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# Automatic Distortion Correction for Efficient Pulse Transmission*

**Abstract:** Automatic correction of pulse distortion provides the prospect for materially extending the efficiency of data communications over telephone networks. Systems that will compensate automatically for deficiencies in the phase and amplitude characteristics of a transmission channel are shown to be technically feasible. Two specific systems are demonstrated: The first, a "time-reversal" system, compensates for distortion in the phase characteristic only, and the second, a "time-domain" system, compensates for distortion in amplitude as well as phase. The theoretical basis of this work is presented and verified by experimental results obtained on simulated lines and on common-carrier channels.

## Introduction

Delay distortion is one of the important limiting factors that prevent more efficient use of telephone networks for data communication. It is next in importance to the fundamental speed limiting factors of bandwidth and noise. This distortion extends the duration of the received data pulses several pulse periods beyond the duration of the properly shaped, band-limited transmission pulse. Since telephone networks were originally designed for human listeners, delay distortion in the networks has been kept within boundaries acceptable to speech communication, but not within boundaries that permit the most efficient data communication to be realized.

The amplitude characteristic of a given line can also adversely influence transmission efficiency and is, therefore, considered in this paper.

The most common method of controlling distortion in a private-line channel (as opposed to a switched line) is to measure the amplitude and delay characteristics of the line and to design correction networks to optimize the line for data transmission. Such a method, which is a frequency-domain approach, is time consuming but quite acceptable for a private line, where the time required for design and construction of correction networks is consistent with the particular application.

A method corresponding to this frequency-domain approach exists in time-domain equalization. Instead of directly correcting the frequency response of a given line, one can operate on the received data pulse by using a series of delay lines to superimpose portions of the delayed, received data pulse in opposition to the overhsoots of the received data pulse. In this way, the ringing effect produced by delay and amplitude distortion can almost be eliminated and a nearly optimum pulse shape achieved.

Both equalization methods described are extensively used in data communications. Instead of attempting perfect equalization of a given line, one may achieve acceptable results by using commercial equalizers. These equalizers are designed by restricting the number of variables and making these variables manually adjustable. In the frequency domain, the central portion of the frequency spectrum is delayed in order to improve the over-all delay of the pulse spectrum. The frequency, as well as the amount of delay introduced, can be varied independently.

One finds that even simplified methods of equalization require complex instrumentation. Procedures, however, are simple, are rapidly accomplished and good results can be obtained within a few minutes once the equipment has been properly set up.

Although manual methods are well suited to private line operations, they do not lend themselves very well to automated procedures required for switched-network applications. In these applications, links are established automatically and are in service for a relatively short period of time. It is for this type of application that automatic optimization is most desirable. This paper addresses itself

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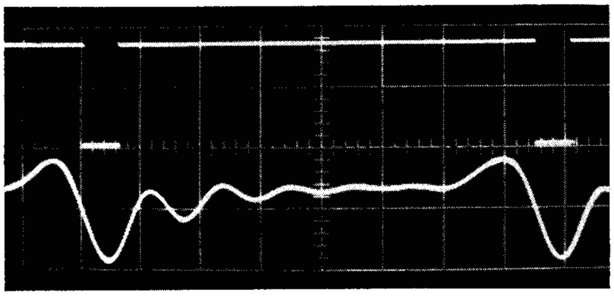


**Figure 1** Effect of differential delay on single pulse response. Upper trace, transmitted pulse; lower trace, received pulse.

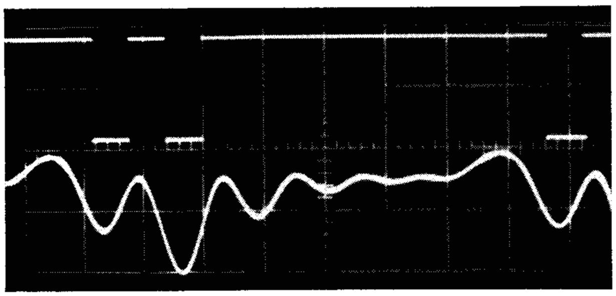


**Figure 2** Effect of differential delay distortion on a pulse pair. Upper trace, transmitted pulse; lower trace, received pulse.

to this subject of automatic distortion correction for pulse transmission. The two general methods considered are referred to as *time reversal* and *time domain* techniques.

## Differential delay-distortion correction by time reversal

The transfer function of a communication circuit can be written as $A(\omega)e^{-j\phi(\omega)}$ in which $A(\omega)$ and $\phi(\omega)$ describe the amplitude and phase characteristics, respectively. Phase can be written $\phi(\omega) = f(\omega) + n\pi$. In a distortionless system, $A(\omega)$ is constant and $\phi(\omega) = k\omega + n\pi$, in which $n$ is an integer and $k$ is a constant. The envelope delay time, $d\phi(\omega)/d\omega$, is constant in a distortionless channel and is variable when $f(\omega)$ does not vary linearly with frequency. The effect of this differential delay is shown in Fig. 1, in which a 250-$\mu$sec pulse was transmitted via an "$L$" carrier system and a vestigial sideband modem. The breadth of the received signal demonstrates the variation in transmission delay for the frequency components of the test pulse. Efficient pulse transmission is not possible over this type of circuit without compensation for differential delay distortion. Amplitude distortion, of course, also has a degrading effect upon the shape of the output pulse. However, the primary role in extending the pulse duration, insofar as data transmission is concerned, is played by delay distortion. Since we are dealing here with efficient transmission, the output waveform must also have overshoots due to band limiting but, as shown later, in a properly compensated system these overshoots can be made to occur at times which do not interfere with the transmission of data. Figure 2 shows the intersymbol interference resulting from superposition of the components of a pulse pair.

Consider the following conceptual experiment: The received, distorted signal of Fig. 1 is recorded on magnetic tape. The tape is carried to the transmission point and played into the line in reverse, such that the frequency components which encountered the greatest delay are transmitted first and the components which were delayed least are transmitted last. The result of this operation will be *an exact compensation of delay distortion.* The transmission of the reversed message is referred to here as time-reversal transmission.

The transmission frequency characteristic is

$$T_1(j\omega) = A(\omega)e^{-j\phi(\omega)}, \tag{1}$$

which gives the ratio of output to input of the transmission system. If the output signal is reversed in time after period $\tau$ ($\tau$ being greater than signal duration), a reversal of phase angle takes place and the time-reversed transfer characteristic may now be expressed as follows:

$$T_2(j\omega) = A(\omega)e^{j\phi(\omega)-j\omega\tau}. \tag{2}$$

This process is depicted by the block diagrams in Fig. 3 which, in effect, are equivalent to each other.

