Echo Cancellation and Applications

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ECHO IS THE PHENOMENON IN WHICH A DELAYED and distorted version of an original sound or signal is reflected back to the source. In this article, we review practical echo cancellation techniques—in particular, those used in telecommunications.

The theoretical basis for echo cancelers is in the field of adaptive filtering. This field has been extensively studied for the past few decades, and practical adaptive echo cancellation was well conceived by the mid-1960s [1] [2]. After that, in the 1970s, several organizations evaluated the subjective transmission quality of the echo canceler [3] [4]. However, due to the intensive processing required, widespread implementation had to wait for advances in Large-Scale Integration (LSI). The first practical Very Large-Scale Integration (VLSI) echo canceler appeared in 1980 [5], and rapid improvements in performance and functionality and reduction in cost and size ensued.

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The scope of echo cancellation has also broadened. Beginning with the original concept of canceling adverse echo in long-haul voice frequency links, echo cancelers have found wide use in full-duplex data transmission over two-wire circuits [6] [7] and are now being applied to control echo and howl in electronic conferencing applications.

This article first examines the various situations in which echoes are generated. Then, echo path modeling techniques and adaptive algorithms for coefficient control are reviewed. Current international standardization activities are discussed after that, and echo canceler implementation considerations are stated.

Basic Principle

Echoes arise in various situations in the telecommunications network and impair communication quality.

Figures 1 and 2 are examples of how an echo canceler is placed within a network.

In Figure 1, echo is generated at the hybrid transformer used for two-wire-to-four-wire conversion. An end-to-end connection of the current telephone network consists of both two-wire links, as in a subscriber loop, and four-wire links, as in a long-haul repeatered link. A hybrid transformer is used at the connection point, and due to imperfect impedance matching, echoes are generated as shown. An echo canceler placed on the four-wire side combats these echoes.

Figure 2 illustrates acoustic echo. This echo results from the reflection of sound waves and acoustic coupling between the microphone and loudspeaker. This echo disturbs natural dialog and, at its worst, causes howling. Voice switches and directional microphones have been conventional solutions to these problems, but have placed physical restrictions on the speaker.

For either type of echo, acoustic or electronic, the echo canceler first estimates the characteristics of the echo path, then generates a replica of the echo. The echo is then subtracted from the received signal. Adaptive Digital Filtering (ADF) is required to obtain a good echo replica, since the echo path is usually unknown and time-varying.

Table I shows some echo canceler applications.

Basic Filter Structures and Algorithms

The echo canceler must accurately estimate the echo path characteristic and rapidly adapt to its variation. This involves the selection of an adaptive filter and an algorithm for the adaptation. The best selection depends on the particular application and on performance requirements. In this section, various alternatives for this selection are outlined.

Filter Structures

Figure 3 lists various filter structures that are of practical importance, and Table II summarizes their characteristics. *A(Z)* and *B(Z)* are polynomials of *Z* having the coefficients $\{a_i\}$ *and* $\{b_i\}$. These coefficients are adaptively controlled for maximum echo cancellation.

The adaptive Finite Impulse Response (FIR) filter shown in Figure 3(a), also referred to as a tapped delay line, is by far the most widely used. The convergence property of the coefficients to the optimum value is well understood [8] [9]. The major drawback is that as the echo duration becomes longer, the number of taps increases proportionately and the convergence speed decreases. For telephone speech transmission, the echo duration is usually several tens of ms; thus, the required number of taps is on the order of several hundred, which is within a manageable range. For the acoustic echo cancelers used in teleconferencing systems, echoes are generated by sound waves

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Table I. Echo Canceler Applications

Echo Source	Application	Example
1. Hybrid transformer impedence mismatch	Voice communication • Long-haul transmission	• Satellite communication • Automatic call transfer • Electronic meeting with telephone
	Data communication • Voice-band full-duplex data transmission	V. 32 data modem
	• Baseband full-duplex data transmission	ISDN subscriber loop
2. Sound-wave reflection and acoustic coupling	• Speaker/microphone system	Teleconferencing system Hands-free telephone

with a much longer propagation delay, and the number of required taps is one order of magnitude larger. FIR implementation meets some difficulty in terms of both performance and hardware complexity for this case.

Research is currently going in several directions in an attempt to find new filter structures that reduce complexity and improve performance. One approach is to adapt Infinite Impulse Response (IIR) filters [10–12]. The rationale behind the IIR is that if the echo path can be better modeled by a combination of zeros and poles, then the IIR is more suitable. The key points here are to guarantee stability by confining the poles within the unit circle and to obtain an unbiased estimate of the coefficients that provide the optimum global performance.

