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17085 DSP

Implementing DSP on the PIC32

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Class Objectives

When you walk out of this class you will….

- **Create a DSP application for the PIC32**
- **Design and simulate DSP elements with free, multi-platform tools**
- **Measure performance and understand maximum capacity on the PIC32**
- **Implement a DSP based system on the PIC32**

Agenda

- A practical DSP system example
- DSP Constructs on the PIC32
- Filters using Octave
- Lab 1 – Creating Silence and then filter it
- DSP Ideals vs. Implementation
- Lab 2 – Create a LR-4 Crossover
- Real-World Correction
- Lab 3 – Implement a Correction Filter
- Summary

A practical DSP system example

Bi-amped 2.1 Speaker System

- **Speaker System with built-in subwoofer**
- **Bi-amped**
  - Two amplifiers
  - Class D
- **Active Digital Crossover**

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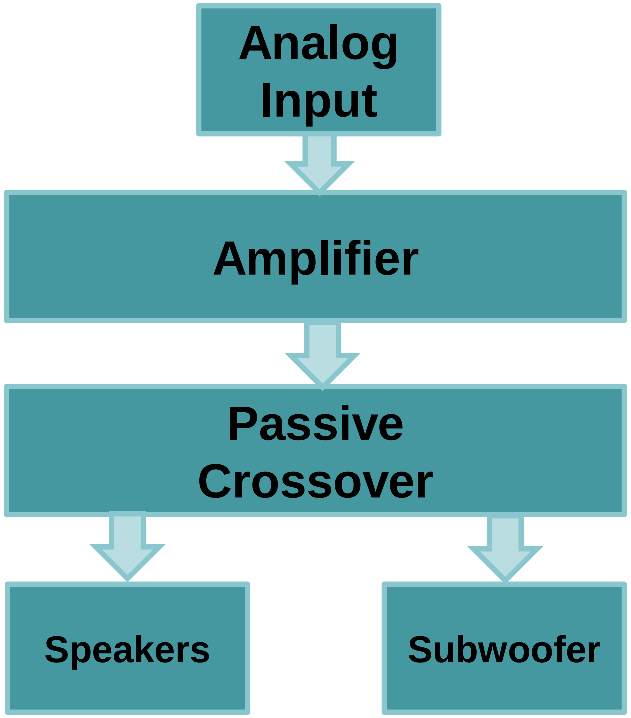
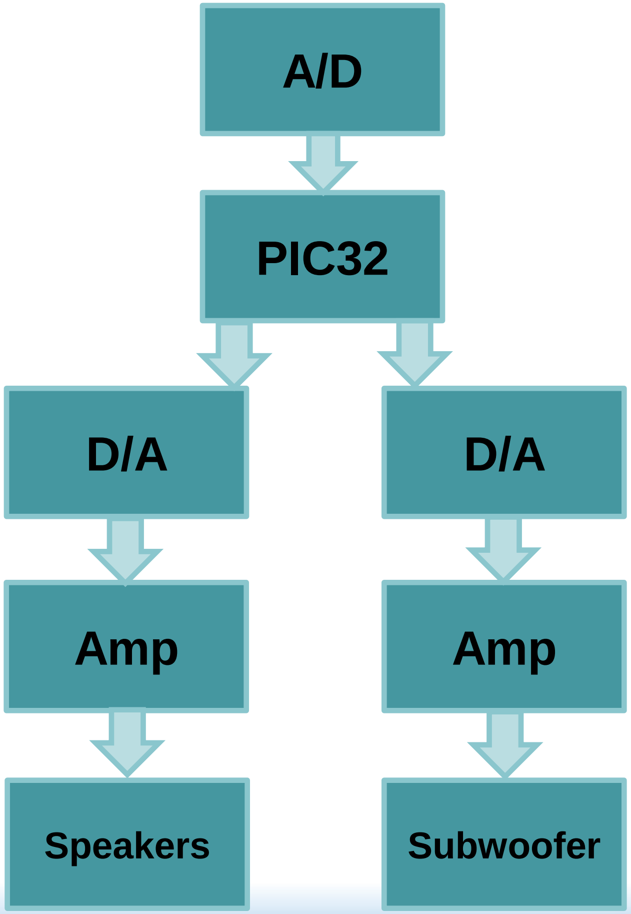
Block Diagrams

Analog System



Digital System



Comparison

Analog System

- **Pros**
  - Simple and inexpensive
  - Fewer active components
- **Cons**
  - Higher cost speakers
  - Higher power amplifier required