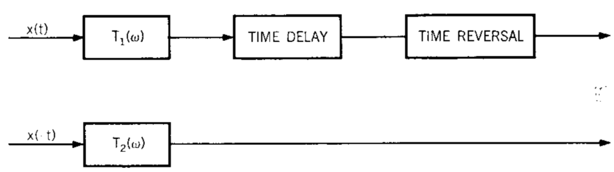
After a second pass of the signal through the transmission system, the over-all transfer characteristic becomes

$$T_1(j\omega)T_2(j\omega) = A^2(\omega)e^{-j\omega\tau}, \tag{3}$$

which corresponds to the response of a system having a fixed delay and no phase distortion, but twice the amplitude distortion (in dB) of the original system.

The fact that the received signal pulse is time reversed

**Figure 3** Process of time reversal affecting the over-all transmission frequency characteristic.



is of minor importance in a digital data system for which the correct waveform for the non-phase-distorted transmission of a single bit has been determined. In general, data signals are rectangular pulses and the time-reversed waveshape of each pulse is indistinguishable from its normal version.

Bogert[1] and Scott[2] have suggested time-reversal schemes for lines with known and accessible electrical midpoints. The interest of the authors of this paper is the switched network (dial system) in which case it is not feasible to consider bisectable lines.

Consider a system such as shown in Fig. 4 in which the following sequence of operations takes place:

(*a*) A test pulse representative of ONE is transmitted, received, sampled, quantized and stored.
(*b*) The stored, quantized samples are retransmitted at low speed to the originating station, where they are stored in a digital waveform generator.
(*c*) Transmission of coded messages at high speed may now take place using the data stored in the waveform generator (time reversed) as a ONE.

Step 1 corresponds to the recording of a received pulse and Step 2 corresponds to carrying the recording back to the original transmission point. The third step requires further discussion.

The time-reversed signal to be transmitted whenever a ONE is called for is many times the width of the test pulse. If a data sequence is to be transmitted, means must be provided to generate a composite waveform that is a linear superposition of combinations of the stored signal corresponding to the data sequence.

A digital waveform generator was designed and built with the following characteristics:

1. Resolution: 1 part in 32
2. Samples per data bit: 4
3. Length of data sequence: 12 bits

The generator consists of a group of digital-to-analog (D/A) converters, a summing amplifier, a timing clock and a shift register to control the D/A converters according to the data input sequence.

The waveform of Fig. 5 is the filtered output of the generator which is a time-reversed replica of the system's response to a single pulse such as shown in Fig. 1.

Received signals for a single and a double time-reversed pulse sent over common-carrier channels are shown in Figs. 6 and 7 respectively (compare with Figs. 1 and 2). Although overshoots still remain in the time-reversed waveforms, these waveforms are essentially symmetric about the main peak(s), which indicates that phase distortion has been corrected but not amplitude distortion.

Figures 8 and 9 show the system response to quasi-random data sequences before and after time reversal.

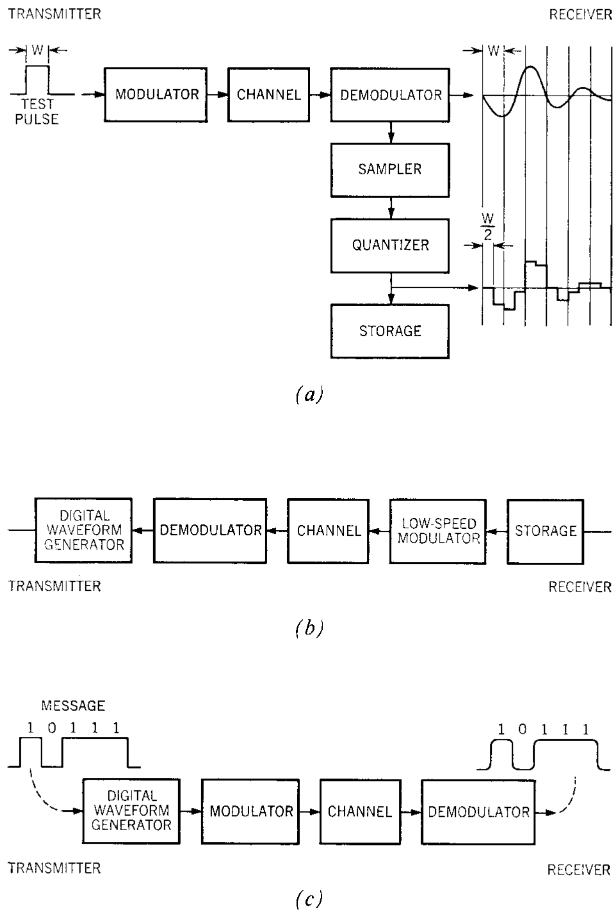



**Figure 4** Distortion correction via a feedback channel. (a) .... (b) .... (c) ....

The response is displayed by means of an "eye diagram," which is a superposition of the waveforms for all ONE's and ZERO's in the data sequence. A good "eye" opening is an indication of adequate margin for reliably differentiating ONE's and ZERO's.

It has been demonstrated in this section that it is feasible to determine automatically an appropriate pre-distorted waveform which compensates for differential delay distortion on randomly selected communication channels.

### Distortion correction in time domain

- *Fundamental considerations*

The scheme described in the foregoing section demonstrates how degrading effects of a transmission system may be compensated for, in part, by "pre-distorting" the input waveform. In order to synthesize automatically a pre-distorted waveform that will compensate for *amplitude as well as phase imperfections,* it is helpful to approach

