Chapter 7

Real-Time FIR Digital Filters

Introduction

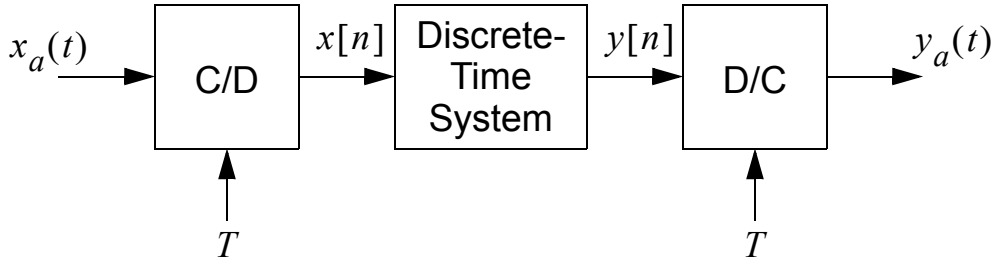
Digital filter design techniques fall into either finite impulse response (FIR) or infinite impulse response (IIR) approaches. In this chapter we are concerned with just FIR designs. We will start with an overview of general digital filter design, but the emphasis of this chapter will be on real-time implementation of FIR filters using C and assembly.

- Basic FIR filter topologies will be reviewed
- To motivate the discussion of real-time implementation we will begin with a brief overview of FIR filter design
- Both fixed and floating-point implementations will be considered
  - The MATLAB signal processing toolbox and filter design toolbox will be used to create quantized filter coefficients
- The use of circular addressing in assemble will be considered
- Implementing real-time filtering using analog I/O will be implemented using the AIC23 codec interface and associated ISR routines

Basics of Digital Filter Design

- A filter is a frequency selective linear time invariant (LTI) system, that is a system that passes specified frequency components and rejects others
- The discrete-time filter realizations of interest here are those LTI systems which have LCCDE representation and are causal
- Note that for certain applications noncausal filters are appropriate
- An important foundation for digital filter design are the *classical* analog filter approximations
- The filter design problem can be grouped into three stages:
  - Specification of the desired system properties (application driven)
  - Approximation of the specifications using causal discrete-time systems
  - System realization (technology driven - hardware/software)

- A common scenario in which one finds a digital filter is in the filtering of a continuous-time signal using an A/D-$H(z)$-D/A system



- Strictly speaking $H(z)$ is a discrete-time filter although it is commonly referred to as a *digital filter*
- Recall that for the continuous-time system described above (ideally)

$$H_{\text{eff}}(j\Omega) = \begin{cases} H(e^{j\Omega T}), & |\Omega| < \pi/T \\ 0, & \text{otherwise} \end{cases} \tag{7.1}$$

- Using the change of variables $\omega = \Omega T$, where T is the sample spacing, we can easily convert continuous-time specifications to discrete-time specifications i.e.,

$$H(e^{j\omega}) = H_{\text{eff}}\left(j\frac{\omega}{T}\right), |\omega| < \pi \tag{7.2}$$

Overview of Approximation Techniques

- Digital filter design techniques fall into either IIR or FIR approaches

- General Approximation Approaches:
  - Placement of poles and zeros (ad-hoc)
  - Numerical solution of differential equations
  - Impulse invariant (step invariant etc.)
  - Bilinear transformation
  - Minimum mean-square error (frequency domain)
- FIR Approximation Approaches
  - Truncated impulse response of ideal *brickwall* responses using window functions
  - Frequency sampling desired response using transition samples
  - Optimum equiripple approximations (use the *Parks-McClellan algorithm* via the *Remez exchange algorithm*, `firpm()` in MATLAB)
  - Minimum mean-square error in the frequency domain
- In the FIR approaches the additional constraint of *linear phase* is usually imposed

Basic FIR Filter Topologies

- Recall that a causal FIR filter containing $M+1$ coefficients has impulse response

$$h[n] = \sum_{k=0}^{M} a_k \delta[n-k] \tag{7.3}$$

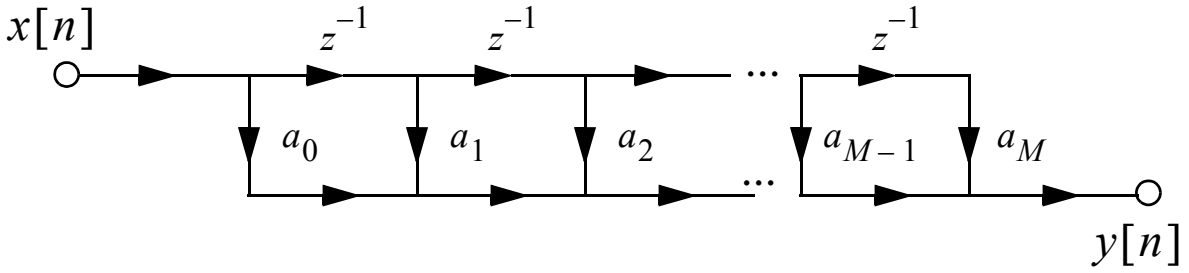
- The corresponding z-transform of the filter is

$$H(z) = \sum_{k=0}^{M} a_k z^{-k} = a_0 + a_1 z^{-1} + \ldots + a_{M-1} z^{-M} \tag{7.4}$$

- The frequency response of the filter is

$$H(e^{j\omega}) = \sum_{k=0}^{M} a_k e^{-j\omega k} \tag{7.5}$$

- The classical direct form topology for this filter is shown below

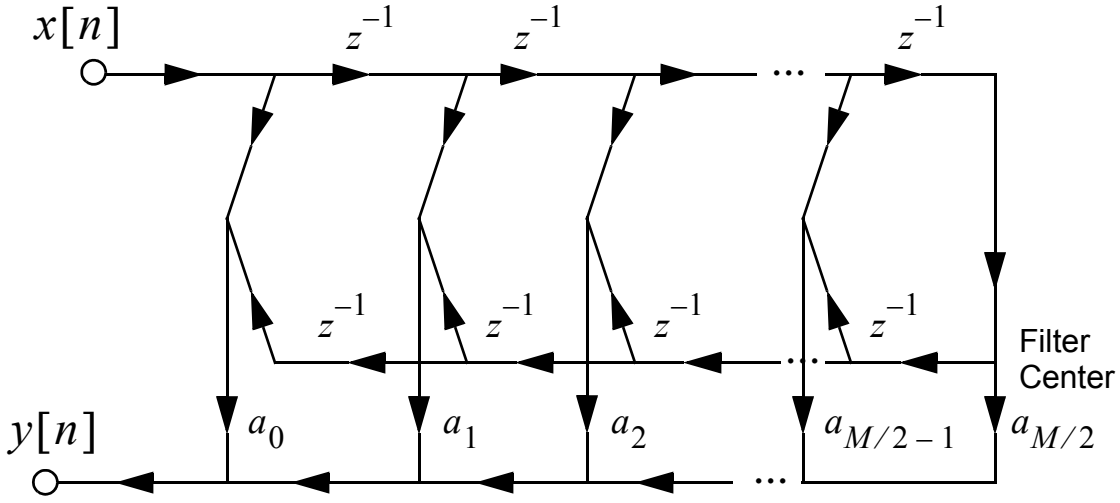


Direct Form FIR Structure

- Often times we are dealing with linear phase FIR designs which have even or odd symmetry, e.g.,

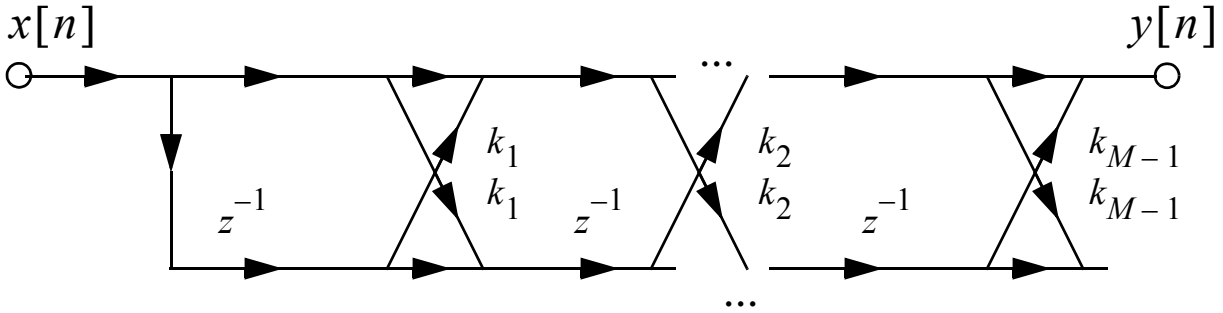
$$h[n] = h[M-n] \text{ even}$$
$$h[n] = -h[M-n] \text{ odd} \tag{7.6}$$

- In this case a more efficient structure, reducing the number of multiplications from $M+1$ to $M/2+1$, is



Modified Direct Form FIR Structure for *M* Even

- Another structure, not as popular as the first two, is the FIR lattice



FIR Lattice Structure

  – Although not discussed any further here, the *reflection coefficients*, k_i, are related to the a_k via a recurrence formula (in MATLAB we use `k=tf2latc(num)`)

- The lattice structure is very robust under coefficient quantization

Overview of FIR Filter Design

Why or Why Not Choose FIR?

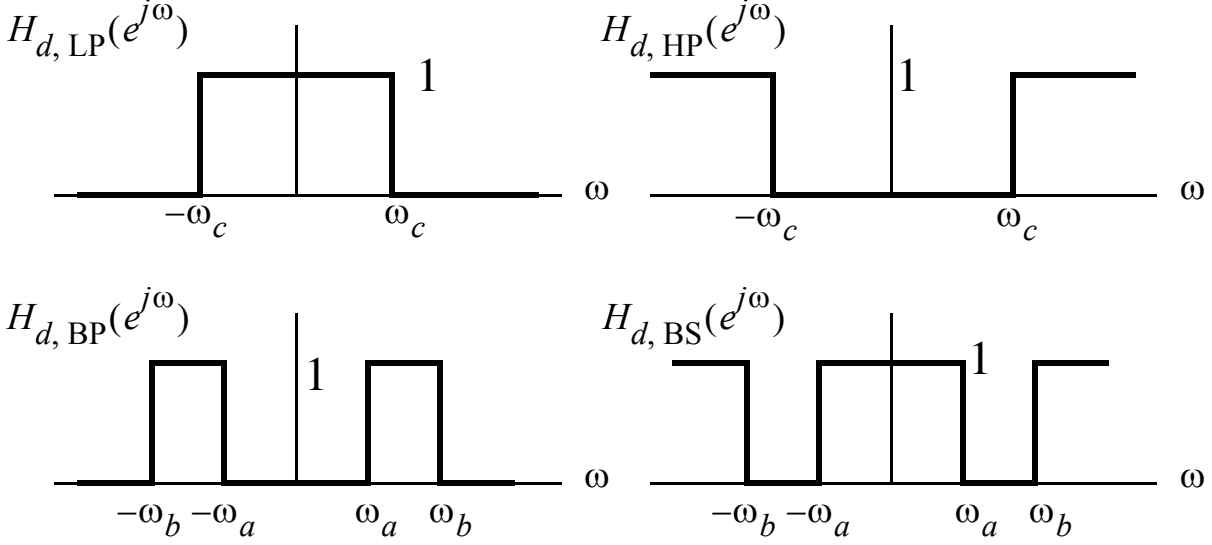
- FIR advantages over IIR
  - Can be designed to have exactly linear phase
  - Typically implemented with nonrecursive structures, thus they are inherently stable
  - Quantization effects can be minimized more easily
- FIR disadvantages over IIR
  - A higher filter order is required to obtain the same amplitude response compared to a similar IIR design
  - The higher filter order also implies higher computational complexity
  - The higher order filter also implies greater memory requirements for storing coefficients

FIR Design Using Windowing

- The basic approach is founded in the fact that ideal lowpass, highpass, bandpass, and bandstop filters, have the following noncausal impulse responses on $-\infty < n < \infty$

$$
\begin{aligned}
h_{d,\,\mathrm{LP}}[n] &= \frac{\sin(\omega_c n)}{\pi n} \\
h_{d,\,\mathrm{HP}}[n] &= \delta(n) - \frac{\sin(\omega_c n)}{\pi n} \\
h_{d,\,\mathrm{BP}}[n] &= \frac{\sin(\omega_b n) - \sin(\omega_a n)}{\pi n} \\
h_{d,\,\mathrm{BS}}[n] &= \delta(n) - \frac{\sin(\omega_b n) - \sin(\omega_a n)}{\pi n}
\end{aligned}
\tag{7.7}
$$

where the ideal frequency responses corresponding to the above impulse responses are



Ideal *Brickwall* Filters

- Consider obtaining a causal FIR filter that approximates $h_d[n]$ by letting

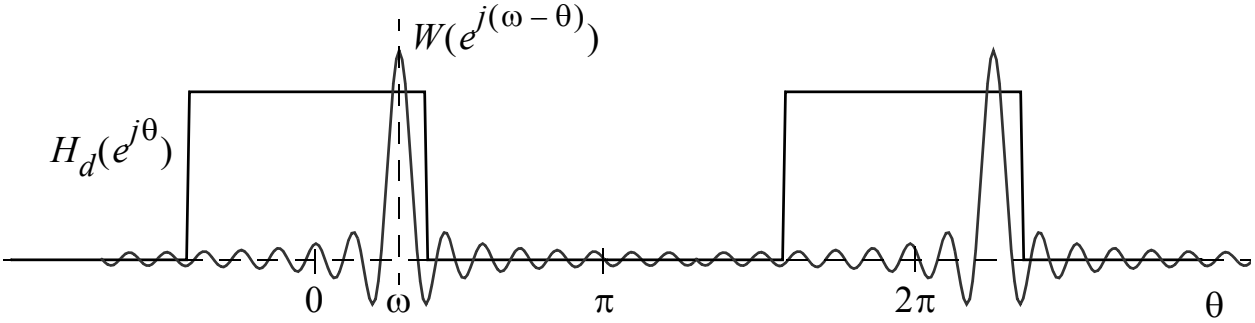
$$
h[n] = \begin{cases} h_d[n - M/2], & 0 \le n \le M \\ 0, & \text{otherwise} \end{cases}
\tag{7.8}
$$

- Effectively we have applied a rectangular window to $h_d[n]$ and moved the saved portion of the impulse response to the right to insure that for causality, $h[n] = 0$ for $n < 0$
  - The applied rectangular window can be represented as