Digital System

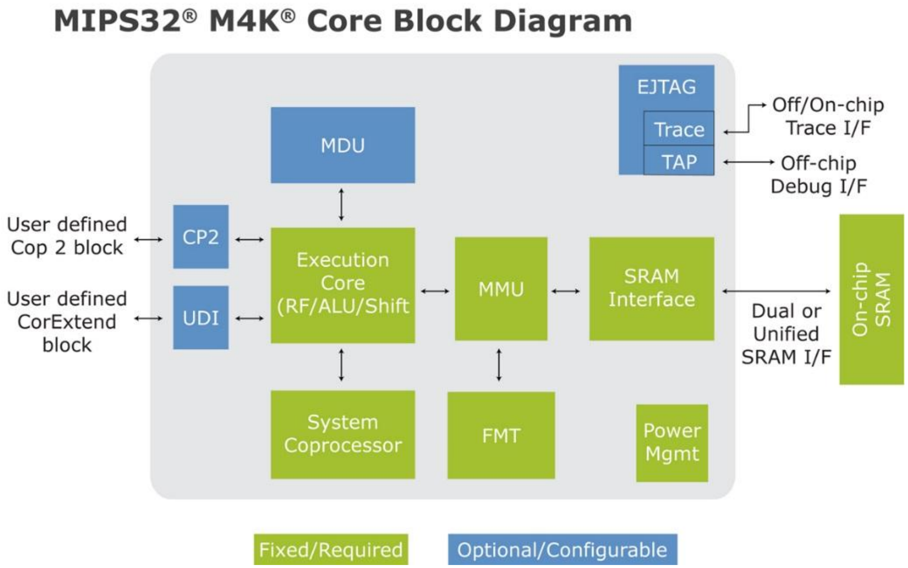
- **Pros**
  - Higher order filtering
  - Equalization
  - Higher SPL with less power
- **Cons**
  - More complex system
  - Software in the signal path

Class Hardware

- **PIC32MX1 & MX2**
  - SPI w/ I^{2}S support
  - 40MHz (50MHz and higher available)
- **AK4645A**
  - Stereo codec
    - **A/D 16 bits**
    - **D/A 16-24 bits**
- **One codec on board**

PIC32 DSP Capabilities

- **32 bit data path and registers**
- **32x16 multiplier with 64 bit result (HI and LO)**
- **Specific DSP instructions**
  - MADD & MSUB
    - **Multiplies and then adds the result to HI/LO**
  - MUL
    - **Multiplies and sends 32 bit result to register**
- **Maximum issue rates**
  - One 32x16 multiply per clock
  - One 32x32 multiply every other clock



DSP Constructs on the PIC32

DSP Constructs

- Filter
- Equalizer
- Delay
- Attenuation

Filter

Selectively attenuates and passes frequencies

Cut off frequency (f_0)

-3dB down point

Order

6dB per octave per order

2^{nd} order = 12dB/octave

Q

Quality factor

0.707 = Butterworth

Octave

Interval between frequencies of ½ or double

100 - 200Hz

1000 - 2000Hz

DSP Constructs

- Filter
- **Equalizer**
- Delay
- Attenuation

Equalizer

A band pass filter with change in gain

Center frequency (f_0)

Bandwidth (BW)

Defined in octaves (1, ½, 1/3, 1/24, etc.)

dB gain

Can be positive or negative

DSP Constructs

- Filter
- Equalizer
- Delay
- Attenuation

Delay

Time shift

FIFO structure

Delay measured in sample periods

Used to align signals in time

DSP Constructs

- **Filter**
- **Equalizer**
- **Delay**
- **Attenuation**

Attenuation

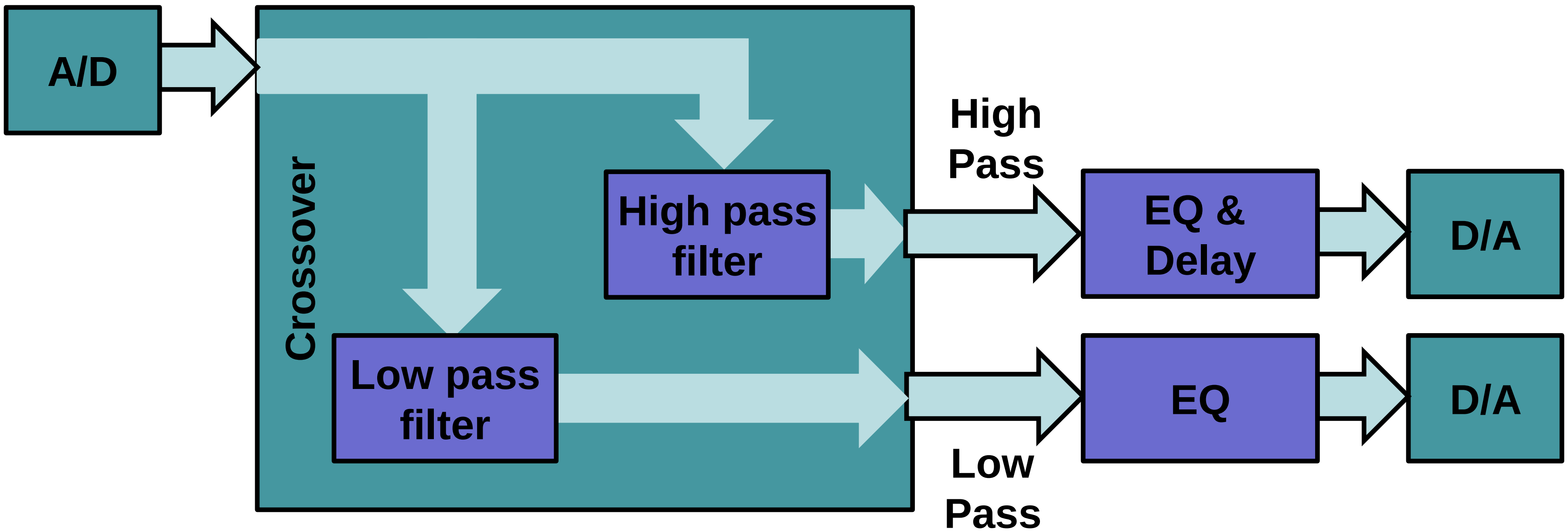
Used to match amplitudes of signals

Always implemented as a reduction

Fractional multiply

Reduces resolution

PIC32 Signal Path



Number Formats

- **AK4645A format**
  - Two's Complement

Examples:

0000 0000 0000 0000 = 0
0000 0000 0000 0001 = 1
1111 1111 1111 1111 = -1
0111 1111 1111 1111 = 32767
1000 0000 0000 0001 = -32767

- **DSP Library format**
  - Q1.15
  - Fractional format
  - Represented by int16
  - No conversion needed

Examples:

0000 0000 0000 0000 = 0
0000 0000 0000 0001 = 0.000030517578125
1111 1111 1111 1111 = -0.000030517578125
0111 1111 1111 1111 = 0.999969482421875
1000 0000 0000 0001 = -0.999969482421875

Math and Number Formats

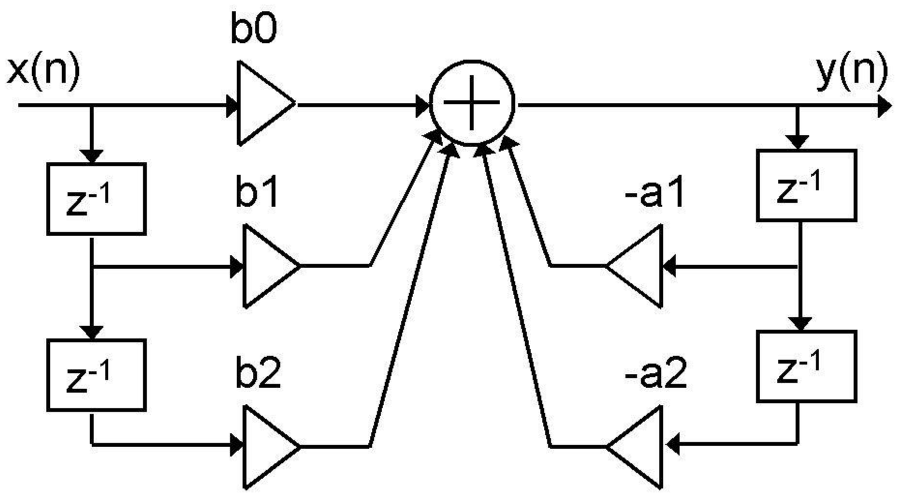
- **Add**
  - Add two numbers (16 bit + 16 bit)
  - Result is 16 bits
  - Q1.15
- **Multiply**
  - Multiply two numbers (16 bit * 16 bit)
  - Result is 32 bits
  - Q2.30 shifted to Q1.31
- **Precision**
  - Q1.15 – Single precision
  - Q1.31 – Double precision

Filter Choice

- **Butterworth filter**
  - Maximally flat pass band
  - 6 dB/octave roll-off per pole (or order)
- **IIR chosen for digital implementation**
  - Can approximate a Butterworth response
  - Lower order than equivalent Finite Impulse Response (FIR)
  - Recursive with potential for instability

Filter Implementation

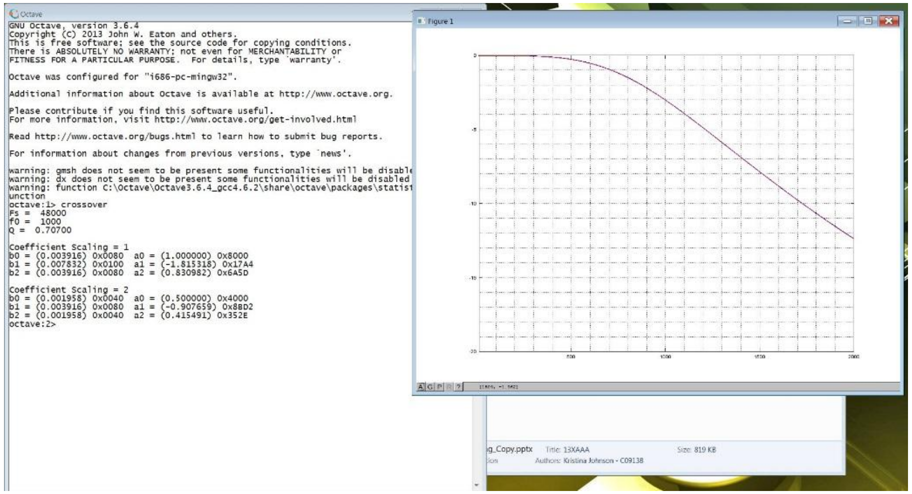
- 2^{nd} order Bi-quad
- Direct Form I
- Preferred for audio due to single sum
- 5 multiplies and 4 delay elements



Filters using Octave

GNU Octave

- High-level interpreted language for complex math
- Open source alternative to MATLAB®
- Available on Windows, Mac, and Linux
- www.gnu.org/software/octave