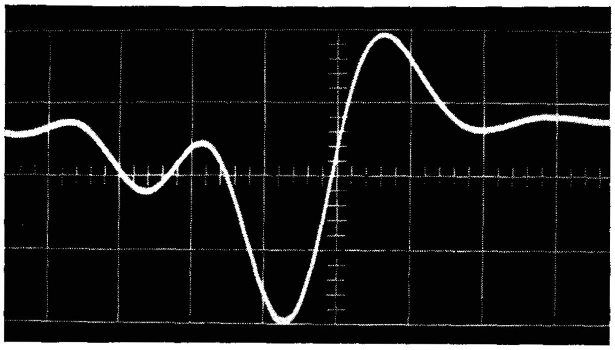


**Figure 5** Generated time-reversed waveform.

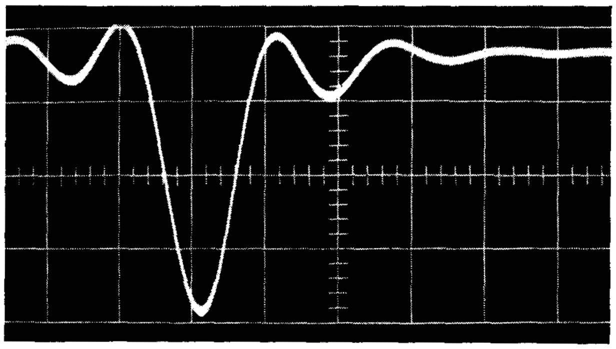


**Figure 6** System response to time-reversed single pulse.

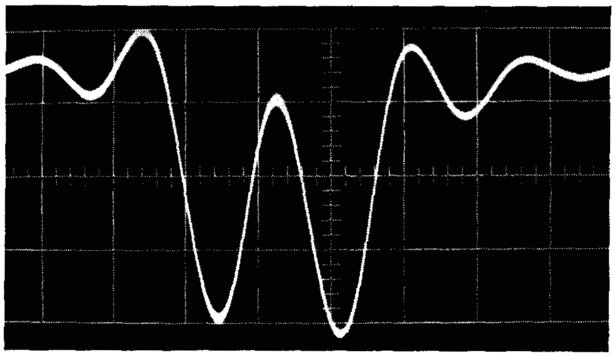


**Figure 7** System response to a time-reversed pulse pair.

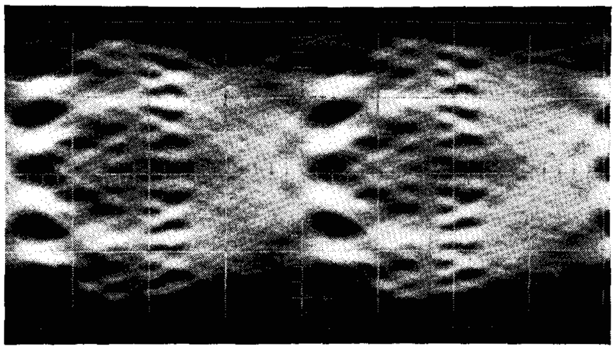


**Figure 8** Effect of differential delay on quasi-random data response.

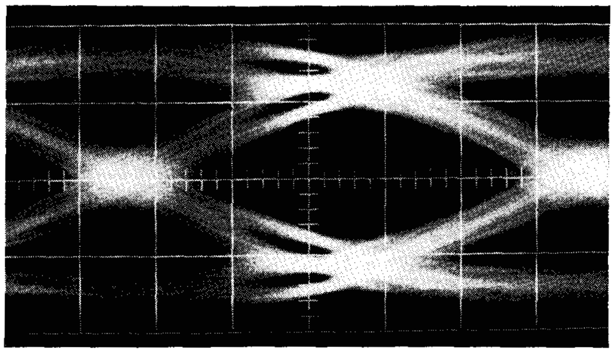


**Figure 9** System response to time-reversed quasi-random data.

the problem in the time domain. The automatic systems described in this section depend upon information directly extracted from the time-domain response. The distortion correction is performed at the transmitter in the initial description and at the receiver in the subsequent discussion.

Consider the response of a given linear transmission system to a pulse of unit amplitude and unit duration. The resulting output will be a waveform having amplitudes $\cdots a_{-1}, a_0, a_1, a_2 \cdots$ at points spaced successively in unit time periods. Let the time reference at the output be chosen arbitrarily so that one of the $a$'s, $a_0$, corresponds to the point of greatest amplitude in the output waveform. The objective of the equalization procedure is to distort the input signal in such a way that the main peak, $a_0$, becomes the midpoint of the output pulse and all other $a$'s are reduced to zero.* After this pre-distorted waveform has been determined, it may be used to represent all ONE's in messages subsequently sent through that transmission system. The result converges on an "eye diagram" which satisfies Nyquist's first criterion[3] in which the received wave, at the mid-time of each binary digit, has one of only two possible values. It can be shown that this result is equivalent in the frequency domain to the creation of an over-all amplitude characteristic that has a real part that is odd-symmetric about half the bit rate, $f_d/2$, and an imaginary part that is even-symmetric about $f_d/2$. The over-all characteristic of the transmission system includes

* The concept of distortion correction by input signal shaping was set forth by Nyquist[3] in his classic paper in 1928.

the input waveform characteristics together with the characteristics of the transmission channel.†

The particular type of input waveform which is dealt with in this discussion is one composed of rectangular pulses having unit duration and adjustable amplitudes. (This type, of course, is not the only one that can be used to satisfy the criterion for distortionless transmission.) Let the components of the input wave be labelled $\cdots C_{-1}$, $1, C_1, C_2 \cdots$, where the subscripts correspond to those used for the same relative output positions. If the output is normalized by automatic gain control, the main peak of the output wave becomes unity and the sampled values of the output wave become $\cdots d_{-1}, 1, d_1, d_2 \cdots$. It is evident that, if the output wave deviates from zero by an amount $d_k$ at one of the sampling points, $(k \neq 0)$, the deviation may be reduced to zero simply by adding a unity-duration pulse having amplitude $-d_k$ in the $k$-position of the input wave. The distortion-correction procedure becomes a simple iterative process of consistently inserting corrective pulses in the input waveform at digit positions corresponding to deviations in output. If the original input-output relationship is expressed as

$$[\cdots 0, 1, 0, 0, \cdots] \rightarrow [\cdots d_{-1}, 1, d_1, d_2, \cdots], \quad (4)$$

then the final relationship, which is the goal reached after all deviations have been reduced to zero, may be expressed as

$$[\cdots C_{-1}, 1, C_1, C_2, \cdots] \rightarrow [\cdots 0, 1, 0, 0, \cdots]. \quad (5)$$

The response after $n$ iterations may be written generally

$$[\cdots C_{-1,n}, 1, C_{1,n}, C_{2,n}, \cdots] \rightarrow [\cdots d_{-1,n}, 1, d_{1,n}, d_{2,n}, \cdots]. \quad (6)$$

This iteration process converges on the solution to a set of simultaneous equations. For example, if we can ignore the effects of all input components other than $C_{-1}$ through $C_2$, then the equations to be solved are as follows:

$$\begin{aligned} C_{-1} + d_{-1} + d_{-2}C_1 + d_{-3}C_2 &= 0 \\ d_1C_{-1} + 1 + d_{-1}C_1 + d_{-2}C_2 &= 1 \cdot G_r \\ d_2C_{-1} + d_1 + C_1 + d_{-1}C_2 &= 0 \\ d_3C_{-1} + d_2 + d_1 C_1 + C_2 &= 0 \quad , \end{aligned} \quad (7)$$

in which the factor $G_r$ is the amplification factor introduced by automatic gain control to normalize the output.

