Stretch: A Time-Transformation Technique

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Abstract

Stretch is a passive, linear, time-variant technique for performing temporal operations on many classes of signals. The technique employs three dispersive networks and a mixer. Signal slowdown, speedup, or time reversal can be attained by choice of network slopes. These temporal operations are performed within a signal “window,” and the duration of the window is determined by the network time-bandwidth products. Both heuristic argumentation and rigorous analysis are presented, as are the results of a simple laboratory experiment.

Stretch is a general technique for transforming the time scale of signals which are confined within a defined interval of time. Slowdown or speedup of signals is accompanied by an increase or decrease in the time interval over which they occur. The principles of this technique allow temporal operations to be performed on electrical, electromagnetic, optical, or acoustical signals. The system is linear in the sense that the principle of superposition applies and good waveform fidelity is obtained. This technique is a general solution to the problem of matching data rates of signals to receivers when the frequency characteristics of both are fixed.

Impedance matching to obtain maximum transfer of power is a well-known concept. The analogous concept of matching the data rate to the information transmission characteristics may be more often neglected. Mismatch results in either a loss of signal information or an inefficient use of the transmission system or receiver.

Consider an experiment in which the rise time associated with a nonrepetitive nanosecond transient is to be measured. If the signal is applied directly to the input of an oscilloscope, inefficient performance and cost results. This is due to the fact that an expensive wide-band oscilloscope is required, despite the fact that the duration of the transient is short and the total information content is small. If we had a convenient way of reducing the bandwidth of the signal by slowing down the waveform before the signal is displayed, we could use an inexpensive instrument. The purpose of this paper is to describe a technique that can be used to provide this function for a wide variety of applications.

The body of this paper consists of five sections. The first section reviews dispersive techniques and provides a heuristic explanation of the general concept as applied to the slowdown of signals. In the second section, some of the approximations used in the previous section are discussed. Particular emphasis is placed on describing the concept of a signal “window” within which transformations are performed. The third section extends the description to the speedup and time-reversal operations. The fourth section establishes the validity of the technique by means of a general rigorous analysis. In this section a closed-form solution to the response of the system to an arbitrary input signal is obtained. The fifth section describes a laboratory experiment and presents some test results.

I. Description

The basic elements of a Stretch system are the heterodyne mixer and the dispersive delay device. Although dispersive elements have been used widely in pulse compression radar [1] and their operation in this and related applications is well understood, a brief summary of their basic principles will be given here, to set the framework for the analysis.

We will define a dispersive element to include all devices in which phase shift varies in a nonlinear manner with frequency. As an example, let us consider a device that has
a linear dispersive characteristic; that is, one in which the relationship between delay through a device and the frequency of the applied signal is linear and of frequency/delay slope \( \sigma \). The response of this device to a short pulse (on a constant frequency carrier) of bandwidth \( B \) and time duration \( \tau \approx 1/B \) will be a longer pulse of length \( B/\sigma \). This response will, however, retain the bandwidth \( B \) by virtue of a frequency modulation of the carrier. The pulse power is reduced by the same proportion that the pulse width is increased to conserve the pulse energy.

It has been shown [2] - [4] that the output signal can be represented quite well by a line in the frequency/time plane as long as the time-bandwidth product (the product of the time duration of the dispersed pulse and its bandwidth) is considerably greater than unity. This representation will be used extensively in the following sections.

The compression process involves introducing the dispersed pulse into a device of slope \(-\sigma\), obtaining a short pulse (length \( \tau \)). The two networks when connected in cascade have a constant total delay and constitute nothing more or less than a delay line. The original pulse shape is therefore recovered.

Notice that if a frequency shift of \( \Delta f \) is applied to the dispersed signal prior to compression, a shift \( \Delta \tau \) in the time position of the output pulse is obtained [5] where \( \Delta \tau = \Delta f/\sigma \). Thus the time of the output pulse is determined by the time at which the dispersed signal passes through any particular frequency and by the delay of the compressor at that frequency. The time of the output pulse is not influenced by the start and finish times of the dispersed signal.

Dispersive devices have been used most extensively in pulse compression radar. They have been constructed to make use of acoustic [6], electrical [7], and electromagnetic [8] dispersion. The principles which give rise to dispersion and the construction techniques are outlined in the referenced papers.