Figures 3(b) and 3(c) show two different formulations of adaptive IIR echo cancellation. In the first configuration, called the serial-parallel structure, pole adaption is achieved, essentially in the all-zero domain, by adding a parallel adaptive FIR filter. Thus, the well known adaptation procedure for FIR can be applied. The coefficients of the parallel filter are then copied to the all-pole filter in the serial path. However, it is known that the convergent coefficient values do not necessarily coincide with the optimum. The structure in Figure 3(c) is a more direct formulation. In this structure, convergence characteristics are not affected by near-end noise, but there is a possibility of convergence to the local minimum. Convergence speed is very slow and stability testing is required. Although the potential of the IIR filter seems to be great, much more study is needed before it becomes practical.

Another promising approach is to convert signals to the frequency domain using the Discrete Fourier Transform (DFT) and carry out echo cancellation in the frequency domain [13] [14]. Convolution for a block of time-domain signals becomes simply coefficient multiplication, substantially reducing complexity. Figure 3(e) shows an example in which an echo canceler is provided for each frequency bin.

In the FIR filter, convergence is the fastest for white (uncorrelated) signals, and the rate decreases for colored (correlated) signals. This can be a serious problem for FIR filters with voice input, particularly when a large number of taps is required. In such a case, the use of a white noise training signal must be considered.

To circumvent this problem, a lattice-type pre-filter, as shown in Figure 3(d), has been proposed [9] [15] [16]. The weighted sum of signals obtained at each stage of the lattice gives the echo replica. The weights are the filter coefficients, adapted in the same way as for the FIR filter. In effect, the lattice whitens the input signal so that rapid convergence is obtained.

Another interesting type of echo canceler consists basically of memory, as shown in Figure 3(f). This structure is suitable for data transmission, especially when the echo duration is short. Echo replicas corresponding to each sequence of transmitted data are stored in memory. Therefore, it is not necessary to compute the echo replica; just read it out of memory, using the data sequence as the address. This structure has an advantage in that nonlinear effects in the echo path can also be included, since table look-up is not restricted to linear functions. But the required amount of memory grows exponentially as the data sequence size become large; hence, it is not suitable for long echo durations. A typical application is in digital subscriber loop transmission for ISDN access.

Adaptation Algorithm

There are two basic categories of algorithms for echo canceler shown in Table III.

The first algorithm is called the Least Squares (LS) algorithm. Based on the information of the past reference signals and the corresponding echoes, this algorithm determines the

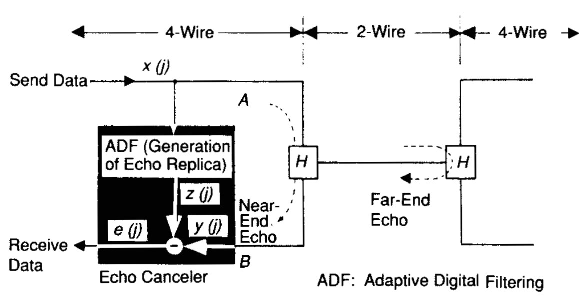

Fig. 1. Echo generation at hybrid transformers.

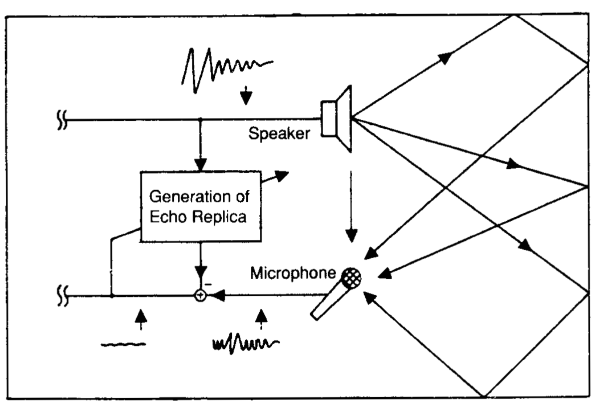

Fig. 2. Acoustic echo canceler.

coefficients that minimize the squared error summed over time [17–19]. To gradually fade out the past data and enable finite dimension arithmetic, exponentially decreasing weights are sometimes assigned to the past errors.

