# Comparing LTE Channelizers Implemented with Linear Phase Recursive Filters and FIR Filters

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## ABSTRACT

Cell sites repeaters may receive a composite signal containing a mix of LTE channels with bandwidths of 5, 10, 15, and 20 MHz and be required to rearrange the frequency plan of the channels or to drop and insert specific channels prior to transmitting the altered composite signal. The straight forward approach to this task is to down-convert, and down-sample each channel in the mix and then up-sample and up-convert and merge the new traffic mix. The filters applied to the up and down conversion task as well as the up and down sampling task would likely be linear phase FIR filters because of the ease with which the resampling task can be embedded in the filtering task. We present an alternate filter structure formed from linear phase recursive filters and compare their performance and computational complexity with their FIR filter counterparts. We will show that the recursive filter version of the channel extractor requires significantly few arithmetic operations and actually outperforms the non-recursive version as demonstrated by the error vector magnitude (EVM) of the two options.

## 1. INTRODUCTION

The most traditional approach to extracting and inserting a single channel from a composite signal containing a group of narrowband signals is shown in figure 1. The input half band Hilbert transform filter reduces the bandwidth by a factor of 2 and reduces the sample rate by the same factor of 2 as it converts the real input data stream into a complex output data stream. The center frequency of the desired channel aliases to a new center frequency due to the down sampling, but since we knew its original center frequency we know the output center frequency. The spectra at the input to the system has a known bandwidth and known center frequency with the spectrum confined to a specified frequency span. We use a Direct Digital Synthesizer (DDS) to heterodyne the center frequency of the down sampled signal to baseband and present the base

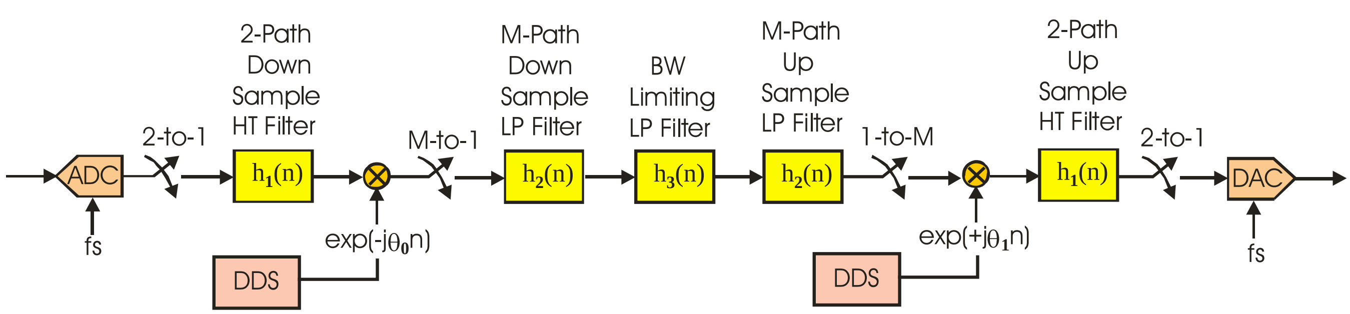


Figure 1. Digital Down Converter Formed from a 2-Path Down-Sampling Hilbert Transform Filter, a DDS Down Conversion, an M-Path Down-Sampling Low-Pass Filter, a BW Limiting Low-Pass Filter, an M-Path Up-Sampling Filter, Reducing and Sample Rate Reducing Filter, a DDS Up Conversion, and a 2-Path Up-Sampling Hilbert Transform Filter.

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banded complex signal to a pair of real filter paths that efficiently reduce bandwidth in a cascade of multirate filters.

The first filter in this cascade is an M-path filter that reduces bandwidth and sample rate M-to-1. The sample rate reduction permits subsequent processing to be performed with minimum workload at the reduced sample rate. The reduced sample rate is chosen to be approximately twice the channel signal's bandwidth. The output of the down sampling filter is processed by the bandwidth limiting filter designed to meet the spectral mask specification of the process. The properly bandwidth limited signal is then presented to a second M-path filter that performs the 1-to-M up-sampling or interpolation process. The up-sampled signal is then heterodyned to the center frequency from which it was down-converted. The frequency offset signal is finally processed by the 2-path Hilbert transform filter that up-samples 1-to-2 and converts the complex input signal to a real output signal. The processing blocks between the input and output Hilbert transform filters is repeated for each of the center frequency bands and bandwidths to be extracted and reinserted by the channel selection process.

