A Self-Adaptive Echo Canceller

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Conventional echo suppressors combat echoes generated at hybrid junctions in long distance telephone connections by interrupting the return path according to some decision scheme based upon the relative levels of the outgoing and return signals. In this brief, a new device is described for *cancelling* the echo without interrupting the return path. We call this device an echo canceller to distinguish it from conventional echo suppressors. It generates a replica of the echo (which is then subtracted from the return signal) by synthesizing a linear approximation to the echo transmission path. It is self-adapting in that it automatically tracks variations in the echo path which may arise during a telephone conversation (e.g., connection or disconnection of extension phones, etc.).

A schematic of such a self-adapting echo canceller is shown in Fig. 1. It is based upon an idea originally proposed by J. L. Kelly, Jr. and B. F. Logan, and incorporates modifications which simplify and improve the implementation and performance.

With reference to Fig. 1, let $x(t)$ be the input speech signal and $y(t)$ the return signal. The return signal consists of an echo $z(t)$ (which is the result of convolving $x(t)$ with the impulse response $h(t)$ of the echo path) and a noise $n(t)$ (which may include a second speech signal). An estimate $\hat{z}(t)$ of $z(t)$ is substracted from the return signal and the error signal $e(t) = y(t) - \hat{z}(t)$ is continuously used to improve this estimate. The signal $\hat{z}(t)$ is given by a linear expansion with time varying coefficients, $g_i(t)$. Thus,

$$\hat{z}(t) = \sum_{i=1}^{N} g_i(t)x_i(t), \tag{1}$$

where

$$x_i(t) = \int_0^t w_i(\tau)x(t - \tau)\, d\tau, \tag{2}$$

and the $w_i(\tau)$ $(i = 1, 2, \cdots)$ form a complete set of orthogonal functions. The dynamic behavior of the system of Fig. 1 is then governed by the set of equations

$$\frac{d}{dt}\, g_i(t) = KF[e(t)]x_i(t) \qquad i = 1, \cdots N. \tag{3}$$

Here K is a positive constant and the function $F[e]$ is chosen to be an odd

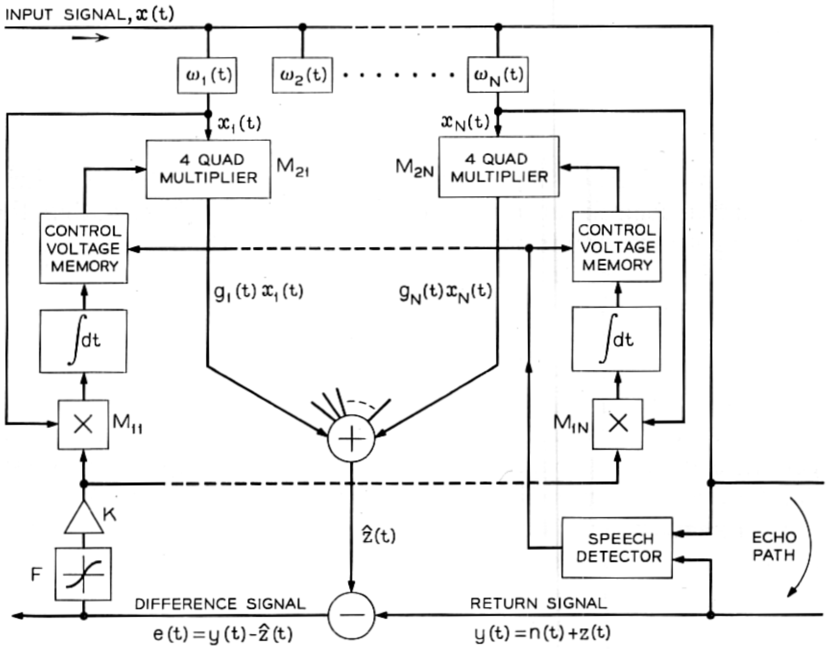

Fig. 1 — Schematic of echo canceller.

and non-decreasing function of the error e. It can then be shown that if $h(t)$ can be well approximated by the expansion

$$h(t) \cong \sum_{1}^{N} h_i w_i(t), \tag{4}$$

where the h_i are constant or slowly varying, then $g_i(t)$ converges to h_i ($i = 1, \cdots N$) in the limit as $t \to \infty$. That is, $\hat{z}(t)$ approaches $z(t)$. The quantity

$$\sum_{i=1}^{N} (g_i(t) - h_i)^2$$

converges monotonically to zero if $n(t) \equiv 0$ and if the expansion given by (4) is exact. However, convergence takes place even in the presence of relatively large amounts of noise. The amount of noise that can be tolerated decreases as the speed of convergence increases. The speed of convergence depends upon the constant K, the choice of function F, and the level and statistical properties of the signal $x(t)$. The proof of convergence, estimates of convergence rate, factors affecting choice of F, and the set $w_i(t)$ as well as results of computer simulations of such echo-cancellers are the subject of a paper under preparation.

A prototype of such a system has been implemented. In this implementation the $x_i(t)$ are obtained from taps on a delay line so that $\omega_i(t) = \delta(t - iT)$, where T is approximately 0.1 msec and N in (4) is 50. The function F has been chosen to be an infinite clipper. This allows the multipliers M_{1i} in Fig. 1 to be replaced by simple transistor switches. The integrators also pose no particular problem and simple operational amplifiers were found satisfactory. However, the multipliers M_{2i} must satisfy more stringent requirements. Their outputs $g_i(t)x_i(t)$ must be strictly proportional to the $x_i(t)$, while strict proportionality to the $g_i(t)$ is of secondary importance. A four quadrant multiplier was designed using current-controlled photo-resistors in a feedback circuit. This circuit exhibits a nonlinearity of less than 10 percent with respect to the $g_i(t)$ and less than 0.5 percent with respect to the $x_i(t)$ over a 50-dB dynamic range.

During time intervals when the input $x(t)$ is zero, there is no corrective feedback and it is important that the gain settings $g_i(t)$ be unaffected by drift in the integrators. It is also desirable (though not absolutely essential) that the feedback path be opened when the noise $n(t)$ (which, as previously noted, may include a second speech signal) is considerably larger than the echo. The box designated as speech detector in Fig. 1 achieves this dual objective. It disconnects the operational amplifiers from the integrating capacitors whenever

$$\langle \mid x(t) \mid \rangle - \langle \mid y(t) \mid \rangle < \varepsilon$$

where ε is a predetermined positive threshold and $\langle \quad \rangle$ indicates time averaging for about 0.5 sec.

Further details of the implementation will be described in a forthcoming publication. The results of computer simulations and tests on the prototype may be summarized as follows:

(*i*) The system converges in about 0.5 second for normal speech levels. This convergence time increases to about 5 seconds for a speech signal 20 dB lower.

(*ii*) When the echo canceller and the echo path were simulated on the computer, cancellations of 60 dB and higher were easily achieved. However, in the case of echoes generated on laboratory simulated telephone connections, the cancellation was about 20–25 dB on both the computer simulation and the prototype.

(*iii*) The system converges in the presence of noise which is up to 8 or

10 dB higher than the echo. After the system has converged, however, even a much larger noise does not appreciably degrade the cancellation.

The implementation must be extensively tested in a variety of telephone circuits before the merits of the proposed system can be fully evaluated.

An accompanying brief by F. K. Becker and H. R. Rudin describes a different instrumentation of an echo canceller based upon similar principles.