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# DesignCon 2006

## Group Delay and its Impact on Serial Data Transmission and Testing

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## Abstract

This paper is an extension of last years paper entitled “Eye Pattern Measurements in Scopes”. Last years paper pointed out the effects of bandwidth, roll-off rate, flatness, and group delay characteristics along with probe loading and return-loss. It only glossed over the more complicated topic of phase response and group delay. This paper goes into greater detail on this topic and explains what group delay is, what its impact is, what theoretically correct group delay is, and how group delay is compensated in both serial data test equipment and serial data transmitters and receivers.

## Author(s) Biography

Pete Pupalaikis is Principal Technologist at LeCroy working on new technology development. Previously he was Product Marketing Manager for high-end scopes. He has six patents in the design of measurement instruments. He holds a BSEE from Rutgers University and is a member of Tau Beta Pi, Eta Kappa Nu and the IEEE Signal Processing and Communications Societies.

## Introduction

Last year, I presented a paper entitled “Eye Patterns in Scopes”. That paper covered many topics related to serial data characteristics, mainly bit-rate and rise-time and oscilloscope characteristics such as bandwidth, roll-off rate, flatness, phase response, etc. The paper explained how the characteristics interact to determine what you see on your scope screen when measuring eye-patterns.

The topic of phase response (or group delay response – I’ll use these interchangeably) was briefly introduced. Group delay, while more complicated to understand and compensate than its brother magnitude response, is equally important.

The first part of this paper deals with oscilloscope responses. I will discuss linear phase systems and minimum phase systems and contrast the two. I will discuss allpass filters and how they work. I will explain how linear phase and minimum phase systems are related by allpass filter networks. I will also explain minimum phase systems from a zero location standpoint and show you how to make minimum phase systems out of any impulse response. I will show the difference between the two response characteristics where serial data measurement is concerned and show why linear phase might be preferable in this case.

The second part of this paper deals with channel group delay characteristics and how these affect serial data transmission. I will show a simple simulation along with a very simple model for understanding the primary effect of group delay on the eye. I will briefly discuss how group delay equalization can be implemented in hardware.

## Scope Group Delay Characteristics

For the last five years, LeCroy has been applying magnitude and phase response compensation to our oscilloscopes utilizing digital signal processing (DSP). The result is an excellent response, as illustrated by the SDA 11000, whose step response is shown in Figure 1.

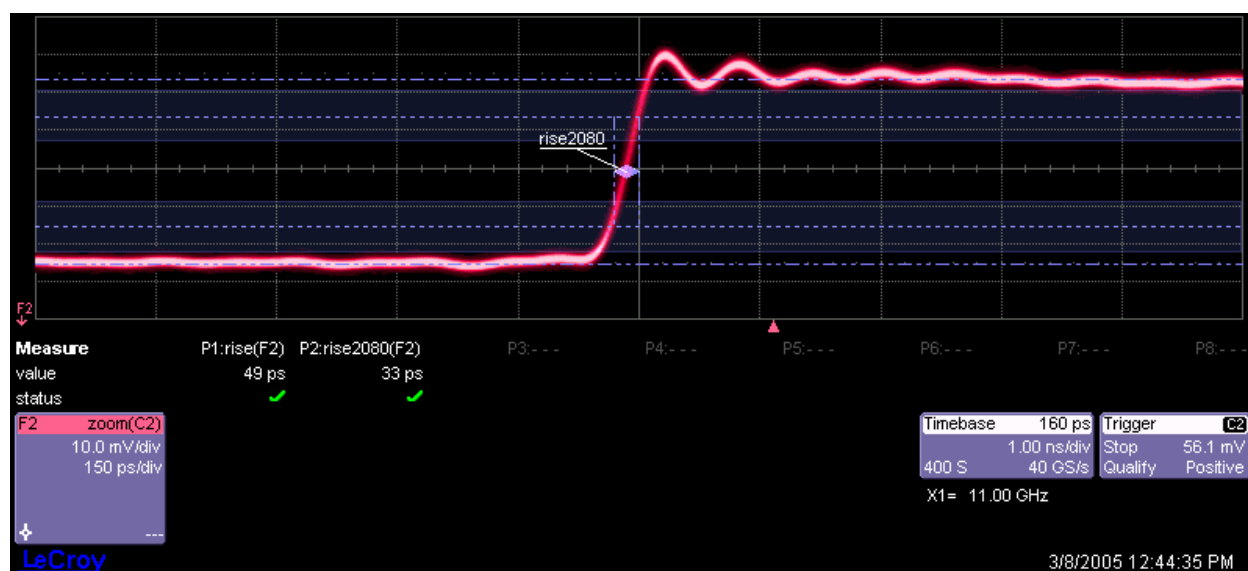
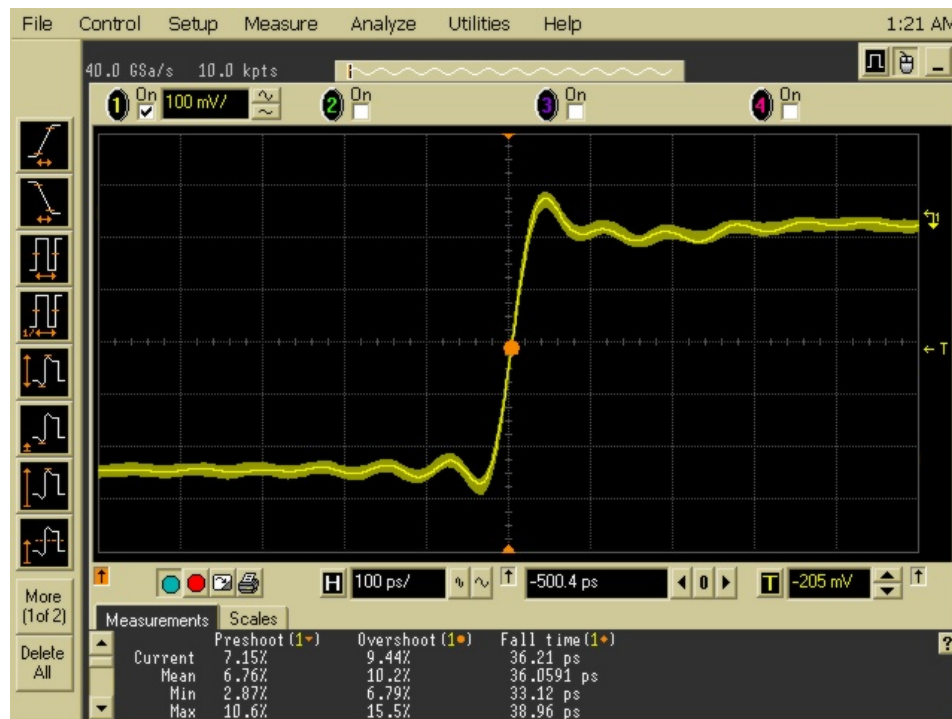


Figure 1 – LeCroy SDA 11000 Step Response

Lately, our competitors began adopting these techniques as well in their real-time scopes. At present, you will not find any high-end real-time oscilloscope manufactured that does not apply DSP for magnitude and phase correction.

A few years ago, scopes started showing up on the market with step responses as shown in Figure 2. My first reaction was that this is the type of response to avoid<sup>1</sup>.



**Figure 2 – Competitors High Bandwidth Scope Step Response**

It turns out that the scopes in Figure 1 and Figure 2 have nearly the same bandwidth, and almost the same magnitude response characteristic. The difference is mainly in the phase responses. In trying to articulate the difference in these responses, I learned (or perhaps clarified my thinking) about a lot of things and it is the result of this learning that I would like to share with you.

## Brick-Wall Responses

All high-end real-time DSOs exhibit a sharp roll-off in magnitude response beyond the bandwidth point<sup>2</sup>. This is a result of scope manufacturers struggling to keep pace with the bandwidth needs of the industry. The sharp roll-off in response has an adverse effect on the step response. The roll-off rates are usually so fast after the bandwidth point that the scopes can be considered to have a “brick-wall” response.

There are two ways to model a brick wall response. One way is to design a finite impulse response (FIR) filter by taking the inverse Fourier transform of the sinc pulse, which provides the classic  $\text{Sin}(x)/x$  impulse response<sup>3</sup>. The step response of such a brick wall model exhibits the familiar Gibbs oscillations. Anyone who has taken a basic DSP course has encountered this. As the reference cited points out, the sharp cutoff is undesirable in that it introduces overshoot and

ringing, but this is unavoidable when the scope runs out of response. But he also points out that were it not for the *delay*, the response would appear to have an instantaneous step response with anticipatory ripples – a violation of real-world causality.

Another way to model a brick wall response is to create a Butterworth filter and design it with ever increasing filter order – by introducing more poles<sup>4</sup>. In the limit, this filter takes on the brick wall response, but you will note that this system never preshoots.

Both of these models of a brick wall system are valid and the step responses for these two types of models are shown in Figure 3. It turns out that the first model is *linear phase* and the second model is *minimum phase*.