For simplicity, the algorithm will be described for the adaptive FIR Filter in Figure 1. The echo replica $z(j)$ can be expressed using the reference x(j) as follows:

$$Z(j) = \sum_{i=0}^{N-1} h_i(j) \cdot x(j-i) \tag{1}$$

where $[h_i(j)]$ is the estimate of the echo-path impulse response at time j. N denotes the tap length. Denoting the received echo signal as $y(j)$, the estimated error $e(j)$ becomes

$$e(j) = y(j) - z(j) \tag{2}$$

$$= \sum_{i=0}^{N-1} [C_i - h_i(j)] \cdot x(j-i)$$

where c_i is the unknown echo impulse response.

In the LS algorithm, the criterion function $D(j)$ is defined as

$$D(j) = \sum_{\ell=-\infty}^{j} e^2(\ell) \cdot w(j-\ell) \tag{3}$$

The tap coefficients are adapted to minimize $D(j)$, with weighting function $w(j-l)$ taken as an example, as follows:

$$W(\ell) = (1-\lambda)^{\ell}, \quad 0 < \lambda < 1 \tag{4}$$

This weighting function smooths out the influence of the past data. Taking the derivative of $D(j)$ with respect to $h_i(j)$, $\{j = 0,1,\ldots,N-1\}$, and setting it equal to 0, the following matrix equations are obtained:

$$H(j+1) = H(j) + \lambda R^{-1}(j) X(j) e(j) \tag{5}$$

$$R(j) = (1-\lambda) R(j-1) + \lambda X(j) X^T(j) \tag{6}$$

Here, $H(j+1)$ is the coefficient vector of the adaptive filter, $X(j)$ is the input signal vector, and $R(j)$ is the autocorrelation matrix of $X(j)$.

The advantage of this algorithm is fast convergence, irrespective of the correlation characteristics of the input signals. However, obtaining the optimum coefficient value involves computation of the inverse matrix and results in complex implementation. There are several algorithms proposed to simplify this computation, such as the Fast Kalman method [20], but we will not go into too much detail here.

The other adaptation algorithm category is the Least Mean Squares (LMS) algorithm [8]. Here, the criterion function is taken to be the expected value of the squared error, and the taps are adapted according to the stochastic steepest descent algorithm. The criterion function can be expressed as

$$D(j) = E[e^2(j)] \tag{7}$$

For practical reasons, the instantaneous value of the squared error is used in place of Equation (7) and the coefficients are controlled using its derivative with respect to the tap coefficients. It turns out that convergence can be proved for such an approach [21]. Using Equation (2),

$$\frac{dD(j)}{dh_i(j)} = -2e(j) \cdot x(j-i) \tag{8}$$

Therefore, the recursive tap coefficient adaption becomes

$$h_i(j+1) = h_i(j) + 2\mu e(j) \cdot x(j-i) \tag{9}$$

with μ being a constant.

The value of μ plays an important role in determining the convergence speed, stability, and residual error after convergence. For a larger value of μ, the convergence becomes faster, but it results in a larger residual error and is more prone to instability. Moreover, the effect of μ depends on the signal power level. To overcome this last problem, a modification to Equation (9) is quite effective, where the correction term is normalized by the input signal power. This method is called the Learning Identification (Normalized LMS) method [22], and

Table II. ADF Structure and Characteristics

ADF Structure		Characteristics
FIR		• Basic structure. • In the LMS algorithm, the numbers for operation and convergence speed are approximately proportional to the tap length.
IIR	Serial-parallel	• The same adaptation algorithm as in the FIR structure can be used. • In general, stability testing is required. • Performance is limited by near-end noise.
	Parallel	• The convergence characteristic is not affected by near-end noise. • There is a possibility of convergence to the local minimum. • Convergence speed is very slow. • Stability testing is required.
Lattice		• Convergence speed is fast because input signal is orthogonalized. • Stability testing can be easily done. • LMS algorithm is not suitable to this structure when the input signal changes rapidly.
Frequency-domain structure		• A transform operation is required. • The required number of operations is small. • The echo canceler is provided to each frequency bin.
Echo replica memorization		• Nonlinearity of the echo path can be canceled.

