Introduction

THE NEED

In today's complex environment, data is frequently composed of a mixture of analog and digital components spread over a broad range of frequencies. In many applications, the relevant data is encoded or obscured. Capturing the right signals becomes a challenge. Engineers find it increasingly difficulty to examine only those parts of the data they are interested in. Traditional (or even smart) oscilloscope triggering cannot always provide a satisfactory answer.

For example, servo motors from disk drives add a low frequency component to the high frequency data output. It is hard to achieve an accurate analysis of data unless the low component is removed.

Another common example is switched power supply units, which inject the switching frequency component into many system parts. Viewing digital signals mixed with this switching frequency component could be very difficult. Filtering is definitely required.

Yet another example is in ADSL residential connectivity, where data is transmitted over 256 narrow bands. Each band is only 4.7kHz wide, and the gap between two adjacent bands is also 4.7kHz. Examining such complex waveforms with regular DSOs is almost impossible; filtering out unwanted frequency components is necessary.

THE SOLUTION

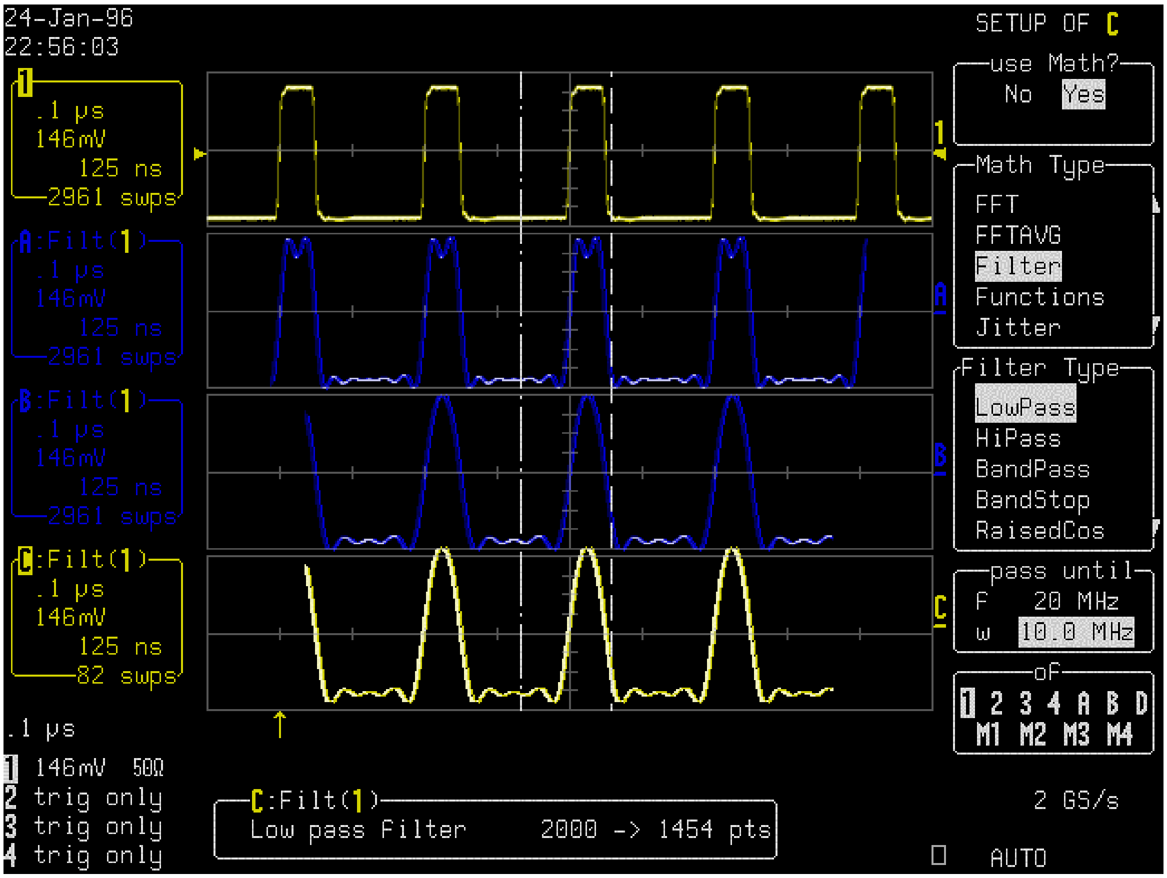
At present, these needs are addressed in two ways. One way is building analog filters and placing them in front of the oscilloscope, providing an already filtered signal to the DSO. The disadvantages of this approach are many. Analog filters depend heavily on the accuracy and stability of analog components. Although in some cases analog filters are easily implemented, they are quite impractical for low (< 100 HZ) or high (> 100 MHz) frequency ranges. In comparison, digital filters can provide the desired results in those cases.