Using Octave

- **Text based input**
- **.m files are collections of commands (basic scripting language)**
- **All .m files are in Octave folder inside class directory**
- **Used to design and simulate DSP functions**

```
# Filter Parameters
Fs = 48000  #Hz
f0 = 1000 #Hz
dBGain = 0; #dB - only for peaking filters
Q = .707    #Filter shape for Butterworth = .707 (Q=0.54 and Q=1.31 for 4th order Butterworth)

#Intermediate Calculations
A = sqrt(10^(dBGain/20));
w0 = 2*pi*f0/Fs;

alpha = sin(w0)/(2*Q);

#LPF

# Coefficients
blpf = [ ((1-cos(w0))/2)/(1+alpha), (1-cos(w0))/(1+alpha), ((1-cos(w0))/2)/(1+alpha)];
alpf = [ (1+alpha)/(1+alpha), (-2*cos(w0))/(1+alpha), (1-alpha)/(1+alpha) ];

# Coefficients - Scaled down by 2
blpfs = blpf/2;
alpfs = alpf/2;

# Quantized Q15 Coefficients
blpfq = fix(blpf * 2^15)/2^15;
alpfq = fix(alpf * 2^15)/2^15;

# Quantized Q15 Coefficients - Hex
blpfhex = ndec2hex(fix(blpf * 2^15),16);
alpfhex = ndec2hex(fix(alpf * 2^15),16);

printf("\nCoefficient Scaling = 1\n");
for n=1:3
    printf("b%u = (%f) 0x%s\ta%u = (%f) 0x%s\n",n-1,blpf(n),ndec2hex(fix(blpf(n) * 2^15),16),n-1,
end
printf("\nCoefficient Scaling = 2\n");
for n=1:3
    printf("b%u = (%f) 0x%s\ta%u = (%f) 0x%s\n",n-1,blpfs(n),ndec2hex(fix(blpfs(n) * 2^15),16),n-
```

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Tools for Your Project

DSP17085lib.c

- **Q15iirQ15_DF1**
  - IIR Filter – Direct Form I
  - Single Precision
- **Q31iirQ31_DF1**
  - IIR Filter – Direct Form I
  - Double Precision
- **Q15toQ31**
  - Convert Q1.15 to Q1.31
- **Q31toQ15t**
  - Truncate Q1.31 to Q1.15
- **Q31toQ15r**
  - Round Q1.31 to Q1.15

M Files

- **lpf.m**
  - Low pass filter coefficients
  - Single & Double precision
- **hpf.m**
  - High pass filter coefficients
  - Single & Double precision
- **eqf.m**
  - EQ filter coefficients
  - Single & Double precision

Lab 1: Creating Silence and then filter it

Lab 1 Objectives

- Set up Audio_Start and get silence
- Design a low pass filter using Octave
- Implement filter on PIC32
- Estimate maximum, simultaneous filters on PIC32

Lab 1 Summary

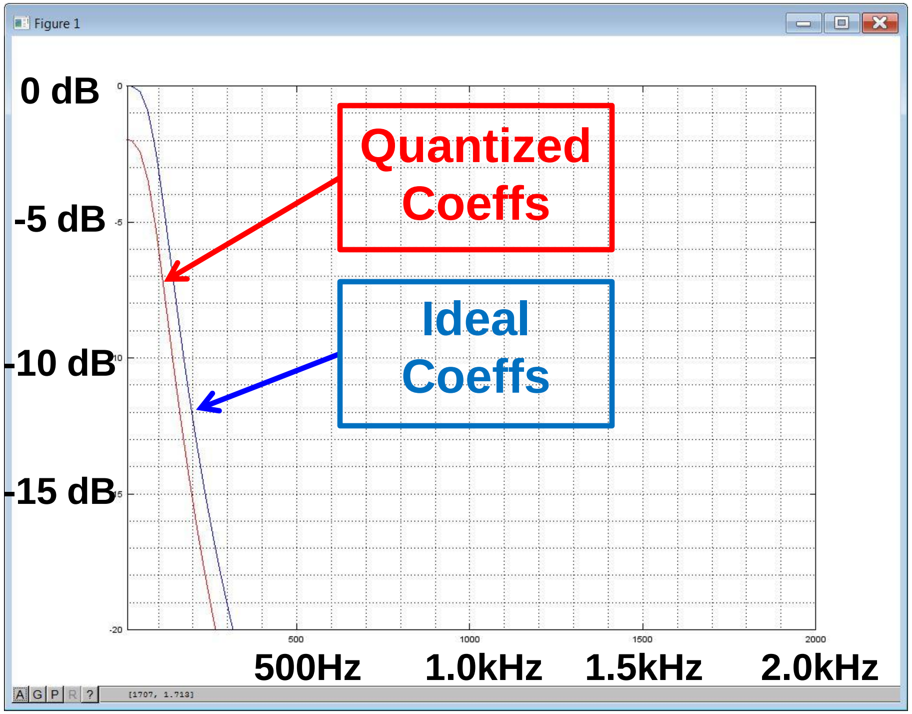
- Set up Audio_Start and heard silence
- Designed a low pass filter using Octave
- Implemented filter on PIC32
- Estimated maximum, simultaneous filters on PIC32