Table III. Adaptation Algorithms and Characteristics

Algorithm	Characteristics
Least Squares (LS) method	• Calculation of tap coefficients, which minimize the squared error summed over time • Fast convergence speed independent of input signal • Large computation number
Least Mean Squares (LMS) method	• Gradient tap adjustment to reduce the estimate of error • Convergence speed dependent on input signal • Small computation number

the iterative recursive equation becomes

$$h_i(j+1) = h_i(j) + \alpha\left[e(j)x(j-i)\right] / \sum_{k=0}^{N-1} x^2(j-k) \tag{10}$$

$$0 < \alpha < 2$$

This LMS algorithm is widely used due to its comparatively easy implementation and its well-established stability characteristics. Its major drawback is the dependence on correlation of the reference signal; the convergence slows for highly correlated signals such as voice. In some applications with a large number of taps, such as acoustic echo cancelers, the use of a whitened training signal or a pre-whitening filter (lattice structure, linear predictive filter [23], etc.) becomes necessary.

Standardization Activities

Since echo cancelers influence the quality of telephone connections, particularly for circuits employing satellite links, performance requirements and evaluation methods have been studied by such international standards bodies as CCITT.

CCITT Recommendation G.165 [24] outlines the basic requirements for echo cancelers used in telephone circuits. The prime objective here is to specify the maximum residual echo level, L_{res}. L_{res} is a function of the input signal level, as shown in Figure 4, and includes the echo suppression effect on the hybrid as well as the echo cancelers. G.165 also specifies that the convergence time must be less than 500 ms. The convergence here is defined to be the state in which L_{res} is 27 dB below the reference signal level. G.165 includes other items such as initial setup time and degradation in a double-talk situation, i.e., the situation in which both parties talk at the same time.

CCITT has also studied acoustic echo cancelers [25]. Since their performance depends heavily on the acoustic environment, the recommendation covers the reverberation characteristics of the conference room as well as echo cancellation and other performance requirements.

Due to the complexity and diversity of echo cancellation quality specifications, work towards a more thorough definition continues.

Implementation of Echo Cancelers

Echo Cancelers for Telephone Circuits

As described earlier, FIR implementation is widely used for this application since the echo duration is rather short—on the order of 10–60 ms. The sampling rate being 8 kHz, the required number of taps is below 500, and the LMS algorithm can be conveniently used for adaptation.

The required computation for each tap is one multiplication and one addition for the convolution, and one multiplication and one addition for the recursive coefficient update of Equation (9). The multiplication by μ can be regarded as common overhead. Since multiplications take up most of the hardware, the additions can be ignored for a rough estimation of the total size. Assuming a tap length of 512, the required multiplications rate becomes

$$512 \times 2 \times 8 \text{ kHz} = 8.2 \times 10^6 \text{ multiplications/s.}$$

The required memory amounts to 1,024 words, or 512 words each for data and coefficients. The processing requirements can easily be met using a single VLSI chip with a multiplying time of 100 ns.

To realize a practical echo canceler, many peripheral functions need to be implemented. Double-talk control is necessary to prevent erroneous adaptation when both parties on the line talk at the same time, since the other party's voice can be mis-

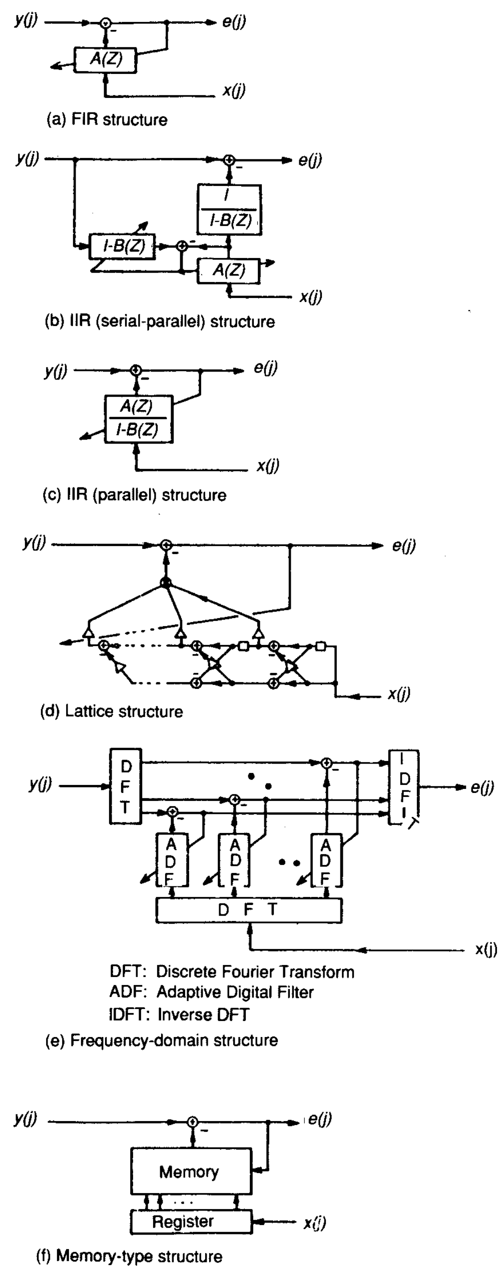

Fig. 3. Adaptive digital filter structures.

