Echo Canceler with Two Echo Path Models

KAZUO OCHIAI, TAKASHI ARASEKI, AND TAKASHI OGIHARA

***Abstract*—An adaptive echo canceler with two echo path models is proposed to overcome the false adaptation problem for double-talking. The echo canceler possesses two separate echo path models (EPMs), one (background EPM) for adaptively identifying echo path transfer characteristics and the other (foreground EPM) for synthesizing an echo replica to cancel out echo. The parameter values of the foreground EPM are refreshed by those of the background EPM, according to a transfer control logic, when the logic determines that the background EPM is giving a better approximation of echo path transfer characteristics than the foreground EPM. Completely digital hardware implementation is described. Using the hardware, it is shown that virtually complete double-talking protection is actually realizable by the new method.**

I. INTRODUCTION

ECHO in telecommunication network is induced mostly by impedance mismatch at the hybrid coil connecting the two wire and the four wire systems. This echo problem has become increasingly more important with the sharp increase of large loop delay telecommunication channels, particularly with the advent of satellite communication. In a channel with large loop delay, it is clear that smooth conversation is difficult without proper echo control [1], [11], [12].

Although an echo suppressor [14], [15] has been utilized as an echo control device, it inherently includes the well known shortcoming of speech clipping. It also has a problem in that echo is little controlled or little suppressed during double-talking.

To overcome the shortcomings of an echo suppressor, the echo canceler has been investigated [1], [3]-[9]. An echo canceler has some means of measuring echo path characteristics and of synthesizing a replica of the echo using the measured characteristics. In other words, it has an echo path model (EPM). The echo replica generated is utilized for canceling out the actual echo. The measurements of echo path characteristics are carried out by adjusting parameter values within the EPM so that EPM transfer characteristic conforms to that of the echo path. Among the measurement methods, the one that adaptively measures or identifies the echo path characteristic from the input and output speech signals [3]-[9] is considered most favorable, because an actual echo path is not perfectly constant [5].

The adaptive method, however, raises a new problem. The problem derives from the fact that, for correctly identifying the characteristics, the output signal of an echo path must originate solely from its input signal. During double-talking, however, the output signal contains not only the echo of the input signal, but the near-end talker's speech signal as well. Under this condition, the EPM transfer characteristics may be greatly disturbed unless adaptation is disabled prior to initiation of double-talking [10].

A simple and most direct approach for this would be to use a double-talking detector and to enable or disable adaptation according to the detector output. As yet no detector adequate for the present purpose has been designed. For example, using the conventional method of comparing receive-in and send-in signal levels, it is well known that a considerable portion (sometimes a whole syllable) of the initial part of break-in near-end talker's speech is lost. If such a method were used for an echo canceler, adaptation would proceed for a considerable amount of time in the presence of double-talking, before the adaptation is finally disabled by the detector.

The purpose of this paper is to propose an adaptive echo canceler with a new structure, overcoming the double-talking problem. The proposal is to have two separate echo path models (EPMs), one (background EPM) for adaptively identifying echo path transfer characteristics and the other (foreground EPM) for synthesizing the echo replica to cancel out the echo. The parameter values of the foreground EPM are refreshed by those of the background EPM, only when the latter is determined, by some criteria, to give a better echo path transfer characteristics approximation than the former.

In this paper: the structure and main functions of the proposed echo canceler are described; its performance characteristics are illustrated through computer simulations: a completely digital hardware implementation is described; and experimental results are given for double-talking protection capability, convergence characteristics and a subjective evaluation.

II. SYSTEM DESCRIPTION

A basic structure of the proposed echo canceler is depicted in Fig. 1. The system operates on 8-kHz sampled data. Denoting discrete time by k, the incoming signal $z(k)$ to the echo canceler send-in terminal is the sum of the echo $y(k)$ and near-end talker's speech $n(k)$. The circuit noise considered is included in near-end talker's speech signal $n(k)$. Estimation of the echo path transfer characteristic is carried out, as a background operation, by the background EPM and ADAPTATION PROCESSOR, using the background error signal $e^{(b)}(k)$ and receive-in signal $x(k)$. Specifically, the background EPM possesses a parameter set $h_i^{(b)}(k)(i = 0, 1, \cdots, N-1)$ to approximate the echo path transfer characteristic and a means for synthesizing an echo replica $(y^{(b)}(k))$ corresponding to this set of parameter values. The background echo replica $(y^{(b)}(k))$ is subtracted from the send-in signal $z(k)$ and the

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The authors are with the Nippon Electric Company, Kawasaki, Japan.