Fig. 1 is a functional block diagram of one implementation of the Stretch technique. The input signal that is to be transformed in time is applied to a dispersive device, mixed against a sweep, and then applied to a second dispersive device which is used as a compressor. For purposes of explanation, let us assume that an arbitrary input waveform (Fig. 2) is to be slowed down in time to reduce the bandwidth of the signal.

Shannon’s sampling theorem [9] allows us to resolve the general signal of Fig. 2(A) into a series of component pulses. In accordance with the sampling theorem, the elementary pulses used to represent the waveform must have a bandwidth at least equal to the bandwidth of the input signal (i.e., pulse width commensurate with the rise time of the signal). If rectangular pulses are visualized, then it is easy to see [Fig. 2(B)] how the complex signal can be approximately represented by the summation of the individual component pulses.

Consider first a “reference” pulse at the center \((t = t_R)\) of the signal described in Fig. 2. This pulse, shown in Fig. 3(A), has a pulse length \( \tau_1 \) and an amplitude \( P_{R1} \) as determined by the bandwidth of the sampled waveform and the amplitude of the waveform at the time \( t_R \). If the bandwidth of the input dispersive device is at least equal to that of the signal, a dispersed pulse will be obtained. The dispersed pulse length and the slope of the frequency modulation under this envelope is determined by the characteristics of the device. Fig. 3(B) shows the output signal from the device. The bandwidth over this pulse length is the same as the bandwidth of the sample, \( 1/\tau_1 \). The dispersed pulse length is

\[
\tau_2 \approx 1/\sigma_1 \tau_1
\]

where \( \sigma_1 \) is the frequency/delay slope of the device. The amplitude of the pulse is

\[
P_{R2} \approx \frac{\tau_1}{\tau_2} P_{R1}.
\]

This dispersed reference pulse is now mixed with a wide-band sweep. The sweep is a linearly frequency-
modulated voltage [as shown in Fig. 3(B)] which is very similar to the signal obtained at the output of the dispersive device. The sweep can be generated either actively, if components are available, or passively by impulsing a third dispersive device with a trigger.

If the input signal is to be slowed down, the wide-band sweep frequency/time slope ($\alpha_2$) is set slightly below that of the dispersive network ($\alpha_1$). The timing of the sweep [shown as a dotted line in Fig. 3(B)] is adjusted so that the sweep is on during the time duration of the dispersed reference pulse.

The difference-frequency mixer output is chosen in this example; hence, the mixer output signal [shown in Fig. 3(B)] is at a lower carrier frequency and has a reduced slope, $\alpha_3$, where

$$\alpha_3 = \alpha_1 - \alpha_2.$$  

This mixer output is now applied to the input of the second dispersive device (the compressor).

The characteristics of the second dispersive device are chosen so that the signal applied to the device is compressed. Since the slope of the applied signal is determined by the characteristics of the input dispersive device and of the sweep, this network can be designed independently of the input signal. The result of compressing the signal is shown in Fig. 3(D). The compressed pulse width ($\tau_3$) is the reciprocal of the bandwidth of the applied signal. This bandwidth is determined by the slope ($\alpha_3$) and the pulse length ($\tau_2$) of the applied signal:

$$\tau_3 = \frac{1}{\sigma_3 \tau_2} = \frac{\alpha_1}{\alpha_3} \tau_1.$$  

The amplitude of the output pulse (assuming no mixer loss) is

$$P_{R3} = \frac{\tau_2}{\tau_3} P_{R2} = \frac{\tau_1}{\tau_3} P_{R1} = \frac{\alpha_3}{\alpha_1} P_{R1}.$$  

Notice that both the output pulse width and the amplitude are proportional to the input pulse width and the amplitude, with a proportionality constant determined by the design of the input and output dispersive devices. Furthermore, since the output signal energy is always the same as the input signal energy, no change will result in the signal-to-noise ratio of a noisy input signal (the noise power is also reduced by the ratio of the input-to-output bandwidth).

Now let us consider a second “test” pulse in the train of sampling pulses that represents the input waveform. The test pulse, which is similar to the reference pulse, is positioned at an arbitrary time ($t_R = t_f + \Delta t_1$) relative to the reference pulse. Its amplitude ($P_{R1}$) is equal to the amplitude of the input signal at time $t_f$.

Fig. 4(A) shows the two pulses separated by the time $\Delta t_1$. Fig. 4(B) shows the output of the dispersive network corresponding to these two pulses; the two dispersed linear FM signals are displaced in time by $\Delta t_1$. While passing through the mixer, the two signals are mixed against the sweep and, as before, are reduced in slope and center frequency [Fig. 4(C)].