Figure 2 presents a stylized version of the input and output spectra of the digital down converter. Citing a specific example, we assume the input sample rate is 192 MHz, the channel bandwidth of interest, confined to a 80 MHz frequency span about the quarter sample rate of 48 MHz, is a 20.00 MHz Band centered at 50 MHz. We want to down convert this band and reduce the sample rate 6-to-1 to obtain an output rate of 32 MHz, slightly below twice the channel bandwidth. The 6-to-1 sample rate reduction happens in two stages, the 2-to-1 input half band filter and the 3-to-1 3-path output filter. We determined that the FIR half band input filter required 59 taps to extract the signal from the 80 MHz frequency span with 80 dB out-of band attenuation. Twenty nine of the 59 coefficients of the half band filter are zeros leaving 30 non-zero coefficients. These coefficients are applied once for every two input samples which means the input filter is operating at 15 multiplies per input sample.

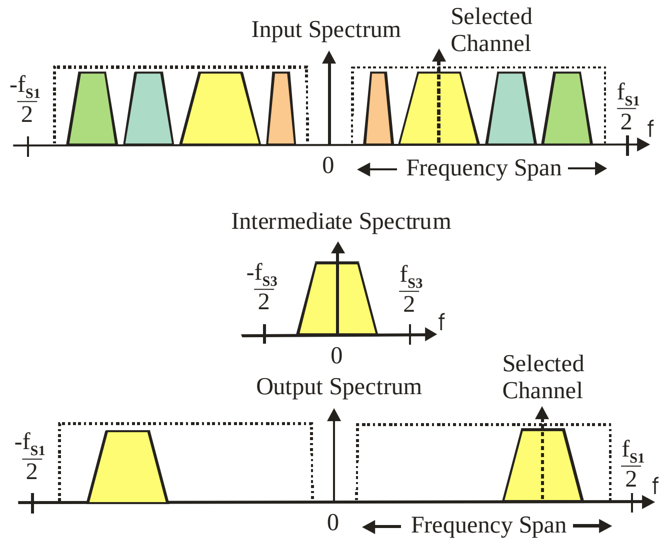


Figure 2. Typical Spectra at Input and Output of General Digital Down Converter and Up-Converter

Coefficient symmetry can also be used to reduce this workload by a factor of 2. We also determined the 3-path FIR filter required 33 weights, which when distributed over the 3-paths is 11-multiplies per input sample which becomes 22 multiplies per complex input sample at its 96 MHz sample rate or 11 multiplies per input sample referred back to the 196 MHz input sample rate.

The composite frequency response requirements of the 20 MHz bandwidth filter are presented in Table 1. We can extract from this table the frequency response specification of the bandwidth limiting filter following the 3-to-1 down sample filter. These extracted specifications are shown in table 2. Here we allocate 0.1 dB pass band ripple and 0.5° to the pair of input and output resampling filters on either side of the bandwidth limiting filter.

Table 1. 20 MHz Bandwidth Filter Specification

| 20 MHz Filter | Frequency, MHz | Attenuation/ Ripple |
|---|---|---|
| Pass Band | 0-to-9.8775 MHz, fs=192 MHz | ± 0.25 dB, ± 1.0 ° |
| Stop Band | 10.18 MHz | -42 dB |
| Stop Band | 10.98 MHz | -57 dB |
| Stop Band | 19.98 MHz | -67 dB |
| Stop Band | 96.00 MHz | -67 dB |

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Table 2. 20 MHz Bandwidth Low Pass Filter Specification

| 20 MHz Filter | Frequency, MHz | Attenuation/ Ripple |
|---|---|---|
| Pass Band | 0-to-9.8775 MHz, fs=32 MHz | ± 0.15 dB, ± 0.5 ° |
| Stop Band | 10.18 MHz | -42 dB |
| Stop Band | 10.98 MHz | -57 dB |
| Stop Band | 16.00 MHz | -63 dB |

We designed the 20 MHz bandwidth limiting filter to meet the specifications of Table 2 using the Remez (or PMFIR) equal ripple design routine. The number of taps required to meet the pass band and stop band mask limits was 261 taps. At the 32 MHz sample rate, the rate at which this filter operates, the 130 sample group delay of this filter is 4.06 μsec. Figure 3 shows the impulse response and frequency response of the composite filter chain described in this section. We first see that the group delay of the composite filter 6.69 μsec so that the delay introduced by the pair of input down sampling filters and pair of output up sampling filers is 0.63 μsec or 121 samples of the 901 samples at 192 MHz. We note that the stop band of the filter frequency meets the stop band spectral mask. We also see that the pass band ripple is equal ripple and its level easily satisfies the pass band mask. Since all the filters in the FIR cascade were designed as linear phase filters we are confident that the cascade is also linear phase.

