boB Gudgel's notes for the January 2007 NW chapter AES DSP meeting

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EQUATION 6-1
The convolution summation. This is the formal definition of convolution, written in the shorthand: $y[n] = x[n] * h[n]$. In this equation, $h[n]$ is an M point signal with indexes running from 0 to M-1.

$$y[i] = \sum_{j=0}^{M-1} h[j]\, x[i-j]$$

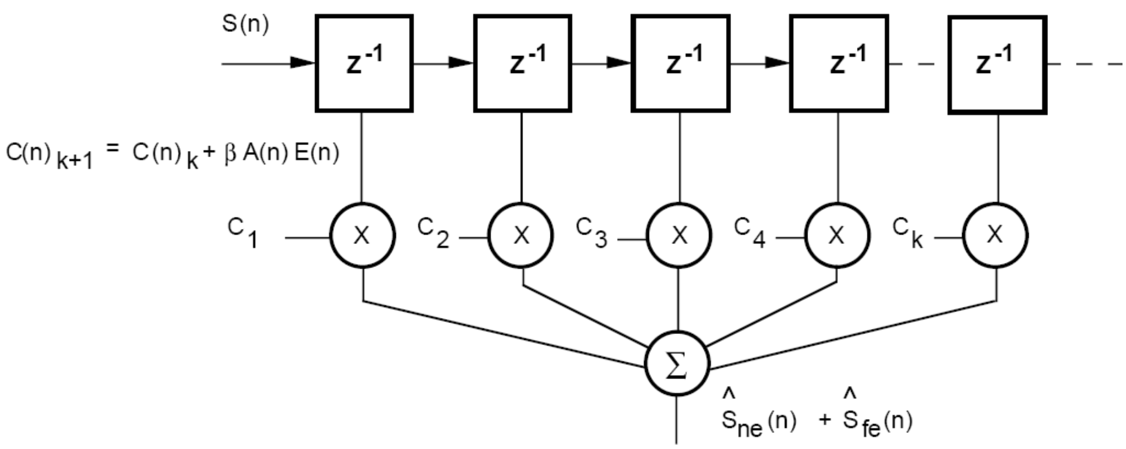

Figure 2.20 LMS Adaptive Filter

(Analog Devices pic) Fig 2.20 is a FIR filter where C1 thru Ck are coefficients (impulse)

ADSP-2181 MAIN FIR code… The one line, fir1loop, happens 256 times for the convolution of the signal and entire impulse response (kernel)

```
        taps = 256;  {taps is the same length as the impulse kernel for the filter type}
fir:    cntr=taps-1;        {taps = 256 for this FIR, cntr decrements}
        mr=0, mx0=dm(i0,m0), my0=pm(i4,m4); {initialize the registers}

        do fir1loop until ce;  {ce means Counter Empty (cntr empty) }

fir1loop:   mr=mr+mx0*my0(ss), mx0=dm(i0,m0), my0=pm(i4,m4); ←The loop

        mr=mr+mx0*my0(rnd);         {I think that rnd means round}
        if mv sat mr;
        sr = ashift mr1 by 1 (lo);  {amplify filtered signal by 2 }
        mr1 = sr0;

        jump output;
```