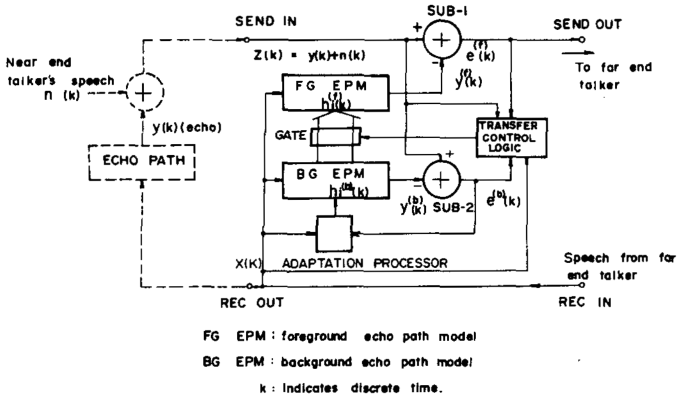

Fig. 1. Functional block diagram showing basic proposed adaptive echo canceler.

error signal $e^{(b)}(k)$ is fedback to correct the set of parameter values $h_i^{(b)}(k)$. Parameter values $h_i^{(f)}(k)$ of the foreground EPM are supplied by those of the background EPM, by means of the TRANSFER CONTROL LOGIC, when the logic determines that the background EPM is better than the foreground EPM.

In the following, a transfer control logic and adaptation algorithm, which are the two main functional elements of the echo canceler, are described.

A. Transfer Control Logic

As criterions determining transfer actuation, the following three conditions were experimentally selected to be simultaneously satisfied:

$$\text{(i)}\quad L_j(e^{(b)}) < \beta L_j(e^{(f)}) \tag{1}$$

$$\text{(ii)}\quad L_j(e^{(b)}) < \gamma L_j(z) \tag{2}$$

$$\text{(iii)}\quad L_j(z) < L_j(x) \tag{3}$$

with hangover time T for the condition

$$L_j(z) > L_j(x)$$

where $L_j(A)$ indicates a short time power of a signal $A(k)$ for j-th time interval, containing consecutive M samples, and is approximated by

$$L_j(A) = \sum_{i=0}^{M-1} |A(jM-i)| \qquad (j = 1, 2, \cdots). \tag{4}$$

β and γ are constant positive values less than $1 (\beta < 1, \gamma < 1)$, respectively. Another condition to further restrict the transfer was also provided. Conditions (i) ~ (iii) were required to be simultaneously satisfied over D consecutive time intervals, for the transfer actuation.

Of the three conditions, (i) indicates that the error signal level from the background EPM is smaller than that from the foreground EPM, by more than $-20 \log \beta$ dB. Condition (ii) indicates that the degree of send-in signal (identical with 'echo' for nondouble-talking condition) cancelation by the background EPM is larger than $-20 \log \gamma$ dB. Condition (iii) means that the transfer is prohibited for the clear double-talking condition, easily detectable by the conventional signal level comparison method.

It appears that condition (i) alone might work well for the present purpose. This is because it is generally considered that the smaller the error power is, the better the model characteristics are. Actually, condition (i) was found to work well, as long as the break-in near end talker's speech signal ($n(k)$) is assumed to be random noise. Consequently, condition (i) is effective for a break-in of such noise-like speech sounds as voiceless consonants, for example /s/, /h/, etc. For voiced sounds, in particular for vowels, however, condition (i) alone was not satisfactory and the two additional conditions (ii) and (iii) were introduced.

Finally $\beta = 0.875$, $\gamma = 2^{-3}$, $M = 128$ (16 ms), $D = 3$ and $T = 128$ ms were selected by preliminary experiments for the values of β, γ, M, D and T. It might appear that 18 dB echo cancelation ($\gamma = 2^{-3}$) is too large for a transfer condition. However, experiments showed that smooth transfer is actually possible, as is shown later by an example through a computer simulation.

B. Adaptation Algorithm

The learning identification method of Ref. [2] was adopted as the adaptation algorithm. Using the algorithm, the echo path transfer characteristics are identified in the form of an impulse response. Thus, the echo replica is synthesized by a transversal digital filter. The coefficients of the filter or the tap gains correspond, in this case, to the aforementioned parameters of an echo path model.