## 2. LINEAR PHASE IIR FILTER OPTION

Here we consider replacing the down sampling input filters and the up-sampling output filters with linear phase recursive all-pass filter equivalents. We will see that we can implement these filters with reduced computational burden as well as exhibit reduced composite group delay and reduced levels of pass band ripple.

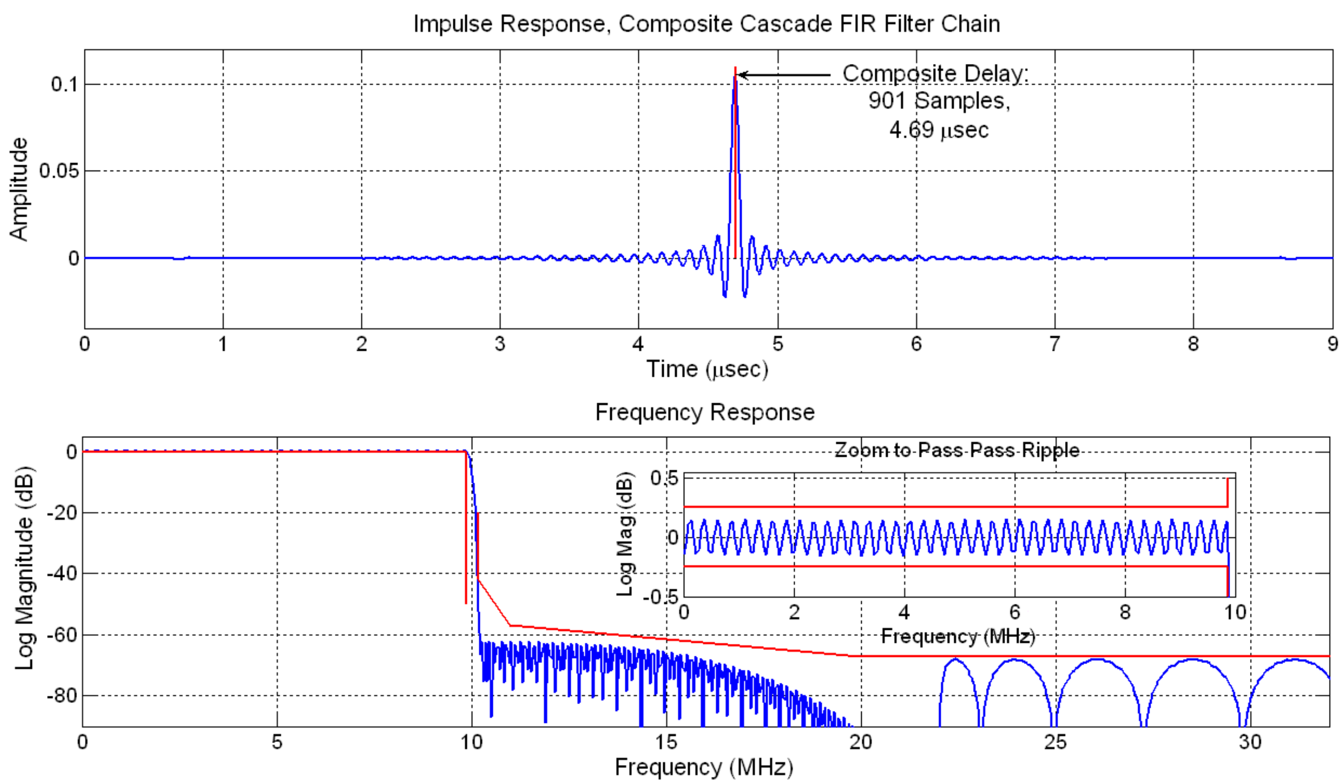


Figure 3.Impulse Response and Frequency Response of 20 MHz Composite FIR Filter Chain

The recursive IIR version of the half band filter required 12-coefficients which when amortized over the 2-input samples results in 6 multiplies per input sample, a slight improvement over the FIR implementation. Figure 4 shows the frequency response pass band ripple, pass band phase response, and phase peak to peak phase ripple with respect to de-trended phase response. The figures are remarkable: First we see the peak-to-peak in-band log magnitude ripple is 0.02 μ–dB (that's millionths of a dB). We also see the phase response appears to be linear, and when we de-trend the phase to obtain the non causal phase response we see the peak-to-peak phase ripple is approximately 0.006 degrees or about 1/10 of a milliradian. The pass band response of this filter filter is pretty good.