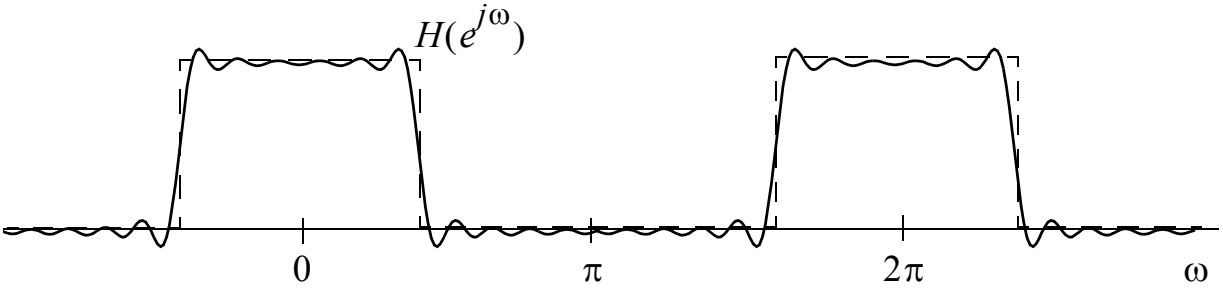
$$h[n] = h_d[n - M/2]w[n] \tag{7.9}$$

  where here $w[n]$ is unity for $0 \le n \le M$ and zero otherwise
- Since the rectangular window very abruptly shuts off the impulse response, Gibbs phenomenon oscillations occur in the frequency domain at the band edges, resulting in very poor stopband response
- This can be seen analytically with the aid of the windowing Fourier transform theorem

$$H(e^{j\omega}) = \frac{1}{2\pi}\int_{-\pi}^{\pi} e^{-j\theta M/2} H_{\mathrm{d}}(e^{j\theta}) W(e^{j(\omega-\theta)}) d\theta \tag{7.10}$$



(a) Circular convolution (windowing)



(b) Resulting frequency response of windowed sinc(x)

  - Use of the rectangular window results in about 9% peak ripple in both the stopband and passband; equivalently the peak passband ripple is 0.75 dB, while the peak sidelobe level is down only 21 dB

- To both decrease passband ripple and lower the sidelobes, we may smoothly taper the window to zero at each end
  - Windows chosen are typically symmetrical about the center $M/2$
  - Additionally, to insure linear phase, the complete impulse response must be either symmetric or antisymmetric about $M/2$

Example: Use of the Hanning Window

- The Hanning window is defined as follows

$$w[n] = \begin{cases} 0.5 - 0.5\cos(2\pi n/M), & 0 \le n \le M \\ 0, & \text{otherwise} \end{cases} \tag{7.11}$$

- Consider an $M = 32$ point design with $\omega_c = \pi/2$
- The basic response $h_d[n]$ for a lowpass is of the form

$$h_d[n] = \frac{\sin(\omega_c n)}{\pi n} = \frac{\omega_c}{\pi}\text{sinc}\left(\frac{\omega_c n}{\pi}\right) \tag{7.12}$$

since $\text{sinc}(x) = \sin(\pi x)/(\pi x)$

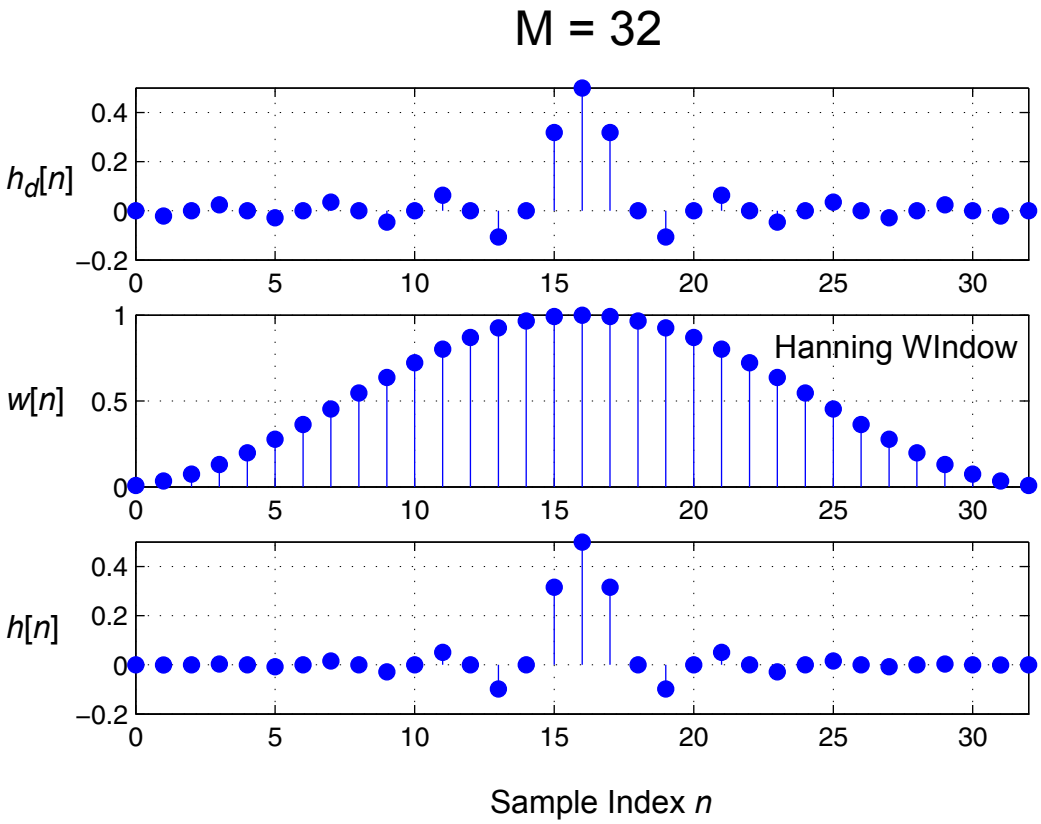
- Ultimately we will use MATLAB filter design functions, but for now consider a step-by-step approach:

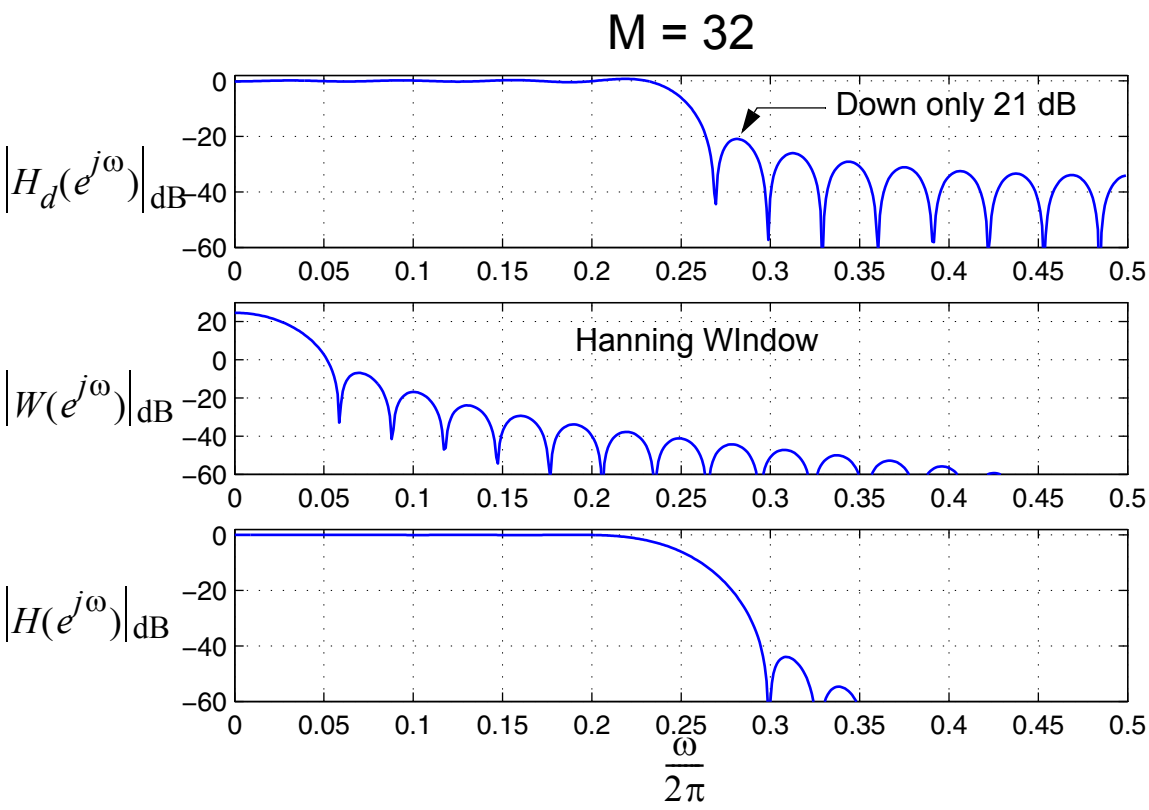
```
>> n = 0:32;
>> hd = pi/2/pi*sinc(pi/2*(n-32/2)/pi);
>> w = hanning(32+1);
>> h = hd.*w';
>> [Hd,F] = freqz(hd,1,512,1);
>> [W,F] = freqz(w,1,512,1);
>> [H,F] = freqz(hd.*w',1,512,1);
```

- The time domain waveforms:
- The frequency response:
- In general for an even symmetric design the frequency response is of the form

$$H(e^{j\omega}) = A_e(e^{j\omega})e^{-j\omega M/2} \tag{7.13}$$

where $A_e(e^{j\omega})$ is a real and even function of ω


M = 32
$h_d[n]$
$w[n]$
$h[n]$
Hanning WIndow
0.4
0.2
0
−0.2
1
0.5
0
0
5
10
15
20
25
30
Sample Index n

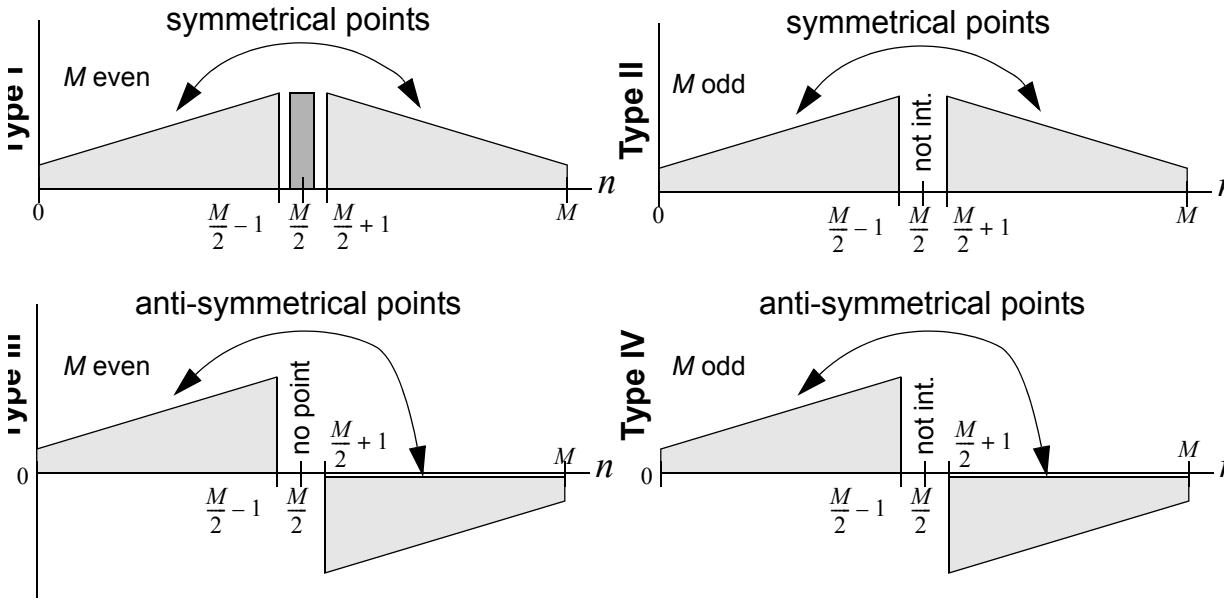

M = 32
$|H_d(e^{j\omega})|_{\text{dB}}$
$|W(e^{j\omega})|_{\text{dB}}$
$|H(e^{j\omega})|_{\text{dB}}$
Down only 21 dB
Hanning WIndow
0
−20
−40
−60
20
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5
$\frac{\omega}{2\pi}$

- For an odd symmetric design

$$H(e^{j\omega}) = jA_o(e^{j\omega})e^{-j\omega M/2} \tag{7.14}$$

where $A_e(e^{j\omega})$ is a real and even function of ω

- Note that the symmetric case leads to type I or II generalized linear phase, while the antisymmetric case leads to type III or IV generalized linear phase



- For $h[n]$ symmetric about $M/2$ we can write the magnitude response $A_e(e^{j\omega})$ as

$$A_e(e^{j\omega}) = \int_{-\infty}^{\infty} H_e(e^{j\theta})W_e(e^{j(\omega-\theta)})d\theta \tag{7.15}$$

where $H_e(e^{j\omega})$ and $W_e(e^{j\omega})$ are real functions correspond-

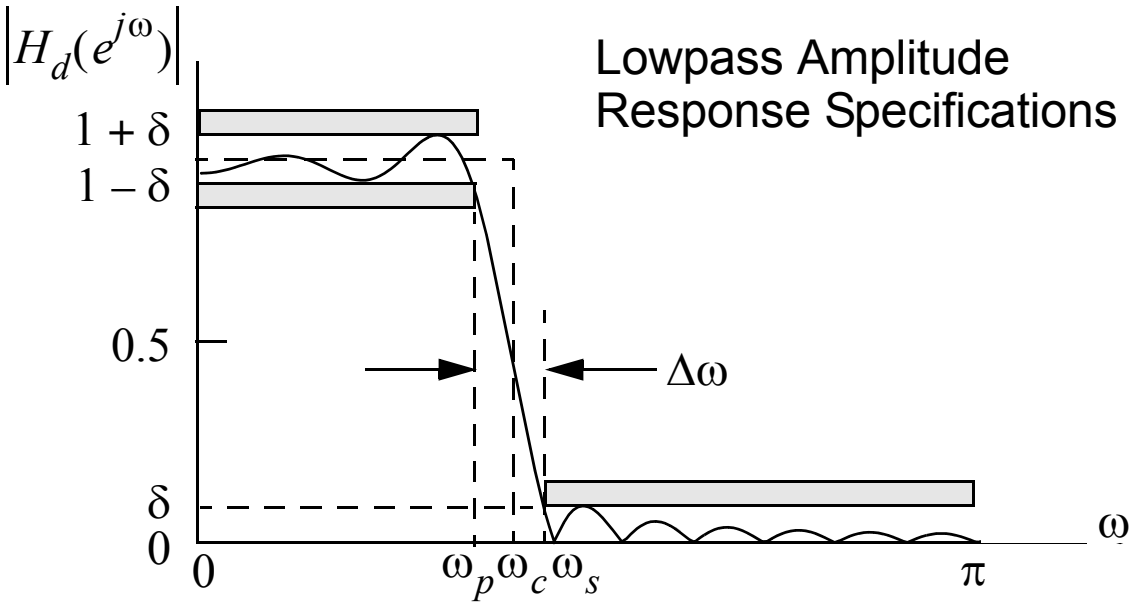
ing to the magnitude responses of the desired filter frequency response and window function frequency response respectively

Lowpass Design

- For a lowpass design having linear phase we start with

$$h[n] = \frac{\sin[\omega_c(n - M/2)]}{\pi(n - M/2)} w[n] \qquad (7.16)$$

- The lowpass amplitude specifications of interest are



- The stopband attenuation is thus $A_s = -20\log\delta$ dB and the peak ripple is $\varepsilon_{\text{dB}} = 20\log(1 + \delta)$
- For popular window functions, such as rectangular, Bartlett, Hanning, Hamming, and Blackman, the relevant design data is contained in the following table (the Kaiser window function is defined in (7.19))

Table 7.1: Window characteristics for FIR design.