The second approach, practiced by many engineers, is using the DSO as a digitizer. The digitized data output is then transferred to a PC for processing. This solution frequently provides the required results, but it might be too slow or too limited in flexibility for some applications.

With the Digital Filter Package, LeCroy provides a solution that combines the best of both worlds. This package includes seven of the most useful filter types, in addition to a custom design feature. You can easily set the edge (or corner) frequency in addition to the transition region width for each filter. It is possible to use single filters or multiple (up to four) filters cascaded for even more complex filtering. Once filtered, waveforms include mostly relevant frequency components, undesired parts being greatly attenuated.

If filters with special characteristics are desired, the custom design feature allows you to design unique filters tailored to your specific needs. The required filter can be designed with a digital filter design or math package such as MATLAB® or Mathcad®. Filter coefficients can be directly downloaded from the program into the scope, using the DSOFilter utility. It is also possible to specify the filter coefficients on an Excel spreadsheet and to use DSOFilter to download them from the spreadsheet to the scope.

DFP's flexibility and ease of operation are demonstrated by the following example:



1: a signal composed of 5 MHz pulses is filtered by:

A: 40 MHz low-pass filter with a 20 MHz transition region width

B: 30 MHz low-pass filter with a 15 MHz transition region width

C: 20 MHz low-pass filter with a 10 MHz transition region width

Enhanced Solutions

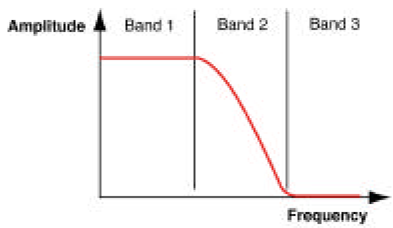
DFP can be coupled with other LeCroy software products — such as AORM, JTA, PMA1, or DDA — to enhance the capabilities of these products and to provide improved solutions.

For example

- Jitter Measurement: the DFP Band-pass Filter can be coupled with the JTA package to measure jitter over a narrow frequency range.
- Optical Recording: In some cases equalization with a different response is required. In such a case, the DFP custom filter feature can be used in conjunction with AORM. The required custom filter can be easily implemented to provide the necessary equalization, while all other AORM functions remain unchanged.
- Power Measurement: A Band-stop filter can be coupled with the PMA1 package to eliminate the switching power supply frequency component from power lines. The Band-stop filter can be tuned to match a specific power supply switching characteristic.

FILTER TYPES[1]

Low-pass Filter



Low-pass filters are useful for eliminating accumulated high frequency noise and interference, and for canceling high frequency background noise.

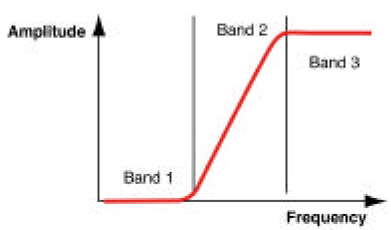
Sample applications are in datacom, telecommunications, and disk drive and optical recording analysis for accurate RF signal detection.

Band 1: Pass Band — DC to top of the transition region; signal passes unattenuated.

Band 2: Transition Region — edge frequency to edge frequency plus width; increasing attenuation.