The 3-path linear phase recursive filter required 8-coefficients across the 3-paths of the filter, requiring less than 3-multiplies per input sample at the 96-MHz input sample rate per I-Q filter path. This represents about one-fourth of the workload for the corresponding FIR filter. Figure 5 shows the frequency response pass band ripple, pass band phase response, and peak to peak phase ripple with respect to de-trended phase response. These figures are also pretty remarkable: First we see the peak-to-peak in-band log magnitude ripple is 1. μ–dB. We also see the phase response appears to be linear, and when we de-trend the phase to obtain the non causal phase response we see the peak-to-peak phase ripple is approximately 0.06 degrees or about a milliradian. The pass band response of this filter is not bad at all.

The final filter we examine is the low pass filter that performs the required bandwidth shaping and bandwidth reduction. Due to the very narrow transition bandwidth of this filter the best structure for this filter is a linear phase tapped delay line FIR filter. We note that the Remez algorithm does not take advantage of the relaxed mask levels near the filter pass band. A modified version of the algorithm permits a stop band tilt and allows us to meet the design specifications with a fewer number of coefficients. The equal ripple version of the design algorithm requires 261 coefficients while the tilted stop-band (from our MATLAB code myfrf_2) can meet the spectral mask requirements with a 221 tap filter.

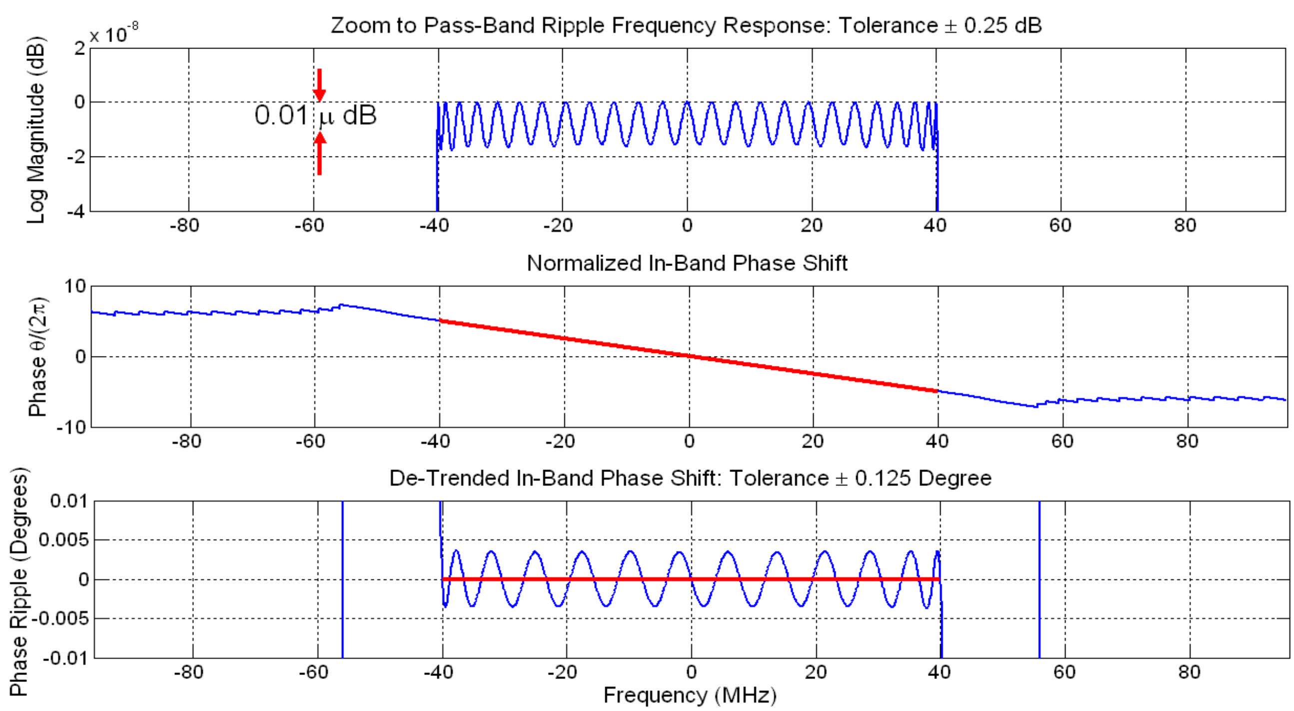


Figure 4. Log Magnitude Frequency Response, Phase Response, and Peak-to-Peak Phase Ripple of Recursive Linear Phase, 2-Path Half Band Filter, Pass band ± 40 MHz, fs = 192 MHz