Type	Transition Bandwidth $\Delta\omega$	Minimum Stopband Attenuation	Equivalent Kaiser β
Rectangle	$1.81\pi/M$	21 dB	0
Bartlett	$1.80\pi/M$	25 dB	1.33
Hanning	$5.01\pi/M$	44 dB	3.86
Hamming	$6.27\pi/M$	53 dB	4.86
Blackman	$9.19\pi/M$	74 dB	7.04

The general design steps:

1. Choose the window function, $w[n]$, that just meets the stopband requirements as given in the Table 7.1
2. Choose the filter length, M, (actual length is $M+1$) such that

$$\Delta\omega \leq \omega_s - \omega_p \tag{7.17}$$

3. Choose ω_c in the truncated impulse response such that

$$\omega_c = \frac{\omega_p + \omega_s}{2} \tag{7.18}$$

4. Plot $\left|H(e^{j\omega})\right|$ to see if the specifications are satisfied
5. Adjust ω_c and M if necessary to meet the requirements; If possible reduce M

- By using the Kaiser window method some of the *trial and error* can be eliminated since via parameter β, the stopband attenuation can be precisely controlled, and then the filter length for a desired transition bandwidth can be chosen

Kaiser window design steps:

1. Let $w[n]$ be a Kaiser window, i.e.,

$$w[n] = \begin{cases} I_0\left[\beta\left(1-\left[\dfrac{(n-M/2)}{M}\right]^2\right)^{1/2}\right], & 0 \le n \le M \\ 0, & \text{otherwise} \end{cases} \tag{7.19}$$

2. Chose β for the specified A_s as

$$\beta = \begin{cases} 0.1102(A_s - 8.7), & A_s > 50\text{dB} \\ 0.5842(A_s-21)^{0.4} + 0.07886(A_s-21), & 21 \le A_s \le 50 \\ 0.0, & A_s \le 21 \end{cases} \tag{7.20}$$

3. The window length M is then chosen to satisfy

$$M = \frac{A_s - 8}{2.285\Delta\omega} \tag{7.21}$$

4. The value chosen for ω_c is chosen as before

Optimum Approximations of FIR Filters

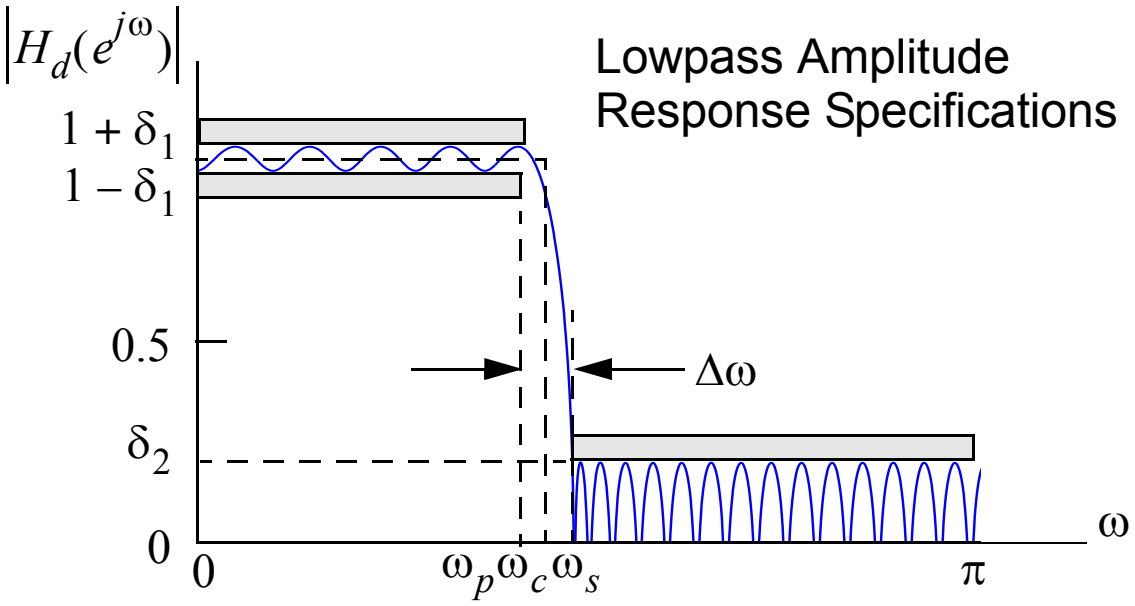
- The window method is simple yet limiting, since we cannot obtain individual control over the approximation errors in each band
- The optimum FIR approach can be applied to type I, II, III, and IV generalized linear phase filters
- The basic idea is to minimize the weighted error between the desired response $H_d(e^{j\omega})$ and the actual response $H(e^{j\omega})$

$$E(\omega) = W(\omega)[H(e^{j\omega}) - H_d(e^{j\omega})] \tag{7.22}$$

- The classical algorithm for performing this minimization in an equiripple sense, was developed by Parks and McClellan in 1972
  - Equiripple means that equal amplitude error fluctuations are introduced in the minimization process
  - In MATLAB the associated function is `firpm()`
- Efficient implementation of the Parks McClellan algorithm requires the use of the *Remez exchange algorithm*
- For a detailed development of this algorithm see Oppenheim and Schafer[1]
- Multiple pass- and stopbands are permitted with this design formulation

1. A. Oppenheim and R. Schafer, *Discrete-Time Signal Processing*, second edition, Prentice Hall, Upper Saddle River, New Jersey, 1999.

- Each band can have its own tolerance, or ripple specification, e.g., in a simple lowpass case we have



- An equiripple design example will be shown later

MATLAB Basic Filter Design Functions

The following function list is a subset of the filter design functions contained in the MATLAB signal processing toolbox useful for FIR filter design. The function groupings match those of the toolbox manual.

Filter Analysis/Implementation	
`y = filter(b,a,x)`	Direct form II filter vector `x`
`[H,w] = freqz(b,a)`	z-domain frequency response computation
`[Gpd,w] = grpdelay(b,a)`	Group delay computation
`h = impz(b,a)`	Impulse response computation
`unwrap`	Phase unwrapping
`zplane(b,a)`	Plotting of the z-plane pole/zero map

Linear System Transformations	
`residuez`	z-domain partial fraction conversion
`tf2zp`	Transfer function to zero-pole conversion
`zp2sos`	Zero-pole to second-order biquadratic sections conversion
`tf2latc`	Transfer function to lattice conversion

FIR Filter Design	
`fir1`	Window-based FIR filter design with standard response
`kaiserord`	Kaiser window FIR filter design estimation parameters with `fir1`

FIR Filter Design (Continued)	
`fir2`	Window based FIR filter design with arbitrary sampled frequency response
`firpm`	Parks-McClellan optimal FIR filter design; optimal (equiripple) fit between the desired and actual frequency responses
`firpmord`	Parks-McClellan optimal FIR filter order estimation

Windowed FIR Design From Amplitude Specifications

$$\omega_p = 0.4\pi \text{ , } \omega_s = 0.6\pi \text{ and } \delta = 0.0032 \Rightarrow A_s = 50 \text{ dB}$$

- A Hamming window or a Kaiser window with a particular β value will achieve the desired stop band attenuation

$$\Delta\omega = 0.2\pi \geq \frac{6.27\pi}{M}$$

or

$$M \geq \lceil 31.35 \rceil = 32$$

- The cutoff frequency is

$$\omega_c = \frac{0.2 + 0.3}{2} \times 2\pi = 0.25 \times 2\pi$$

- Another approach is to use a Kaiser window with

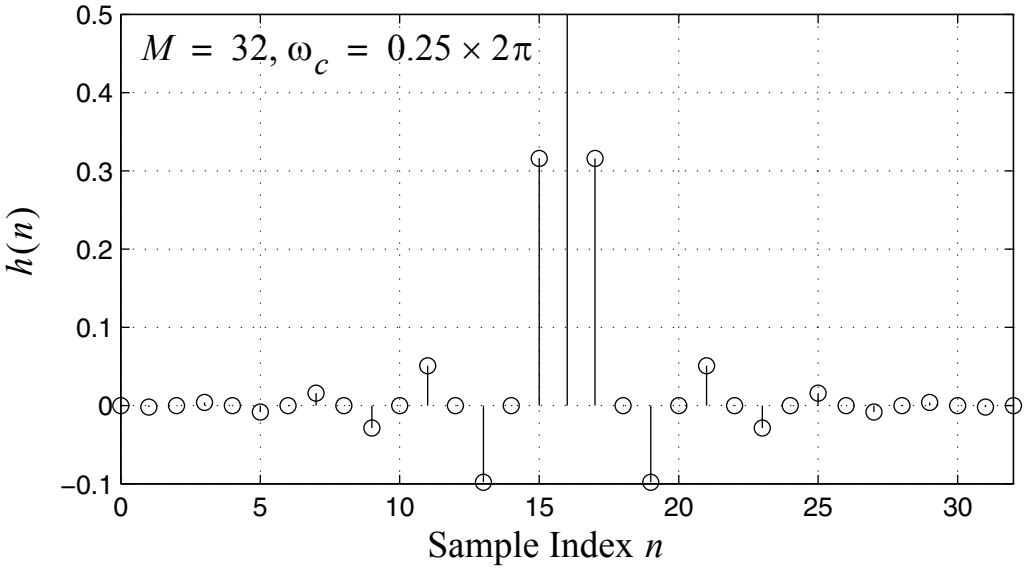
$$\beta = 0.5842(50 - 21)^{0.4} + 0.07886(50 - 21) = 4.5335$$

and

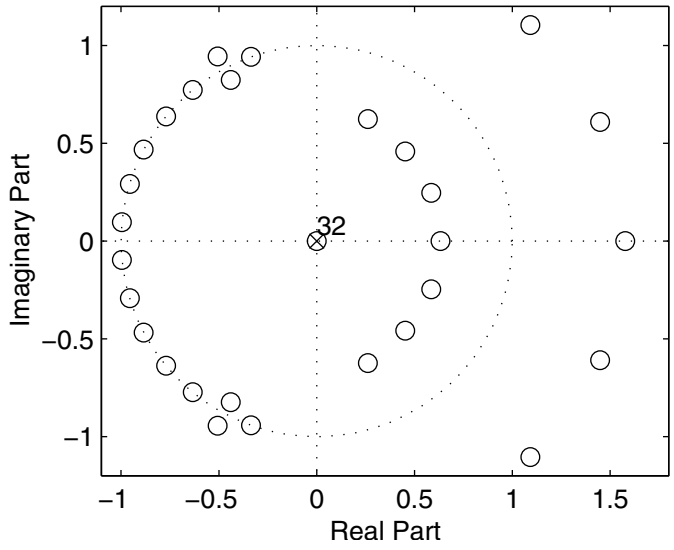
$$M = \left\lceil \frac{50 - 8}{2.285(0.2\pi)} \right\rceil = \lceil 29.25 \rceil = 30$$

- Verification of these designs can be accomplished using the MATLAB filter design function `fir1()`

```
» % Hamming window design
» % (note fir1 uses hamming  by default)
» b = fir1(32,2*.25,hamming(32+1));
» stem(0:length(b)-1,b)
» grid
» zplane(b,1)
» [H,F] = freqz(b,1,512,1);
» plot(F,20*log10(abs(H)))
» axis([0 .5 -70 2])
» grid
» % Kasier window design
» bk = fir1(30,2*.25,kaiser(30+1,4.5335));
» [Hk,F] = freqz(bk,1,512,1);
» plot(F,20*log10(abs(Hk)))
» axis([0 .5 -70 2])
» grid
```



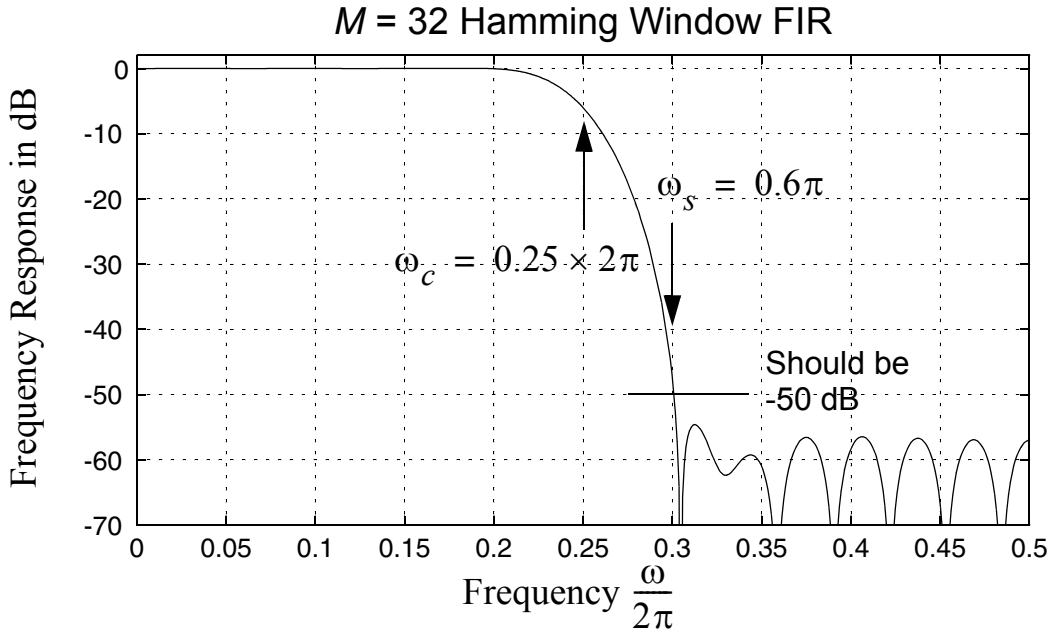
M = 32 impulse response with Hamming window



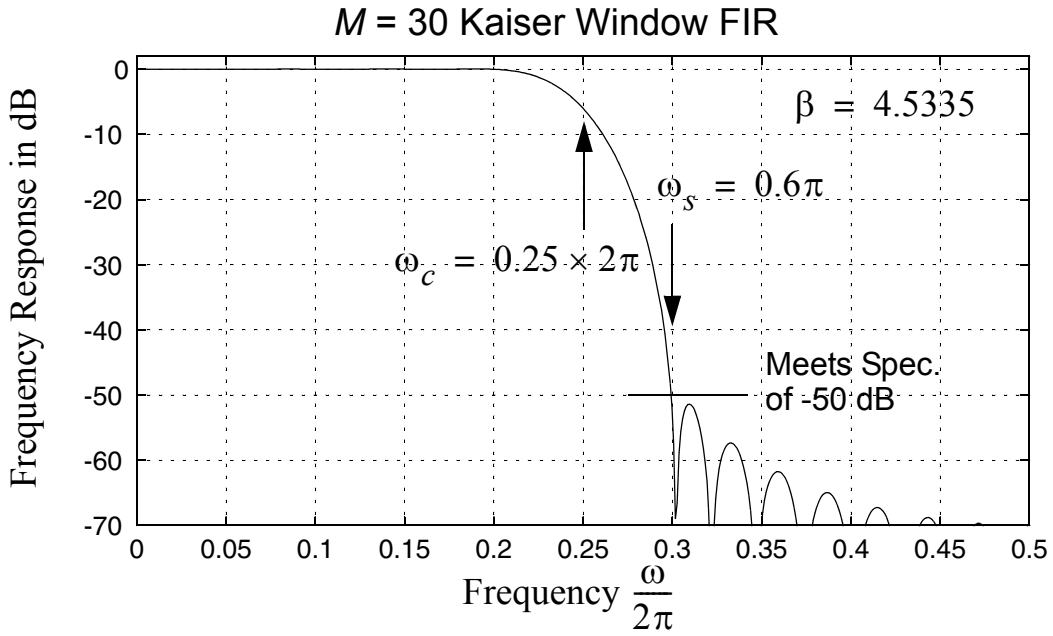
Note: There are few misplaced zeros in this plot due to MATLAB's problem finding the roots of a large polynomial.