Band 3: Stop Band — above end of transition region; signal is highly attenuated.

High-pass Filter



High-pass filters are useful for eliminating DC and low frequency components. Sample applications include Disk Drive and Optical Recording analysis (emulation of the SLICING function).

Band 1: Stop Band — DC to bottom of the transition region; highly attenuated.

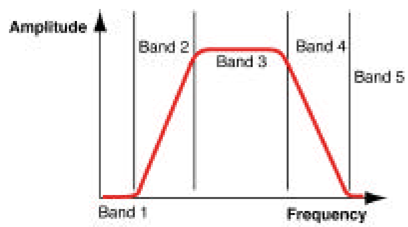
Band 2: Transition Region — edge frequency minus width to edge frequency; decreasing attenuation.

Band 3: Pass Band — above edge frequency; signal passes unattenuated.

1. Filters are optimal FIR filters of less than 1000 taps, according to the Parks-MacLellan algorithm described in *Digital Filter Design and Implementation* by Parks and Burros, John Wiley & Sons, Inc., 1987

LeCroy Digital Filter Package

Band-pass Filter



Band-pass filters are useful for emphasizing a selected frequency band. Sample applications include radio channel identification, broadband transmission, ADSL, clock generators (i.e., eliminating the central frequency and displaying harmonics only), and telecommunications (Jitter measurement over a selected frequency range).

Band 1: First Stop Band — DC to bottom of first transition region; highly attenuated.

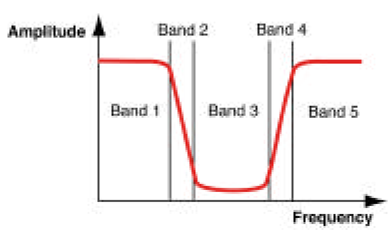
Band 2: First Transition Region — lower corner minus width to lower corner; decreasing attenuation.

Band 3: Pass Band — signal passes unattenuated.

Band 4: Second Transition Region — upper corner to upper corner plus width; increasing attenuation.

Band 5: Second Stop Band — signal highly attenuated.

Band-stop Filter



Band-stop filters are useful for eliminating a narrow band of frequencies. Sample applications include medical equipment, such as ECG monitors where the dominant ripple at 50/60 Hz is rejected, leaving the low energy biological signals intact. Digital troubleshooting: the inherent frequency of the switched power supply is blocked, revealing power line voltage drops and glitches caused by the system clock generator.

Band 1: First Pass Band — DC to bottom of first transition region; signal passes unattenuated.

Band 2: First Transition Region — lower corner minus width to lower corner; increasing attenuation.

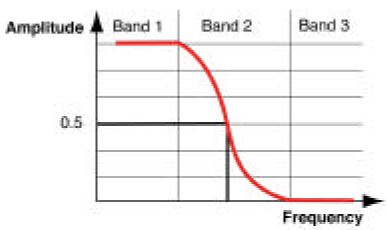
Band 3: Stop Band — signal is highly attenuated.

Band 4: Second Transition Region — upper corner to upper corner plus width; decreasing attenuation.

Band 5: Second Pass Band — signal passes unattenuated.

COMMUNICATION CHANNEL FILTERS

Raised Cosine (a low-pass filter)



These filters belong to the low-pass filter category (with a variety of shapes). Raised cosine is one of a class of filters used to minimize intersymbol interference: the time domain impulse response crosses zero at all bit time intervals except the one with the impulse.

Applying raised root cosine twice (or at the sending and receiving end of a signal, for example) results in a raised cosine filter effect. Sample applications include wireless cellular communications such as WCDMA, datacom, telecommunications, disk drive and optical drive analysis.

Band 1: Pass Band — DC to corner frequency minus half width; signal passes unattenuated.

Band 2: Transition Region — corner minus half width to corner plus half width; attenuation increases with frequency with a rolloff shape of *0.5cos(a) + 0.5*, where *a* ranges from 0 to ? over the transition region. This region is determined by ?, which is specified as a percentage of the corner frequency.