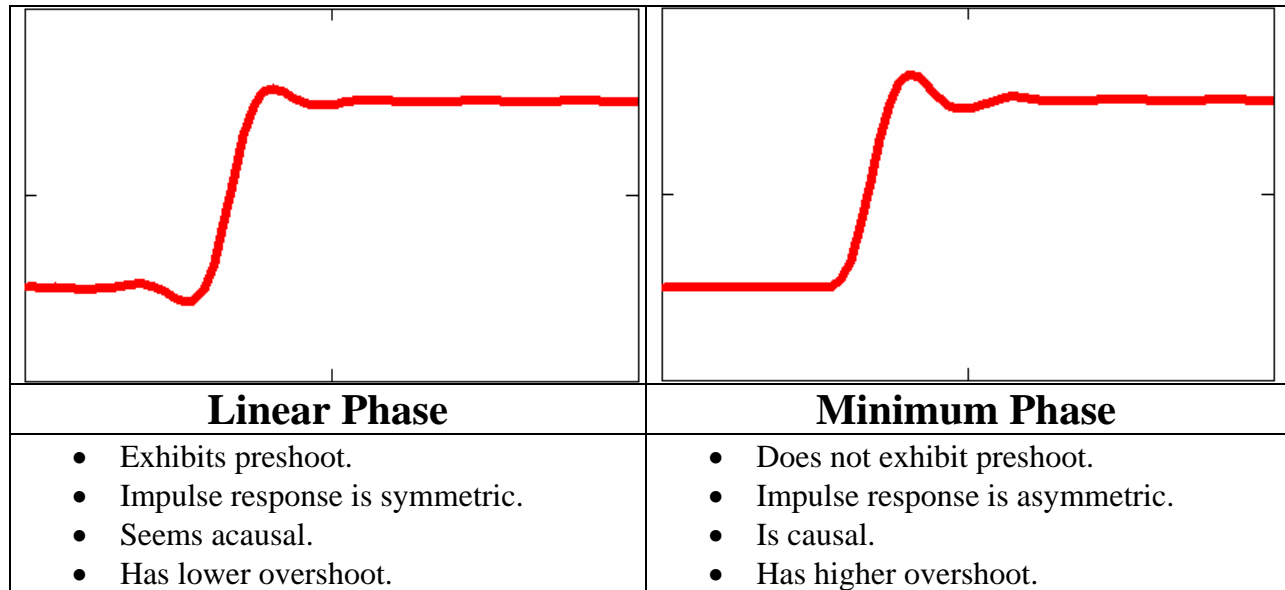


Figure 3 – Step Response of Linear Phase vs. Minimum Phase System

## Linear Phase

The concepts of linear phase and certainly minimum phase are often ill understood concepts. Let me start by explaining the concept of linear phase. In my explanation, I will try to debunk several myths.

First, the concept of the ideality of linear phase derives from the concept of group delay. Group delay is sometimes called envelope delay and should not be confused with phase delay. Both group delay and phase delay are related to the phase of the system by the following formulae<sup>5</sup>:

$$P(\omega) \triangleq -\frac{\Phi(\omega)}{\omega} \text{ or } P(f) = -\frac{\Phi(f)}{2 \cdot \pi \cdot f}$$

Equation 1 – Phase Delay

$$D(\omega) \triangleq -\frac{d}{d\omega} \Phi(\omega) \text{ or } D(f) = -\frac{1}{2 \cdot \pi} \cdot \frac{d}{df} \Phi(f)$$

## Equation 2 – Group Delay

The phase delay is the time delay of a sinusoid at frequency  $f$  assuming that the sinusoid is constant for all time. The group delay is the time delay of the amplitude envelope of a narrow group of frequencies around  $f$ . You can see that when the phase  $\Phi(f)$  is linear with frequency, both the phase delay and the group delay evaluate to a constant delay. When the phase is non-linear with frequency, neither the phase delay nor group delay is constant with frequency.

In a typically encountered band-limited system, the group delay rises near the band edge. The reason for this is that the band edge is defined by the presence of one or more poles that, while adding 20 dB/decade to the roll-off rate, also add 90 degrees of phase lag. At the center frequency of a complex conjugate pair of poles, where the magnitude response tends to be peaked, the group delay is also peaked. This means that high frequency components of the system tend to be delayed as they pass through the system. This shows up in the step response as a slower risetime and higher overshoot since the high frequency components don't arrive coincident with the edge but instead arrive after the edge has passed.

Most of us have been taught in school that linear phase is the desired phase response. In audio and to some extent, communications systems, this is true. In control systems, this is not true.

*Myth: Linear phase describes the ideal phase response much as flat describes the ideal magnitude response.*

I call this a myth because it fails to consider one point:

*It is impossible to band-limit a response without introducing delay.  
It is impossible to linearize the phase of a band-limited system without adding more delay.*

The delay of the linear phase system is sometimes intolerable. To illustrate the situation, consider the step in Figure 3. The step on the right was generated using a 5<sup>th</sup> order Butterworth filter. A Butterworth filter is an all-pole filter and is therefore infinite impulse response (IIR) as poles mean that the output depends not only on the input but on some internal storage element that theoretically remembers its past history forever. The step on the left is linear phase and looks like it was generated using the  $\text{Sin}(x)/x$  FIR filter. This is not true, however, and I can't help debunking another common myth:

*Myth: FIR filters are linear phase, IIR filters are not.*

Reality is:

*It is easy to design linear phase FIR filters.  
It is not easy to design linear phase IIR filters.*

And the corollary:

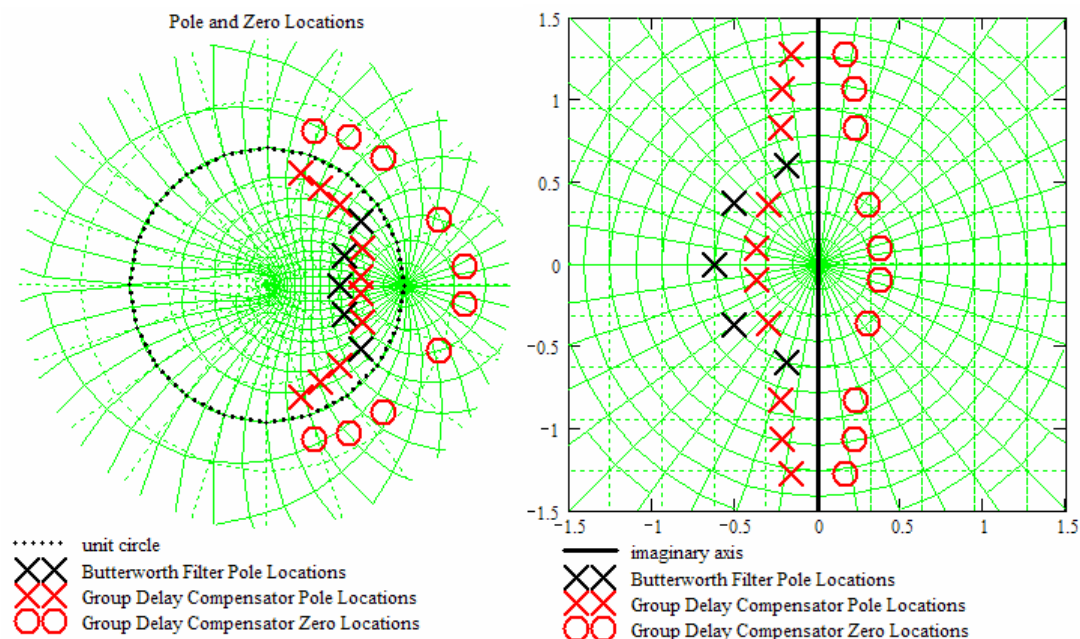
*It is not easy to design minimum phase FIR filters.  
It is easy to design minimum phase IIR filters.*

Of course some of you who are experienced in filter design may quibble with these statements. Forgive me if there is some exaggeration here.

The reality is that most FIR filters designed directly as FIRs are linear phase because you have a choice of what you want to set the phase to, and most often, the choice made is linear phase. Similarly, there is some elegance to a symmetric FIR (one that is symmetric on both sides about some center peak in the impulse response) – the symmetry imposes linear phase. IIR filters are usually designed using analog filter design techniques. Most analog filters are all-pole and will be minimum phase (or certainly not linear phase) unless you do something special.

## Allpass Filters

The step response shown at the left in Figure 3 is actually the 5<sup>th</sup> order Butterworth filter shown to the right with the addition of a phase compensation network designed to linearize the phase.



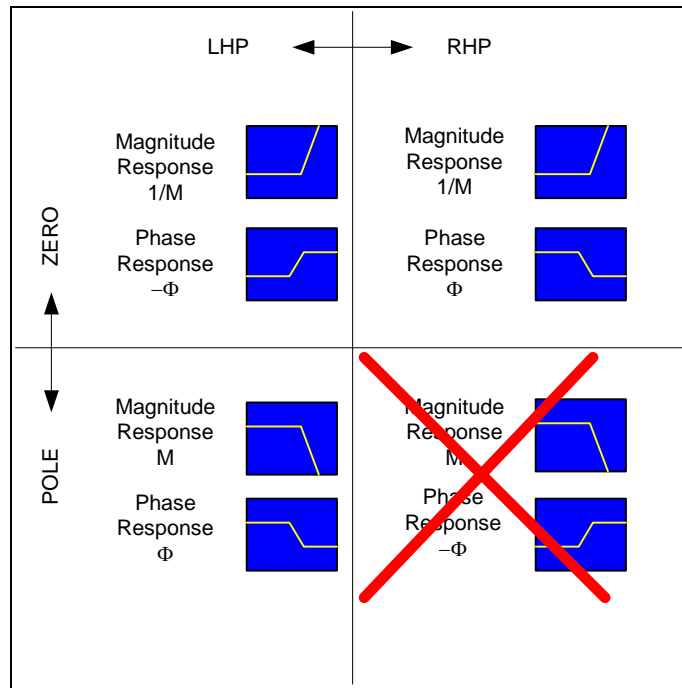
**Figure 4 – Digital and Analog Pole and Zero Locations for the Step Responses in Figure 3**

Figure 4 shows the pole and zero locations of the systems that generate the step responses in Figure 3. The black poles are the poles of the 5<sup>th</sup> order Butterworth filter. The red poles and zeros are the poles and zeros of a phase compensation system. The combination of the red and black poles and zeros form the system that generates the step response on the left in Figure 3. This is an IIR system and has essentially linear phase. It is important to note that both systems have the exact same magnitude response. In other words, the addition of the red poles and zeros in Figure 4 has no effect on the magnitude response.