Note: The many zero quadruplets that appear in this linear phase design.

M = 32 pole-zero plot with Hamming window



M = 32 frequency response with Hamming window



M = 30 frequency response with Kaiser window

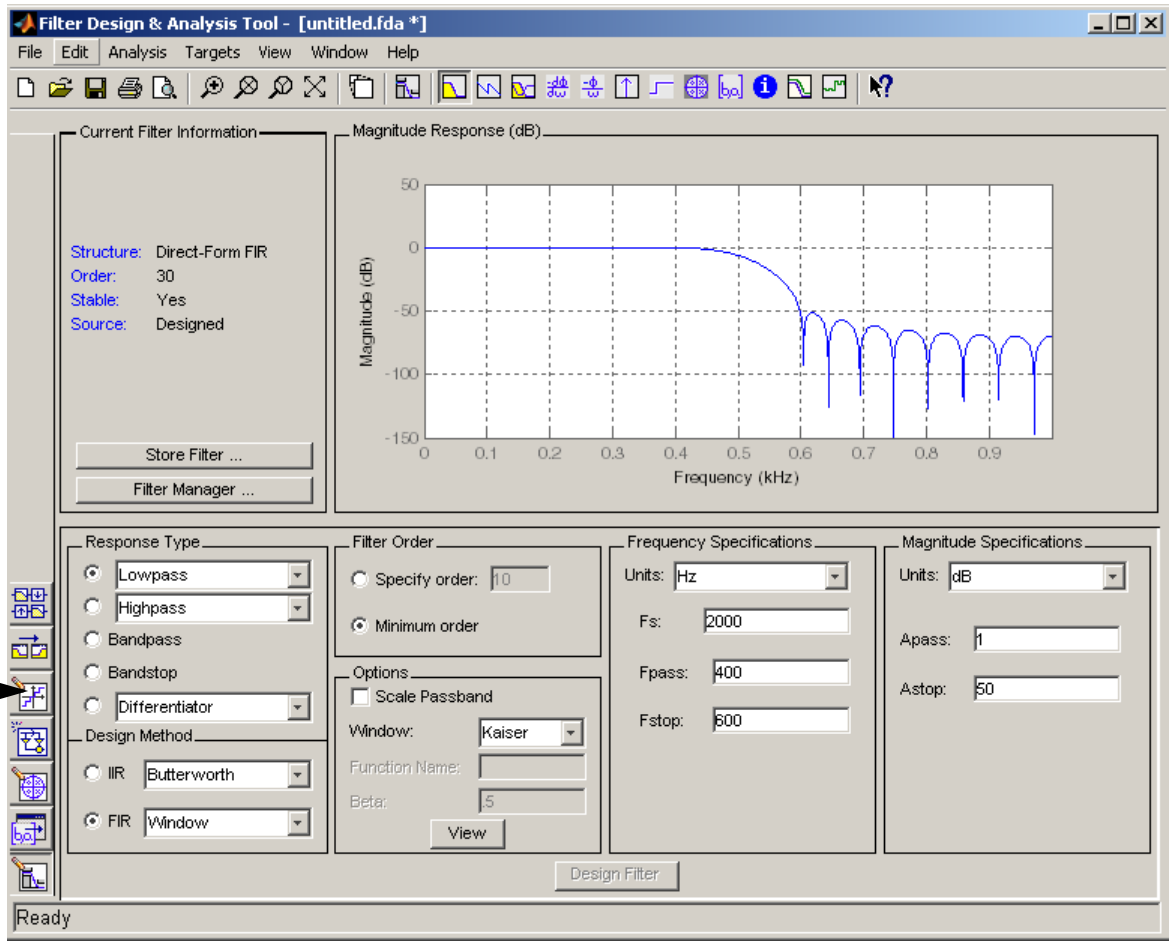
Using MATLAB's `fdatool`

- Digital filter design from amplitude specifications to quantized filter coefficients can be accomplished using the MATLAB tool `fdatool`
  - Note: The basic `fdatool` is included with the signal processing toolbox, and when adding the associated *Fixed-Point Tool Box*, gives one the capability to perform additional filter designs and perform quantization design and analysis

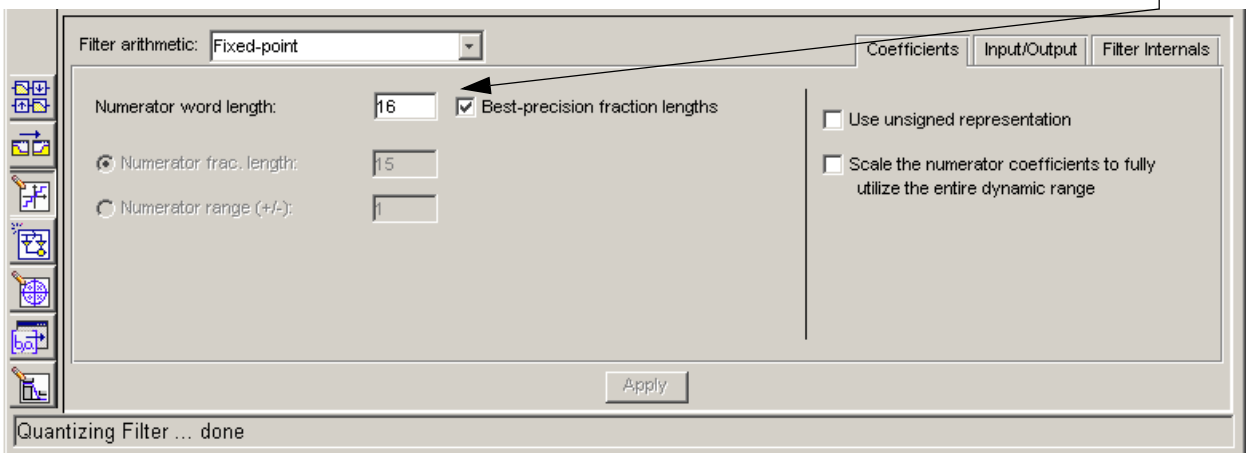
Example: Windowed FIR Design

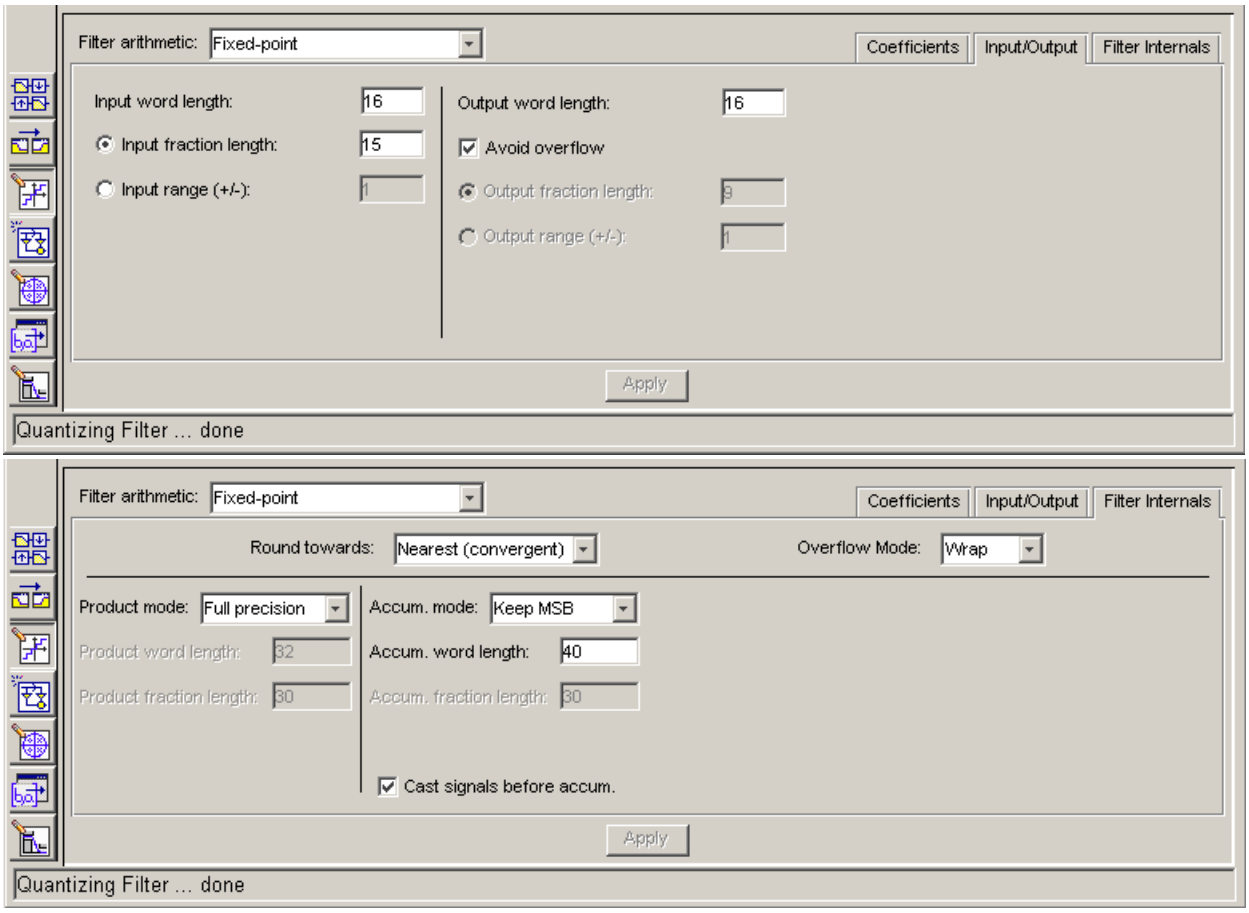
- Consider an example that repeats the previous windowed FIR design using the Kaiser window

- In this example a Kaiser window linear phase FIR filter is designed assuming a sampling rate of 2000 Hz, so the folding frequency is 1000 Hz and the critical frequencies are $f_p = 400\,\text{Hz}$ and $f_s = 600\,\text{Hz}$

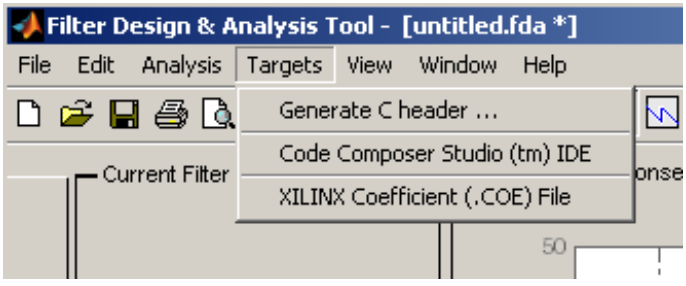


Under *Set Quantization Parameters*, we set coefficient quantize to a Q15 format





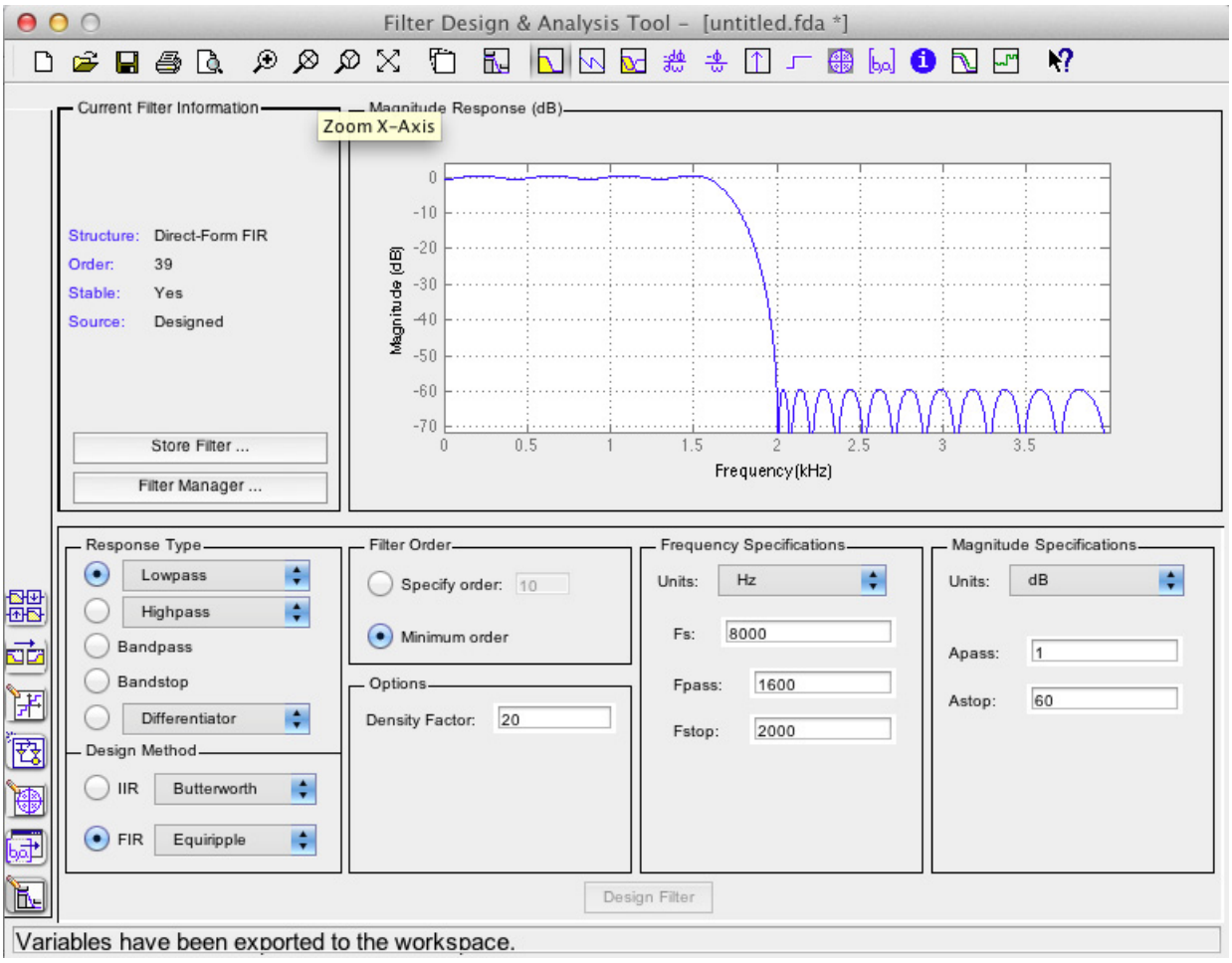
- Many other filter viewing options are available as well
  - Display phase, Gain & phase, Group delay
  - Impulse response, Step response
  - Pole-zero plot
  - Filter coefficients
- Once you are satisfied with a design, the filter coefficients can be exported to a .h file, or to the MATLAB workspace



```
/ * Filter Coefficients (C Source) generated by the Filter Design
and Analysis Tool
 *
 * Generated by MATLAB(R) 7.0.1 and the Filter Design Toolbox 3.1.
 *
 * Generated on: 15-Mar-2005 14:59:26
 *
 */

/*
 * Discrete-Time FIR Filter (real)
 * -------------------------------
 * Filter Structure  : Direct-Form FIR
 * Filter Order      : 30
 * Stable            : Yes
 * Linear Phase      : Yes (Type 1)
 * Arithmetic        : fixed
 * Numerator         : S16Q15
 * Input             : S16Q15
 * Output            : S16Q9
 * Product           : S32Q30
 * Accumulator       : S40Q30
 * Round Mode        : convergent
 * Overflow Mode     : wrap
 * Cast Before Sum   : true
 */

/* General type conversion for MATLAB generated C-code  */
#include "tmwtypes.h"
/*
 * Expected path to tmwtypes.h
 * C:\MATLAB701\extern\include\tmwtypes.h
 */
const int N_FIR = 31;
const int16_T h[31] = {
   -39,     0,    123,    0,   -275,     0,    529,      0,
  -943,     0,   1662,    0,  -3208,     0,  10338,  16384,
 10338,     0,  -3208,    0,   1662,     0,   -943,      0,
   529,     0,   -275,    0,    123,     0,    -39
};
```