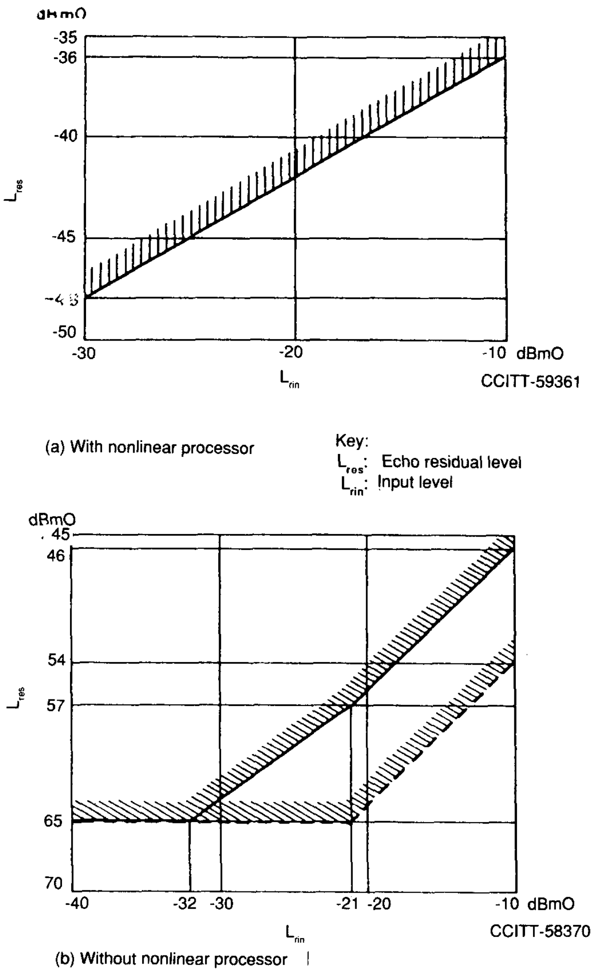

Fig. 4. Required echo residual levels vs. input levels.

interpreted as an echo. A center clipper is used to remove the low-level echo signal caused by line noise, codec quantization error, etc., which cannot be canceled by the echo canceler. This function is disabled when the send signal exists. A tone disabler must also be provided to facilitate full-duplex data transmission over voice channels. This disconnects the canceler by a tone indication signal sent prior to data transmission. These miscellaneous functions can be integrated on the same chip with the main canceler [26] [27], or can be separately realized by a general-purpose signal processor [28]. A hardware configuration example for the latter case is shown in Figure 5. Echo cancelers for three channels are commonly controlled by a single DSP.

Echo Cancelers for Full-Duplex Data Transmission over Voice Channels

Full-duplex data modems, such as the one specified in CCITT Recommendation V.32 [29], require echo cancelers to be placed at the line interface where hybrids connect the modems to two-wire loops. A considerably higher level of echo cancellation is required as compared to telephone applications. Since the received signal can be about 40 dB below the transmitted signal level, the echoes must be suppressed to about 70 dB to obtain the necessary signal-to-noise ratio of 30 dB. Therefore, high-precision arithmetic (usually more than 24 bits) is needed.

Far-end echo due to the far-end hybrid must also be taken into account. In this case, the echo delay depends on the entire length of the connection, which in some cases includes a satellite link. Therefore, two echo cancelers, one for the near-end echo and another for the far-end echo, are required. Moreover, careful design is needed to prevent undesirable interference with automatic equalizers. Figure 6 shows an example of a modem configuration employing full-duplex transmission.

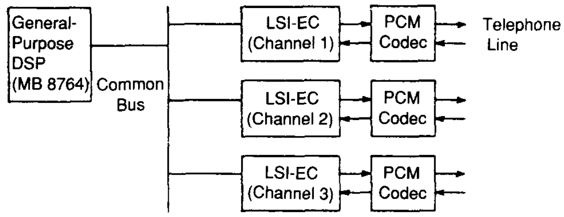

Fig. 5. Echo canceler for three-channel processing.