To understand how this works, consider Figure 5. Figure 5 shows the effect of an individual pole or zero in the left or right half plane (in the  $s$  domain). If you remember from Bode approximations, a pole introduces 3 dB attenuation in magnitude and a 45 degree phase lag at the frequency of the pole. It introduces essentially no magnitude effect and no phase lag far before the pole and essentially 20 dB/decade drop and 90 degrees of phase lag far above the pole frequency. Poles can only exist in the left half plane (LHP), otherwise the system is unstable. A LHP zero has the exact opposite effect of a pole and will cancel a pole if placed directly on top



of it. Right half plane (RHP) zeros are perfectly legal. A RHP zero has the same magnitude response effect of the LHP zero, but opposite phase effect. A RHP zero has the opposite magnitude effect of a pole, but the same phase effect.



**Figure 5 – Characteristics of Poles and Zeros in the Left-Half and Right-Half Plane**

It is this last effect that is often exploited in phase compensation in the design of *allpass* filters. Allpass filters utilize combinations of RHP zeros and LHP poles strategically located to cancel each others magnitude effects. The phase effect, it turns out, is doubled. In other words, a LHP pole coupled with a RHP zero can provide no magnitude response effect, but a 90 degree phase lag at the pole/zero frequency and 180 degrees of phase lag far up in frequency. You can see that the poles and zeros used for phase compensation in Figure 4 are arranged as an allpass filter network. Table 1 and Table 2 outline some digital allpass filter characteristics.

Transfer Function	$H(z) = \frac{-\zeta + z^{-1}}{1 - \zeta \cdot z^{-1}}$
Group Delay	$D\left(x = \cos\left(2 \cdot \pi \cdot \frac{f}{F_s}\right)\right) = \frac{1}{F_s} \cdot \frac{1 - \zeta^2}{1 + \zeta^2 - 2 \cdot \zeta \cdot x}$
Frequency of Maximum Group Delay	$f_{gd\max} = 0$
Zero Frequency Group Delay (and Maximum Group Delay)	$D_0 = \frac{1}{F_s} \cdot \frac{1 - \zeta^2}{1 + \zeta^2 - 2 \cdot \zeta}$

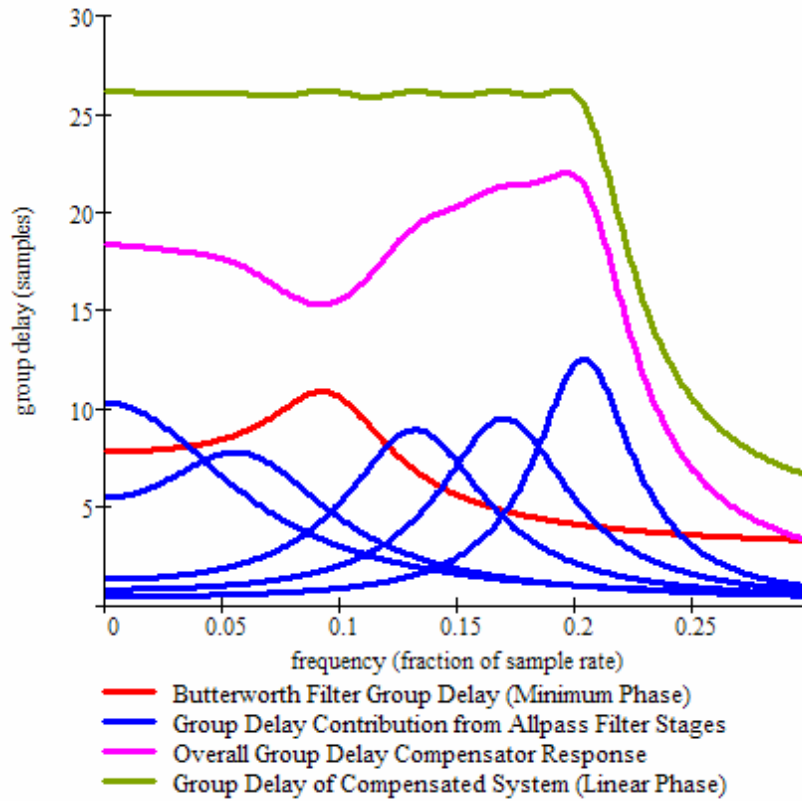
**Table 1 – First Order Allpass Filter Information**

	$\alpha = -2 \cdot \text{Re}(\zeta), \beta =  \zeta ^2$
Transfer Function	$H(z) = \frac{ \zeta ^2 - 2 \cdot \text{Re}(\zeta) \cdot z^{-1} + z^{-2}}{1 - 2 \cdot \text{Re}(\zeta) \cdot z^{-1} +  \zeta ^2 \cdot z^{-2}}$
Group Delay	$D\left(x = \cos\left(2 \cdot \pi \cdot \frac{f}{Fs}\right)\right) = \frac{-2}{Fs} \cdot \frac{(-\alpha + \alpha \cdot \beta) \cdot x + (\beta^2 - 1)}{4 \cdot \beta \cdot x^2 + (2 \cdot \alpha \cdot \beta + 2 \cdot \alpha) \cdot x + (1 + \alpha^2 - 2 \cdot \beta + \beta^2)}$
Frequency of Maximum Group Delay	$f_{gd\max} = \frac{Fs}{2 \cdot \pi} \cdot \cos^{-1}\left(\frac{-8 \cdot \beta - 8 \cdot \beta^2 + 4 \cdot \sqrt{4 \cdot \beta^2 + 8 \cdot \beta^3 + 4 \cdot \beta^4 - \alpha^2 \cdot \beta^3 - 6 \cdot \alpha}}{8 \cdot \alpha \cdot \beta}\right)$ $f_{gd\max} \cong \arg(\zeta) \cdot \frac{Fs}{2 \cdot \pi} = \cos^{-1}\left(\frac{-\alpha}{2 \cdot \sqrt{\beta}}\right) \cdot \frac{Fs}{2 \cdot \pi}$
Zero Frequency Group Delay	$D_0 = \frac{-2}{Fs} \cdot \frac{-\alpha + \alpha \cdot \beta + \beta^2 - 1}{2 \cdot \beta + 2 \cdot \alpha \cdot \beta + 2 \cdot \alpha + 1 + \alpha^2 + \beta^2}$

**Table 2 – Second Order Allpass Filter Information**

Group delay correction is performed by strategically cascading multiple allpass filter sections. Figure 6 shows how this is done. In Figure 6, the red trace shows the group delay characteristic of the 5<sup>th</sup> order Butterworth filter. The blue traces show the group delay effects of the multiple allpass filter sections. The magenta trace shows the combined effect of the allpass filter network. Finally, the gold trace shows the final system group delay.

Figure 6 shows some things about the strategy of linearizing phase that are good to understand. First, you see that the all-pole Butterworth response introduces not only delay, but non-linear delay. The only way to correct the non-linear effect of the delay is to introduce a system that delays some frequency components less than others, but it is important to see that this always causes more delay in the system.



**Figure 6 - Group Delay Linearization Strategy**

### Minimum Phase

Going back to Figure 4, and using the understanding that allpass filters provide no magnitude change, you can see that there are an infinite number of combinations of all-pass filter sections that can be added to the 5<sup>th</sup> order all-pole Butterworth filter that will have identical magnitude response, but different group delay response, but all combinations will introduce more overall delay to the signal than the Butterworth filter alone. Plus, the only way to add poles and zeros to the Butterworth response without affecting the magnitude response is to add all-pass sections, which always have RHP zeros (or zeros outside the unit circle in the digital domain). This leads to the definition of the minimum phase system.<sup>6</sup>

*In control theory and signal processing, a linear, time-invariant system is minimum-phase if the system and its inverse are causal and stable.*

From this definition, you can infer that all of the zeros of the minimum phase system must be in the LHP (inside the unit circle) otherwise it is impossible to invert the system since the zeros will turn into poles in the inverse system. It is easy to generate the minimum phase system provided the pole and zero locations of the system. Figure 5 shows that the magnitude effect of a zero does not depend on whether it is in the LHP or RHP, so to generate the minimum phase system, simply move any RHP zeros to the LHP. In the  $s$  domain, this means simply changing the sign of the real part of the zero. In the digital domain, this means taking the complex conjugate of the inverse of the zero location. In Figure 3, you can see that if we do this, all of the RHP zeros

would land on top of the added poles for phase compensation, and the minimum phase system is simply the original 5<sup>th</sup> order Butterworth filter.