Example: Equiripple FIR Design Using float

- As a second example of using FDATool consider an equiripple design using a sampling frequency of 8 kHz to match a AIC3106, $f_p = 1600\,\text{Hz}$, $f_s = 2000\,\text{Hz}$, a passband ripple of $\pm 0.5\,\text{dB}$ (1 dB total), and a stopband attenuation of 60 dB



- Since the lab is no longer equipped with the fixed-point toolbox, we will export the coefficients using the function

```
function filter_C_header(Hq,mode,filename,N)
%        filter_C_header(Hq,mode,filename,N): Used to create a C-style header
file
%       containing filter coefficients. This reads Hq filter objects assuming
a
%        direct-form FIR design is present
```

```
%
%        Hq = quantized filter object containing desired coefficients
%      mode = specify 'fixed', 'float', or 'exponent'
% file_name = string name of file to be created
%         N = number of coefficients per line
```

Writing float C Coefficient Files

An alternative to using FDA tool to write header files, we can use the function

```
function filter_C_header(Hq,mode,filename,N)
%        filter_C_header(Hq,mode,filename,N): Used to create a C-style
header file
%        containing filter coefficients. This reads Hq filter objects
assuming a
%        direct-form FIR design is present
%
%        Hq = quantized filter object containing desired coefficients
%      mode = specify 'fixed', 'float', or 'exponent'
% file_name = string name of file to be created
%         N = number of coefficients per line
```


```
>> filter_C_header(Hd,'float','fir_fltcoeff.h',3)
```

- The result is

```
//define the FIR filter length
#define N_FIR 40
/*****************************************************************/
/*                   Filter Coefficients                         */
float h[N_FIR] = {-0.004314029307,-0.013091321622,-0.016515087727,
                  -0.006430584433, 0.009817876267, 0.010801880238,
                  -0.006567413713,-0.016804829623, 0.000653253913,
                   0.022471280087, 0.010147131468,-0.025657740989,
                  -0.026558960619, 0.023048392854, 0.050385290390,
                  -0.009291203588,-0.087918503442,-0.033770330014,
                   0.187334796517, 0.401505729848, 0.401505729848,
                   0.187334796517,-0.033770330014,-0.087918503442,
                  -0.009291203588, 0.050385290390, 0.023048392854,
                  -0.026558960619,-0.025657740989, 0.010147131468,
                   0.022471280087, 0.000653253913,-0.016804829623,
                  -0.006567413713, 0.010801880238, 0.009817876267,
                  -0.006430584433,-0.016515087727,-0.013091321622,
                  -0.004314029307 };
/*****************************************************************/
```

- Real-time code based on `ISRs_Plus.c` is the following

```
// Welch, Wright, & Morrow,
// Real-time Digital Signal Processing, 2011
// Modified by Mark Wickert February 2012 to include GPIO ISR start/stop postings

////////////////////////////////////////////////////////////////////////
// Filename: ISRs_fir_float.c
//
// Synopsis: Interrupt service routine for codec data transmit/receive
//           floating point FIR filtering with coefficients in *.h file
//
////////////////////////////////////////////////////////////////////////

#include "DSP_Config.h"
#include "fir_fltcoeff.h"   //coefficients in float format

// Function Prototypes
long int rand_int(void);

// Data is received as 2 16-bit words (left/right) packed into one
// 32-bit word.  The union allows the data to be accessed as a single
// entity when transferring to and from the serial port, but still be
// able to manipulate the left and right channels independently.

#define LEFT  0
```

```
#define RIGHT 1

volatile union {
	Uint32 UINT;
	Int16 Channel[2];
} CodecDataIn, CodecDataOut;


/* add any global variables here */
float x_buffer[N_FIR];	//buffer for delay samples


interrupt void Codec_ISR()
////////////////////////////////////////////////////////////////////////
// Purpose:   Codec interface interrupt service routine
//
// Input:     None
//
// Returns:   Nothing
//
// Calls:     CheckForOverrun, ReadCodecData, WriteCodecData
//
// Notes:     None
////////////////////////////////////////////////////////////////////////
{
	/* add any local variables here */
	WriteDigitalOutputs(1); // Write to GPIO J15, pin 6; begin ISR timing pulse
	int i;
	float result = 0; //initialize the accumulator


	if(CheckForOverrun())// overrun error occurred (i.e. halted DSP)
		return;			// so serial port is reset to recover

	CodecDataIn.UINT = ReadCodecData();// get input data samples

	//Work with Left ADC sample
	//x_buffer[0] = 0.25 * CodecDataIn.Channel[ LEFT];
	//Use the next line to noise test the filter
	x_buffer[0] = 0.125*((short) rand_int());//scale input by 1/8

	//Filtering using a 32-bit accumulator
	for(i=0; i< N_FIR; i++)
	{
		result += x_buffer[i] * h[i];
	}
	//Update filter history
```

```
        for(i=N_FIR-1; i>0; i--)
        {
                x_buffer[i] = x_buffer[i-1];
        }

        //Return 16-bit sample to DAC
        CodecDataOut.Channel[ LEFT] = (short) result;
        // Copy Right input directly to Right output with no filtering
        CodecDataOut.Channel[RIGHT] = CodecDataIn.Channel[ RIGHT];
        /* end your code here */
        WriteCodecData(CodecDataOut.UINT);// send output data to  port
        WriteDigitalOutputs(0); // Write to GPIO J15, pin 6; end ISR timing pulse
}

//White noise generator for filter noise testing
long int rand_int(void)
{
        static long int a = 100001;

        a = (a*125) % 2796203;
        return a;
}
```

- In this C routine the filter is implemented using a buffer to hold present and past values of the input in normal order

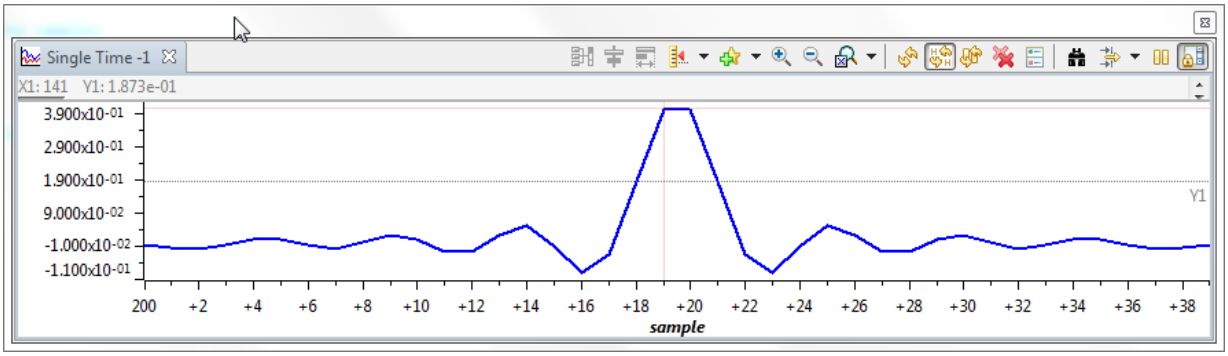
  ```
  float x_buffer[N_FIR];
  ```

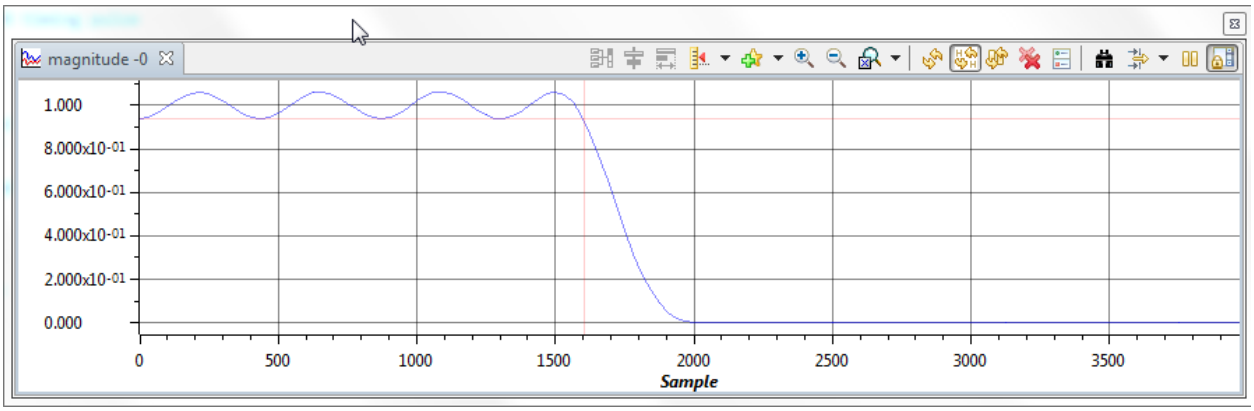
- In particular `x_buffer[0]` holds the present input, `x_buffer[1]` holds the past input and so on
- The filter coefficients are held in the short array `h[]` as a_0= `h[0]`... a_{39}= `h[39]`
- The filter output is computed by placing

  ```
  result += x_buffer[i] * h[i];
  ```

  in a loop running from `i=0` to `i=N_FIR-1`, here `N_FIR` = 40

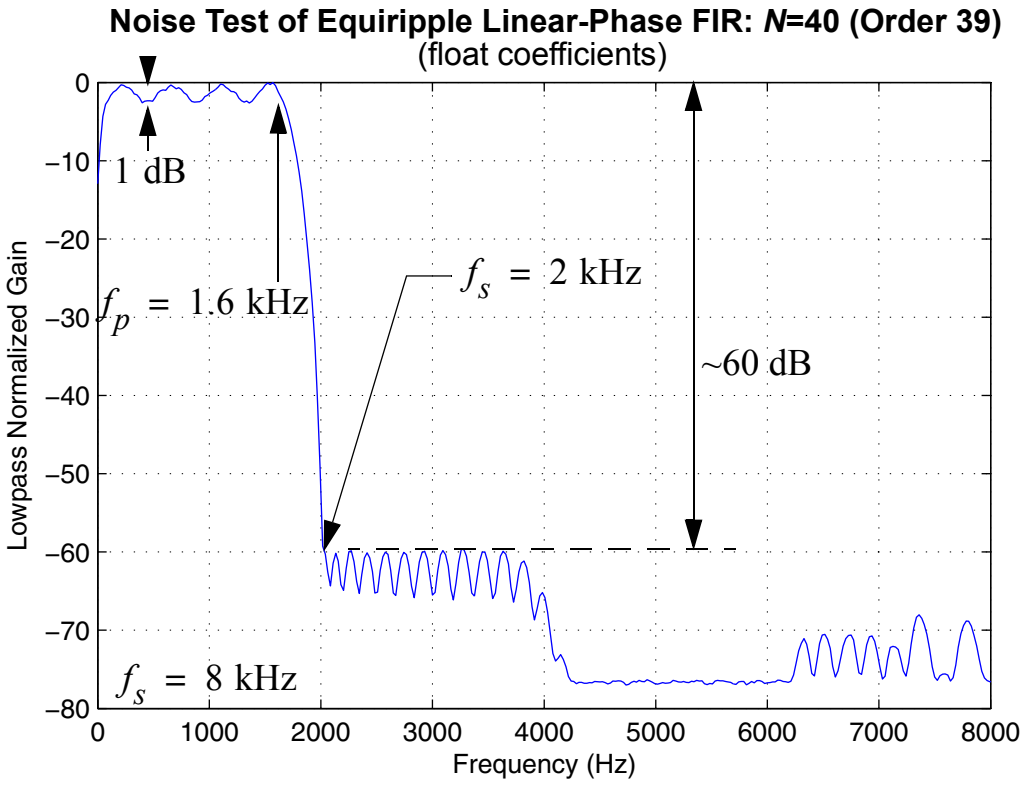
- Once the program has been loaded and run, we can verify that the `float` coefficients contained in `fir_flt coeff.h` are loaded into memory properly using the graph capability of CCS



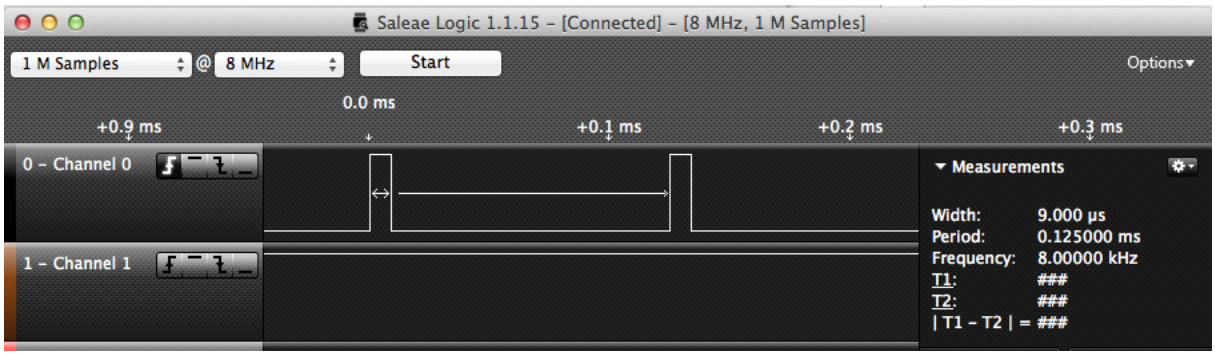


- A 30 second sound file created at 44.1 ksps, 16-bits, was imported into MATLAB and then the power spectrum was estimated using `psd()`

```
>> [Px,F] = simpleSA(x(:,1),2^11,48000,-90,10,'b');
>> plot(F,10*log10(Px/max(Px)))
```