Denote the i-th tap gain or i-th coefficient of the filter at time k by $h_i^{(f)}(k)$ and $h_i^{(b)}(k)$ for foreground and background EPM, respectively. Then, the adaptation algorithm is expressed exclusively in terms of $h_i^{(b)}(k)$, as described above, and is given by

$$h_i^{(b)}(k+1) = h_i^{(b)}(k) + \Delta h_i^{(b)}(k) \tag{5}$$

$$\Delta h_i^{(b)}(k) = \alpha \frac{x(k-i) \cdot e^{(b)}(k)}{\sum_{i=0}^{N-1} (x(k-i))^2} \tag{6}$$

for $i = 0, 1, 2, \cdots, N-1$ where N is the number of the digital filter taps, α is a constant and

$$e^{(b)}(k) = z(k) - y^{(b)}(k) \tag{7}$$

$$y^{(b)}(k) = \sum_{i=0}^{N-1} h_i^{(b)}(k) \cdot x(k-i). \tag{8}$$

Convergence (more correctly, nondivergence) is guaranteed for a value of $0 < \alpha < 2$ [2].

III. COMPUTER SIMULATION

The following computer simulations show examples of detailed operations of the echo canceler, for both convergence and double-talking conditions. Simulations were conducted for the echo canceler with 324 filter taps, 15 bits filter coefficient

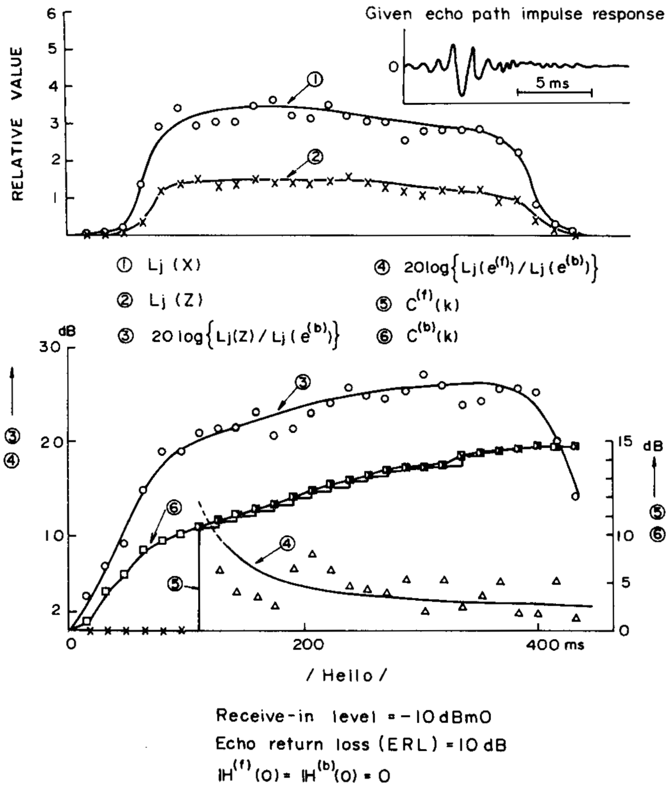

Fig. 2. A computer simulation example for convergence. The curves were visually fitted to data. Characters with sidebars are boldface in text.

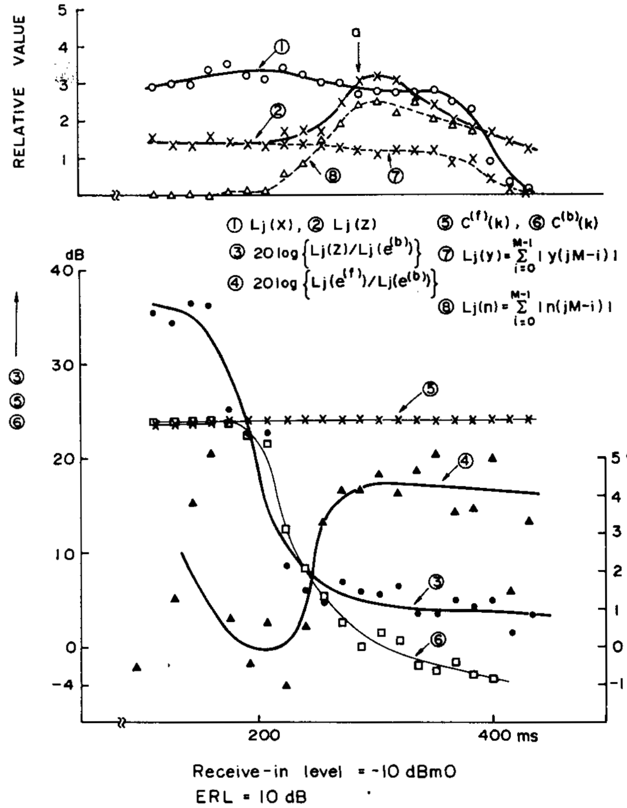

Fig. 3. A computer simulation for a double-talking condition. The curves were visually fitted to data.