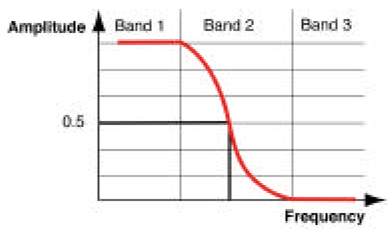
Band 3: Stop Band — above corner frequency plus half width; highly attenuated.

The impulse function for the raised cosine filter is:

$$h(t) = \frac{\left[\dfrac{\sin\left(\pi \dfrac{t}{T_s}\right)}{\pi \dfrac{t}{T_s}}\right] \cos\left(\pi \beta \dfrac{t}{T_s}\right)}{1 - \left(2 \beta \dfrac{t}{T_s}\right)^2}$$

LeCroy Digital Filter Package

Raised Root Cosine (a low-pass filter)



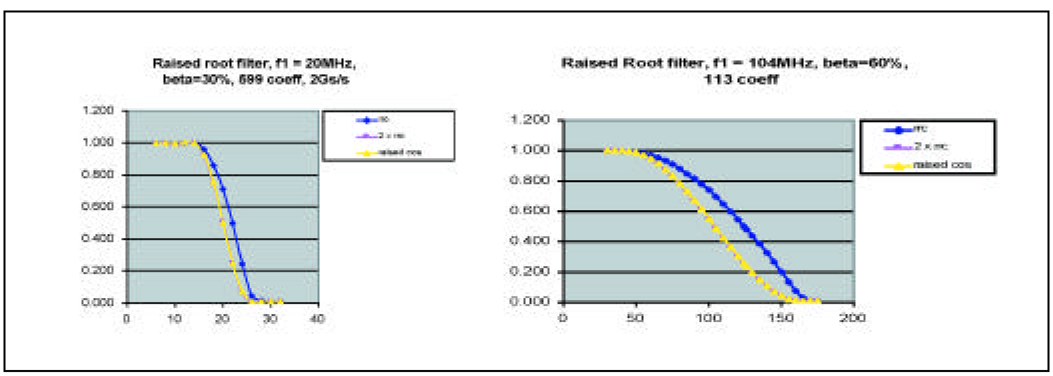
Band 1: Pass Band — DC to corner frequency minus half width; signal passes unattenuated.

Band 2: Transition Region — corner minus half width to corner plus half width; attenuation increases with frequency with a rolloff shape of *0.5[cos(a) + 0.5]*^{½}, where *a* ranges from 0 to ? over the transition region. This region is determined by ?, which is specified as a percentage of the corner frequency.

Band 3: Stop Band: — above corner frequency plus half width; signal is highly attenuated.

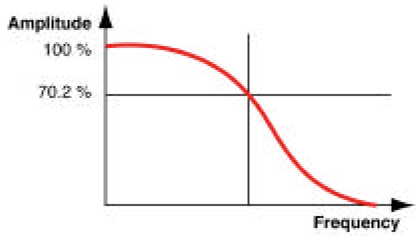
The impulse function for the square-root raised cosine filter is:

$$h(t) = \frac{4\beta}{\pi\sqrt{T_s}} \; \frac{\cos\left((1+\beta)\pi\frac{t}{T_s}\right) + \dfrac{\sin\left[(1-\beta)\pi\frac{t}{T_s}\right]}{4\beta\frac{t}{T_s}}}{1-\left(4\beta\frac{t}{T_s}\right)^2}$$



Illustrated above are two raised root cosine filters with different beta. The diagrams show that applying this filter twice results in a raised cosine response.

Gaussian



Band 1: Pass Band — DC to half power bandwidth% times modulation frequency, pass; 3 dB down at half power bandwidth.

The shape of a Gaussian filter's frequency response is a Gaussian distribution centered at DC. The signal becomes more attenuated with increasing frequency. It is not possible to specify a transition region or a stop band for Gaussian filters. However, the BT value, a fraction of the symbol frequency, determines the filter's width, where:

B = half power bandwidth

T = bit (or modulation period)

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