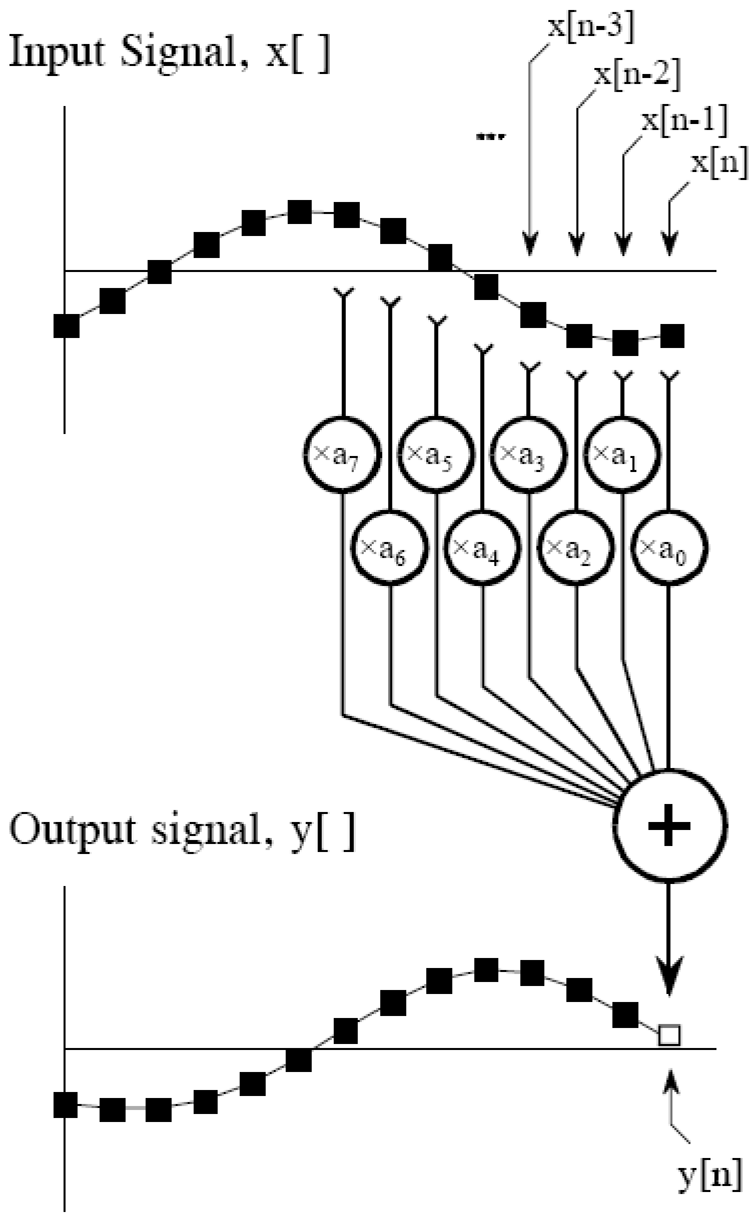

FIGURE 28-2
FIR digital filter. In FIR filtering, each sample in the output signal, y[n], is found by multiplying samples from the input signal, x[n], x[n-1], x[n-2], ..., by the filter kernel coefficients, a_0, a_1, a_2, a_3 ..., and summing the products.

```
MR = MR + MX1*MY1;
```

performs a multiply/accumulate operation. It multiplies the input values in registers MX1 and MY1, adds that product to the current value of the MR register (the result of the previous multiplication) and then writes the new result to MR

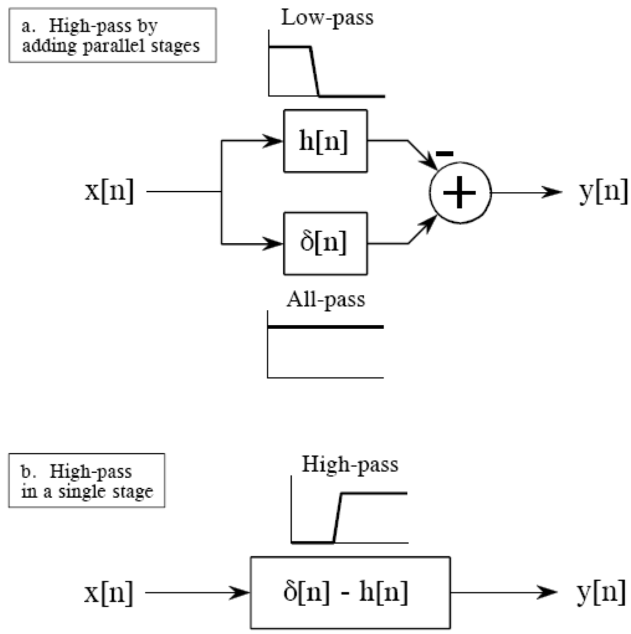

FIGURE 14-6
Block diagram of spectral inversion. In (a), the input signal, $x[n]$, is applied to two systems in parallel, having impulse responses of $h[n]$ and $\delta[n]$. As shown in (b), the combined system has an impulse response of $\delta[n] - h[n]$. This means that the frequency response of the combined system is the *inversion* of the frequency response of $h[n]$.