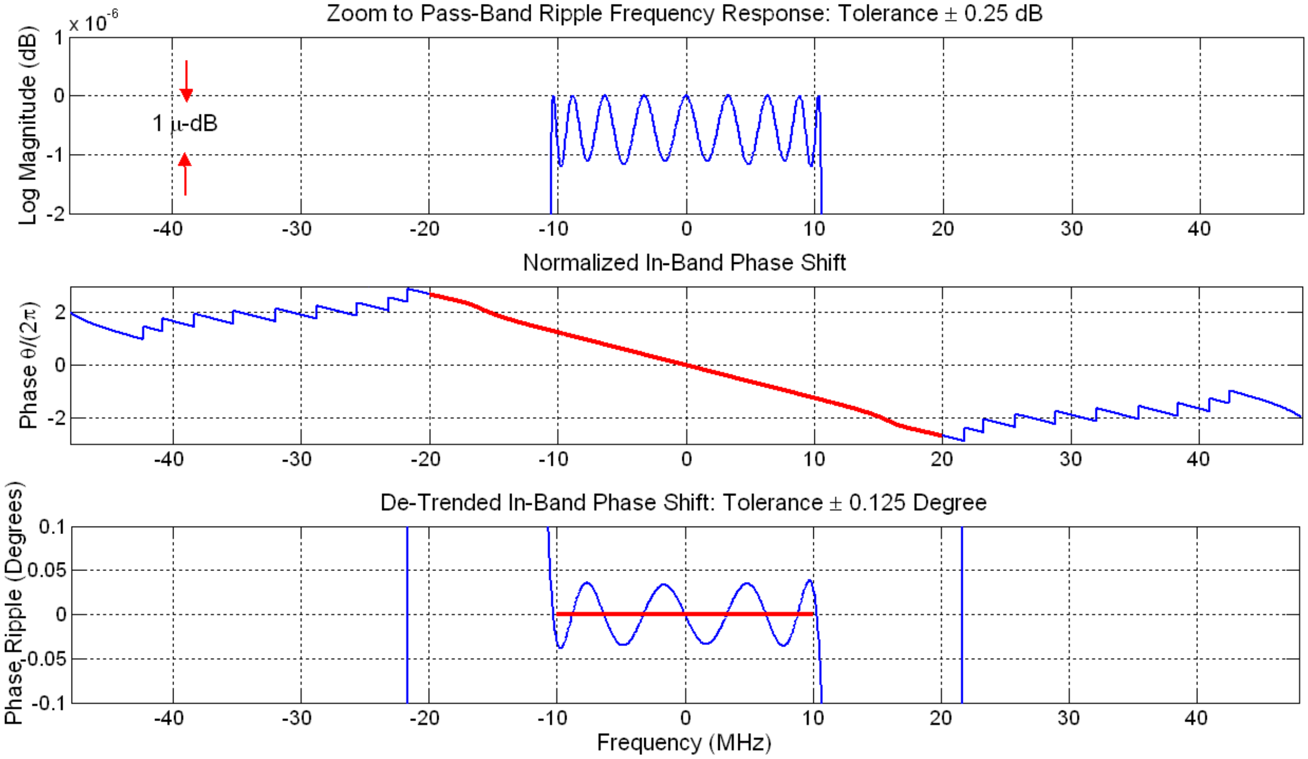


Figure 5. Log Magnitude Frequency Response, Phase Response, and Peak-to-Peak Phase Ripple of Recursive Linear Phase, 3-Path, 3-to-1 Down Sample Low-Pass Filter, Pass band ± 10 MHz, fs = 96 MHz

The 221 taps represent a group delay 110 samples at the 32 MHz sample rate, the rate at which the filter operates for a delay of 3.44 μsec. This is a 20 sample reduction or 0.52 μsec reduced time delay. The bulk delay in the bandwidth limiting filter is reduced by approximately 15% by taking advantage of the realed mask levels at the pass band edge. We can expect additional reduction in time delay due to the recursive pre-and-post linear phase IIR filters.