This works for all-zero systems as well. If we have the impulse response of a system, we have the filter coefficients of a FIR filter that defines the system:

$$H(z) = a_0 + a_1 \cdot z^{-1} + a_2 \cdot z^{-2} + \dots + a_N \cdot z^{-N} \text{ or } H(z) = \frac{a_0 \cdot z^N + a_1 \cdot z^{N-1} + a_2 \cdot z^{N-2} + \dots + a_N}{z^N}$$

### Equation 3 – Polynomial Representation of the Impulse Response

If we take the first 50 points of the impulse response (the derivative of the step response) of the linear phase step response shown in Figure 3 and think of it as a polynomial in the form of Equation 3, we can find the roots of this polynomial and these are the zeros of the all-zero FIR filter that creates this response. The zero locations are shown in Figure 7. In Figure 7, I've taken the liberty of also plotting the minimum phase zero locations by simply taking the zeros located outside the unit circle and placing them appropriately inside, re-emphasizing that this has no effect on the magnitude response characteristics of the system.

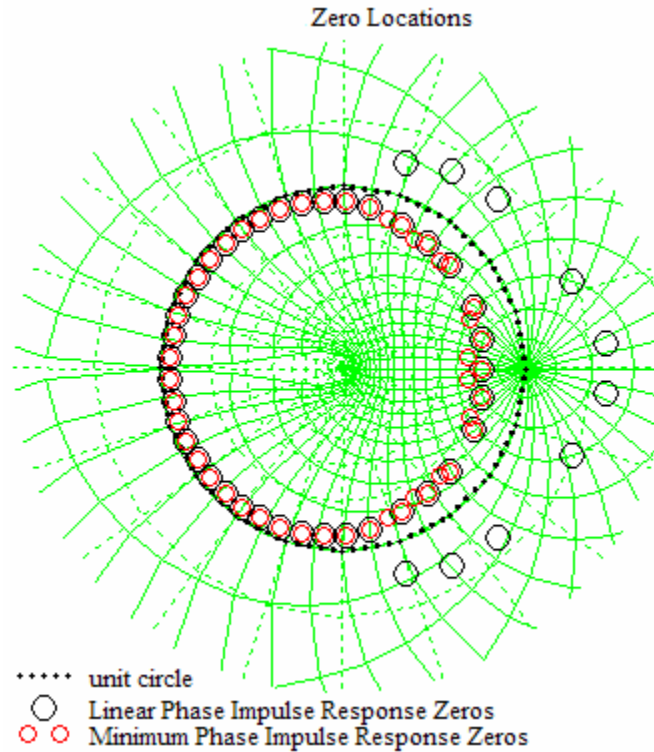
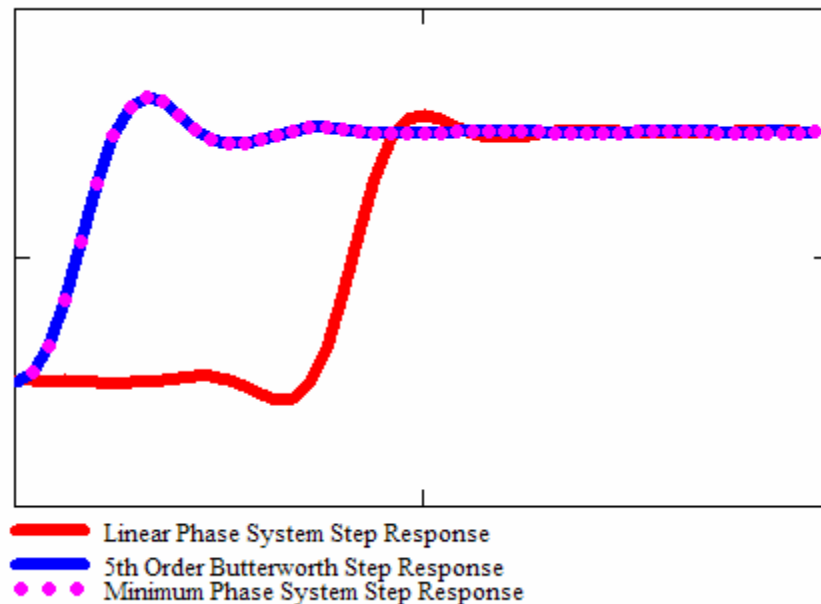


Figure 7 – Zero Locations of a Minimum Phase and Non-minimum Phase All-Zero System

We then take the zeros of the minimum phase system found and put this back in the form of Equation 3 by turning the multitude of roots back into a polynomial. Equation 3 implies that this process has produced the minimum phase system impulse response. Using this as a FIR filter and passing a step through it, we find the step response of the minimum phase system. This is

shown in Figure 8. Figure 8 shows the step responses of the linear phase and Butterworth step response. It shows these responses in the correct time location relative to a stimulus applied at time zero. Not surprisingly, the minimum phase response matches the Butterworth response identically.



**Figure 8 – Step Response Comparison of the Minimum Phase and Linear Phase System**

Figure 8 illustrates several points. First, it exposes the actual relationship between the edge in the step response and the time that the stimulus was applied. In the linear phase case, the response to the step occurs much later than the step. The reason is that the phase linearizing system delayed the low frequency components to allow time for the high frequency components to catch up. The minimum phase step reacts immediately after the application of the stimulus. In Figure 3, the location where the input step occurs is easy to locate for the Butterworth system. It is difficult to locate in the linear phase case – it turns out that the actual step occurred much earlier. This is what leads to the seemingly acausal or anticipatory behavior of the linear phase system. In the scope, when group delay compensation is applied, the delay of the filter network is accounted for in the trigger placement.

Figure 8 also illustrates the fact that the minimum phase system has the step response concentrated back at the location of the actual step. This system has the minimum delay in reacting to the step for the given magnitude response. This is why minimum phase systems are also referred to as minimum delay systems\*.

Aside from the apparent anticipatory behavior of the linear phase system, there are other reasons why linear phase is not the best response and it all boils down to the extra delay. Linear phase systems are systems where all of the frequency components arrive at the output simultaneously

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\* The word *minimum phase* is not accurate. It should be referred to as minimum phase lag systems as they add the minimum amount of phase lag.

but with additional delay. The simultaneity of the components is desirable from a distortion standpoint and is clearly desirable in an audio system where the added delay is unimportant. In communications systems the added delay may or may not be important. In control systems, however, the added delay is deadly and the minimum phase system is clearly desirable from a controllability aspect.

## Generating Minimum Phase Responses

I mentioned in the previous section that it is easy to generate minimum phase systems provided the pole and zero locations of the system – simply move the zeros to the LHP (or inside the unit circle). A general problem is that usually you don't know the pole and zero locations. All classical IIR filters are minimum phase (Butterworth, Chebyshev, etc.) because they are all-pole filters, which is why I stated that minimum phase IIR design is easy. In FIR filter design, a problem presents itself in that FIR filters are generally designed by directly determining the impulse response. This is analogous to working with a polynomial, as pointed out by Equation 3. The way to generate a minimum phase design is to design the FIR filter, find the roots of the polynomial describing the impulse response, move the outlying zeros inside the unit circle, and re-inflate the polynomial. Root finding is in itself problematic, but many methods exist<sup>7</sup>. The problem is that these methods only work well for reasonably sized polynomials. My experience is that MathCAD begins having trouble around 30<sup>th</sup> order. My own code that I've built out of numerous methods has trouble around 50<sup>th</sup> order. As a general statement, I can say that finding the roots of a 200 tap FIR is not a good thing to do.

In looking for ways to do this, I found many papers that attempted to address root finding directly and managed to stumble across a really great recipe<sup>8</sup>. This recipe is an algorithm that converts any FIR filter into a minimum phase FIR in a very simple manner. This algorithm is shown in Equation 4. You'll have to read the reference cited to understand how the algorithm works, but suffice it to say, it produces the exact same impulse response that generates the minimum phase step response as shown in Figure 8 without the effort of root finding.