These equations are linearly independent in a realizable system. An iteration technique that reduces the deviations in any consistent serial or parallel order will cause convergence.

† Non-automatic means of accomplishing distortion correction by input signal shaping have been suggested by Gibson,[10] who also describes equalizers designed upon the principle of adjusting the signal received during each time unit as a function of the signal received during nearby time units.

The correct solution to these equations insures that the output wave is distortionless within the time period $-1$ through $+2$. The equations do not define the deviations outside this period. Hence, the number of digit positions (i.e., the correction period) required in the implementation of this technique is a function of the number of positions during which the response to a unit input pulse has significant deviations from zero. Experience indicates that a ratio of at least two is required in comparing the former number to the latter. In some cases of severe phase distortion in which the pulse response contains large overshoots preceding or following the main pulse, the distortion can be corrected adequately only by using a much larger number of positions for the correction period. Of course, no finite number of positions will permit complete correction if the transmission system does not have the required bandwidth.

The time at which the peak of the main pulse occurs was chosen as the output time reference. It is evident that, during the iteration process, the position of this output peak may shift somewhat relative to the timing clock of the waveform generator. Although the magnitude of the shift normally is a small fraction of a unit interval, the distortion-correction system can follow this shift automatically. That is, it can sample the output at points spaced in time by unit intervals from the *current* peak point. The resulting final solution then becomes the same as would be obtained from a set of equations like (7) in which the $d$'s are measured at the sampling times established by the final solution.

The distortion-correction procedure outlined in the previous paragraphs is performed by applying a main unit pulse to the transmitter input. Neighboring unit-duration pulses, with their amplitudes proportioned so that the receiver output is distortionless, are then appended to the main unit pulse. Since we are dealing with linear transmission systems, a similar procedure could be followed for distortion correction at the receiver instead of at the transmitter. The counterpart of a transmitted, square-wave correction pulse is a received pulse, distorted by the line. Replicas of the distorted output of the receiver can be modified in amplitude and displaced respectively in time by unit intervals and then added together so that the final output is distortionless. The differences in implementing automatic systems for these two procedures are considered subsequently in this paper.

All of the foregoing analysis requires that the *system* be *time invariant* and *linear* from input point to output point. It follows also that the analysis applies to synchronous carrier systems insofar as any frequency spectrum shift in the channel has been properly compensated and the phase of the carrier recovered at the receiver is independent of the particular data pattern being transmitted. Carrier recovery becomes difficult in double-sideband systems and vestigial-sideband systems if, within the band used for carrier recovery, the phase vs frequency characteristic of the channel has a significant component that is even-symmetric about the carrier frequency. In this case, the recovered carrier tends to change phase as the data pattern changes and care must, therefore, be taken to keep the change within tolerable bounds. This care is required regardless of whether the automatic distortion correction is done at the transmitter or at the receiver, since the automatic system operates on the baseband signal. Although the effects of even symmetry can usually be kept small, the problem might be reduced by the use of compromise equalization about the carrier frequency. Simple, automatic, carrier distortion correction is also conceivable.

• *Automatic systems*

If a linear system is compensated to give undistorted transmission of a single pulse corresponding to a given data rate, then the compensation will be suitable for any linear combination of pulses at the same data rate. It was demonstrated earlier that it is only necessary to introduce corrections at bit sampling times. There are at least two systematic routines which may be employed to compensate for amplitude and phase distortion by observing a single pulse on an oscilloscope.

*Routine 1. a.* Establish time marks at the bit rate such that one marker coincides with the peak of the main pulse (readjustment may be necessary as a result of subsequent steps).

*b.* Examine the response at each bit time sequentially and adjust each response (except the main pulse) to zero. It is usually necessary to repeat this operation several times.

*Routine 2. a.* Same as *1.a.* above.

*b.* Seek the response at a bit time (except the main pulse) which differs greatest from zero and adjust that response until it no longer represents the greatest deviation from the base line. Continue until each desired response at bit times is zero.

Adjustments can be introduced either at the receiver (after demodulation) or at the transmitter (prior to modulation). In either case, the signal is examined at the receiver.

The pre-distorted waveform generator at the transmitter was used for both time-reversal and time-domain experiments. It consists of a group of digital-to-analog (D/A) converters, a summing amplifier, a timing clock and a shift register to control the D/A converters according to the data input sequence. The pre-distortion system works as follows: An undistorted pulse is transmitted and sampled at the receiver, where either the first undesired response or the greatest undesired response (dependent upon the logic employed) is sampled. The address (bit

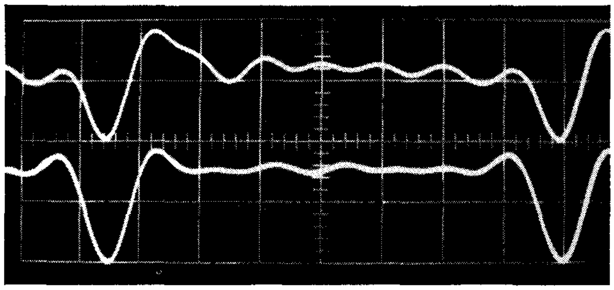


**Figure 10** Pre-distorted input and system response to a single pulse.

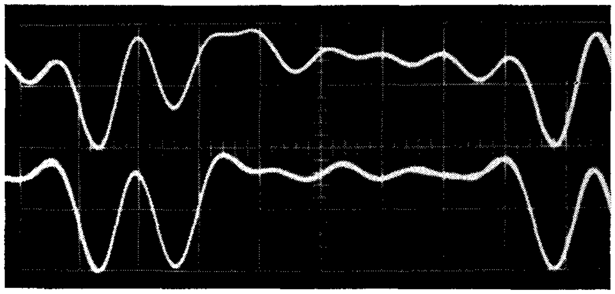


**Figure 11** Pre-distorted input and system response to a pulse pair.

time number) and the sign of the error are stored and retransmitted at a slow speed to the predistorted waveform generator where an incremental correction is made. The procedure continues iteratively until all undesired responses are minimized. Unlike the post-distortion approach, this scheme has the disadvantage of relatively low speed and the need for a return channel.

