
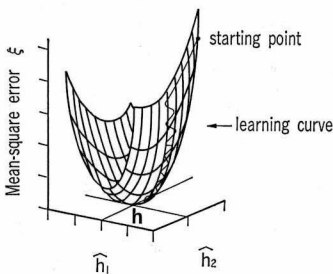
Fig. 6 Adaptive transversal filter.

Fig. 7 Two-dimensional quadratic performance surface.

tion, fast convergence speed and high ERLE. This adaptation is performed only when echo $y(j)$ is far greater than the near-end talker signal $s(j)$.

3.2.2 Adaptive filter structure

Various adaptive filter structures[9] have been proposed. We will describe an adaptive transversal filter structure[10] because it is capable of real time operation, guarantees stability and can use an adaptive algorithm that is well known for tracking the varying characteristics of the echo path. The configuration of the adaptive transversal filter, which corresponds to the impulse response estimation part in Fig. 5, is shown in Fig. 6.

The filter coefficients of $\hat{h}(j)$ ($=\hat{h}_0(j)$, $\hat{h}_1(j)$, ..., $\hat{h}_{n-1}(j)$) are adjusted to minimize the mean square of the residual echo $e(j)$ ($=y(j)-\hat{y}(j)$) in every sampling interval. The filter coefficients of $\hat{h}(j)$ correspond to the discrete impulse response of the echo path $h(j)$. Therefore, the size of the adaptive

transversal filter should accommodate the impulse response of the echo path.

3.2.3 Adaptive algorithm for the echo canceller

Many kinds of adaptive algorithms[11] have been proposed for echo cancellers. The NLMS (Normalized Least Mean Square) algorithm[12] is the most popular. It can operate in real time since it imposes a small computational load. It also guarantees stability.

Considering the expectation of the square error $\xi(=E[e(j)^2])$ as the measure of estimation performance, ξ becomes the second order function of $\hat{h}_i$. When tap coefficient vector $\hat{h}$ has 2 components $\hat{h}_1$ and $\hat{h}_2$, the ξ surface is illustrated as shown in Fig. 7. The mean-square error ξ formulates a paraboloid. The bottom (minimum point) of ξ corresponds to the optimum solution h.

The method of steepest descent[11] estimates the gradient $\hat{V}(j)$ of ξ and updates filter coefficients according to Eq. (1).

$$\hat{h}(j+1)=\hat{h}(j)+\mu(-\hat{V}(j)) \qquad (1)$$

where j denotes discrete time index and μ is a step gain.

However, the method of steepest descent results in a heavy computational load and requires a large memory to calculate the expectation. When the instantaneous square error $e(j)^2$ is used instead of its expectation ξ to estimate the gradient $\hat{V}(j)$, then, Eq. (2) is obtained.

$$\hat{V}(j)=-2e(j)x(j) \qquad (2)$$

Substituting Eq. (2) into Eq. (1) gives Eq. (3).

$$\hat{h}(j+1)=\hat{h}(j)+2\mu e(j)x(j) \qquad (3)$$

Equation (3) is called the LMS (Least Mean Square) algorithm.[13,14] Normalizing the adjustment vector by the norm of the input vector $x(j)$ and replacing 2μ with α, the NLMS (Normalized LMS) algorithm[12] is obtained as follows.

$$\hat{h}(j+1)=\hat{h}(j)+\alpha\frac{e(j)}{||x(j)||^2}x(j) \qquad (4)$$

The major drawback of the NLMS algorithm is its dependence on correlation of the received input: the convergence slows for a highly correlated signal such as speech. Various proposals, such as a whitening algorithm, have been studied[15] to remedy this drawback.

When near-end talker signal $s(j)$ is added to echo signal $y(j)$, the $s(j)$ is misinterpreted as an echo and

the cancellation performance is degraded. There-
fore, a double-talk detector, as shown in Fig. 5,
must be added to prohibit the adaptation during
double-talking. Although the adaptation is pro-
hibited, the impulse response is stored and the echo
replica is continuously created to cancel the echo.
The near-end talker signal is selectively transmitted
without being affected by the echo canceller.