To determine the position of the mixer’s output signal, we must consider the fact that at any instant of time the mixer’s two inputs and the sweep behave like signals of constant frequency. Since both input signals are mixed against a common instantaneous sweep frequency, the two output signals must have a constant frequency separ-
The test pulse is, as before,

\[ P_{T3} = \frac{\sigma_3}{\sigma_1} P_{T1} \]

Thus we have shown that all of the samples of the input waveform are both broadened and separated by a controllable factor in passing through the system. In addition, no signal energy has been lost. The reconstituted output signal is, of course, the sum of the output sample pulses. We may conclude that the input signal has been slowed down in time with (conceptually) no waveform distortion or loss of signal-to-noise ratio.

II. Approximations

To make the operation of the basic technique easily understood, several approximations have been made in the preceding analysis. Three of the most significant are the following:

1) that the input function can be represented by a set of rectangular pulses;
2) that the compression network characteristics and dispersed signals can be represented adequately by plots in a frequency-time coordinate system;
3) that all of the pulses representing the input waveform are close to the reference pulse, and that the input and output dispersive networks have passbands which are considerably wider than the spectrum of the signals.

The first two approximations are treated extensively in the literature [2], [9]. If the system parameters are conservatively chosen, these approximations represent neglectable error in the analysis. The first approximation is generally avoided by using \( \sin \frac{x}{x} \) (band-limited) sampling pulses. This requires that the phase of the carrier under the pulses be considered, so that during reconstitution the vector sum can be taken where adjacent samples overlap. The second approximation is shown to be valid for any reasonable choice of parameters, that is, for designs in which both input and output dispersive time-bandwidth products exceed about 10. The third approximation is the most significant and leads us to the concept of a signal "window."

In discussing the third approximation, let us assume that the passbands of the dispersive devices are rectangular in shape, that they are exactly equal to the bandwidth of the signals passing through the devices, and that the signal spectra are band limited, allowing the use of \( \sin \frac{x}{x} \) pulse samples. Referring to the previous analysis, the bandwidth of the input dispersive network is \( 1/r_1 \) and the bandwidth of the output network is \( 1/r_2 \). We will assume that the input signal is band limited with a bandwidth of \( 1/r_1 \). If we now review the analysis, we will find that no difficulty arises until we reach the signal that is applied to the output compressive network. Referring to Fig. 4(C), we see that the time separation between the test and reference signals...
has been accompanied by a shift in frequency. As we move the test sample pulse away from the reference position (that is, increase Δt), the linear FM signal applied to the compressor shifts in frequency (that is, Δf increases), as shown in Fig. 5(A) and (B).

The passband of the output compressor is fixed to accommodate the band defined by the reference pulse. Therefore, as the test pulse is moved from the reference position, a loss of spectrum and of pulse energy results. This, in turn, results in a gradual degradation of the test pulse shape and amplitude as a function of the position. This effect is shown in Fig. 5(C).

We can arbitrarily define an input “window” by assuming that pulses which pass through the system are useful if no more than half of their energy is lost. Although this definition allows a considerable loss of power and bandwidth at the edges of the window, it is useful in many applications. The result of this assumption is that the maximum displacement of the test pulse (at the output) is equal to \( \pm \Delta t_{2\text{ max}} \), where

\[
\Delta t_{2\text{ max}} = \frac{1}{2 \tau_3 \sigma_3 (1 - \sigma_3 / \sigma_1)}.
\]

The ratio of the output signal’s maximum displacement divided by the output pulse width (or signal rise time) is equal to the window width \( W \) (expressed in number of resolution elements):

\[
W = \frac{2 \Delta t_{2\text{ max}}}{\tau_3} = \frac{1}{\tau_3^2 \sigma_3 (1 - \sigma_3 / \sigma_1)}
\]

or

\[
W = \frac{\tau_2}{\tau_3} \frac{1}{1 - \sigma_3 / \sigma_1}.
\]

The term \( \tau_2 / \tau_3 \) is the pulse compression ratio of the compressor. In addition, when the parameters are chosen to obtain slowdown of input signals, the term \( \sigma_3 / \sigma_1 \) often becomes negligible.