The post-distortion approach uses a lumped-parameter delay line, tapped at time intervals corresponding to the bit duration; amplifiers at each tap with a digital gain control over the range $-1$ to $+1$; and a summing amplifier. In the post-distortion approach the receiver detection equipment combines a sample-and-hold circuit (which samples at bit times), a digitally-controlled delay line to control the time at which samples are taken, amplitude comparison circuits, and an address-and-sign register.

• *Experimental results*

To test the time-domain equalization technique, the following equipment was used: A vestigal-sideband modulator and demodulator, an "*L*" carrier channel, and the waveform generator discussed earlier. The modem was operated at a 3-kcps carrier frequency. The details of this modem are described in an article by Critchlow, Dennard and Hopner.[4]

Figures 10 and 11 show the system input and output

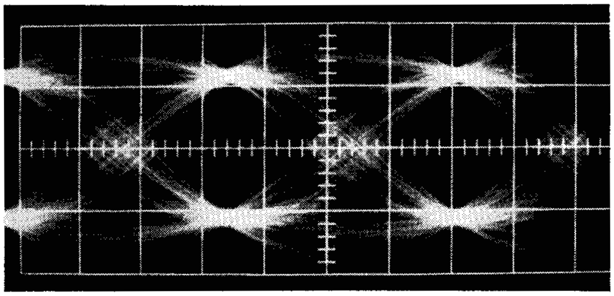


**Figure 12** System response to time-domain equalized quasi-random binary data.

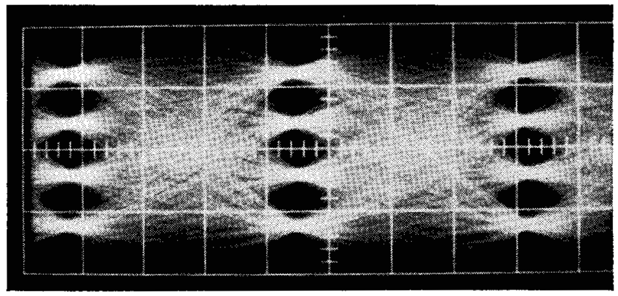


**Figure 13** System response to time-domain equalized quasi-random quarternary data.

waveforms for single and double pulses. If the output pulses are compared with those in Figs. 6 and 7, it can be seen that the time-domain scheme provides equalization superior to that of the time-reversal scheme. This superiority becomes clear if the "eye" of Fig. 12, obtained by time-domain equalization, is compared with the "eye" of Fig. 9. Because the spurious amplitude responses at bit sampling times have been minimized, multilevel modulation is possible with the time-domain equalization technique. Figure 13 shows an "eye" pattern at an 8-kbps data rate.

In order to obtain more comprehensive information, tests were performed on both real and simulated lines. For these tests a VSB modem operating at 2.3-kcps carrier frequency was employed. On some lines, with severe phase distortion near 2.3 kcps, a 2.1-kcps carrier was used.

On the simulated lines with the 2.3-kcps carrier, 4-kbps data transmission was possible in most of the tests. Figure 14 shows the response before and after equalization for a representative case of symmetric-moderate delay distortion. Figure 15 indicates the striking improvement obtained in a case of inverted (over-compensated) line distortion. Another example of especially successful equalization is shown in Fig. 16 for a simulated poor-loaded cable. The delay characteristics of the simulated lines are shown by the graph on each of the illustrations. Altogether, 13 different simulated lines were used in the tests. In all cases of successful equalization only two iterations were required to determine the appropriate test waveform. Only three tests failed at 4 kbps. Failure in two of the tests was due to failure of the modem to retrieve the carrier. When the carrier was patched through from transmitter to receiver, it was possible to equalize the signal in the baseband. The third test failed with a patched-through carrier at 4 kbps but was successful at 3.5 kbps. The most probable cause of the failure was an insufficient number of shift cells in the waveform generator.

Tests were performed over many common-carrier channels, including some loops as long as 4000 miles. The noise and delay distortion on most of the lines tested were too great to permit four-level modulation. In these tests, it was found that the best results were obtained through the combined use of a compromise filter and time-domain equalization. The data rate was 2 kbps. A test over an actual common-carrier facility is shown in Fig. 17. This line could be equalized for two-level operation at 1.75 kbps but not at 3.5 kbps.

Following the work described here, Mohn and Stickler[5] implemented a fully automatic equalization system utilizing digital techniques at the transmitter or receiver such that no analog delay line was required.