4. APPLICATIONS OF CIRCUIT ECHO CANCELLERS

Since the circuit echo path is an electrical path, its
impulse response is shorter than 20 ms. The echo
canceller should also cover the round-trip propaga-
tion delay of 20–40 ms to and from the hybrid coil.
In other words, it should be capable of handling a
40–60 ms echo. Therefore, a 320–480 tap transversal
filter is required for 8-kHz sampling. This specifi-
cation is easily achieved with a single chip LSI using
today's LSI technology. The surrounding conditions
are very good for circuit echo cancellers, i.e., a very
low noise level and an impulse response that does
not vary. Therefore, circuit echo cancellers have
been intensely studied and many applications have
been developed. Some applications are described
here.

4.1 Automatic Call-Transfer Service

An automatic call-transfer service[16] transfers a
received call to another predetermined telephone.
With this service, the received call is transmitted via
a local exchange or a local telephone to any chosen
telephone. Therefore, the total transmission loss
from sender to receiver becomes quite large. To
compensate for this transmission loss, the 2-wire
circuit is transformed into a 4-wire circuit by a
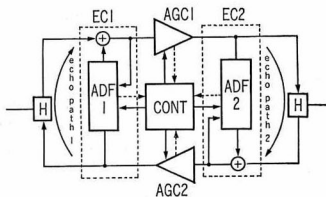
H : hybrid coil   CONT : controller
EC : echo canceller   AGC : automatic gain
ADF : adaptive digital filter   controller

Fig. 8 Configuration of a voiceband both-
way repeater.
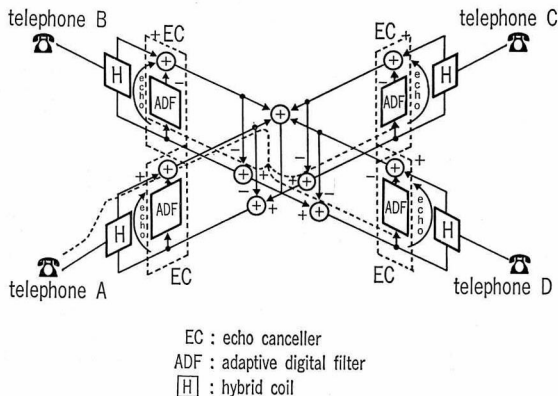
EC : echo canceller
ADF : adaptive digital filter
H : hybrid coil

Fig. 9 Configuration of a multiple-site telephone conference.

hybrid coil, the receive and send signals are separately amplified. Then, the 4-wire circuit is transformed back into a 2-wire circuit by another hybrid coil. These procedures are carried out in the voiceband bothway repeater. Echo cancellers are used to control the echo in the 4-wire loop of the repeater as shown in Fig. 8.

4.2 Multiple-Site Telephone Conference Service

The multiple-site telephone conference service allows several talkers to speak simultaneously from multiple sites by cross-connecting multiple lines. In this system, the send signals from multiple sites are added up and distributed to all the sites involved. Therefore, echoes are also added up and the loop gain of the 4-wire system may exceed unity. Echo cancellers are used in services such as that shown in Fig. 9.

4.3 Push Button Response Service

The push button response service allows a customer to respond to a voice message with the touch of a telephone button such as in automatic reservation services. Occasionally, the button is pushed before the voice message actually finishes. Therefore, an echo canceller is used at the service center to remove the echo of a voice message and to clearly receive the push button signal from the telephone.

5. APPLICATIONS OF ACOUSTIC ECHO CANCELLERS

An acoustic echo is formed by a direct sound from a loudspeaker and many sounds reflected by walls (i.e., reverberation). The impulse response of the acoustic echo path is quite long. The impulse re-

Fig. 10 Impulse response of an acoustic echo path.

sponse varies with talker movement and with the environmental conditions such as temperature and humidity. There are many noises that degrade the performance such as air conditioner noise and surrounding speech. Acoustic echo cancellers should overcome these problems.