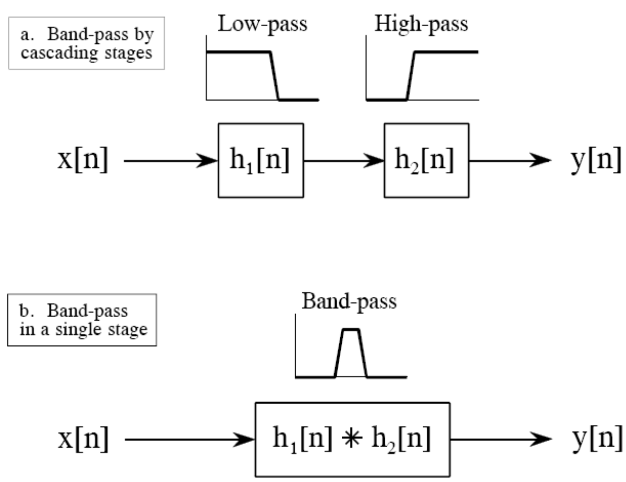

FIGURE 14-8
Designing a band-pass filter. As shown in (a), a band-pass filter can be formed by cascading a low-pass filter and a high-pass filter. This can be reduced to a single stage, shown in (b). The filter kernel of the single stage is equal to the *convolution* of the low-pass and high-pass filter kernels.

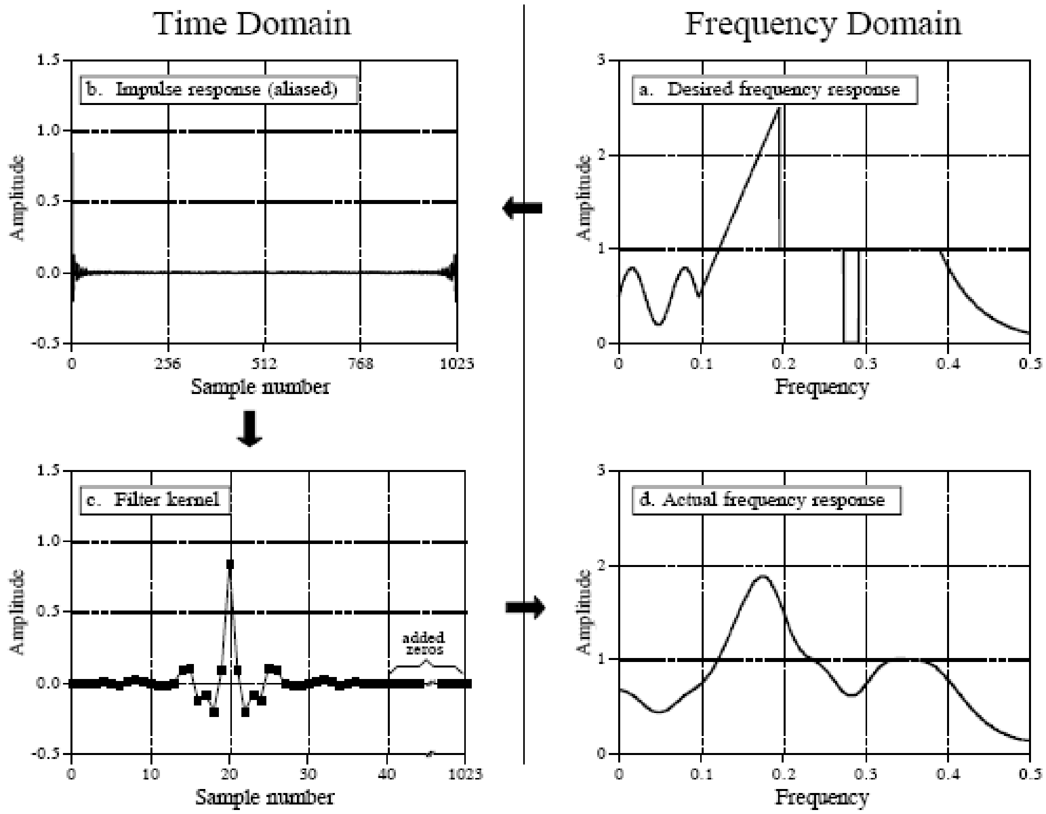

FIGURE 17-1
Example of FIR filter design. Figure (a) shows the desired frequency response, with 513 samples running between 0 to 0.5 of the sampling rate. Taking the Inverse DFT results in (b), an *aliased* impulse response composed of 1024 samples. To form the filter kernel, (c), the aliased impulse response is truncated to $M+1$ samples, shifted to the right by $M/2$ samples, and multiplied by a Hamming or Blackman window. In this example, M is 40. The program in Table 17-1 shows how this is done. The filter kernel is tested by padding it with zeros and taking the DFT, providing the actual frequency response of the filter, (d).