ISR Timing Results



- We see that the ISR running the 40 tap FIR requires a total of 9 µs
- In the limit this says when sampling at 8 ksps a maximum of $125 \times 40/9 \approx 555$ taps can be implemented

Code Enhancements

- The use of C intrinsics would allow us to utilize parallel multiply accumulate operations
  - This idea applies to word (32-bit) operations for the fixed filter and double word operations on the C67 for the float case
- We have not yet used a *circular buffer*, which is part of the C6x special hardware capabilities

A Simple Fixed-Point Implementation

- Simple modification to the previous float FIR example yields a version using short integers

```
// Welch, Wright, & Morrow,
// Real-time Digital Signal Processing, 2011
// Modified by Mark Wickert February 2012 to include GPIO ISR start/stop postings

////////////////////////////////////////////////////////////////////////
// Filename: ISRs_fir_short.c
//
// Synopsis: Interrupt service routine for codec data transmit/receive
//           fixed point FIR filtering with coefficients in *.h file
//
////////////////////////////////////////////////////////////////////////

#include "DSP_Config.h"
#include "fir_fixcoeff.h"   //coefficients in decimal format

// Function Prototypes
long int rand_int(void);

// Data is received as 2 16-bit words (left/right) packed into one
// 32-bit word.  The union allows the data to be accessed as a single
// entity when transferring to and from the serial port, but still be
// able to manipulate the left and right channels independently.

#define LEFT  0
#define RIGHT 1
```

```
volatile union {
	Uint32 UINT;
	Int16 Channel[2];
} CodecDataIn, CodecDataOut;


/* add any global variables here */
short x_buffer[N_FIR];	//buffer for delay samples

interrupt void Codec_ISR()
////////////////////////////////////////////////////////////////////////
// Purpose:   Codec interface interrupt service routine
//
// Input:     None
//
// Returns:   Nothing
//
// Calls:     CheckForOverrun, ReadCodecData, WriteCodecData
//
// Notes:     None
////////////////////////////////////////////////////////////////////////
{
	/* add any local variables here */
	WriteDigitalOutputs(1); // Write to GPIO J15, pin 6; begin ISR timing pulse
	int i;
	int result = 0; //initialize the accumulator


	if(CheckForOverrun())// overrun error occurred (i.e. halted DSP)
		return;			// so serial port is reset to recover

	CodecDataIn.UINT = ReadCodecData();// get input data samples

	/* add your code starting here */
	//Work with Left ADC sample
	//x_buffer[0] = 0.25 * CodecDataIn.Channel[ LEFT];
	//Use the next line to noise test the filter
	x_buffer[0] = ((short) rand_int())>>3;//scale input by 1/8

	//Filtering using a 32-bit accumulator
	for(i=0; i< N_FIR; i++)
	{
		result += x_buffer[i] * h[i];
	}
	//Update filter history
	for(i=N_FIR-1; i>0; i--)
```

```
	{
		x_buffer[i] = x_buffer[i-1];
	}

	//Return 16-bit sample to DAC; recall result is a 32 bit accumulator
	CodecDataOut.Channel[ LEFT] = (short) (result >> 15);
	// Copy Right input directly to Right output with no filtering
	CodecDataOut.Channel[RIGHT] = CodecDataIn.Channel[ RIGHT];
	/* end your code here */

	WriteCodecData(CodecDataOut.UINT);// send output data to  port
	WriteDigitalOutputs(0); // Write to GPIO J15, pin 6; end ISR timing pulse
}

//White noise generator for filter noise testing
long int rand_int(void)
{
	static long int a = 100001;

	a = (a*125) % 2796203;
	return a;
}
```

Writing fixed C Coefficient Files

```
>> filter_C_header(Hd,'fixed','fir_fixcoeff.h',8)
```

```
//define the FIR filter length
#define N_FIR 40
/*****************************************************************/
/*                   Filter Coefficients                         */
short h[N_FIR] = { -141, -429, -541, -211,  322,  354, -215, -551,
                     21,  736,  333, -841, -870,  755, 1651, -304,
                  -2881,-1107, 6139,13157,13157, 6139,-1107,-2881,
                   -304, 1651,  755, -870, -841,  333,  736,   21,
                   -551, -215,  354,  322, -211, -541, -429, -141 };
/*****************************************************************/
```

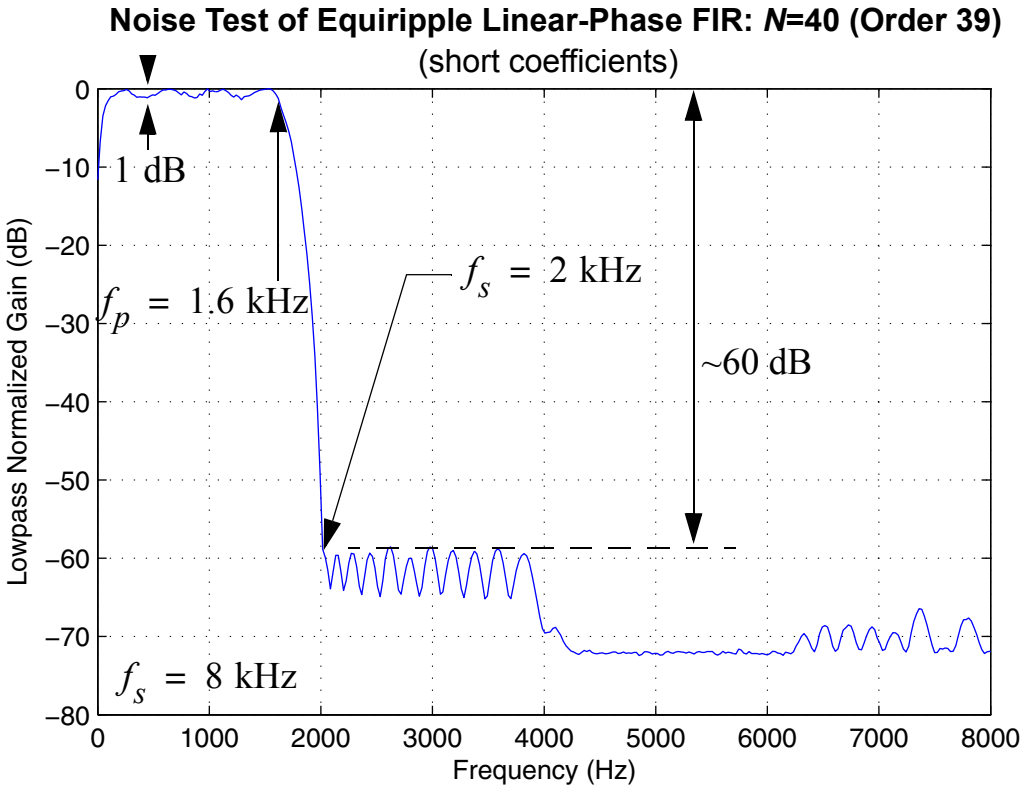
- The filter was tested with the OMAP-L138 board's AIC3106 and a white noise input generated in software

```
x_buffer[0] = ((short) rand_int())>>3;//scale input by 1/8
```

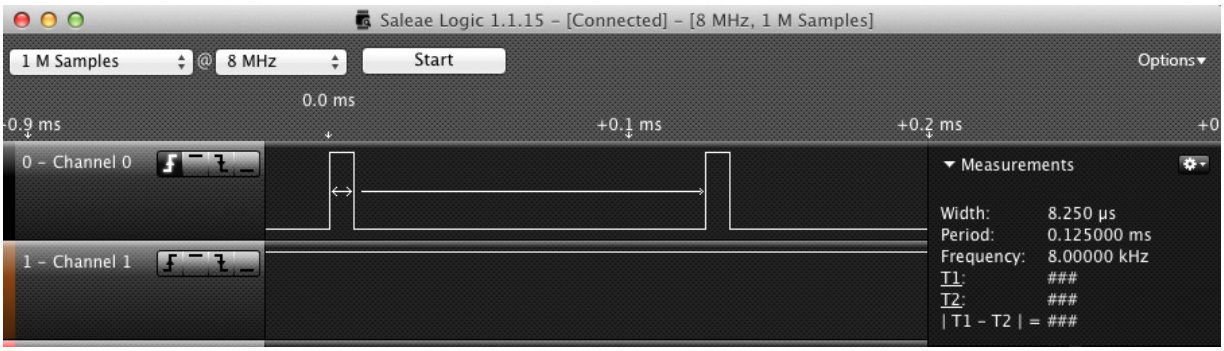
- A 30 second sound file created at 44.1 ksps, 16-bits, was imported into MATLAB and then the power spectrum was

estimated using `psd()`

```
>> [Px,F] = simpleSA(x(:,1),2^11,48000,-90,10,'b');
>> plot(F,10*log10(Px/max(Px)))
```



ISR Timing Results

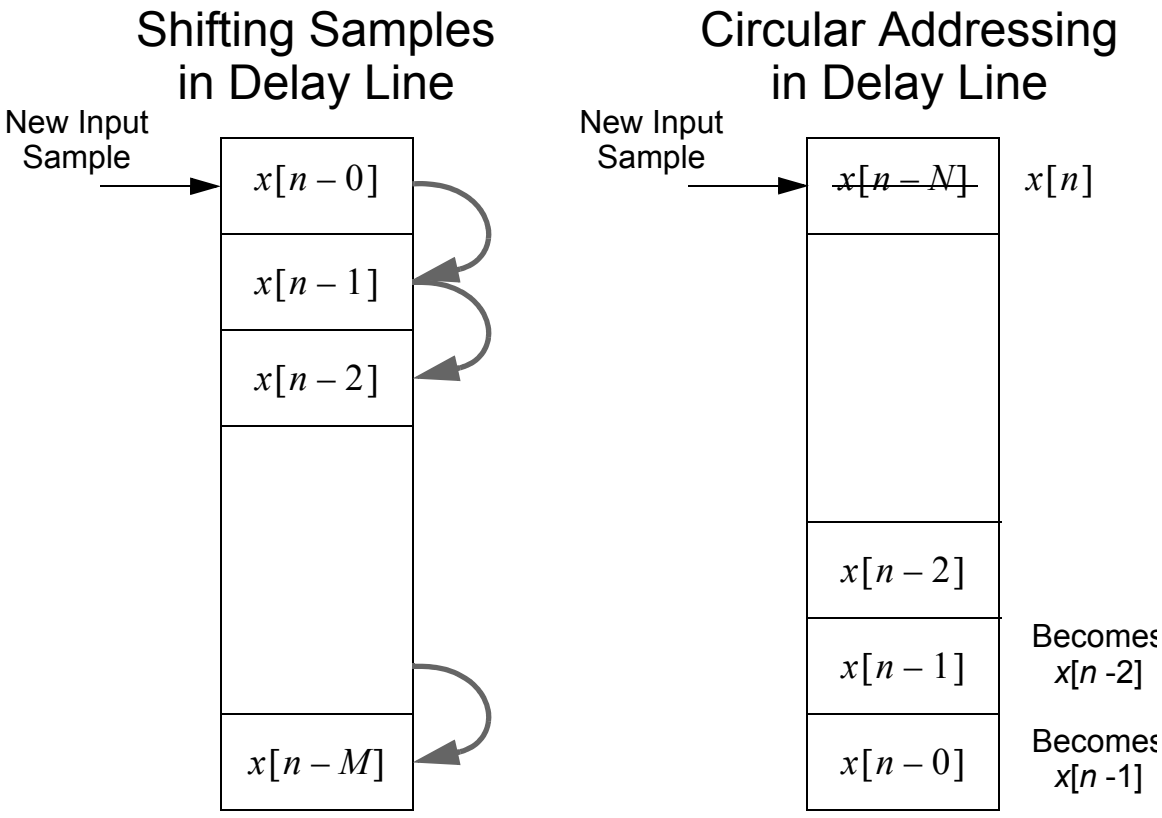


- We see that the ISR running the 40 tap FIR requires a total of 8.25 µs, which is slightly faster than the `float` version

Circular Addressing

Circular Buffer in C

- Up to this point the filter memory or history was managed using a linear buffer array in the fixed and float example routines given earlier
- A more efficient implementation of this buffer (tapped delay line) is to simply overwrite the oldest value of the array with the newest values
- Proper access to the data is accomplished with a pointer that wraps modulo the buffer size



- See text Section 3.4.3

- See also TI *Application Report* SPRA645–June 2002

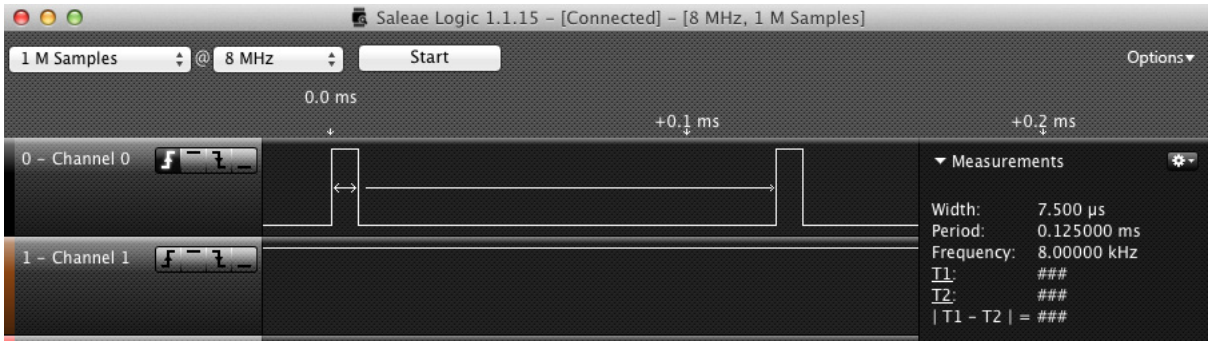
C Circular Buffer

- A circular buffer can be directly implemented in C
- The code given below is a modification of the program `ISRs_fir_short_circ.c`:

```
/* add any global variables here */
short x_buffer[N_FIR];       //buffer for delay samples
short newest, x_index;   //circular buffer index control

interrupt void Codec_ISR()
////////////////////////////////////////////////////////////////////////
// Purpose:   Codec interface interrupt service routine
//
// Input:     None
//
// Returns:   Nothing
//
// Calls:     CheckForOverrun, ReadCodecData, WriteCodecData
//
// Notes:     None
////////////////////////////////////////////////////////////////////////
{
      /* add any local variables here */
      WriteDigitalOutputs(1); // Write to GPIO J15, pin 6; begin ISR timing pulse
      int i;
      int result = 0; //initialize the accumulator


      if(CheckForOverrun())// overrun error occurred (i.e. halted DSP)
            return;                    // so serial port is reset to recover

      CodecDataIn.UINT = ReadCodecData();// get input data samples

      /* add your code starting here */
      //circularly increment newest
      ++newest;
      if(newest == N_FIR) newest = 0;
      //Put new sample in circular buffer from ADC
      //Work with Left ADC sample
      //x_buffer[newest] = 0.25 * CodecDataIn.Channel[ LEFT];
      //Use the next line to noise test the filter
```

```
        x_buffer[newest] = ((short) rand_int())>>3;//scale input by 1/8
        //Filtering using a 32-bit accumulator
        x_index = newest;
        for(i=0; i< N_FIR; i++)
        {
                result += h[i]*x_buffer[x_index];
                //circularly decrement x_index
                --x_index;
                if(x_index == -1) x_index = N_FIR-1;
        }

        //Return 16-bit sample to DAC; recall result is a 32 bit accumulator
        CodecDataOut.Channel[ LEFT] = (short) (result >> 15);
        // Copy Right input directly to Right output with no filtering
        CodecDataOut.Channel[RIGHT] = CodecDataIn.Channel[ RIGHT];
        /* end your code here */

        WriteCodecData(CodecDataOut.UINT);// send output data to  port
        WriteDigitalOutputs(0); // Write to GPIO J15, pin 6; end ISR timing pulse
}
```