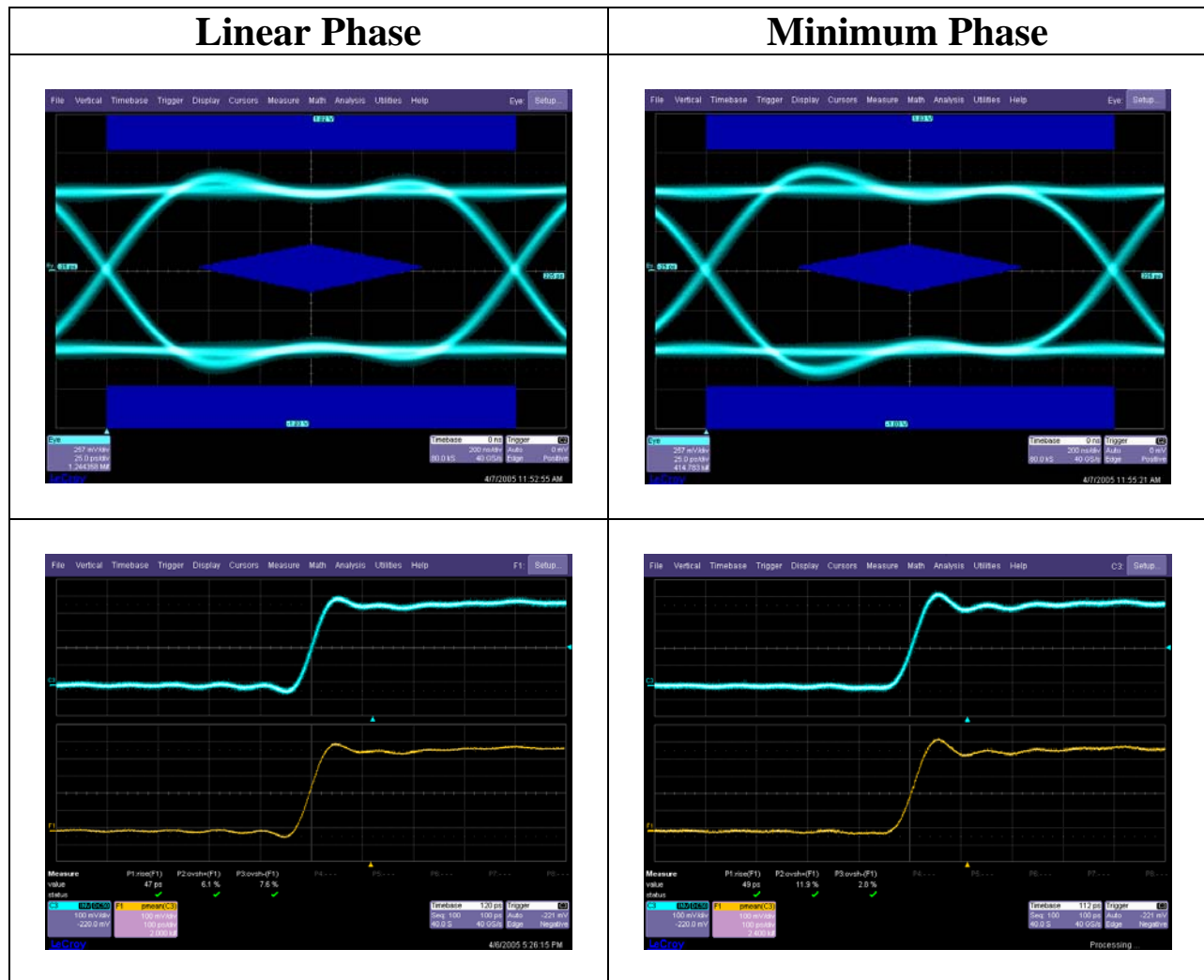
$$\begin{aligned}
& \text{ir}_{\min}(\text{ir}) := \begin{aligned} & K \leftarrow \text{last}(\text{ir}) + 1 \\ & \text{for } k \in 0..K-1 \\ & \quad s_k \leftarrow \begin{cases} 0 & \text{if } (k=0) \vee \left(k = \frac{K}{2}\right) \\ 1 & \text{if } (0 < k) \wedge \left(k < \frac{K}{2}\right) \\ -1 & \text{if } \left(\frac{K}{2} < k\right) \wedge (k < K) \end{cases} \\ & H \leftarrow \text{CFFT}(\text{ir}) \cdot K \\ & \theta \leftarrow -1j \cdot \text{CFFT}\left(\overrightarrow{(s \cdot \text{ICFFT}(\ln(|H|)))}\right) \\ & N \leftarrow \frac{K}{2} \\ & \text{for } n \in 0..N \\ & \quad H_{\min_n} \leftarrow |H_n| \cdot e^{1j \cdot \theta_n} \\ & \text{for } nn \in 1..N-1 \\ & \quad H_{\min_{N+nn}} \leftarrow \overline{H_{\min_{N-nn}}} \\ & \text{ir}_{\min} \leftarrow \frac{(\text{Re}(\text{ICFFT}(H_{\min})))}{K} \\ & \text{ir}_{\min} \end{aligned}
\end{aligned}$$

**Equation 4 – Method of converting a FIR into a minimum phase FIR**

## Minimum vs. Linear Phase in Scopes

At this point in your reading, you will understand the difference between minimum phase and linear phase responses. LeCroy has always strived to provide systems with minimum phase responses because the response is more natural and avoids the causality issue. Our competition strives to provide linear phase responses, assumedly because they believe this is the best response. It's easy to state that linear phase is the ideal response.

My opinion is that the minimum phase response is the best response that a band-limited system deserves. That being said, a good argument can be made for linear phase when viewing serial data signals and eye patterns. The reason for this is illustrated in Figure 9. Figure 9 shows the response of the LeCroy SDA 11000 Serial Data Analyzer to both a 30 ps step and a 5 Gb/s serial data signal. You will notice that the apparent acausal step response of the linear phase system translates to a symmetric eye pattern. The more natural step response of the minimum phase system translates into a slightly asymmetric eye pattern. Note that the mask used for the eye pattern test is not designed to deal with any asymmetry – a typical situation in standards compliance testing. It is for these reasons that LeCroy scope users now have a choice of minimum phase and linear phase response.



**Figure 9 – Comparison of 5 Gb/s Serial Data Signal Acquired By LeCroy SDA 11000 With Minimum Phase and Linear Phase Response**

In summary, linear phase might be a better response characteristic for measurements involving eye-patterns. Also, where some tests specify scope magnitude response characteristics, either by bandwidth, flatness or some specific magnitude characteristic, they typically don't specify phase response. It turns out that the phase or group delay characteristic is very difficult to understand from a frequency domain perspective, *unless the phase is linear*. In other words, it's easy to predict the time domain response of a linear phase system and it is difficult to say anything about the time domain response of a system whose phase is not linear (without simply looking at the step or impulse response).

## Group Delay Effects on Serial Data

### IRIS

Last year, I introduced a simulation tool called IRIS. It is a tool created by LeCroy for the purpose of understanding how scope characteristics interact with serial data characteristics. Scope characteristics are interchangeable with channel characteristics with regard to what you



will see on the scope screen. Therefore, the IRIS tool is useful for examining the effects of serial data channel characteristics on the eye pattern, as well.

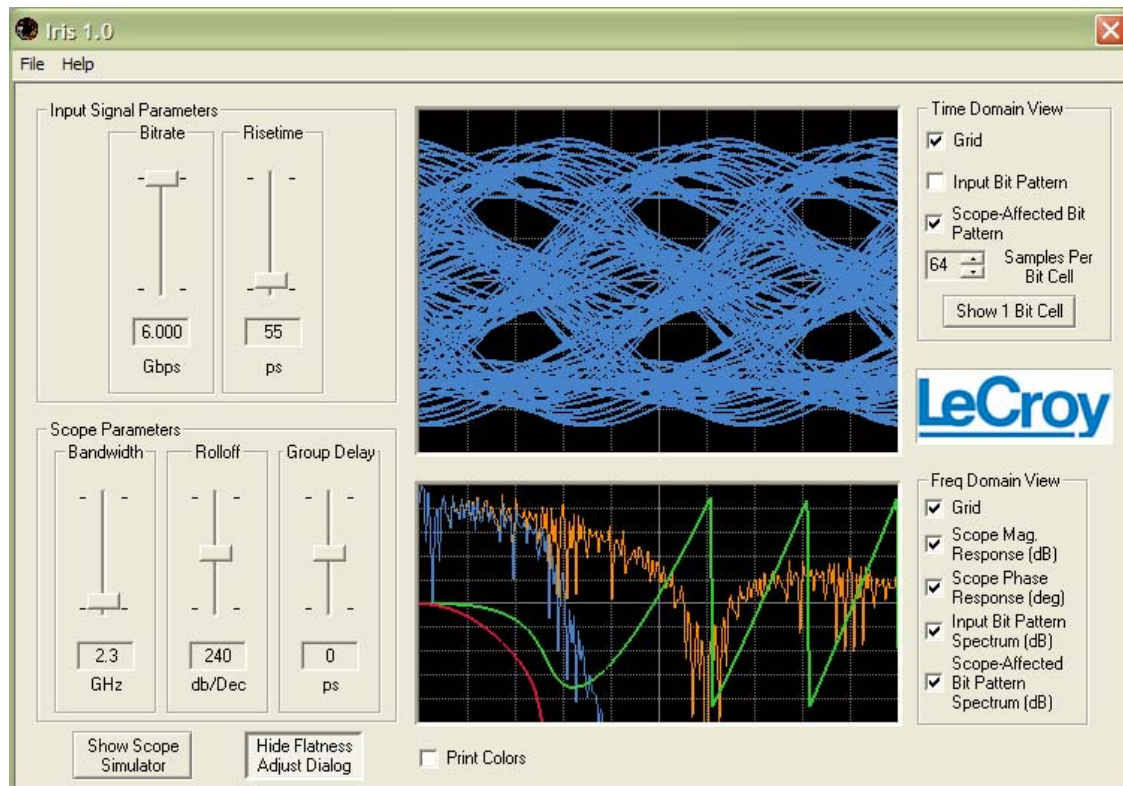
Figure 10 shows a 6 Gb/s serial data signal that is passed through a channel with 2.3 GHz of bandwidth and a 240 dB/decade drop in response after the bandwidth point. The eye is closed. The response of the channel is assumed to be an all-pole response.

Figure 11 shows the effect of providing some coarse compensation to the group delay. You can see that the eye opens and becomes distinguishable.