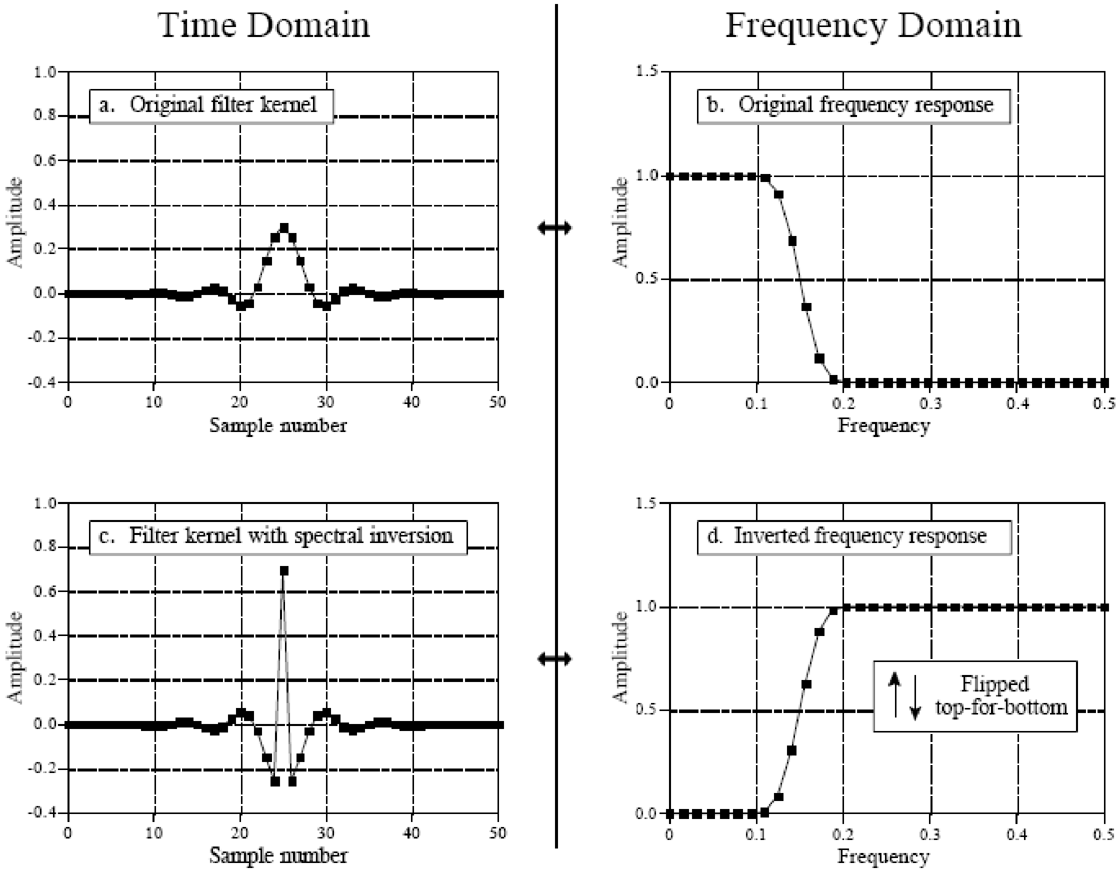

FIGURE 14-5
Example of spectral inversion. The low-pass filter kernel in (a) has the frequency response shown in (b). A high-pass filter kernel, (c), is formed by changing the sign of each sample in (a), and adding one to the sample at the center of symmetry. This action in the time domain *inverts* the frequency spectrum (i.e., flips it top-for-bottom), as shown by the high-pass frequency response in (d).