Hence, the window width \( W \) becomes approximately equal to the time-bandwidth product of the output dispersive device when the minimum bandwidth compressor is used. If the compressor bandwidth is increased, \( W \) will also increase. In general, the limits over which good performance can be obtained are related to the engineer’s ability to control and predict the performance of the input and output dispersive devices, since wide input bandwidth, a large slowdown ratio, and a wide window all require high network time-bandwidth products. In a practical implementation these devices can be constructed to obtain linear dispersion within sufficient tolerance to obtain spurious response and distortion of the order of 30 to 50 dB below peak signal. The maximum ratio of input-to-output time scale is generally determined by the engineer’s ability to match the sweep to the input dispersion characteristic. By using special error-correcting techniques, time slowdown ratios of the order of several hundred have been achieved.¹

III. Extension to Speedup and Time Reversal

It has been shown that when the slope of the mixer sweep is adjusted to be less than the slope of the input dispersive device, and when the difference output frequency of the mixer is used, a slowdown of the input signal waveform results. The output time scale is controlled by adjusting the sweep slope. When zero slope is used (that is, when a CW signal is applied), unity time expansion is obtained. The expansion ratio increases as the sweep slope approaches the input dispersive device slope.

By extending the discussion of Section I, it can be shown that various combinations of time expansion, time compression, and time reversal can be obtained by controlling the slope of the sweep.

If the sweep slope is negative with respect to the slope of the dispersive device, then time compression or speedup will result. This can be demonstrated by considering that the mixer signal output slope is greater than the signal input slope (Fig. 6). The amount of time compression can be controlled by adjusting the absolute value of the sweep slope, with zero slope yielding unity compression ratio.

When the sweep slope is positive but greater than the input signal slope to the mixer, a time reversal of the input waveform is obtained. This is true because the output signal slope from the mixer is negative relative to the input; thus, the downward frequency shift of the mixer output

¹For example, the signal dispersive network can also generate the reference sweep of Fig. 1, by slightly predispersing the reference impulse, adding it to the signal, and replacing the mixer by a detector diode. This results in partial cancellation of smooth nonlinearities in the dispersion characteristic of the network.
Fig. 6. Speedup; double-pulse response.

Fig. 7. Time reversal.

Fig. 8. Network slope selection diagram.
associated with delay in the test pulse position results in an advancement of the output pulse position. This effect is shown in Fig. 7.

When reversal is desired, the output time scale can still be adjusted. Greatest time expansion or slowdown occurs when the sweep slope is slightly greater than the input signal slope. The expansion ratio decreases to unity when the sweep is equal in slope to twice the input signal slope. Further increase in the sweep slope causes compression of the output signal time scale. These relationships are summarized in Fig. 8.

IV. Closed-Form Analysis

The previous discussion is most useful as a heuristic exposition of operating principles. Since many approximations were made during the course of the analysis, an effort was undertaken to perform an “exact” analysis. The closed-form analysis presented in this section does not depend upon the sampling theorem, and it includes the effects of signal phase. We therefore will now derive the response of the system to a general complex input signal of the form

\[ m(t) \exp i \omega t. \]

It was shown by Klauder [3] that a complex signal of the form

\[ V_1(t) = m(t) \exp i (\omega_0 t \mp k t^2) \]

when applied to a network with transfer function

\[ H(\omega) = \exp i \left( \frac{\omega - \omega_0}{4k} \right)^2 \]

produces the complex output signal

\[ V_2(t) = \sqrt{\frac{k}{\pi}} \tilde{m}(2k) \exp i \left( \omega_0 t \mp k t^2 \pm \frac{\pi}{4} \right) \]

where \( \tilde{m}(\omega) \) is the Fourier transform of \( m(t) \).

In the Stretch system, the input signal \( m(t) \) is modulated onto a carrier frequency, \( \omega_1 + \omega_2 \), greater than the single-sided bandwidth of \( m(t) \) to produce the signal

\[ m(t) \exp i (\omega_1 + \omega_2)t. \]

For purposes of analysis it is useful to rewrite the signal in the form

\[ m'(t) \exp i [(\omega_1 + \omega_2)t - (a + b)t^2] \]

where

\[ m'(t) = m(t) \exp i (a + b)t^2. \]

This is applied to the first dispersive network whose transfer function for positive frequencies is

\[ H_1(i\omega) = \exp i \left( \frac{(\omega - \omega_1 - \omega_2)^2}{4(a + b)} \right) \]

producing the output

\[ \sqrt{\frac{a + b}{\pi}} \times \tilde{m}'[2(a + b)t] \exp i \left( \omega_1 + \omega_2 \right) + (a + b)t^2 - \frac{\pi}{4} \] \cdot (8)