5.1 Audio Teleconference System

An example of the impulse response of an acoustic echo path is shown in Fig. 10. Since the reverberation time of a normal conference room is 100–400 ms, a transversal filter with several thousand taps for 8-kHz sampling is required to model the room impulse response. A 7-kHz bandwidth is required for a high quality audio teleconference system. Therefore, hardware implementation of an acoustic echo canceller is far more complex than that of circuit echo cancellers.

To solve this problem, a cascade organization[17] of convolution processors is used. This parallel processing technique in the time domain reduces the computational load and the memory required for a single chip processor. Furthermore, multiple frequency subband division and down-sampling[18] is used. This parallel processing technique in the frequency domain increases the sampling interval and reduces the number of taps required for each subband. The received input in each subband is whitened; therefore, the convergence speed is enhanced. Various algorithms[19-21] for increasing the convergence speed of the canceller have been studied. Recently, characteristics of room impulse responses have been taken into consideration in the adaptive algorithm of the acoustic echo canceller.[22] This algorithm doubled the convergence speed of the canceller.

5.2 Hands-Free Mobile Telephone

The reverberation time inside a car is shorter than that in normal rooms because the volume of a car is smaller and the interior materials have a high absorption ratio. In addition, the frequency range for mobile telephones is narrower than that for audio teleconference systems. Therefore, an acoustic echo canceller for a mobile telephone requires far less hardware than one for audio teleconference systems. Many studies have been made recently.[23]

However, the surrounding conditions are very severe here for the echo canceller, i.e., motor noise is very high and the impulse response often varies
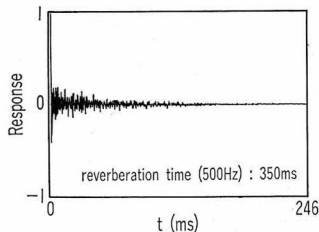
H : hybrid coil
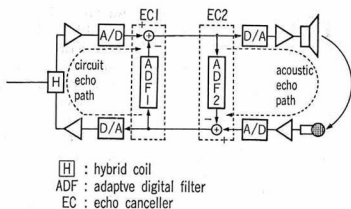ADF : adaptve digital filter
EC : echo canceller

Fig. 11 Configuration of a speakerphone.

greatly as the talker moves in the small compartment. The nonlinearity of radio communication also causes a problem in circuit echo cancellation.

5.3 Speakerphone

A speakerphone allows one to converse using a microphone and a loudspeaker without holding a handset. Speakerphones suffer from both acoustic echo and circuit echoes. Since the loudspeaker and microphone are both mounted in a single body for convenience, there is a stronger acoustic echo in speakerphones than in audio teleconference systems. Since speakerphones should be low in cost, the transversal filters of the cancellers must be short. Thus, circuit and acoustic echo cancellers are mainly used to prevent howling. A speakerphone configuration with circuit and acoustic echo cancellers is shown in Fig. 11.

6. CONCLUSION

The principles and applications of echo control techniques for telecommunications were described with the focus on echo cancellers. Circuit echo cancellers have been well studied and implemented in LSIs for many applications. On the other hand, acoustic echo cancellers require large adaptive filters, which increase the cost and hardware size, and degrade the convergence speed. Double-talk control should be further improved for the echo cancellers.

The IIR (Infinite Impulse Response) filters have been investigated for acoustic echo cancellers. However, it is still unclear whether the IIR filter structure will achieve a large improvement in the echo canceller performance. The RLS (Recursive Least Square) algorithms and lattice filters have also been studied. They will be able to operate in real time and improve the convergence speed of the adaptation in the future

as LSI technology advances.

Acoustic echo canceller techniques still face many problems. However, they are the key to hands-free telecommunications. Therefore, many efforts are being made to overcome these problems at many research and development sites. Problems in the echo cancellers will be solved in the near future.

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