The echo cancelers for this application are normally implemented with high-performance digital signal processors together with the various data modulation-demodulation functions.

Acoustic Echo Cancelers

For this application, due to the long echo duration as described earlier, a straightforward FIR implementation requires more than 4,000 taps and results in a prohibitively complex hardware.

One approach is to split the signal into several frequency bands and provide a separate echo canceler for each band (see Figure 7). Since the sampling rate can be reduced, both the number of taps and the coefficient update frequency can be reduced proportionately. Hence, the total computational requirement decreases in proportion to the number of split bands. Moreover, the tap adjustment can be done in each band independently and consequently, the convergence speed is enhanced. The effect of residual echo at the band gap, however, prohibits more than four or eight bands [30]. An example of a two-band implementation is shown in Figure 8. This system covers an echo duration of 250 ms, with a speech bandwidth of 7 kHz. The total system requires 22 digital signal processor chips [31] [32].

Other forms of filters, such as IIR and frequency domain filters, are now being investigated.

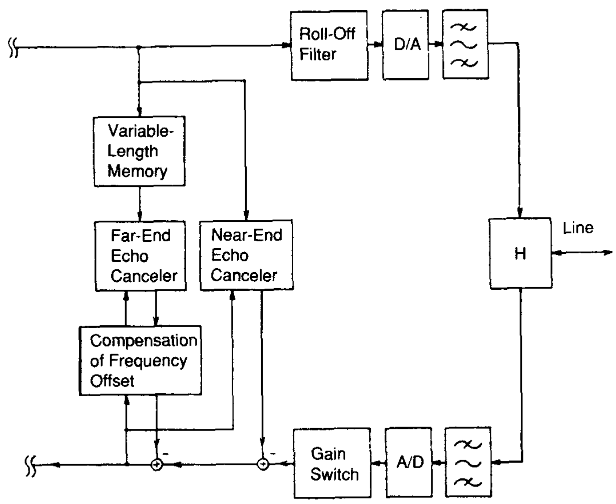

Fig. 6. Echo canceler for full-duplex data transmission.

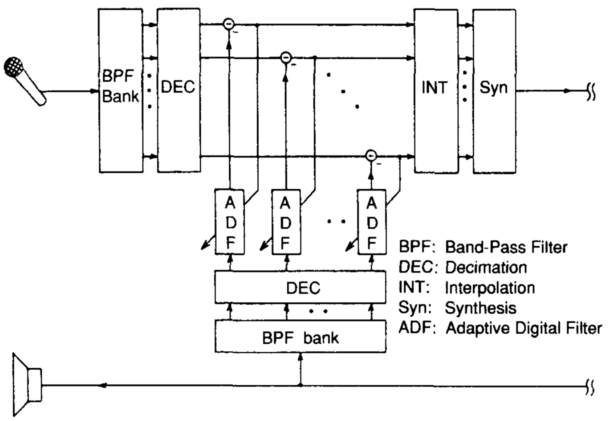

Fig. 7. Sub-band acoustic echo canceler.

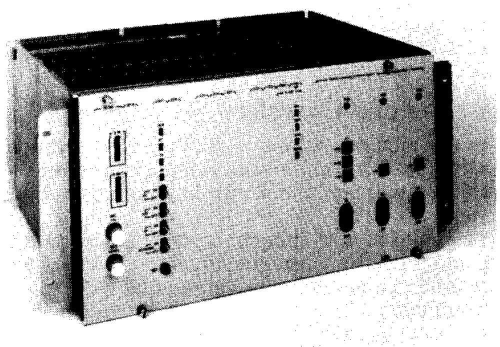

Fig. 8. Acoustic echo canceler using two-band structure.

Echo Cancelers for ISDN Digital Loop Transmission

ISDN access requires 144-kb/s full-duplex data transmission over two-wire subscriber loops. The ANSI T1 committee has adopted a four-level 2B1Q code with a baud rate of 80 kHz for standard baseband transmission, and echo cancelers that operate with this scheme are required.

This application is characterized by its much higher data rate as compared to voiceband applications. However, since the length of the subscriber loop is usually less than 7 km, the echo duration is limited to the order of 100 μs, or equivalent to about an eight-symbol length.

Therefore, either FIR or memory-type echo cancelers are being adopted. The echo cancellation requirement is similar to voiceband data applications and is about 70 dB. LSI implementations using two to three chips are now emerging [33] [34].