Figure 6 shows the impulse response and frequency response of the composite filter chain described in this section. We first see that the group delay of the composite filter is 3.93 μsec so that the delay introduced by the pair of input down sampling filters and pair of output up sampling filers is 0.49 μsec or 94 samples of the 755 samples at 192 MHz. We note that the stop band of the filter frequency meets the stop band spectral mask. We also see that the pass band ripple is also equal ripple and its level easily satisfies the pass band mask. Figure 7 shows the frequency response pass band ripple, pass band phase response, and peak to peak phase ripple with respect to de-trended phase response. First we see the peak-to-peak in-band log magnitude ripple is less than 0.1 dB. We noted from the spectra of the IIR filters in the cascade that their μ-dB magnitude ripple levels could not have contributed to the composite ripple levels hence the ripple of the cascade chain is merely the ripple of the FIR bandwidth limiting filter. We see that the phase response appears to be linear, and when we de-trend the phase to obtain the non causal phase response we see the peak-to-peak phase ripple is approximately 0.12 degrees or about 2 milliradian. We feel pretty confident that that the composite filter chain formed by a pair of linear phase IIR down sampling filters, a linear phase FIR filter, and a pair of linear phase IIR upsampling filters does indeed synthesize a linear phase filter. .

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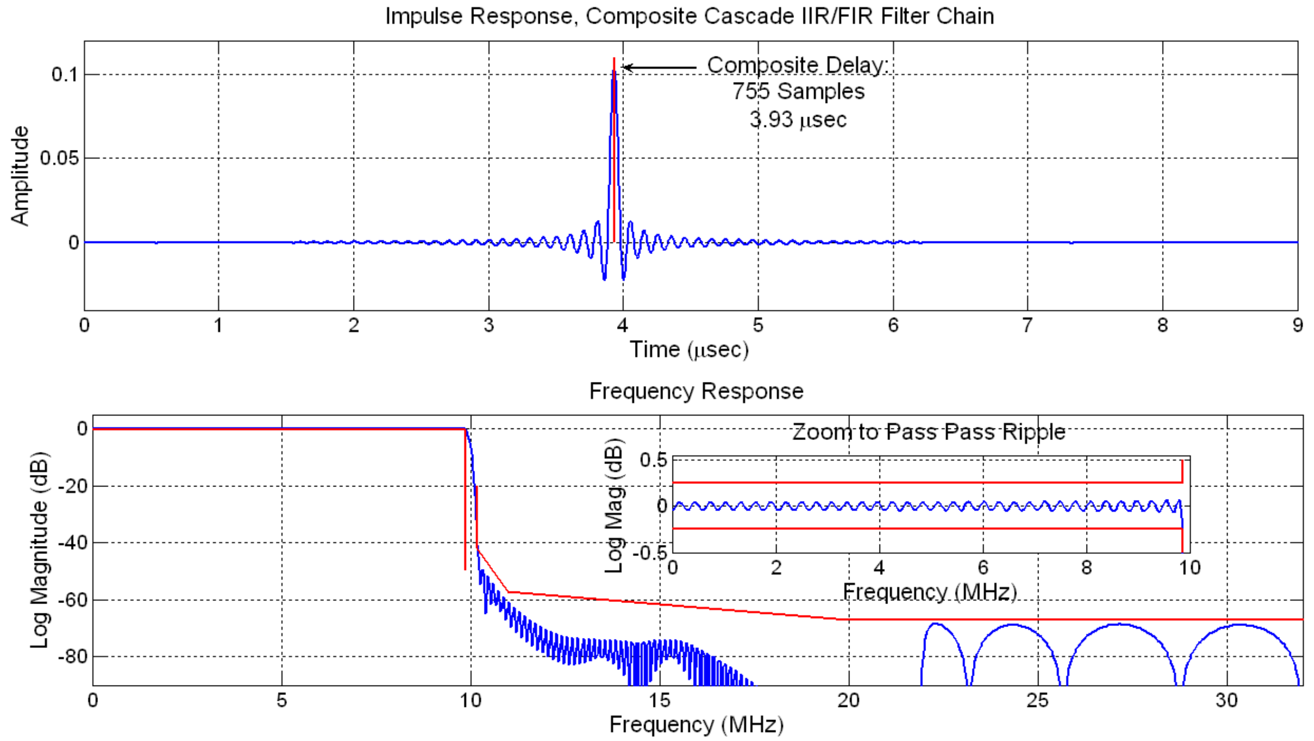


Figure 6.Impulse Response and Frequency Response of 20 MHz Composite IIR-FIR Filter Chain