accuracy and 13 bits sampled data accuracy, comforming to the one actually implemented (described later in details). For an echo path impulse response, a typical one with about 15-ms time duration (shown in Fig. 2) was utilized. To indicate the degree of convergence the following parameters $c^{(f)}(k)$ and $c^{(b)}(k)$ were used for foreground and background EPM, respectively.

$$c^{(f)}(k) = 10 \log \frac{\| \boldsymbol{H} \|^2}{\| \boldsymbol{H} - \boldsymbol{H}^{(f)}(k) \|^2} \tag{9}$$

$$c^{(b)}(k) = 10 \log \frac{\| \boldsymbol{H} \|^2}{\| \boldsymbol{H} - \boldsymbol{H}^{(b)}(k) \|^2} \tag{10}$$

Here, with $N = 324$, $\| \boldsymbol{A} \|^2 = \Sigma_{i=0}^{N-1} a_i^2$ for a vector $\boldsymbol{A} = \{a_0, a_1, \cdots, a_{N-1}\}$, $\boldsymbol{H} = \{h_0, h_1, \cdots, h_{N-1}\}$ indicates 8-kHz sampled sequence of the given impulse response, $\boldsymbol{H}^{(f)}(k) = \{h_0{}^{(f)}(k), h_1{}^{(f)}(k), \cdots, h_{N-1}{}^{(f)}(k)\}$ and $\boldsymbol{H}^{(b)}(k) = \{h_0{}^{(b)}(k), h_1{}^{(b)}(k), \cdots, h_{N-1}{}^{(b)}(k)\}$.

In relation to echo cancelation, $c^{(b)}(k)$ and $c^{(f)}(k)$ indicate average degrees of echo cancelation for the condition that various receive-in speech sounds with a flat average spectrum are used with fixed values of $\boldsymbol{H}^{(b)}(k)$ and $\boldsymbol{H}^{(f)}(k)$. This condition is roughly met during double-talking, assuming that the parameter values are properly frozen to $\boldsymbol{H}^{(b)}(k)$ and $\boldsymbol{H}^{(f)}(k)$. Therefore $c^{(b)}(k)$ and $c^{(f)}(k)$ give estimates of average degrees of echo cancelation during double-talking.

Figure 2 shows typical convergence characteristics for a receive-in speech "Hello". The convergence was initiated with the condition of $\boldsymbol{H}^{(f)}(0) = \boldsymbol{H}^{(b)}(0) = 0$. The figure indicates that the transfer, from the background to foreground EPM, of the parameter (tap gain) values is very smooth, except for a very short period immediately following initiation of convergence. As a reason for this smooth transfer, it should be noted that the degree of echo cancelation by the background EPM, $20 \log \{L_j(z)/L_j(e^{(b)})\}$, is much larger than a corresponding $c^{(b)}(k)$ value. This phenomenon was observed throughout all the computer simulations. It also appeared that the same is true for the hardware echo canceler described later. Although the reason for this is not clear yet, one explanation may be as follows. A short time frequency spectrum of a speech sound, especially of a vowel, is usually concentrated on some frequency regions and an EPM can get large cancelation, for that particular speech, by its transfer characteristics becoming good only for the frequency regions, while discussed above, $c^{(b)}(k)$ exhibits cancelation capability for signals with various frequency spectrum shapes.

Figure 3 shows an example of double-talking. The receive-in signal ($x(k)$) is the same speech "Hello" as in Fig. 2 and the break-in speech ($n(k)$), another person's "Hello," was supplied at the levels $c^{(f)}(k) = 23.6$ dB and $c^{(b)}(k) = 23.9$ dB. As is seen from curves 1 and 2, double-talking is detected, by the signal levels comparison (condition (iii)), at time instant a. Note that the background EPM parameter values (curve 6) have already been contaminated heavily before time a. The foreground EPM (5), however, is shown to retain, undisturbed, the parameter values immediately preceding the initiation of double-talking.