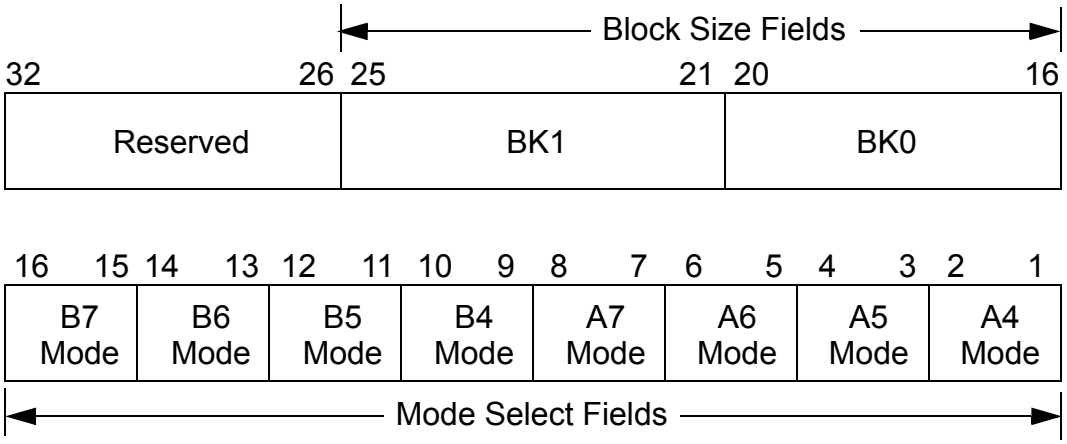
- Notice that C's mod function `x%y` is not used here since it is less efficient than the if statement which does the circular wrapping
- A timing comparison is done using a logic analyzer:



- We see the interrupt time is now 7.5 µs, which is faster by 0.75 µs, compared with the earlier fixed-point FIR

Circular Buffering via C Calling Assembly

- The C6x has underlying hardware to handle circular buffer setup and manipulation
- In the Chassaing text the programs `FIRcirc.c`, and when using external memory for the filter coefficients, `FIRcirc_ext.c`, take advantage of the C6x circular addressing mode
  - Details on the use of hardware circular addressing can be found in Chassaing Example 4.14 and Appendix B
- Two circular buffers can be defined using `BK0` and `BK1` in the address mode register (`AMR`)



- The size of the circular buffers, in bytes, is set into the block size fields as 2^{N+1}, e.g., for 128 bytes (64 short values), $N = 6$ or for `BK0` bits 16–20 would be `00110b`
- The `AMR` is also used to configure the eight register pointers `A4`–`A7` and `B4`–`B7`

- The two mode select bits are set as shown in the table below

Table 7.2: AMR mode register settings.

Mode	Description
00	Linear addressing, the default on reset
01	Circular addressing using BK0
10	Circular addressing using BK1
11	Reserved

- When returning from an assembly routine that uses circular addressing, the AMR setting must be returned to the linear value
- We now consider fir_asmcirc.c and the associated assembly file FIRcircfunc.asm

```
//A circular buffer FIR real-time filtering program
//A modified version of Fir_pcm.c using the AIC23

//project fir_asmcirc.pjt
//fir_short_asmcirc_AIC23.c FIR

int fircircfunc(short x, short *h, int N);

//#include "fir_hcoeff.h"   //N_FIR=40 coeff. in hex
// Assembly assumes 128 taps or some power of 2
#include "bp1750.cof"       //N=128 BP coeff. in decimal
                            //f0 = 1750 at fs = 8k

int rand_int(void);
```

```
#include "DSK6713_AIC23.h"
//set sampling rate {8,16,24,32,44.1,48,96}
Uint32 fs=DSK6713_AIC23_FREQ_8KHZ;
#define DSK6713_AIC23_INPUT_MIC 0x0015
#define DSK6713_AIC23_INPUT_LINE 0x0011
//select input
Uint16 inputsource=DSK6713_AIC23_INPUT_LINE;

int result = 0;         //initial filter output

interrupt void c_int11()  //ISR
{
    short sample_data;

    //Get sample from ADC
    sample_data = input_left_sample();
    //Use the next line to noise test the filter
    //sample_data =  ((short) rand_int())>>1;

    //Filtering using a 32-bit accumulator
    result = fircircfunc(sample_data, h, N);

    //Return 16-bit sample to DAC
    output_left_sample((short) (result >> 15));

  return;         //return from ISR
}

void main()
{
  comm_intr();                 //init DSK, codec, McBSP
  while(1);                   //infinite loop
}
...
```

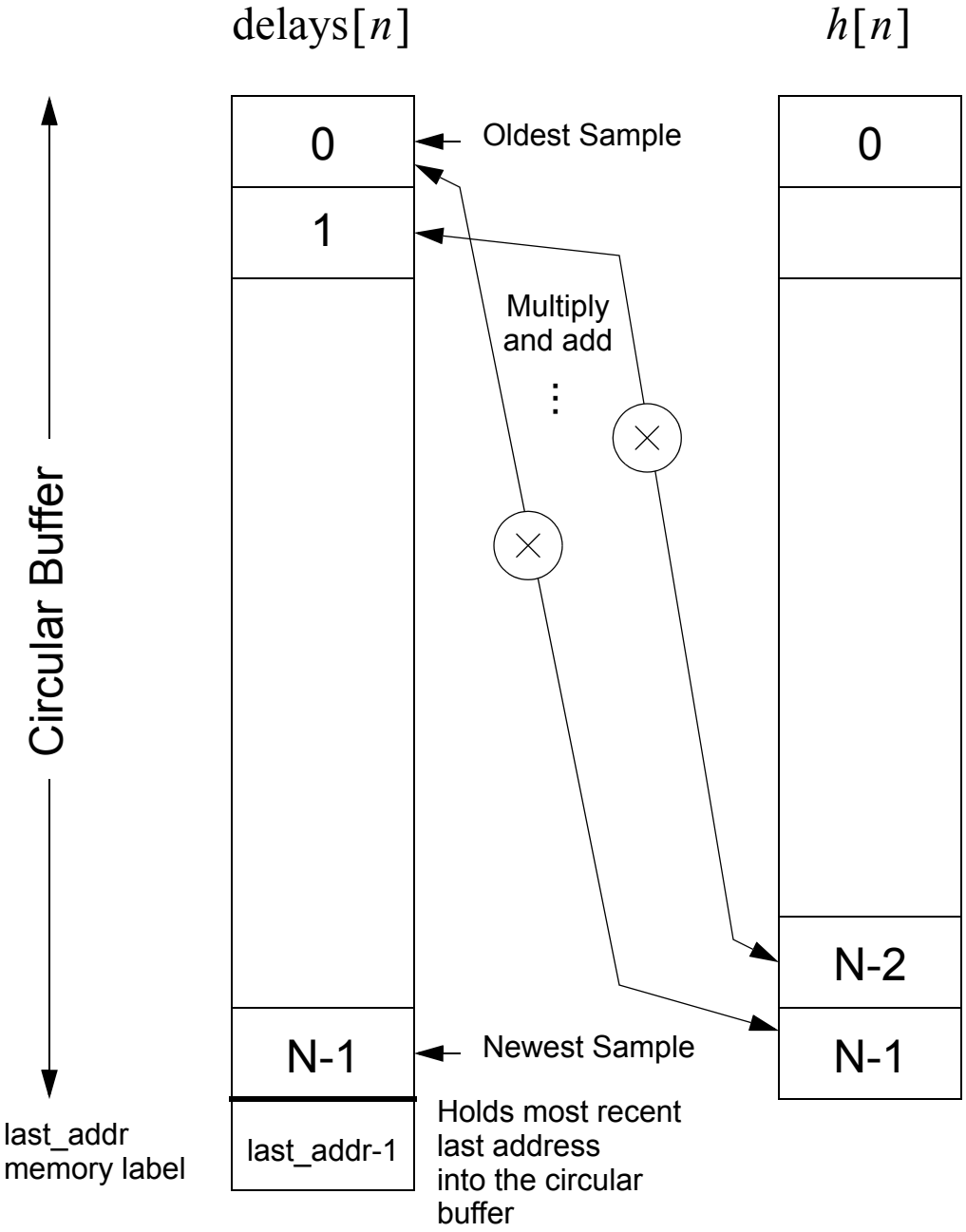
```
;FIRcircfunc.asm ASM function called from C using
;circular addressing. A4=newest sample,
;B4=coefficient address, A6=filter order
;Delay samples organized: x[n-(N-1)]...x[n]
;Coeff as h(0)...h[N-1]

            .def  _fircircfunc
            .def   last_addr
            .def   delays
            .sect "circdata"   ;circular data section
            .align 256    ;align delay buffer
                          ;256-byte boundary
delays      .space 256    ;init 256-byte
                          ;buffer with 0's
last_addr   .int   last_addr-1 ;point to bottom of
                               ;delays buffer
            .text       ;code section
_fircircfunc:       ;FIR function using circ addr
            MV    A6,A1   ;setup loop count
            MPY   A6,2,A6 ;since dly buffer data as
                          ;byte
            ZERO A8    ;init A8 for accumulation
            ADD   A6,B4,B4  ;since coeff buffer
                            ;data as bytes
            SUB   B4,1,B4 ;B4=bottom coeff array h[N-1]

            MVKL 0x00070040,B6 ;select A7 as pointer
                               ;and BK0
            MVKH 0x00070040,B6 ;BK0 for 256 bytes
                               ;(128 shorts)
            MVC  B6,AMR  ;set address mode reg. AMR
            MVK  last_addr,A9  ;A9=last circ addr
                               ;(lower 16 bits)
            MVKH last_addr,A9  ;last circ addr
```

```
                                    ;(higher 16 bits)
            LDW   *A9,A7            ;A7=last circ addr
            NOP   4
            STH   A4,*A7++          ;newest sample-->last
                                    ;address
loop:                                ;begin FIR loop
            LDH   *A7++,A2          ;A2=x[n-(N-1)+i]
                                    ;i=0,1,...,N-1
    ||      LDH   *B4--,B2          ;B2=h[N-1-i]
                                    ;i=0,1,...,N-1
            SUB   A1,1,A1           ;decrement count

    [A1]    B     loop    ;branch to loop if count # 0
            NOP   2
            MPY   A2,B2,A6  ;A6=x[n-(N-1)+i]*h[N-1+i]
            NOP
            ADD   A6,A8,A8  ;accumulate in A8

            STW   A7,*A9    ;store last circ addr to
                            ;last_addr
            B     B3     ;return addr to calling routine
            MV    A8,A4     ;result returned in A4
            NOP   4
```

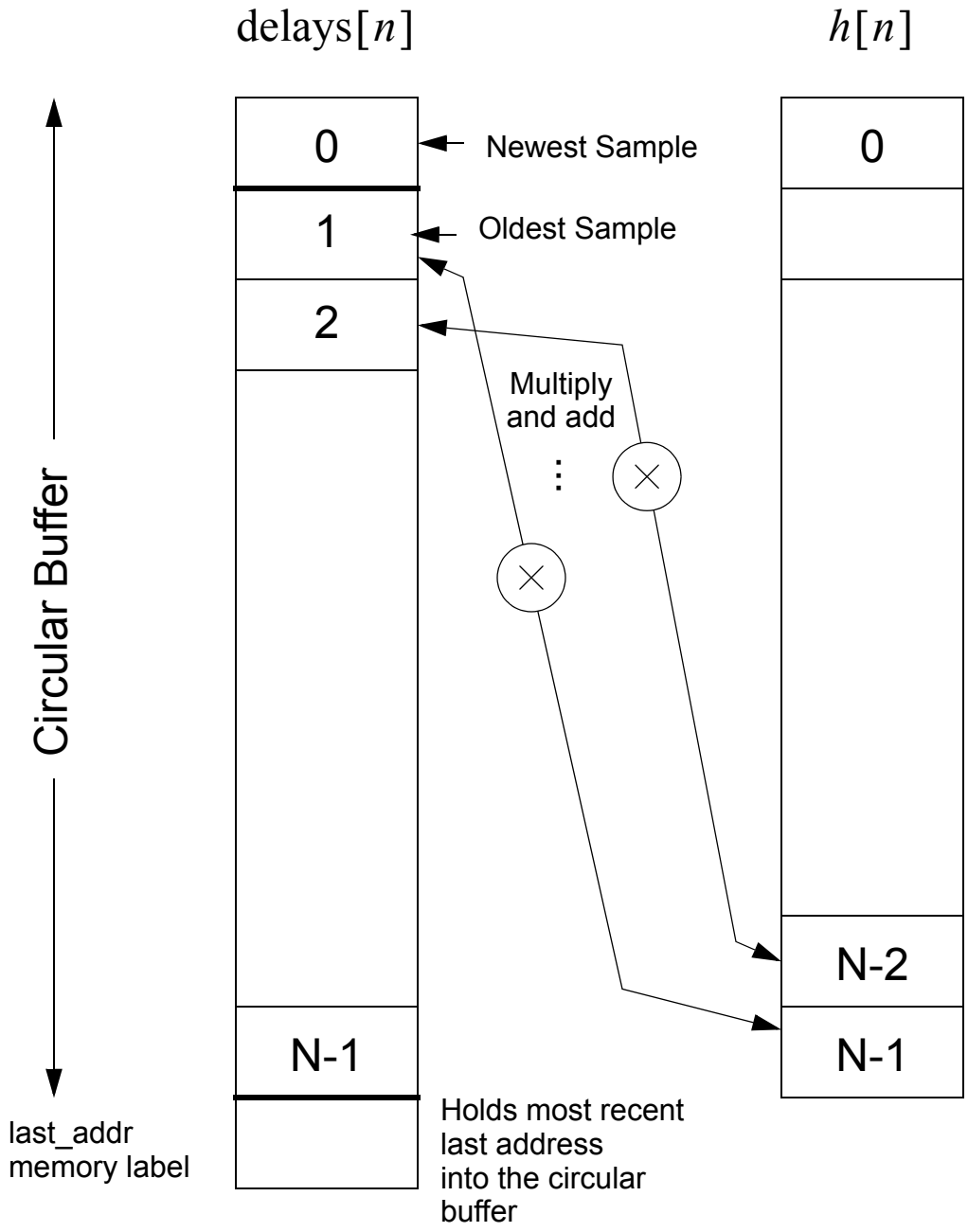
- Looking at the assembly module we see that a block of 256 bytes or 128 short integers is reserved for the circular buffer
  - As currently configured the circular buffer must be a power of 2, so the filter coefficient set used in a previous example will not work, since here `N_FIR = 40`