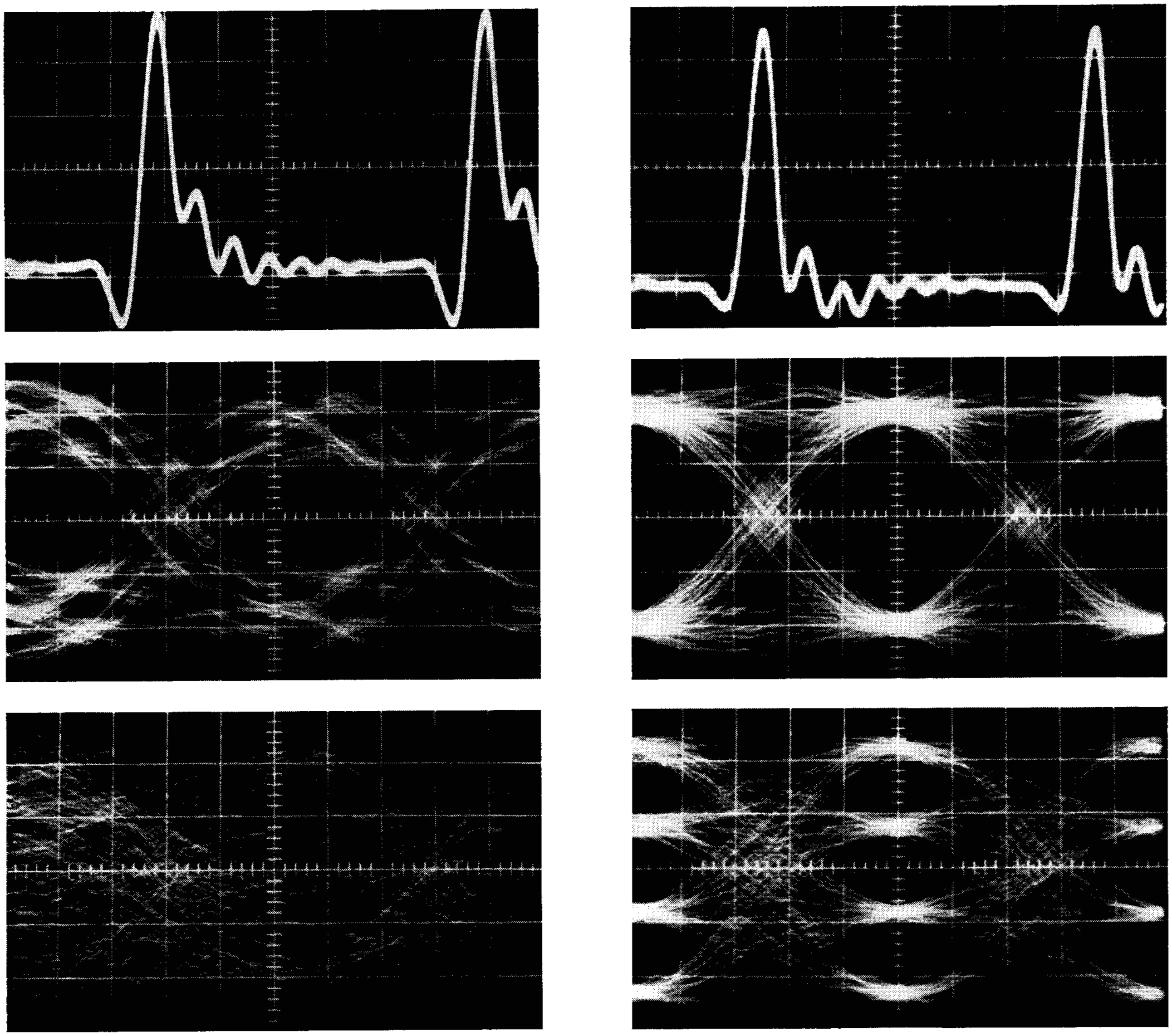
## Summary

The need for improved efficiency in data transmission on switched networks led the authors to study means for automatic distortion correction. Technical feasibility of automatic distortion compensation by two methods was demonstrated, and the experimental results obtained verified the theoretical basis of this work. We believe that systematic approaches to automatic time-domain equalization other than those discussed here may lead to even better results, and hope that the techniques and results reported here will lead other data communication engineers to build upon this study.

## Acknowledgments

The authors acknowledge the valuable contributions of Dr. D. L. Critchlow and Dr. R. Dennard. L. Skarshinski performed much of the experimental work, and M. D. Schettl and J. S. Chomicki constructed the experimental equipment.

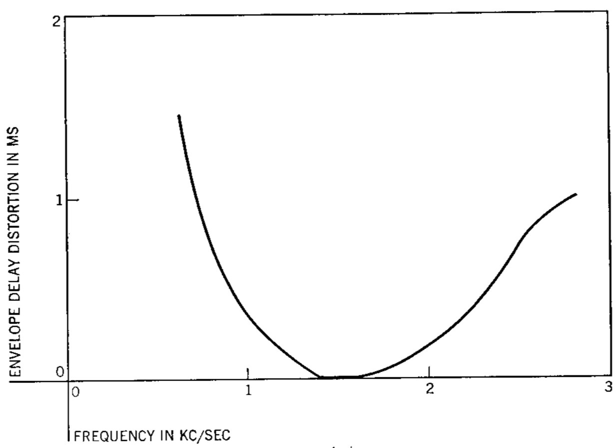
*The references to this paper appear on page 31.*



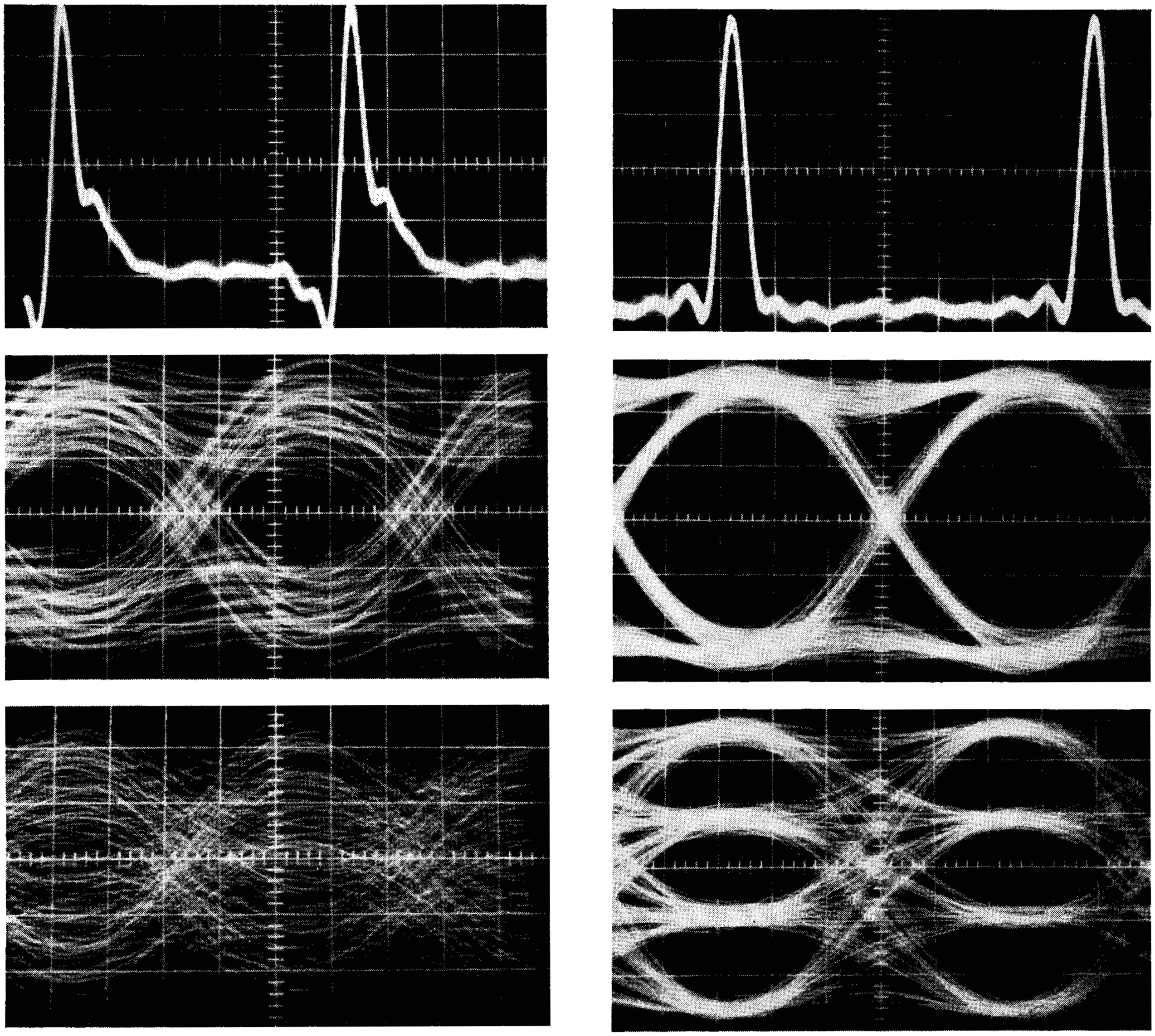
(a)