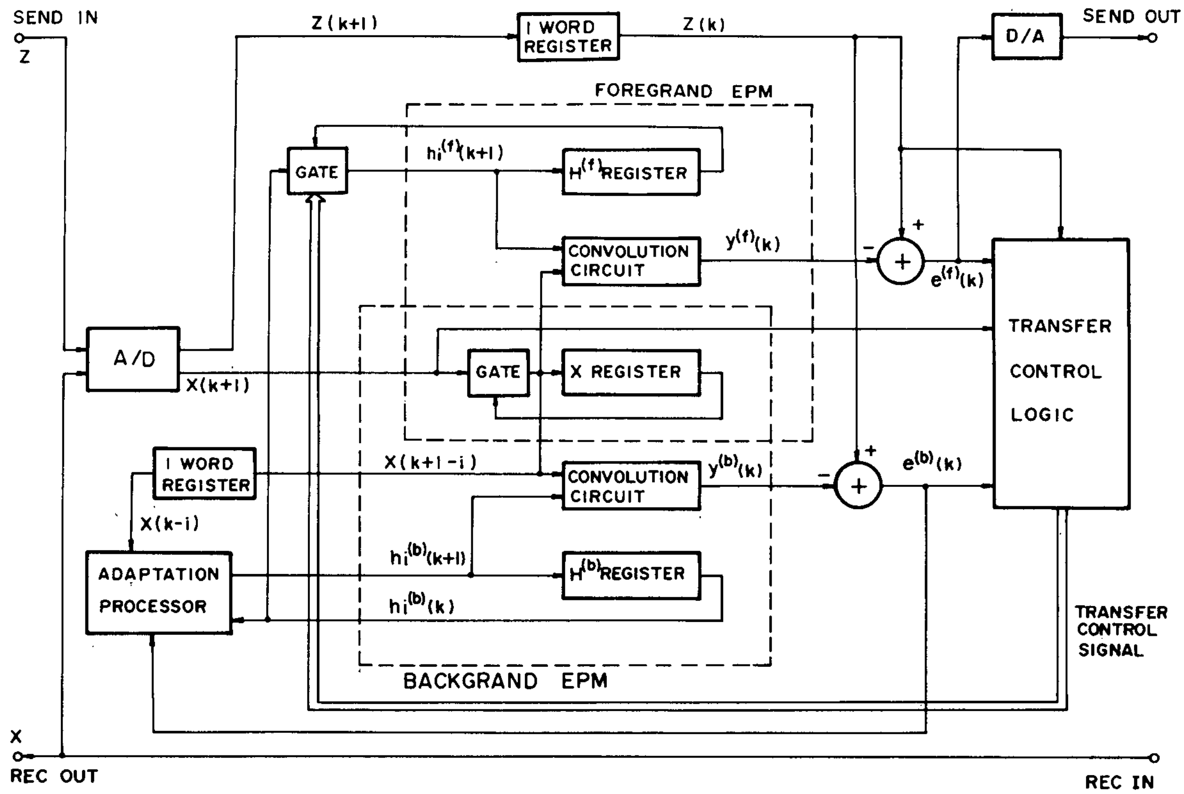

Fig. 4. Functional designed echo canceler block diagram.

IV. IMPLEMENTATION

Average return loss of an echo path is 15 dB with standard deviation of 3 dB [13] and a minimum of about 6 dB [11]. On the other hand, required over-all echo loss is about 31 dB or more [1], [11]. Consequently, 25 dB was selected as a target for the echo return loss enhancement (ERLE) attainable by the echo canceler.

A completely digital implementation was used because of its high reliability and because of the accuracy required in the internal processing, with analog to digital and digital to analog converter at the interface. 324 taps (N) were used on the digital filters. This value corresponds to about 40 ms at the 8-kHz sampling frequency used. The 40-ms value covers echo path end delay up to 25 ms, so as to meet CCITT recommendation (G.161) for an echo suppressor. Another 15 ms is used for the main part of an impulse response which appears after an elapse of time corresponding to the end delay.