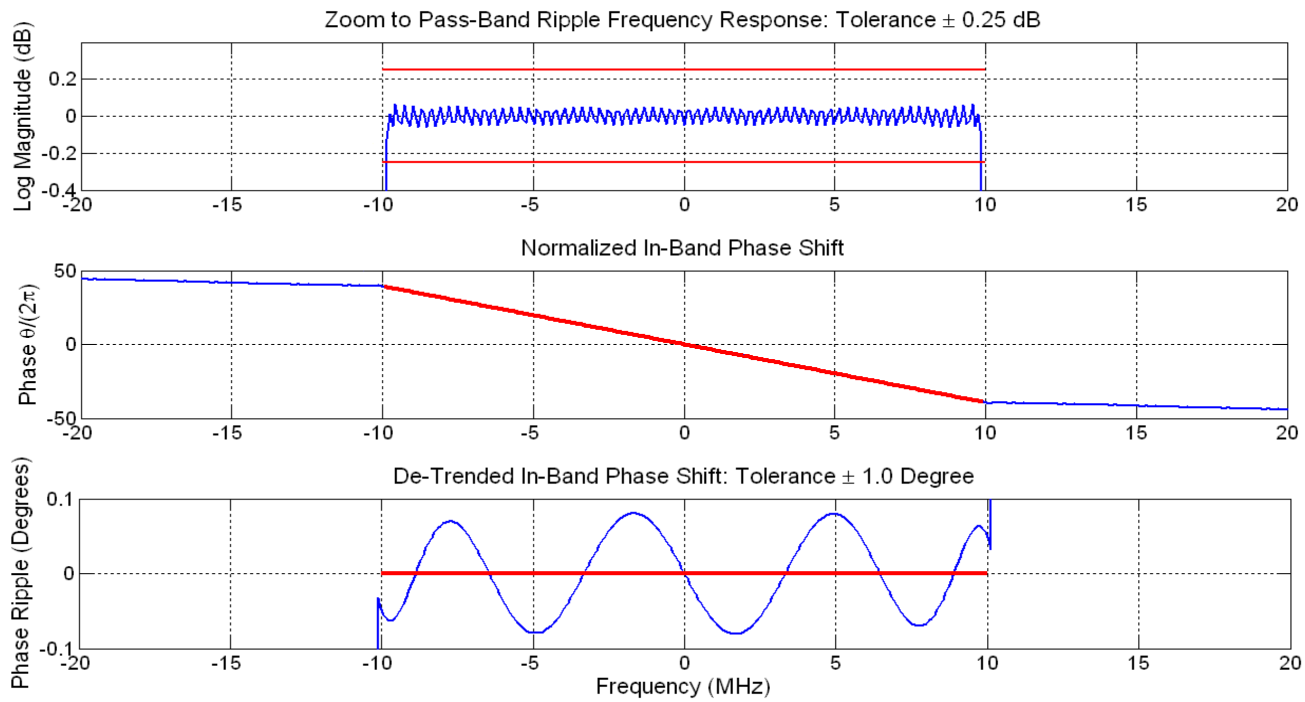


Figure 7. Log Magnitude Frequency Response, Phase Response, and Peak-to-Peak Phase Ripple of Cascade Recursive Linear Phase and Linear Phase FIR Filter, BW ± 10 MHz, fs = 192 MHz.

## 3. COMPARING FIR AND IIR OPTIONS

We first compare the workload of the two filter options. Table 3 lists the number of coefficients for the 5 filters in the channelizer implemented with FIR filters and with IIR and FIR filters. Also listed is the number of operations (multiplies and adds) for each filter referred to the input of the channelizer input or output rate. If more than a single channelizer is formed in these architectures, the input and output half band filters can be shared by the multiple paths. We note that the workload for the two filter options is approximately 100 and 55 operations per input sample point. The linear phase IIR resampling filter option requires 55% of the computational resources of the FIR resampling filter option. Of course part of this improvement is the shortened FIR filter that takes advantage of the stop band spectral mask.

Table 3. Work Load of FIR and IIR Channelizers

| | Fir Coef | Fir Ops/In | IIR Coef | IIR Ops/In |
|---|---|---|---|---|
| Input 2-Path | 59 | 15 | 12 | 6 |
| Input 3-Path | 33 | 11 | 8 | 2.7 |
| BW Filter | 261 | 43.5 | 221 | 36.8 |
| Output 3-Path | 59 | 11 | 8 | 2.7 |
| Output 2-Path | 33 | 15 | 12 | 6 |
| Total Ops/In | - | 99.5 | - | 54.2 |

The second comparison between the two filter options is the group delay between the input and output of the filters. Figures 3 and 6 show the impulse response of the two filters. We had commented earlier that the delays are 901 and 755 samples or 4.69 and 3.93 μsec respectively for the two options. Time delay may be an important parameter in a channelizer system.