Conclusion

This article presented an overview of the principle, structure, and applications of echo cancelers. Echo cancelers for telephone circuits, data transmissions, and ISDN digital loop transmission are now in the state of cost effective realization by using present VLSI technology.

For the acoustic echo canceler, however, much more study is required to make the tracking speed fast and reduce computational complexity.

An echo canceler for a hands-free telephone in the mobile environment is expected to become important hereafter. Techniques of ADF in the echo canceler are also expected to be extended to noise canceling, TV ghost canceling, and FM multipath distortion cancellation.

References

[1] M. M. Sondhi and A. J. Presti, "A Self-Adaptive Echo Canceler," *BSTJ*, vol. 45, p. 1,851, 1966.
[2] F. K. Becker and H. Rudin, "Application of Automatic Transversal Filters to the Problem of Echo Suppression," *BSTJ*, vol. 45, p. 1,874, 1966.
[3] H. G. Suyderhoud et al., "Results and Analysis of an Adaptive Impulse Echo Canceler," *COMSAT Tech. Rev.*, vol. 15, no. 2, p. 253, 1972.
[4] N. Demytko and K. S. English, "Echo Cancellation on Time Variant Circuits," *Proc. IEEE*, vol 65, pp. 444–453, Mar. 1977.
[5] D. L. Duttweiler and Y. S. Chen, "A Single-Chip VLSI Echo Canceler," *BSTJ*, vol. 59, no. 2, pp. 149–160, 1980.
[6] K. H. Mueller, "A New Digital Echo Canceler for Two-Wire Full-Duplex Data Transmission," *IEEE Trans. on Commun.*, vol. COM-24, no. 9, pp. 956–962, 1976.
[7] S. B. Weinstein, "A Passband Data-Driven Echo Canceler for Full-Duplex Transmission on Two-Wire Circuits," *IEEE Trans. on Commun.*, vol. COM-25, no. 7, pp. 654–666, 1977.
[8] B. Widrow *et al.*, "Stationary and Nonstationary Learning Characteristics of LMS Adaptive Filters," *Proc. IEEE*, vol. 64, no. 8, p. 1,151, 1976.
[9] D. G. Messerschmitt, "Echo Cancellation in Speech and Data Transmission," *IEEE J. Sel. Areas in Commun.*, vol. SAC-2, no. 2, pp. 283–297, 1984.
[10] H. Fan and W. K. Jenkins, "An Investigation of an Adaptive IIR Echo Canceler: Advantages and Problems," *IEEE Trans. on ASSP*, vol. 36, no. 12, pp. 1,819–1,834, 1988.
[11] J. J. Shynk, "Adaptive IIR Filtering," *IEEE ASSP Mag.*, vol. 6, no. 2, pp. 4–21, 1989.
[12] K. Kurosawa *et al.*, "Consideration on IIR Type Learning Identification Method," *Trans. IEICE*, vol. J-68-b, no. 11, pp. 1,229–1,232, 1985.
[13] E. R. Ferrara, "Fast Implementation of LMS Adaptive Filters," *IEEE Trans. on ASSP*, vol. ASSP-28, no. 4, pp. 474–475, 1980.
[14] M. R. Asharif *et al.*, "Acoustic Echo Canceler Based on Frequency Bin Adaptive Filtering (FBAF)," *Proc. Globecom '87*, pp. 1,940–1,944, 1987.
[15] B. Friedlander, "Lattice Filters for Adaptive Processing," *Proc. IEEE*, vol. 70, no. 8, pp. 829–867, 1982.
[16] J. Chao and S. Tujii, "A Stable and Distortion-Free Echo and Howling Canceler," *Proc. ICASSP '88*, pp. 1,620–1,623, 1988.
[17] C. W. K. Gritton and D. W. Lin, "Echo Cancellation Algorithms," *IEEE ASSP Mag.*, pp. 30–38, 1984.
[18] L. Ljung, "Analysis of Recursive Stochastic Algorithms," *IEEE Trans. on Auto. Control*, vol. AC-22, no. 4, pp. 551–575, 1977.
[19] M. M. Sondhi and D. A. Berkley, "Silencing Echoes on Telephone Network," *Proc. IEEE*, vol. 68, no. 8, 1980.
[20] L. Ljung, M. Morf, and D. Falconer, "Fast Calculation of Gain Matrices for Recursive Estimations," *Int'l. J. of Control*, vol. 27, no. 1, pp. 1–19, 1978.
[21] H. Robbins and S. Monro, "A Stochastic Approximation Method," *Ann. Math. Stat.*, vol. 22, p. 400, 1951.
[22] J. Nagumo and A Noda, "A Learning Method for System Identification," *IEEE Trans. on Auto. Control*, vol. AC-12, no. 3, p. 282, 1967.
[23] S. Yamamoto *et al.*, "An Adaptive Echo Canceler with Linear Predictor," *Trans. IEICE Japan*, vol. E62, no. 12, p. 851, 1979.
[24] CCITT Recommendation G.165, "Echo Cancelers."
[25] S. Shimada, "International Standardization Trends for Acoustic Echo Canceler," *Proc. IEICE*, vol. EA89-1, pp. 1–5, 1989.
[26] N. Furuya *et al.*, "Custom VLSI Echo Canceler," *Proc. Nat'l. Conv. IEICE Japan*, p. 600, Oct. 1984.
[27] K. Takahashi, T. Sakamoto, and I. Tokisawa, "A Cascadable Echo Canceler," *IEEE ICC '82*, pp. 5F3.1–3.5, 1982.
[28] N. Kobayashi *et al.*, "An Adaptive Transversal Filter VLSI," *Proc. ICASSP '86*, pp. 1,529–1,532.
[29] CCITT Recommendation V.32, "A Family of Two-Wire, Duplex Modems Operating at Data Signalling Rates of up to 9,600 b/s for Use on the General Switched Telephone Network and on Leased Telephone-Type Circuits."
[30] C. P. J. Tzeng, "An Analysis of a Sub-Band Echo Canceler," *Proc. Globecom '87*, pp. 1,936–1,939, 1987.
[31] Y. Sakai *et al.*, "Hardware Implementation of Echo Canceler by Using MB86232," *Proc. IEICE Symposium on DSP*, vol. A4-4, pp. 225–230, 1988.
[32] H. Yasukawa, I. Furukawa, and Y. Ishiyama, "Characteristics of Acoustic Echo Canceler Using Sub-band Sampling and Decorrelation Methods," *Elect. Lett.*, vol. 24, no. 16, pp. 1,039–1,040, Aug. 4th, 1988.
[33] O. Agazzi, D. A. Hodges, and D. G. Messerschmitt, "Large-Scale Integration of Hybrid-Method Digital Subscriber Loops," *IEEE Trans. on Commun.*, vol. COM-30, no. 9. p. 2,095, 1982.