Figure 12 shows the effect of combining the group delay compensation with some coarse magnitude response equalization. You can see that the eye opens further.

Finally, Figure 13 shows the effect of the magnitude compensation only. It is clear that in this case, the group delay compensation has a far greater effect than the magnitude compensation.

I am no expert on serial data equalization and the equalization of both group delay and magnitude was performed in a rather crude manner in this case, but it does illustrate that group delay is also important in the equalization of serial data channels.



**Figure 10 – Serial Data Eye Pattern**

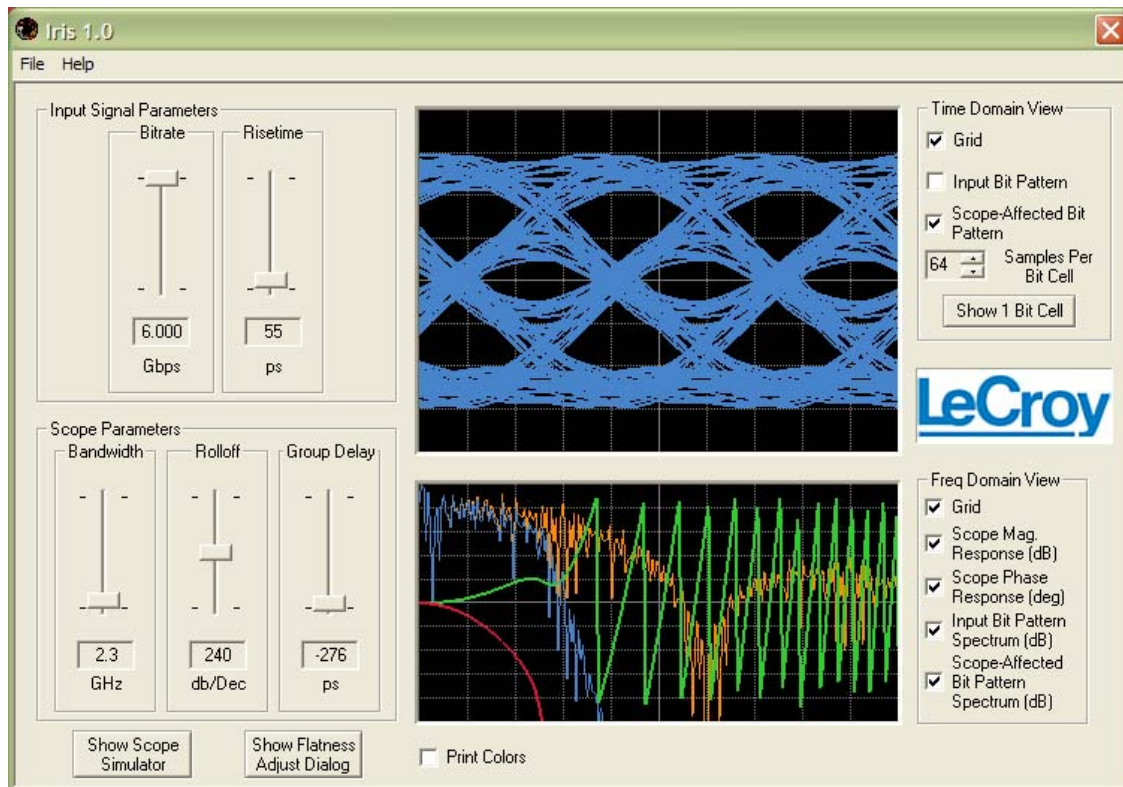


Figure 11 – Serial Data Eye Pattern with Group Delay Equalization

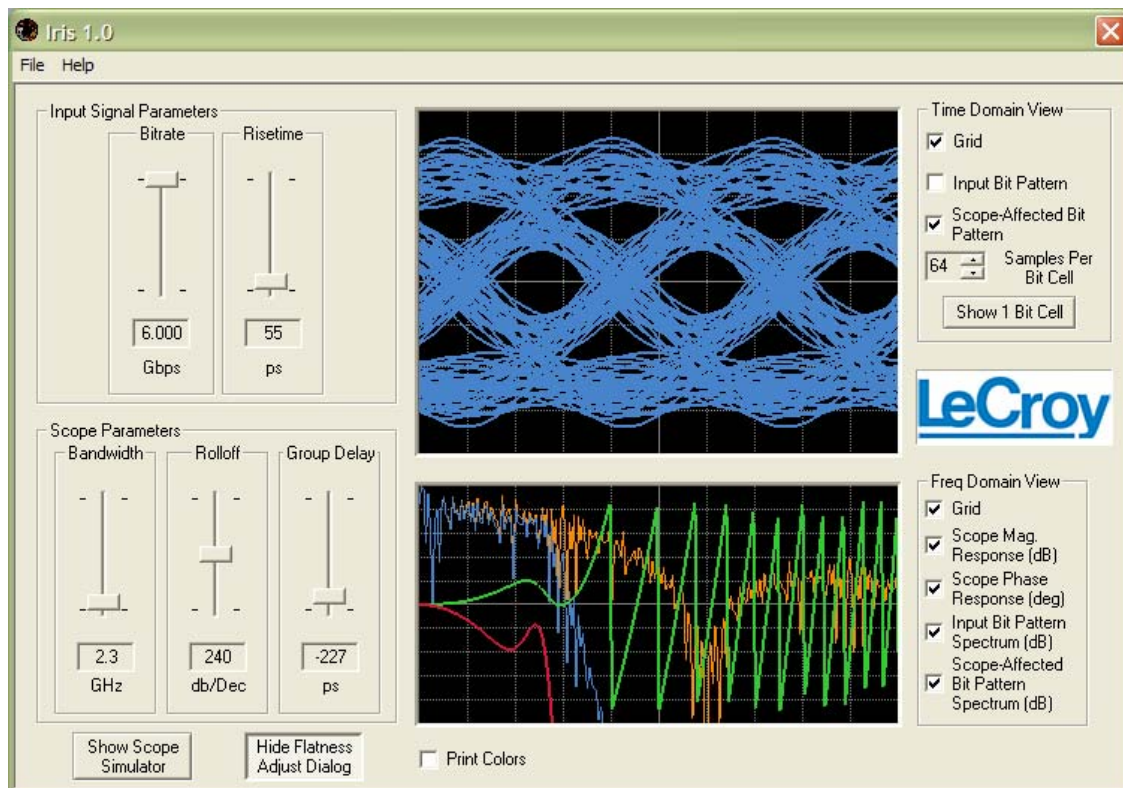
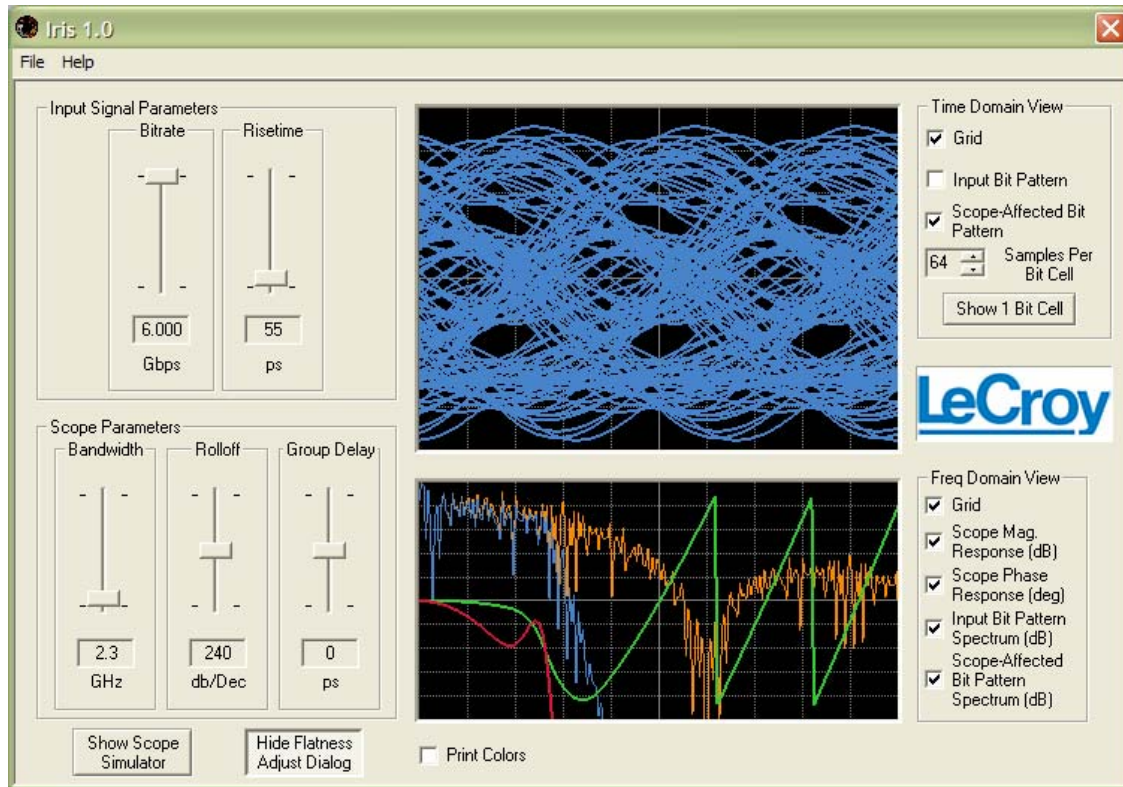


Figure 12 – Serial Data Eye Pattern with Group Delay and Magnitude Equalization



**Figure 13 – Serial Data Eye Pattern with Magnitude Equalization Only**

### Simplified View of Group Delay Effects

Group delay effects can be simply understood by reusing the concept of dispersion penalty, but applying it to phase instead of magnitude. Dispersion penalty is defined as the ratio of the amplitudes of the highest and lowest frequency signals on a transmission line.<sup>9</sup> This definition considers only the magnitude response of the fundamental component of the shortest and longest run-length pattern.