- Once the circular buffer is configured via the AMR settings, all we need to maintain from sample-to-sample is the location of the current newest signal sample; here we use memory location label `last_addr`

On First Pass Through Algorithm



Second Pass Through Algorithm

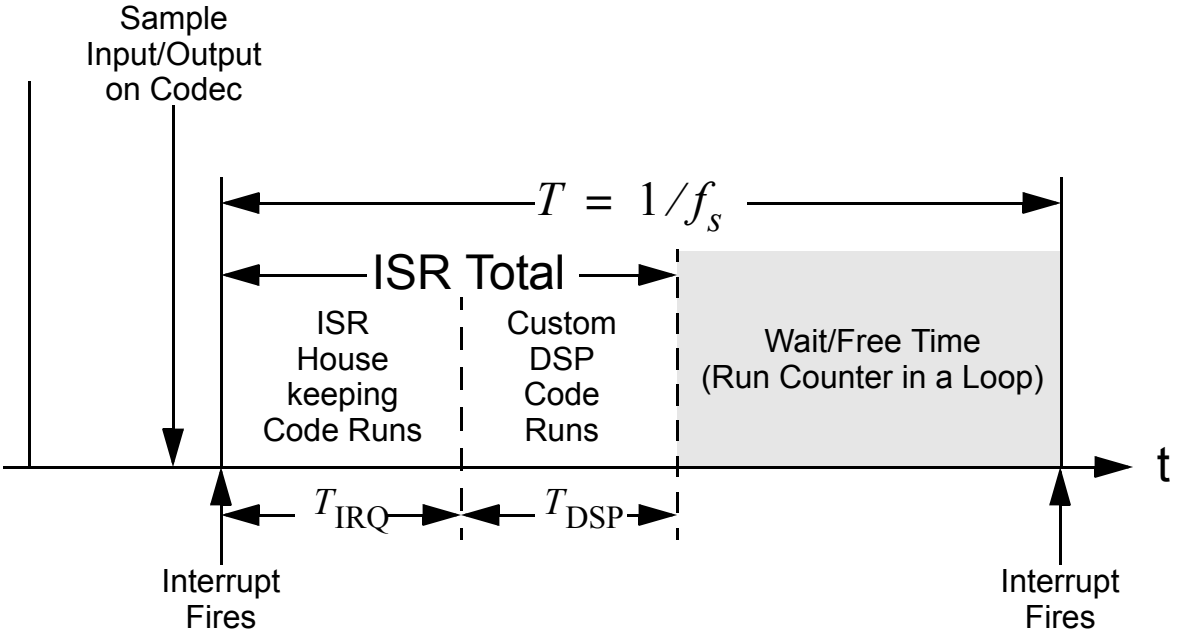


- The process continues with newest sample location moving around the buffer in a circular fashion

Performance of Interrupt Driven Programs

- In the development of real-time signal processing systems, there are clearly performance trade-offs to be made
- The character of all the interrupt driven programs we have seen thus far is that:
  - We first load system parameters into memory, e.g., filter coefficients
  - Next registers are initialized on the CPU and serial port is configured
  - The appropriate interrupt is enabled on the CPU
  - The program then enters an infinite (*wait*) loop
  - While in this infinite loop the CPU is *interrupted* at the sampling rate of f_s samples per second by the AIC23, which has new analog I/O samples to exchange
  - Before returning to the *wait* loop again, the program executes code that carries out the desired signal processing tasks (Note: here we assume for simplicity that there are no other background signal processing tasks to be performed)
- In order for the DSP application to maintain true real-time status, all signal processing computations must be completed before the next codec initiated interrupt is fired
- Debugging interrupt driven programs is not a simple matter

- Pictorially we are doing the following:



The Wait Loop Counter Scheme

C5515 Code Example

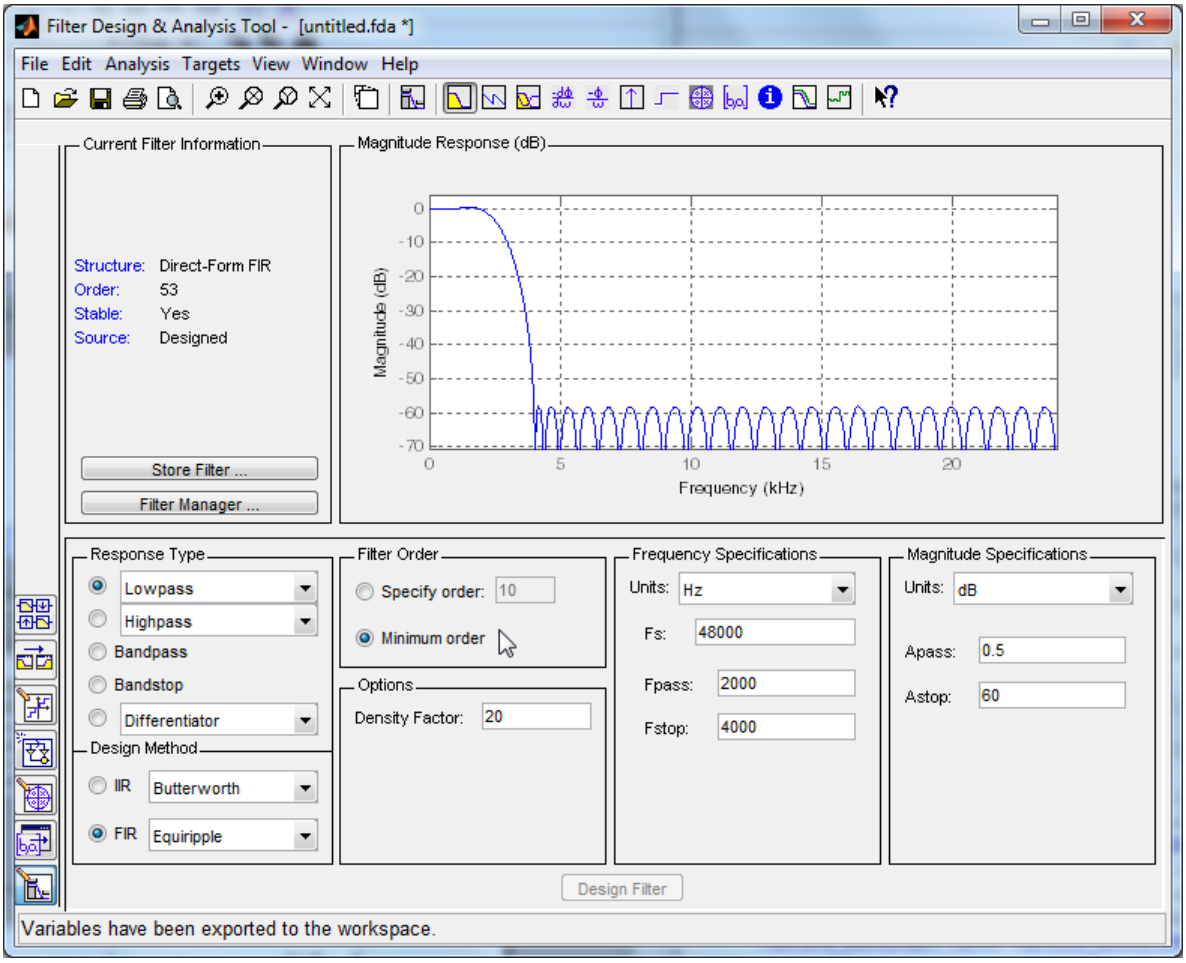
- The C5515 is a fixed-point o processor, so in this FIR filter example we port the earlier OMAP-L138 fixed-point example
- The data types on the VC5505/C5515 are slightly different, so first we consider the differences

Type	Size	Representation	Minimum Value	Maximum Value
char, signed char	16 bits	ASCII	–32 768	32 767
unsigned char	16 bits	ASCII	0	65 535
short, signed short	16 bits	2s complement	–32 768	32 767
unsigned short	16 bits	Binary	0	65 535
int, signed int	16 bits	2s complement	–32 768	32 767
unsigned int	16 bits	Binary	0	65 535
long, signed long	32 bits	2s complement	–2 147 483 648	2 147 483 647
unsigned long	32 bits	Binary	0	4 294 967 295
long long	40 bits	2s complement	–549 755 813 888	549 755 813 887
unsigned long long	40 bits	Binary	0	1 099 511 627 775
enum	16 bits	2s complement	–32 768	32 767
float	32 bits	IEEE 32-bit	1.175 494e–38	3.40 282 346e+38
double	32 bits	IEEE 32-bit	1.175 494e–38	3.40 282 346e+38
long double	32 bits	IEEE 32-bit	1.175 494e–38	3.40 282 346e+38
pointers (data) small memory mode large memory mode	 16 bits 23 bits	Binary	0	 0xFFFF 0x7FFFFF
pointers (function)	24 bits	Binary	0	0xFFFFFF

  - **Note** in particular that `int` is only 16 bits and long is 32 bits
  - There are 40-bit registers available on the processor
  - There are also two MAC units that support 17×17 bit

multiplies

- Design the filter using `fdatool`



- In a polling-based AIC3204 program, we insert a modified version of the OMAP-L138 fixed-point routine presented earlier

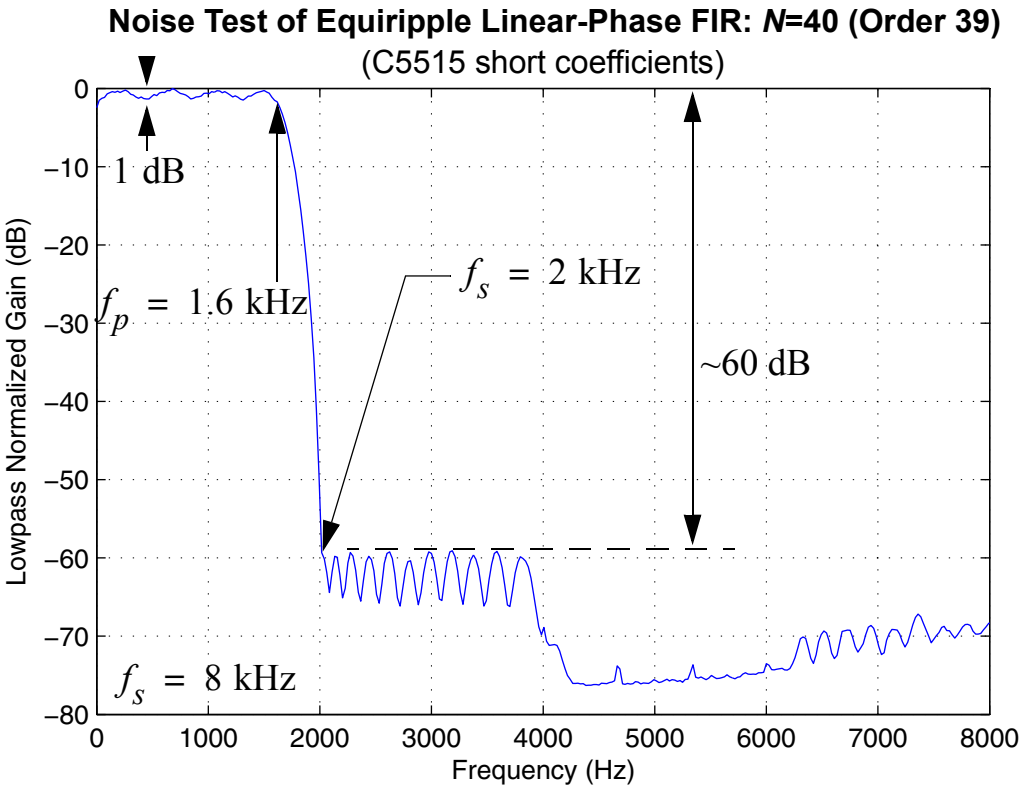
```
//
// main.c in C5515_AIC_3204_polling_FIR
// Mark Wickert March 2012
//
// Codec Library Functions
#include "aic3204.h"
#include "fir_fixcoeff.h"   //coefficients in decimal format

// Function Prototypes
long int rand_int(void);
```

```
short x_buffer[N_FIR];       //buffer for delay samples

/* Sampling Rates:
 * AIC3204_FS_8KHZ
 * AIC3204_FS_16KHZ
 * AIC3204_FS_24KHZ
 * AIC3204_FS_32KHZ
 * AIC3204_FS_48KHZ
 * AIC3204_FS_96KHZ
 */
short fs = AIC3204_FS_8KHZ;

void main(void)
{
    //Define working variables
    short i = 0;
    short l_chan, r_chan;
    short buffer[128];

    //Define filter variables
    long result; //32 bit accumulator

    // Initialize Polling
    comm_poll();

      while(1)
      {
            // Get sample using inputs
            input_sample(&l_chan, &r_chan);

            // Get random sample
            r_chan = ((short) rand_int())>>3;

            /* Fill buffer */
            buffer[i] = r_chan;
            i += 1;
            if (i == 128) i = 0;

            // FIR Filter
            x_buffer[0] = r_chan;
            result = 0;
            for(i=0; i< N_FIR; i++)
            {
                  result += (long) x_buffer[i] * h[i];
            }
            //Update filter history
            for(i=N_FIR-1; i>0; i--)
```

```
            {
                  x_buffer[i] = x_buffer[i-1];
            }

            /* Write Digital audio input */
            output_sample(l_chan, (short)(result>>16));
            //output_sample(l_chan, r_chan);
      }
}

//White noise generator for filter noise testing
long int rand_int(void)
{
      static long int a = 100001;

      a = (a*125) % 2796203;
      return a;
}

// fir_fixcoeff.h
//define the FIR filter length
#define N_FIR 40
/*************************************************************/
/*                  Filter Coefficients                      */
short h[N_FIR] = { -141, -429, -541, -211,  322,  354, -215, -551,
                    21,  736,  333, -841, -870,  755, 1651, -304,
                 -2881,-1107, 6139,13157,13157, 6139,-1107,-2881,
                  -304, 1651,  755, -870, -841,  333,  736,   21,
                  -551, -215,  354,  322, -211, -541, -429, -141 };
/*************************************************************/
```

- Audio noise capture of a 2 kHz lowpass filter, assuming a sampling rate of 48 kHz

```
>> [Px,F] = simpleSA(x(:,2),2^11,48000,-90,10,'b');
>> plot(F,10*log10(Px/max(Px)))
>> axis([0 8000 -80 0])
```



- The results look similar to the earlier OMAP-L138 AIC3106 when using the same fixed-point coefficient set