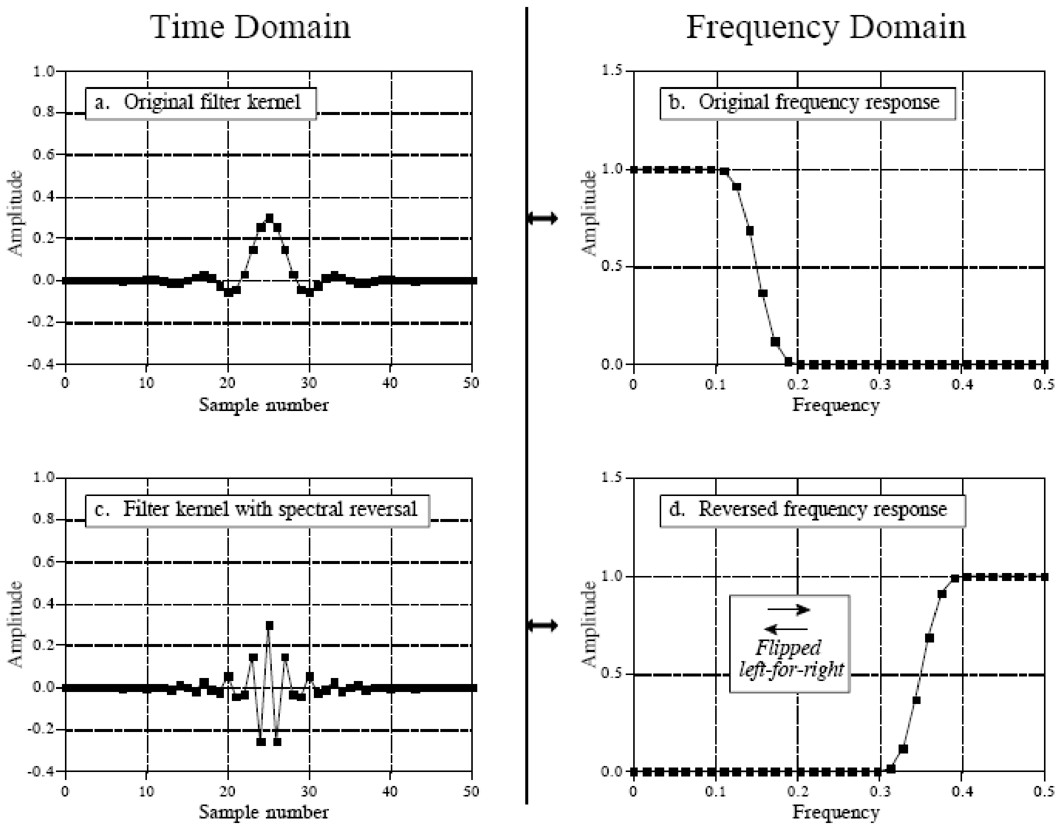

FIGURE 14-7
Example of spectral reversal. The low-pass filter kernel in (a) has the frequency response shown in (b). A high-pass filter kernel, (c), is formed by changing the sign of every other sample in (a). This action in the time domain results in the frequency domain being flipped *left-for-right*, resulting in the high-pass frequency response shown in (d).

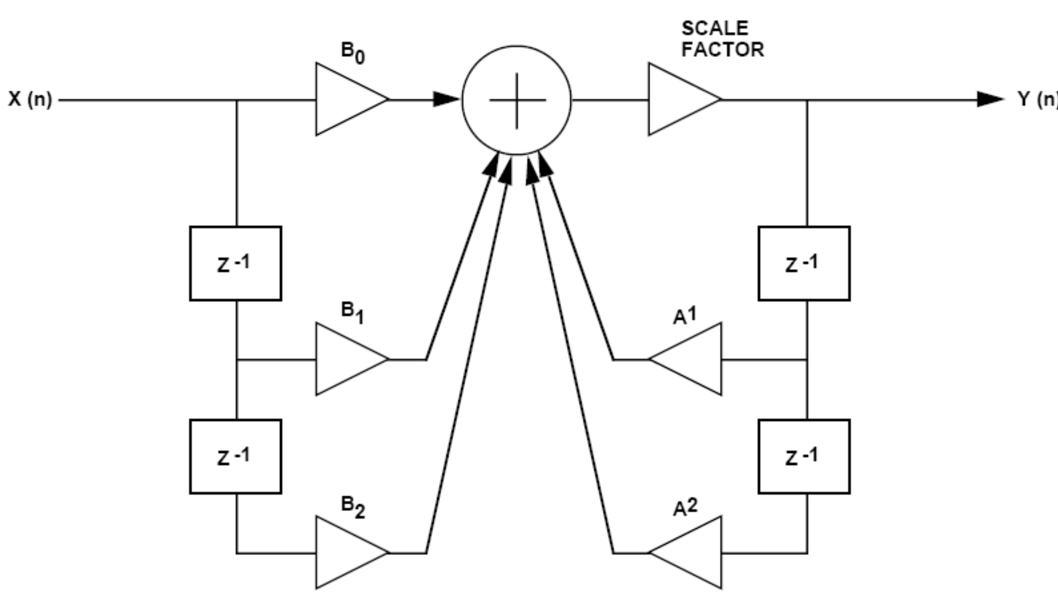

Figure 10.1 Second-Order Biquad IIR Filter Section