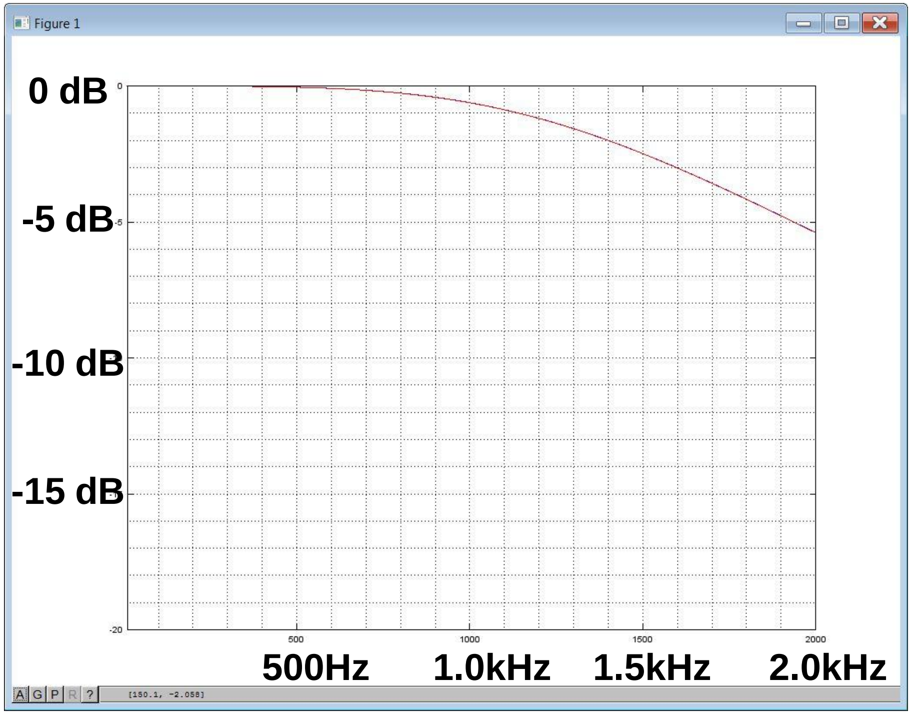
DSP Ideals vs. Implementation

Why 1.6kHz?

- **The filter in Lab 1 implemented a single precision IIR filter**
  - 16 bit coefficients and data
  - 32 bit internal results
- **Quantization imposes limits on pole placement**
- **At 48kHz sample rate, single precision falls apart below 300Hz**

100Hz vs 1.6kHz





Filters below 300Hz?

- **Two methods**
  - Decimation/Interpolation
  - Double precision – 32 bit coefficients / 64 bit results
- **Decimation**
  - Filters work well when away from the extremes
    - **100/48000 = 0.002**
    - **1600/48000 = 0.03**
  - Decimation changes the sample rate
  - Decimate by 16 changes the sample rate to 3kHz
    - **100/3000 = 0.03**

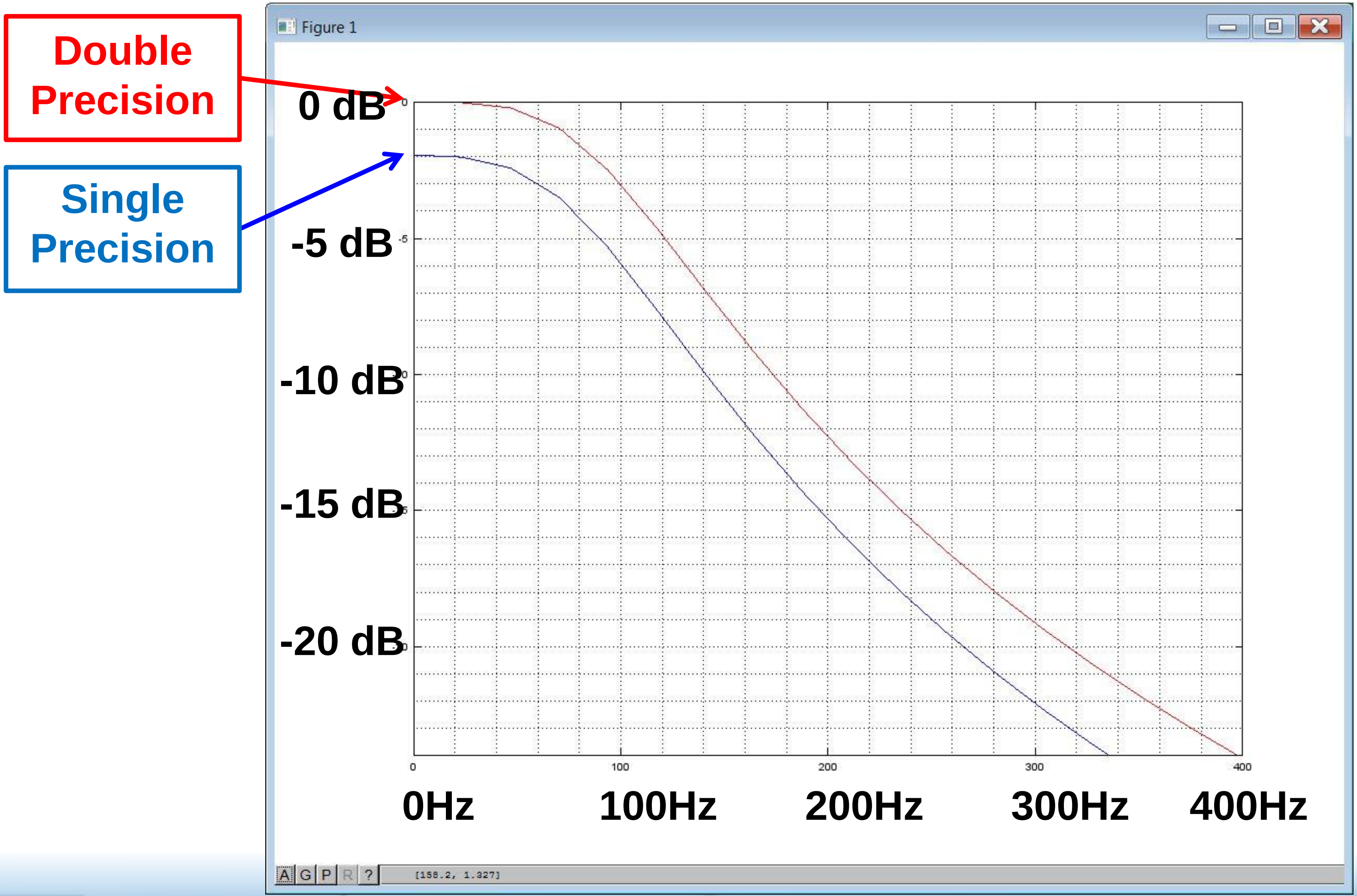
Filters below 300Hz?