A functional block diagram of the designed echo canceler is depicted in Fig. 4. The figure also indicates signal relations within the echo canceler for the time interval where error signals $e^{(f)}(k)$ and $e^{(b)}(k)$ have been generated and new echo replicas $y^{(f)}(k+1)$ and $y^{(b)}(k+1)$ are being generated. The $H^{(f)}$, $H^{(b)}$ and X registers all consist of recirculated shift registers. There are 324 register stages for each of the $H^{(f)}$ and $H^{(b)}$ registers, corresponding to the number of the digital filter taps. The number of the X register stages is 323 so that the relative position of $x(k-i)$ samples and $h_i(k+1)$ values shifts by one sample when the oldest sample $x(k-323)$ is replaced by the new value $x(k+1)$. The system clock, or the clock frequency for the shift register, is 2.592 MHz (= 8 kHz × 324). Other system parameters, such as bit lengths of X and H registers, are shown in Table 1. The ADAPTATION PROCESSOR calculates the correction ($\Delta h_i^{(b)}(k)$) according to relation (6) and generates a new coefficient value $h_i^{(b)}(k+1)$ according to (5). For the gain factor α in (6), $\alpha = 1$ was selected. For the calculation of (6), exponent arithmetic was used due to its simplicity of implementation. This means that only the information on the exponent in a floating point expression of a number is utilized for approximating the magnitude of the number, or that only the information on the position of the most significant binary bit is retained. A minimum value was set for the calculation of $\Sigma(x(k-i))^2$, so that the correction value $\Delta h_i^{(b)}(k)$ becomes smaller for weak receive-in signal. This is because weak receive-in signal induces small echo signal which is most likely contaminated by noise on an echo path.

V. EXPERIMENTS

Experiments were conducted on the hardware echo canceler implemented as above. An echo return loss enhancement (ERLE) value was measured as follows: at the time instant at which the value was measured, the foreground EPM parameter values were frozen by disabling the transfer from the background EPM. Receive-in speech ($x(k)$) was replaced by a band limited (300 ~ 3,400 Hz) white noise. The ERLE value was obtained as send-in ($z(k)$) to send-out ($e^{(f)}(k)$) signal power ratio for the condition. This corresponds to measuring the $c^{(f)}(k)$ value defined for the computer simulation. As an echo path, a bandpass filter of 300 ~ 3,400 Hz was used in combination with an attenuator giving echo return loss values.

TABLE 1
SYSTEM PARAMETERS FOR DESIGNED ECHO CANCELER

1	Sampling rate	8 kHz (125 μs)
2	Size of X and H registers	324 stages (40 ms)
3	Clock	2.592 MHz (386ns)
4	A to D and D to A converter accuracy	13 bits (linear)
5	X register accuracy	13 bits
6	H register accuracy	15 bits
7	Convolution Processor Accuracy	
	Multiply output	19 bits
	Accumulator	24 bits
8	Calculation in Adaptation Circuit	Exponent Arithmetic

A. Convergence Characteristics

As is expected from the computer simulations, convergence characteristics similar to those of a canceler with one EPM [16] are actually obtained.

Figure 5 shows ultimately obtainable or ultimate ERLE values. It also shows that the 25-dB target is almost always realized for the typical echo level of –20 ~ –30 dBmO. The ultimate ERLE value was measured as the ERLE value after a sufficiently long speech (15 words or more) was supplied as the receive-in signal ($x(k)$). Convergence was initiated with both the H registers cleared.

A 5 word 2 second long telephone speech was used to measure convergence rate, or convergence speed, and the ERLE value immediately after cessation of the speech was measured. The results showed that about 20-dB ERLE is realized in this case for the same conditions shown in Fig. 5. Tests were also conducted with end delays up to 25 ms inserted in the echo path in addition to the filter and attenuator. The results were almost identical to those described above.

B. Double-Talking Protection Capability

To evaluate performance for the double-talking protection capability, the circuit shown in Fig. 6 was prepared. This configuration enables automatic detection of double-talking, thus making it possible to count transfer actuations during double-talking, or false transfers.

Transfer of parameter values from background EPM was suspended after completion of convergence. The number of control signal occurrences actuating the transfer under double-talking conditions was counted. Break-in speech signals were added after completion of convergence and suspension of the transfer. The total double-talking length was counted by counter c_1 in 16-ms units. The counter c_2 counted the number of transfer control signals (which were issued synchronously with the 16-ms interval clock pulse) during double-talking. False transfer rate was measured by a_2/a_1, where a_1 and a_2 indicate, respectively, the final output of c_1 and c_2 after all test speeches have been supplied. The speech signals used included dictation of stories and were arranged so that double-talking occurs frequently.

Table 2 shows results of measurements. The table indicates that, for the most frequently encountered condition of equal send-in and receive-in speech level, the false transfer was zero.