To visualize the effect of group delay, I make a very simple model illustrated by Figure 14. I create all of the 16 four bit possibilities.\* We will view the resultant eye pattern in the second bit position. Then, I group the possibilities into three sets. The first set contains the 101010... patterns which, because of the limited bandwidth of the system model, provide only the fundamental at the highest possible frequency. The second set contains the 11001100... patterns which consists of odd harmonics only. In this model, the third harmonic is removed due to limited bandwidth and again, only the fundamental is present. The third set contains the isolated bit patterns – all of the other patterns possible. These patterns contain even and odd harmonics. Because of the limited system bandwidth, only the 2<sup>nd</sup> harmonic is present. This 2<sup>nd</sup> harmonic appears at the same location as the fundamental of the 101010... pattern.

Figure 15 shows a MathCAD worksheet used to generate the bit patterns, measure the frequency content, band limit the content, and allow crude modeling of the effects of group delay by allowing variation of the phase of the highest frequency component.

\* Although I ignore the two DC level cases

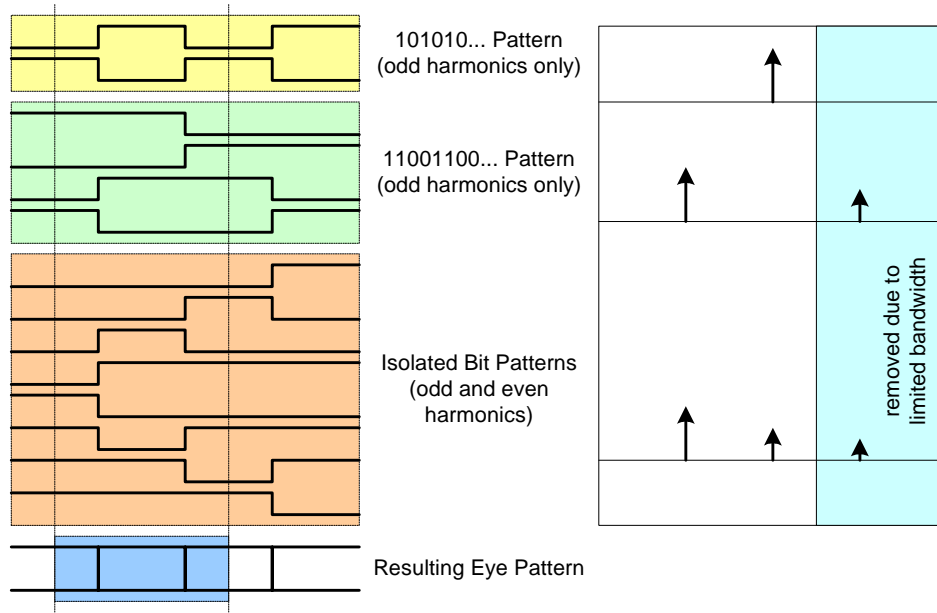


Figure 14 – Simplified Model for Viewing Group Delay Effects

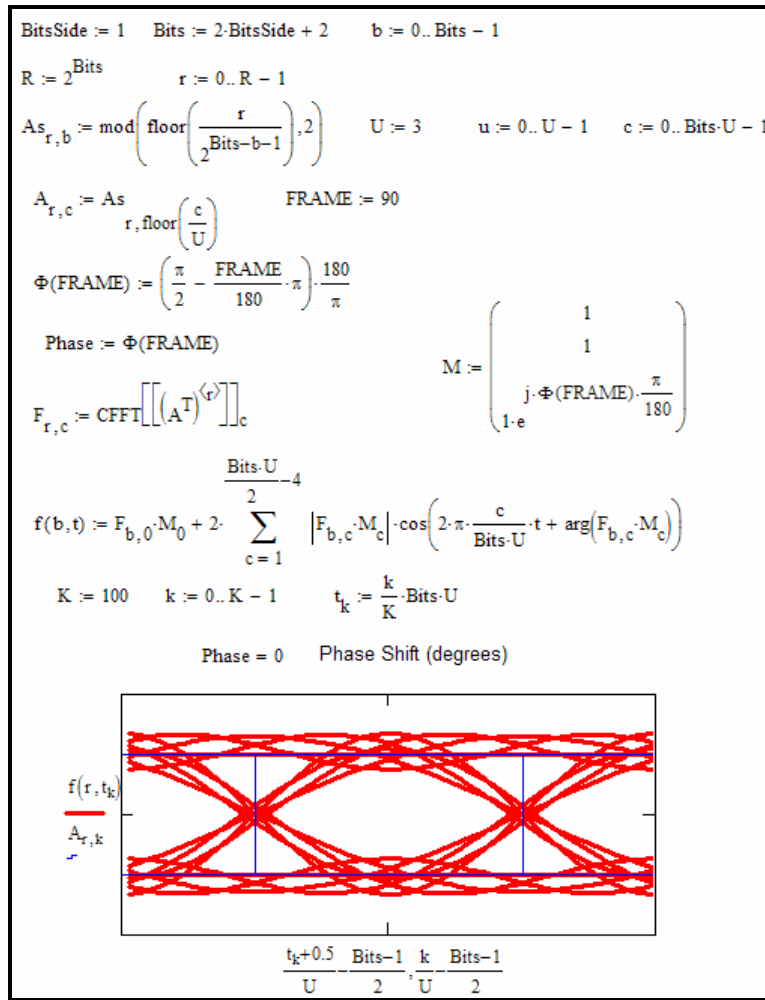
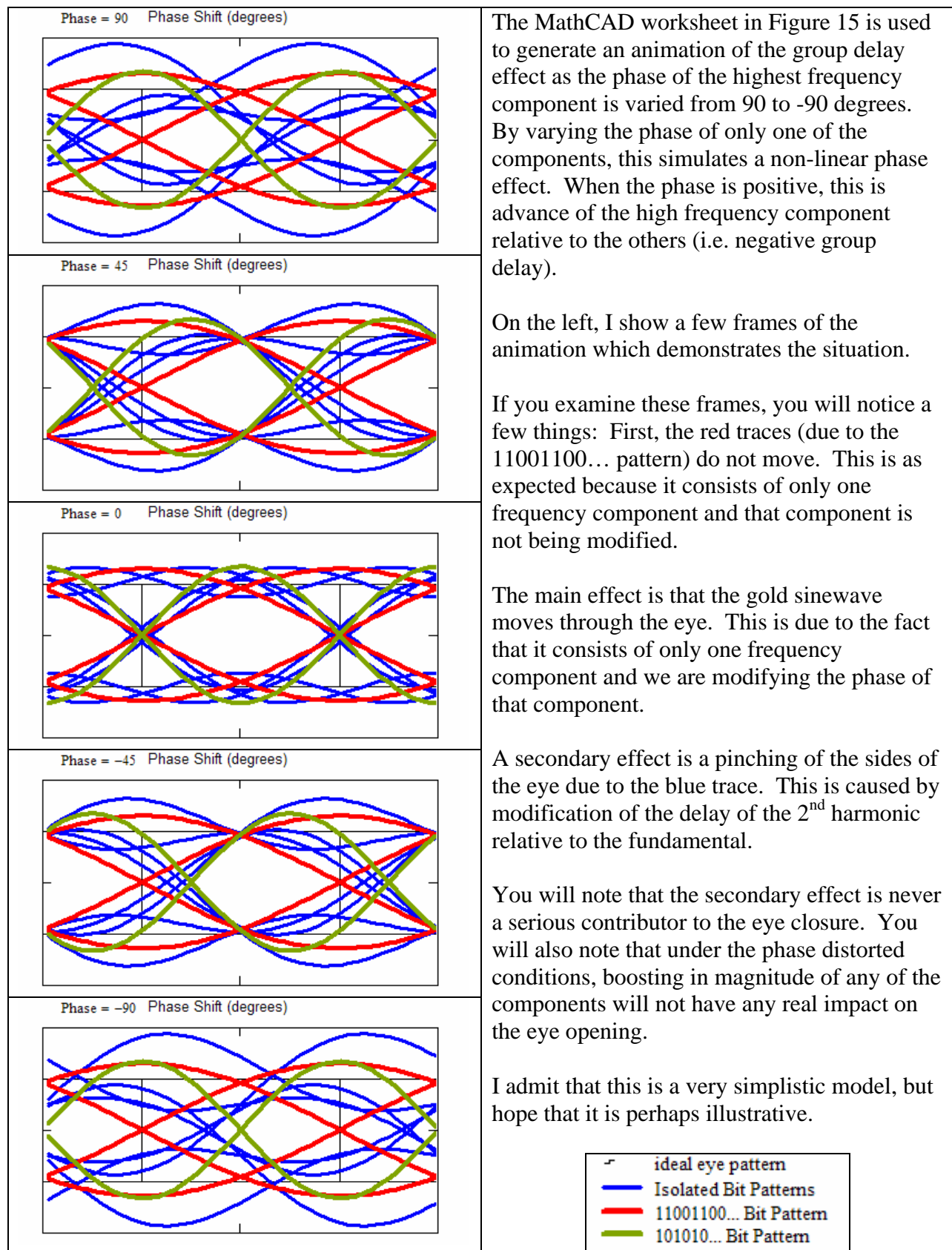