(b)

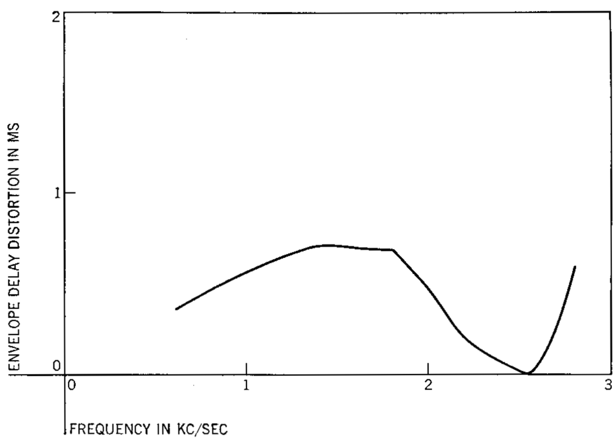
Figure 14 Distortion correction on simulated line having symmetric-moderate delay distortion. (a) Before correction —single pulse response, binary and four-level eye diagrams. (b) After correction—single pulse response, binary and four-level eye diagrams. (c) Delay characteristic of simulated line.




(c)

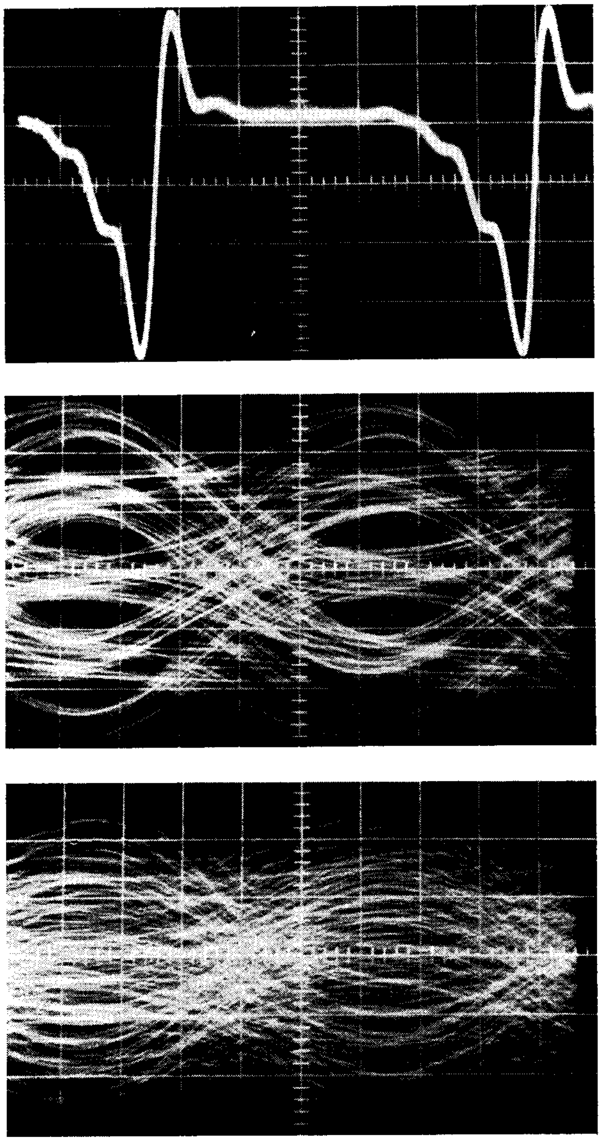


(a) (b)

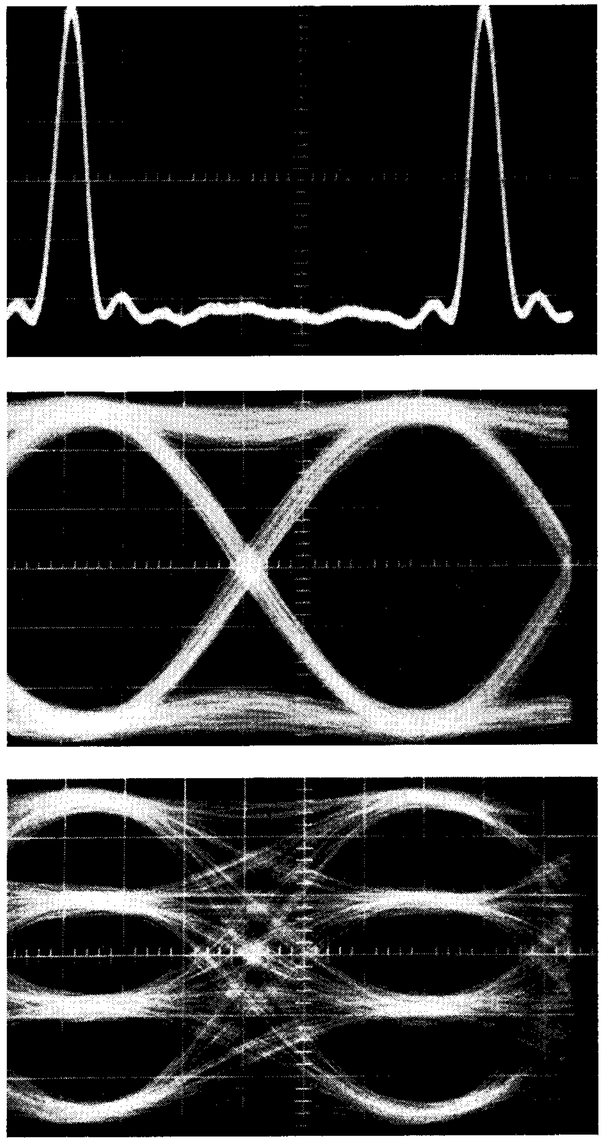



(c)