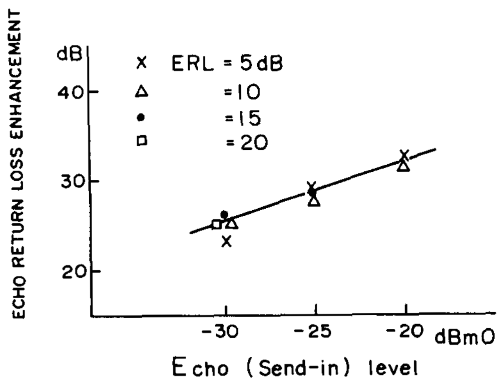

Fig. 5. Measured ultimate ERLE values for speech signals. The line was visually fitted to data.

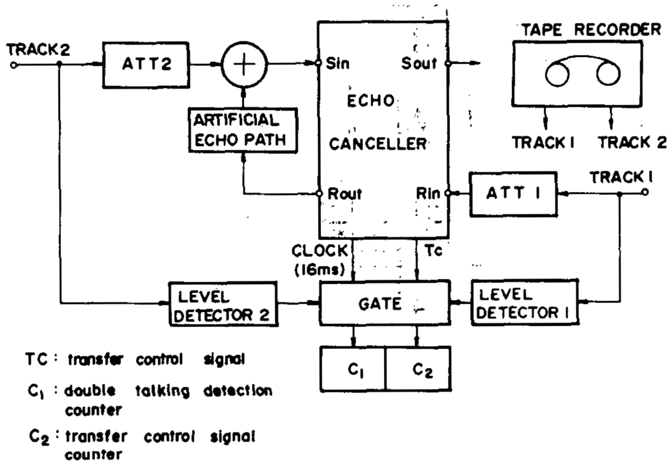

Fig. 6. Experimental setup for measuring false transfer.

TABLE 2
ERROR RATE MEASUREMENTS FOR TRANSFER CONTROL FOR SPEECH. ERL=10 dB. RECEIVE-IN SPEECH LEVEL=–10 dBm0

N-X (dB)		0	-10	-20	-30
Y-N (dB)		-10	0	10	20
FALSE TRANSFER RATE	A	0	0	2.5×10^{-3}	7×10^{-2}
	B	0	0	7×10^{-4}	2×10^{-2}

X: Receive-in speech level
N: Break-in speech level
Y: Echo level
A: Same talker's speech for both the receive-in and the break-in signals. About 40 minutes each for one male and one female speaker. Total, 80 minutes.
B: Different talker's speech for the receive-in and the break-in signal. Total, 40 minutes.

The false transfer was also zero when the break-in signal level was 10 dB smaller than that of the receive-in signal. False transfer occurs when break-in signal level is more than 20 dB lower than the receive-in signal level. However this would be no problem, since such conditions are rarely encountered.

Further, the degree of disturbance for the echo path model will be smaller for such smaller break-in signal conditions, even if a false transfer happens to be actuated.

C. Subjective Evaluation

A subjective evaluation experiment was conducted under laboratory conditions. The newly developed echo canceler with two echo path models (two EPMs E.C.) was evaluated together with an echo canceler with one echo path model (one EPM E.C.). In the latter, the direct method of enabling and disabling adaptation was used as a double-talking protection strategy. The control was actuated according to the output of a double-talking detector, whose logic is identical to condition (iii) of the transfer control logic. Adaptation algorithm, the number of the digital filter taps and accuracies which characterize the one EPM E.C. were the same as those of the two EPMs E.C..

Figure 7 shows the results and clearly indicates that the two EPMs E.C. is by far better rated than the one EPM E.C.. Each point represents the average value of 20 responses. The principal conditions of the experiment were as follows. Model 600 telephone handsets (generally used in commercial service in Japan) were used. Sixty (60) phons (A scale) Hoth room noise, which was measured at the handsets of subjects in telephone booths, was added. Two types of random circuit noise (–8 dB/oct, 0.5 mV; flat, 1 mV), were added to the two wire terminal at the two-four wire junction.

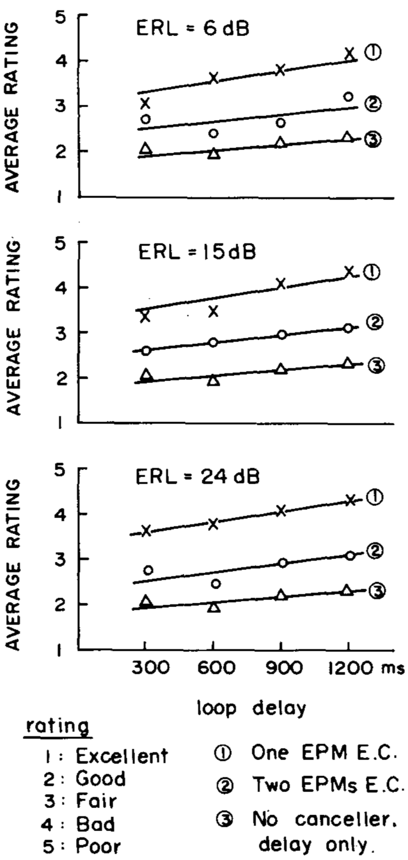

Fig. 7. Subjective evaluation results. The lines were visually fitted to data.