- **Double precision**
  - 32 bit coefficients
  - Slower, but much better filter shape at low frequencies (<20Hz)

$$\mathbf{2^{-15} = 0.000030517578125}$$

$$\mathbf{2^{-31} = 0.000000000465661}$$

Double vs Single Precision



Crossovers

What is a Crossover?

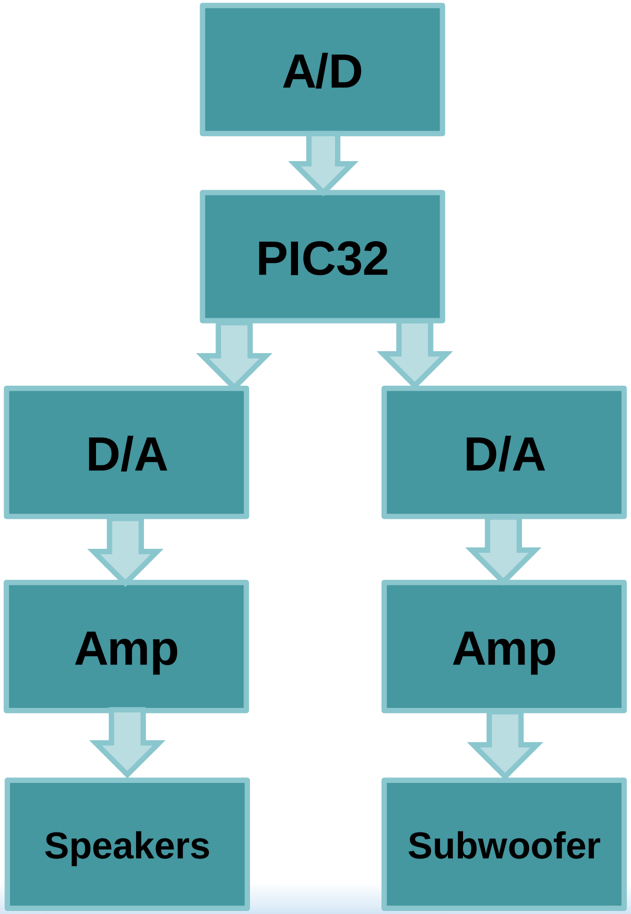
- **A crossover divides the audio spectrum into two or more section**
- **Two-way, three-way, or more**
- **Implementation**
  - Passive or active
  - Analog or digital
- **We are focused on active, digital crossovers**

Remember our example

- **2.1 speaker system**
  - Left & Right speakers
  - Subwoofer
- **Common in computer speaker systems**
- **Left & Right speakers**
  - 250Hz to 20kHz
- **Subwoofer**
  - 20Hz to 250Hz

- **Active means the crossover is applied before the amplifiers**
- **Digital means the crossover is implement mathematically inside the PIC32**