[34] M. Fukuda *et al.*, "An Approach to LSI Implementation of a 2B1Q Coded Echo Canceler for ISDN Subscriber Loop Transmission," *Proc. Globecom '88*, 1988.

Biography

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[29] D. H. Klatt and L. C. Klatt, "Voice Quality Variations Within and Across Female and Male Speakers: Implications for Speech Analysis, Synthesis and Perception," draft of an invited paper in a special session on "The Female Voice," at the 114th Mtg. of the Acoust. Soc. Am., Nov. 1987.

[30] H. Fujisaki and M. Ljungquist, "Proposal and Evaluation of Models for the Glottal Source Waveform," *Proc. ICASSP*, pp. 1,605–1,608, 1986.

[31] K. Itoh and H. Sato, "Pitch Manipulation Method Based on Residual-Excited Linear Prediction Vocoder for Speech Synthesis," *Proc. Acoust. Soc. Jpn. Mtg. '89*, Spring 1989.

[32] K. Hakoda, K. Nagakura, T.Hirahara, and K.Kabeya, "Japanese Text-to-Speech Synthesizer," *Preprints for AVIOS Mtg.*, 1989.

[33] R. Nakatsu and N. Ishii, "Voice Response and Recognition System for Telephone Information Services," *Proc. Speech Tech '87*, pp. 168–172, 1987.

[34] S.R. Hertz, J. Kadin, and K. Karpus, "The Delta Rule Development System for Speech Synthesis from Text," *Proc. IEEE*, vol. 73, no. 11, pp. 1,589–1,601, 1985.

Biography

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Since joining the Musashino Electrical Communication Laboratory of Nippon Telegraph and Telephone Company in 1975, he has been engaged in research on Japanese speech synthesis by rule and text-to-speech conversion. He has been a Senior Researcher in the Speech Processing Department of ATR Interpreting Telephony Research Laboratories since 1986.

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