Our final comparison of the two filter options is the reduction in signal quality introduced as a result of a modulation signal passing through the channelizer. Figure 8 shows the constellation cluster at the output of a matched filter for a signal shaped and band limited by a square-root Nyquist filter. The left most subplot shows the cluster formed by a loop back, modulator followed by demodulator without the channel. The error Vector magnitude (EVM) for the selected shaping filter is seen 20 be -41.3 dB. The center subplot of Figure 8 presents the matched filter cluster obtained when the shaped signal is passed through the FIR based channelizer and then demodulated by a matched filter. We see the EVM has decreased to -33.8 dB. The channelizer has degraded the signal by the inter symbol interference (ISI) caused by pass band ripple. The right most subplot of Figure 8 shows the matched filter cluster obtained with the linear phase IIR channelizer. We see the EVM has decreased to -38.4 dB, a smaller degradation than that inserted by the FIR filter option.

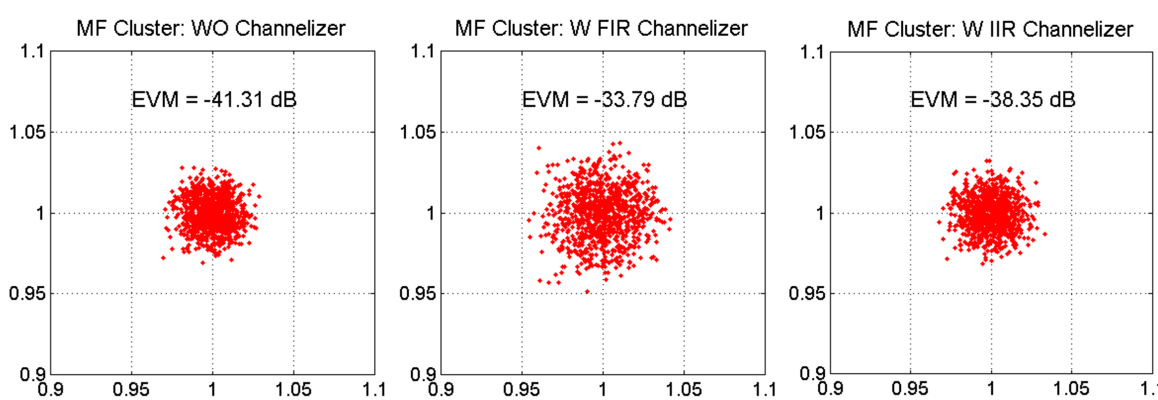


Figure 8. Constellation Cluster at Matched Filter Output: Without Channel, with FIR Filter Channelizer, and with IIR Filter Channelizer

## 4. CLOSING COMMENTS

We have examined a common method, using FIR filters and heterodynes, for extracting selected spectral bands from a wide span of frequencies. The frequency span may contain multiple broadband signals with different bandwidths that have to be assembled or disassembled to obtain different mixes of assigned channel plans or to drop or insert selected channels from a mix. If multiple channels are going to be processed the processing scheme has to be replicated for each selected channel. We are motivated to control cost and to assure signal quality in the processing chain. We are also sensitive to transport or group delay of the processing blocks that manipulate the various different bandwidth channels. Responding to these considerations we examined the use of linear phase recursive filters in place of linear phase non-recursive filters to implement the down-sampling filters and the up-sampling filters in the processing chain of each channel selector.

We designed two versions of each resampling filter in the cascade and compared their relative work load as well as their group delay. The recursive filters had slightly less computational requirements then did their FIR counterparts. The IIR filters exhibited about 15% reduced time delay relative to the FIR filters performing the same filtering tasks.

We also examined the bandwidth limiting FIR filter designed originally by the standard Remez algorithm. The standard Remez algorithm designs filters with equal ripple pass band and equal ripple stop band side lobes. The spectral masks of the filters were shaped and permitted reduced levels of attenuation at the edge of the stop band. By modifying the Remez algorithm to allow $1/f$, or $1/f^2$ stop band side lobe decay rates we were able to reduce the FIR filter lengths by at least 10%. We were cavalier with our design effort and obtained final designs that had significant margins of in-band ripple levels. With another pass of the design activity we are confident we can reduce the margins and obtain filters with reduced number of coefficients and with reduced group delay.

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