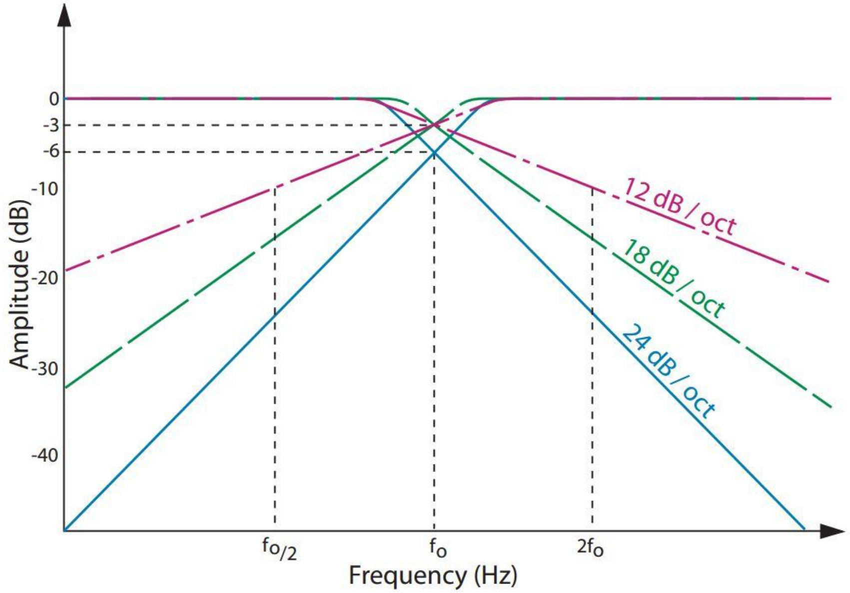
Active Digital Xover



So, how do I make one?

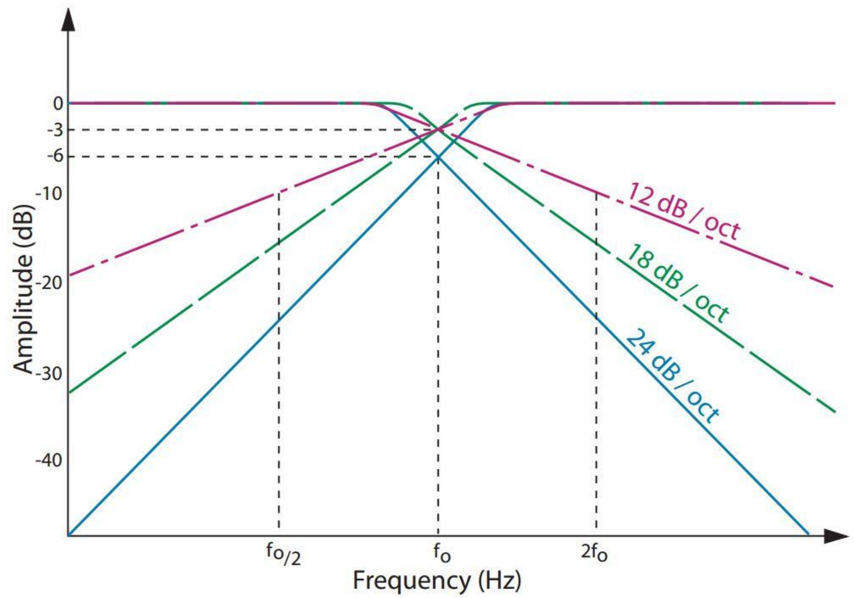
- **Two-way crossover**
  - High pass filter
  - Low pass filter
- **What order?**
- **What frequency?**
- **What type?**

- **Continuing our example….**



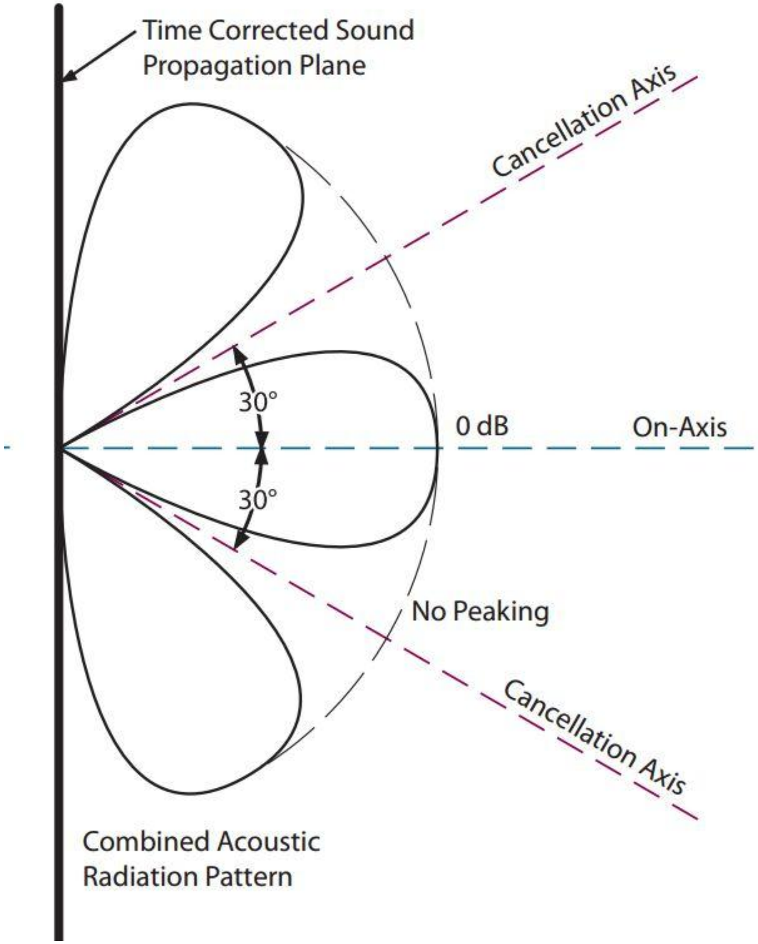
So, how do I make one?