The signal is mixed with an FM ramp,

\[ 2 \cos (\omega_1 t + at^2), \]

and passed through a low-pass filter, producing the difference frequency output

\[ \sqrt{\frac{a + b}{\pi}} \times \tilde{m}'[2(a + b)t] \exp i \left( \omega_2 t + bt^2 - \frac{\pi}{4} \right). \]

This signal is applied to a second dispersive network with positive frequency transfer function

\[ H_2(i\omega) = \exp i \left( \frac{\omega - \omega_2}{4b} \right)^2 \]

producing the output

\[ \sqrt{\frac{a + b}{\pi}} \sqrt{\frac{b}{\pi}} \tilde{m}'[2(a + b)t] \exp i (\omega_2 t - bt^2). \]

The double Fourier transform leads to a signal similar to \( m' \), with a change in argument. This transform is evaluated as follows:

\[ \tilde{m}'[2(a + b)t] \]

\[ = \int m'(\lambda) \exp -i\lambda [2(a + b)t] d\lambda \exp -i\lambda(-2bt) d\tau \]

\[ = \int m'(\lambda) \int \exp -i\tau [2(a + b)\lambda - 2bt] d\tau d\lambda. \]

Integration over \( \tau \) leads to

\[ \int m'(\lambda)2\pi \delta [2(a + b)\lambda - 2bt] d\lambda. \]

Changing variables where

\[ \mu = 2(a + b)\lambda \]

leads to

\[ 2\pi \int m' \left[ \frac{\mu}{2(a + b)} \right] \delta(\mu - 2bt) \frac{d\mu}{2(a + b)} \]

which is easily evaluated to be

\[ \frac{\pi}{a + b} m' \left( \frac{bt}{a + b} \right). \]
Using this result, (12) becomes

$$\sqrt{\frac{a+b}{\pi}} \sqrt{\frac{b}{\pi}} \frac{\pi}{a+b} m' \left( \frac{b}{a+b} \right) \exp \left( \frac{b}{a+b} t \right) \exp \left( \omega_2 t - \frac{b t^2}{\pi} \right).$$

(13)

Define the Stretch ratio as

$$S = \frac{a+b}{b}$$

and substitute (6) for \(m'\) to give the complex representation of the second dispersive network’s output:

$$\sqrt{\frac{1}{S}} m \left( \frac{t}{S} \right) \exp \left( \omega_2 t - b \left( 1 - \frac{1}{S} \right) t^2 \right).$$

(15)

If this signal is envelope-detected, the linear FM term in the output carrier has no effect. Alternatively, it may be removed by mixing the signal with an appropriate FM ramp at the output of the second dispersive network. Assuming envelope detection, the system output is

$$\sqrt{\frac{1}{S}} m \left( \frac{t}{S} \right),$$

which is the input signal stretched in time by the factor \(S\) and scaled in voltage amplitude by the factor \(1/S\). No limits are placed on the value or sign of \(S\). The result is valid over a range of \(t\) such that

$$B_S + 2b \left( 1 - \frac{1}{S} \right) t_M \leq B$$

(17)

where \(B_S\) is the bandwidth of \(m(t/S)\) and \(B\) is the pass-band of the second dispersive network. Rearranging, (17) becomes

$$t_M \leq \frac{B - B_S}{2b \left( 1 - \frac{1}{S} \right)}.$$  

(18)

V. Experimental Results

To verify the operation of the Stretch technique, a modest laboratory experiment was devised to operate within a limited range of parameters. The equipment was constructed, assembled, and aligned, and sufficient test data were taken to verify the relationships developed in the preceding sections.

Fig. 9 is a block diagram of the experimental configuration. The signal source consisted of a laboratory oscillator and a solid-state modulator. The output of the modulator was fed to one input of a dual-trace oscilloscope. This input signal was also fed to the input of the transforming system.

The characteristics of the dispersive networks available at the time of the experiment were a bandwidth of 1/3 MHz, and a frequency/delay slope of 0.02 MHz/μs, resulting in a time-bandwidth product of 6. The networks were constructed of a cascaded series of all-pass networks at 2-MHz center frequency.