**Figure 15** Distortion correction on simulated line having inverted (overcompensated) delay distortion. (a) Before correction—single pulse response, binary and four-level eye diagrams. (b) After correction—single pulse response, binary and four-level eye diagrams. (c) Delay characteristic of simulated line.

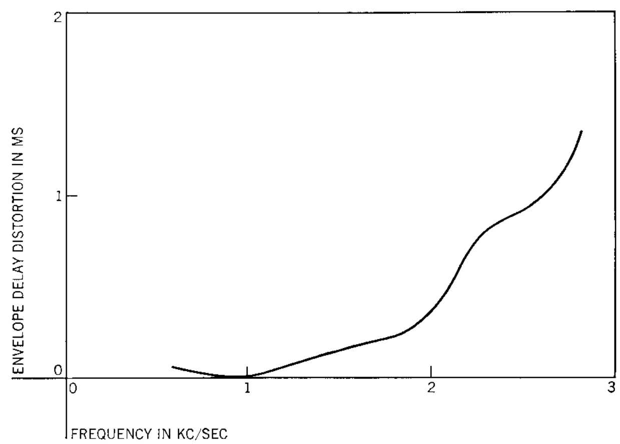


*(a)*

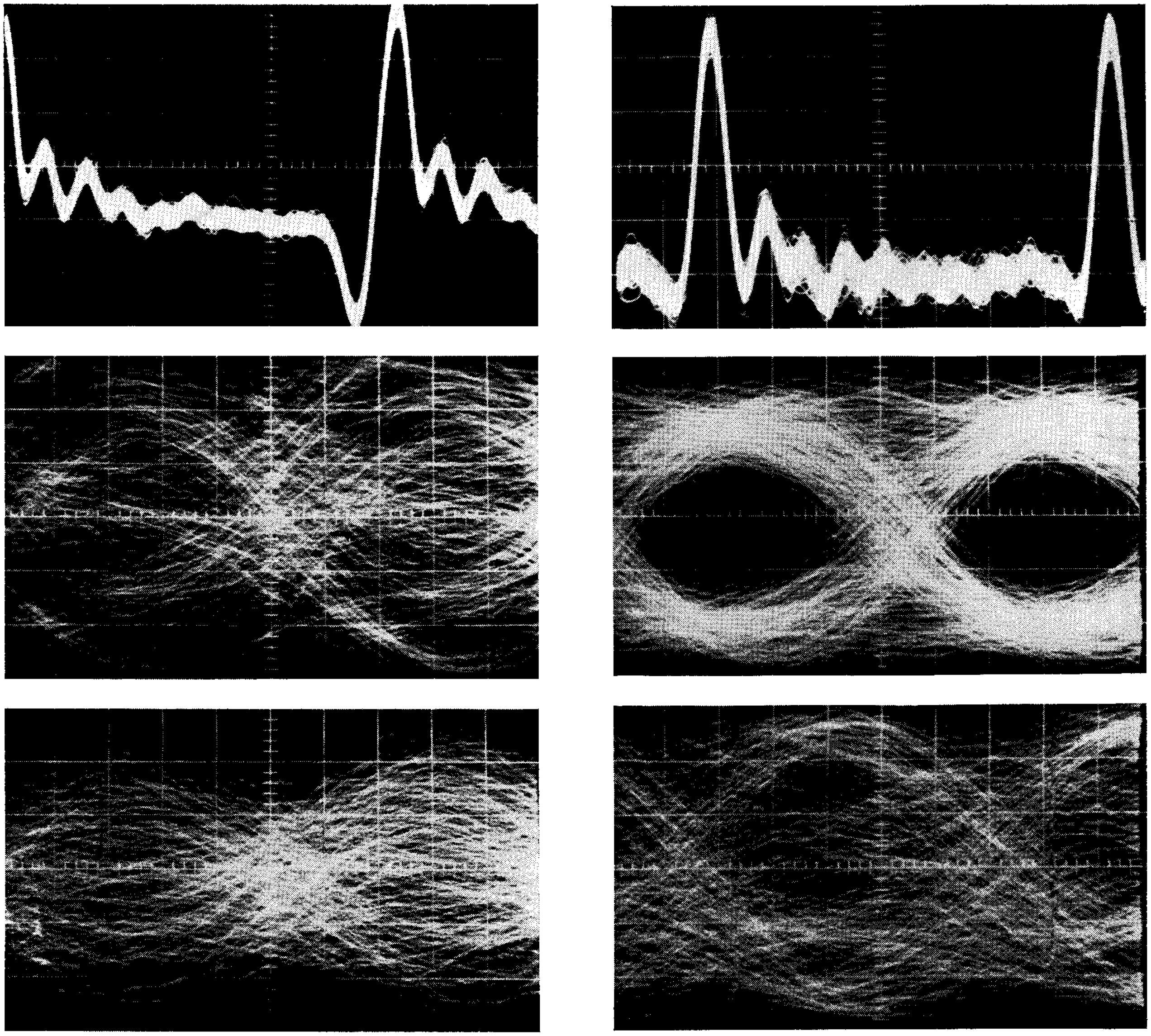


*(b)*

**Figure 16**. Distortion correction on simulated poor-loaded cable. (a) Before correction—single pulse response, binary and four-level eye diagrams. (b) After correction—single pulse response, binary and four-level eye diagrams. (c) Delay characteristic of simulated line.




*(c)*



(a) (b)

**Figure 17** Distortion correction test on a particular loop of the switched network. (a) Before correction—single pulse response at eye diagrams at 1.75 kbps and 3.5 kbps. (b) After correction—single pulse response and eye diagrams at 1.75 kbps and 3.5 kbps.

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