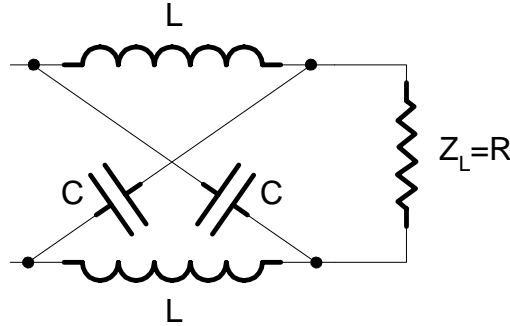
Figure 15 – MathCAD Worksheet for Group Delay Effect Model





## Group Delay Equalization

Group delay equalization can be accomplished by cascading sections of all-pass filters. The all-pass filter has already been described in detail from a pole-zero location standpoint and from a digital implementation standpoint. In the analog domain, the all-pass filter is implemented using sections of the following form shown in Figure 16<sup>10</sup>. This is a lattice filter and can be used in this form or converted to other forms.



**Figure 16 – A first order all-pass filter section**

The easiest way to understand the filter section shown in Figure 16 is look at the low and high frequency behavior. At very low frequencies, the inductor is a short, the capacitor is an open, and the load can be viewed as directly connected to the input. At high frequencies, the inductor is an open and the capacitor is a short and the load sees the input inverted.

The impedance seen looking into this section is:

$$Z_{in} = \frac{s^2 \cdot L \cdot C \cdot Z_L + s \cdot 2 \cdot L + Z_L}{s^2 \cdot L \cdot C + s \cdot 2 \cdot C \cdot Z_L + 1}$$

**Equation 5**

And the transfer function is:

$$\frac{V_{out}}{V_{in}} = \frac{-s^2 \cdot L \cdot C \cdot Z_L + Z_L}{s^2 \cdot L \cdot C \cdot Z_L + s \cdot 2 \cdot L + Z_L}$$

**Equation 6**

In Equation 5, if we set  $Z_{in} = Z_L = R$ , a resistance (like 50 Ohms), we get:

$$L = R^2 \cdot C$$

**Equation 7**

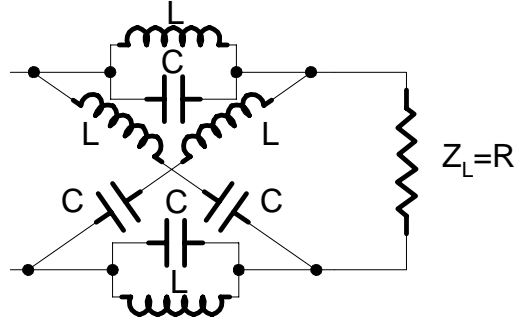
Equation 7 defines the fixed relationship of the inductance to the capacitance in a system with given impedance. When Equation 7 is substituted into Equation 6 and simplified, Equation 8 is obtained.

$$\frac{V_{out}}{V_{in}} = -\frac{s - \omega_0}{s + \omega_0} \text{ where } \omega_0 = \frac{1}{\sqrt{L \cdot C}}$$

**Equation 8 – Transfer Function of First Order Allpass Filter Section**

This is a first order section and consists of only a real pole and a real zero. This is often usable for rough equalization of group delay.

Figure 17 shows a second order section:



**Figure 17 – A second order allpass filter section**

The network in Figure 17 has a transfer function given by

$$C = \frac{1}{\omega_0^2}, L = R^2 \cdot C, R = \sqrt{\frac{L}{C}}, \omega_0 = \frac{1}{\sqrt{L \cdot C}}, Q = \frac{L}{R \cdot \sqrt{L \cdot C}}$$

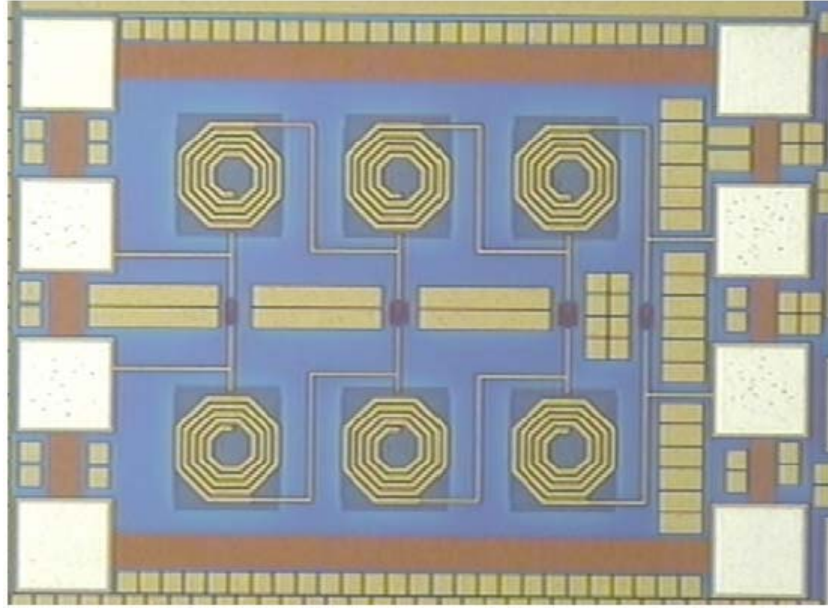
$$H(s) = \frac{s^2 - \frac{\omega_0}{Q} \cdot s + \omega_0^2}{s^2 + \frac{\omega_0}{Q} \cdot s + \omega_0^2}$$

**Equation 9 – Transfer Function of Second Order Allpass Filter Section**

As I understand it, the filter in Figure 17 is a filter whose center frequency can be modified, but not the Q. To perform full group delay compensation, you would need a section whose Q can be varied. I don't know how to do this without adding loss to the network. (Maybe a reader who knows the answer to this would let me know).

### Group Delay Equalizer Realization

Figure 18 is a picture of a passive delay line implemented in a chip.<sup>11</sup> While it is designed to implement a delay line, it illustrates how a group delay equalizer would be implemented (the group delay equalization network can be implemented by crossing the capacitors in the delay line structure). Figure 18 shows the spiral inductors needed. The referenced dissertation goes into great detail in how the inductors and capacitors are modeled and the associated parasitic effects.



**Figure 18 – Die Photo of Passive Delay Line**  
(Courtesy of Georgia Electronic Design Center, Georgia Institute of Technology)

## Summary

I hope I've shed some light on the phase characteristics of scopes and provided some insight into how the scope responses differ. I hope also that you consider phase response in your testing. I also hope you see that phase characteristics are important in serial data transmission and probably need compensation as much as magnitude characteristics.

## References and Credits

While not authors of this paper and not responsible for any errors or omissions I may have made, I have to give credit to some of the LeCroy engineers who have provided me some insight, help, and have patiently endured my interruptions. They are: Bob Talambiras, Dr. Steve Ems and Dr. Jim Mueller.

<sup>1</sup> Peter J. Pupalaikis and Francois LaMarche, Digital Group Delay Compensator, US Patent Appliation 10/678,374, (10/7/2002)

<sup>2</sup> See last years paper – Peter J. Pupalaiks and Eric Yudin, *Eye Patterns in Scopes*, DesignCon 2005

<sup>3</sup> Richard J. Higgins, *Digital Signal Processing in VLSI*, (Prentice Hall, Englewood Cliffs, NJ, 1990), 186-188

<sup>4</sup> T.W. Parks and C.S. Burrus, *Digital Filter Design*, (John Wiley & Sons, 1987), 162-171

<sup>5</sup> Smith, J.O. *Introduction to Digital Filters*, May 2004 Draft, <http://ccrma.stanford.edu/~jos/filters/>.

<sup>6</sup> Wikipedia, [http://en.wikipedia.org/wiki/Minimum\\_phase](http://en.wikipedia.org/wiki/Minimum_phase)

<sup>7</sup> William H. Press, Saul A. Teukolsky, William T. Vetterling, Brian P. Flannery, *Numerical Recipes in C*, Second Edition, (Cambridge University Press, 1992), Section 9

<sup>8</sup> Niranjana Damara-Venkata, Brian L. Evans and Shawn R. McCaslin, *Design of Optimal Minimum Phase FIR Filters Using Discrete Hilbert Transforms*, IEEE Transactions on Signal Processing, Volume 48, Issue 5, May 2000, 1491-1495

<sup>9</sup> Howard Johnson and Martin Graham, *High-Speed Signal Propagation*, (Prentice Hall, 2003), 233

<sup>10</sup> Walter C. Johnson, *Transmission Lines and Networks*, (McGraw Hill, 1950), 325

<sup>11</sup> Soumya Chandramouli, *A Broadband Passive Delay Line Sructure in 0.18 Micron CMOS for a Gigabit Feed Forward Equalizer*, (Georgia Institute of Technology, December 2004)