From the input dispersive network the signal was fed through a series of mixers followed by bandpass filters and isolated by single-stage amplifiers. First, by means of a balanced mixer, the center frequency of the dispersed signal was raised to about 30 MHz. After appropriate filtering and amplification, the signal was mixed with the sweep, which was centered at a frequency of 42 MHz. The resulting narrow-band signal at 12 MHz was bandpass filtered and mixed with a 14-MHz oscillator. A narrow-band dispersed signal at 2 MHz was thus obtained.

The output signals from the third mixer were applied to a network with the characteristics necessary to compress the reduced bandwidth signals. In this experiment the sweeping local oscillator (a varactor-tuned oscillator, driven by a video saw-tooth generator) was adjusted to a slope of 0.01 MHz/μs. The network characteristic (0.01 MHz/μs slope) required to compress the signals was obtained by cascading the remaining two dispersive networks.

The progression of signals through the system results in a two-fold slowdown and a reduction in bandwidth to 166 kHz. The output signal was displayed on the dual-trace oscilloscope. The signal relationships in the system are summarized in Fig. 10. The signal, a simple pulse-modulated carrier [Fig. 10(A)], is dispersed in the first dispersion network, as shown in Fig. 10(B), which also shows the result of the first mixing to 30 MHz; the center frequency of the signals has been changed and the slope inverted in this process. The second mixing, against the sweep, is shown in Fig. 10(C). Since the sweep has less slope than the applied signal and is at a higher frequency, signal slope inversion accompanies the change in bandwidth of the applied signal. The third mixing reduces the center frequency to 2 MHz and inverts the slope again [Fig. 10(D)]. The result of compressing this dispersed signal is the two output pulses shown in Fig. 10(E).

When a test signal was applied to the completed equipment, the results shown in Fig. 11 were obtained. Fig. 11(A) shows that the input is a complex train of pulses following one large pulse, which is 3 μs in duration. The output in this case is shown in Fig. 11(B). The main pulse is 6 μs in duration. Fig. 11(C) shows the input and output for a more complex signal. The output again is a replica of the input, although some narrow-bandening is evident.

Further tests involved moving the single pulse to both sides of the reference position, while measuring the amplitude and pulse width as a function of position. The results of these tests conformed well with theory in that no obvious distortion or attenuation of the signal was observed for input pulse positions within ±5 μs of the window center. Input pulse displacements between 5 and 9 μs from the center resulted in a smooth drop in output amplitude and increasing sidelobes. The time position of the output pulse was double the time position of the input pulse to within a fraction of the pulse width for input displacement to ±9 μs. Input displacement greater than +9 μs resulted in increasing distortion and attenuation.
Fig. 9. Experimental configuration.

Fig. 10. Experiment signal relationships.

Fig. 11. Experimental results.
Conclusions

An approximate analysis has been developed to explain the operation of the Stretch technique, and this analysis has been verified rigorously. Test results have been obtained that support the analysis.

It has been shown that the technique permits convenient linear manipulations of the time and bandwidth coordinates of signals with good waveform fidelity. By means of this technique, signals can be slowed down, speeded up, or reversed in time with corresponding changes in the spectrum. The basic technique permits passive linear operation on electrical, electromagnetic, or acoustic signals over any portion of the spectrum.

In its simplest configuration, two dispersive elements and one mixer are required. Generation of the sweep from a trigger impulse may be accomplished by a third dispersive element. The application of this technique to any particular type of signal therefore depends only on the availability of dispersive elements and nonlinear devices to handle the particular type of energy and the portion of the spectrum involved. For microwave signals, a conventional diode mixer and waveguide operating near cutoff might be used for the essential components. For optical signals, glass is dispersive and photocells operate like mixers. In general, the input bandwidth, the maximum bandwidth reduction, and the amount of information that can be handled by a particular design are determined by the designer’s ability to control and predict the performance and characteristics of the dispersive devices.

The technique provides a general solution to the problem of matching the data rate which is obtained from an experiment or other signal source to that which can be handled by an observer, recorder, display, or transmission line. Several specific applications are currently being pursued or have been successfully solved over the past six years using the principles explained in this paper. Time slowdown factors of up to 75 have been achieved with input signal bandwidths of 200 MHz and window widths of 0.5 µs in presently operational equipment. Time slowdown of 500 MHz bandwidth signals has been achieved in the laboratory, and systems which operate at much higher input bandwidths are in the planning stage.

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References


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