VI. CONCLUSION

A new adaptive echo canceling method, using two echo path models, was proposed to overcome the false adaptation problem for double-talking. Using a control logic for parameter values transfer, from the background EPM to the foreground EPM, it was shown that the method enables virtually complete double-talking protection, retaining convergence characteristics similar to that in case of using one echo path model.

Simpler and more effective transfer control logic should be sought further to simplify the echo canceler structure. Subjective evaluations in comparison with echo cancelers utilizing other double-talking protection strategies should also be conducted further to justify actual effectiveness, since the echo canceler possesses two costly convolution circuits. However, it is possible to consider total cost reduction on a multichannel base. In this case, the present structure is very naturally, as well as effectively, extensible to the multichannel system in which an adaptive echo canceling circuit, that is, the background EPM with adaptation processor, is provided for a plural number of foreground EPMs.

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Kazuo Ochiai received the B.S. degree from the University of Electrocommunications, Chofu, Tokyo, Japan, in 1963.

Since he joined Central Research Laboratories, Nippon Electric Company Limited, Kawasaki, Japan, in 1963, he has been engaged in research and development on speech signal processings, in particular on speech synthesis, speech band compression and echo control. For one year from 1968, he was a Guest Researcher on speech synthesis, at the Phonology Laboratory, Linguistics Department, University of California, Berkeley, CA.

Mr. Ochiai is a member of the Institute of Electronics and Communication Engineers of Japan and the Acoustical Society of Japan.

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Takashi Araseki was born in Hokkaido, Japan, on Oct. 14, 1946. He received the B.S. degree from Hokkaido University, Hokkaido, Japan, in 1969.

He joined, in 1969, Central Research Laboratories, Nippon Electric Company Limited, Kawasaki, Japan, where he has engaged in research and development on speech signal processing, in particular on echo control devices, speech band compression and speech synthesis.

Mr. Araseki is a member of the Institute of Electronics and Communications Engineers of Japan and the Acoustical Society of Japan.

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Takashi Ogihara graduated from the Nippon Electric Technical College, Kawasaki, Japan, in 1966.

In 1960, he joined Nippon Electric Company Limited, Kawasaki, Japan, where he has been engaged in the circuit designs for speech signal processing equipments, such as speech band compression system, echo cancelers and voice operated equipments.

Mr. Ogihara is a member of the Institute of Electronics and Communication Engineers of Japan.

400 Mbit/s Digital Repeater Circuitry Provided by a New Design Method and Beam Lead GHz Transistors

MASAO KAWASHIMA, SENIOR MEMBER, IEEE, ISAO FUDEMOTO, MEMBER, IEEE, YUKOU MOCHIDA, MEMBER, IEEE, AND TAKAO UYEHARA, MEMBER, IEEE

***Abstract*—A stable, compact, and high-performance regenerative repeater circuitry, suitable for digital transmission systems up to several hundred Mbit/s, has been provided through utilization of new devices, such as 7 GHz beam lead shielded junction transistors, and through a new computer-aided design method, and has been successfully applied to the 400 Mbit/s experimental coaxial cable PCM system.**

Major features of this repeater circuitry are:

(1) an equalizing amplifier with low noise figure (7.6 dB), small intersymbol interference (12%), and automatic line equalization of 21 dB tracking range at 200 MHz;

(2) a regenerative output circuit with bipolar pulses of 2.4 Vop amplitude and 700 ps rise time; and

(3) total performance with sufficient noise margin (10 dB for error rate 10^{-11} over a line of 56 dB loss at 200 MHz), small static pattern jitter (20° pp), smaller size (270 × 160 × 52 mm), and lower power (5.8 W).

These have been achieved by use of:

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M. Kawashima and I. Fudemoto are with Fujitsu Limited, Kawasaki, Japan.

Y. Mochida and T. Uyehara are with Fujitsu Laboratories Limited, Kawasaki, Japan.