- **Two-way crossover**
  - High pass filter
  - Low pass filter
- **What order?**
  - 4^{th} order
- **What frequency?**
  - 250Hz
- **What type?**
  - Linkwitz-Riley



Linkwitz-Riley Crossover Alignment

- **2 HP engineers in 1976 publish a paper**
  - Siegfried Linkwitz
  - Russ Riley
- **Benefits**
  - In-phase outputs
  - 24dB/octave slope
  - Perfect combined radiation pattern at the crossover point
- **How to Align**
  - Two 2nd order Butterworth filter in series



See http://www.rane.com/note160.htm for more information. (RaneNote 160 by Dennis Bohn)

Lab 2: Create a LR-4 Crossover

Lab 2 Objectives

- **Implement a double precision crossover**
- **Understand how to create a 4^{th} order Linkwitz-Riley filter using biquads**
- **Measure performance impact of double precision on PIC32**

Lab 2 Summary

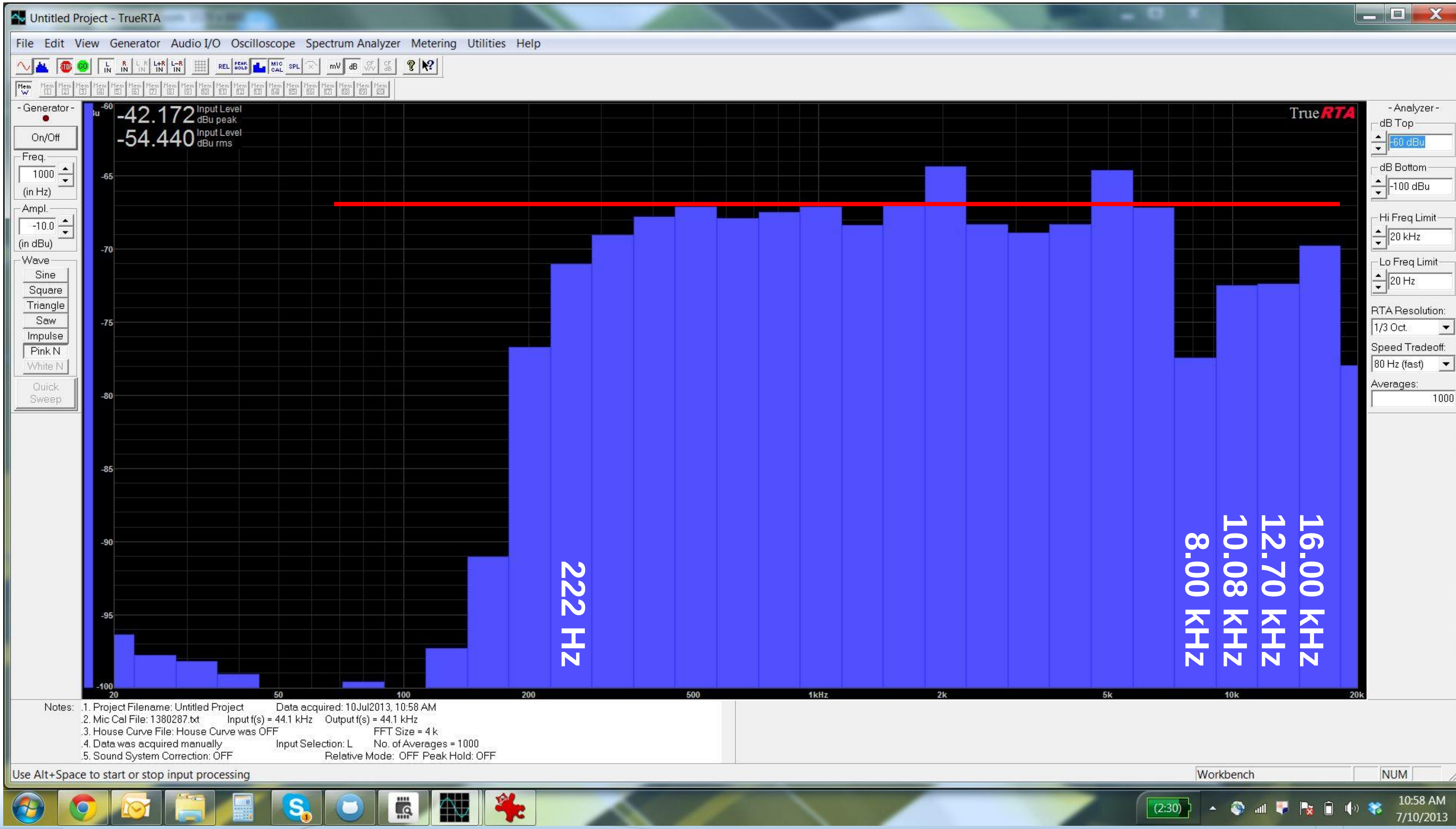
- Implemented a 250Hz, LR-4 crossover
- Used two biquads in series to create 4th order Linkwitz-Riley filter
- Measured the performance impact of double precision filters on the PIC32

Real-World Correction

And then reality hits….

- All of the graphs have had razor sharp lines and perfect slopes
- All speakers are electro-mechanical systems
- Therefore, imperfect

Actual driver measurement



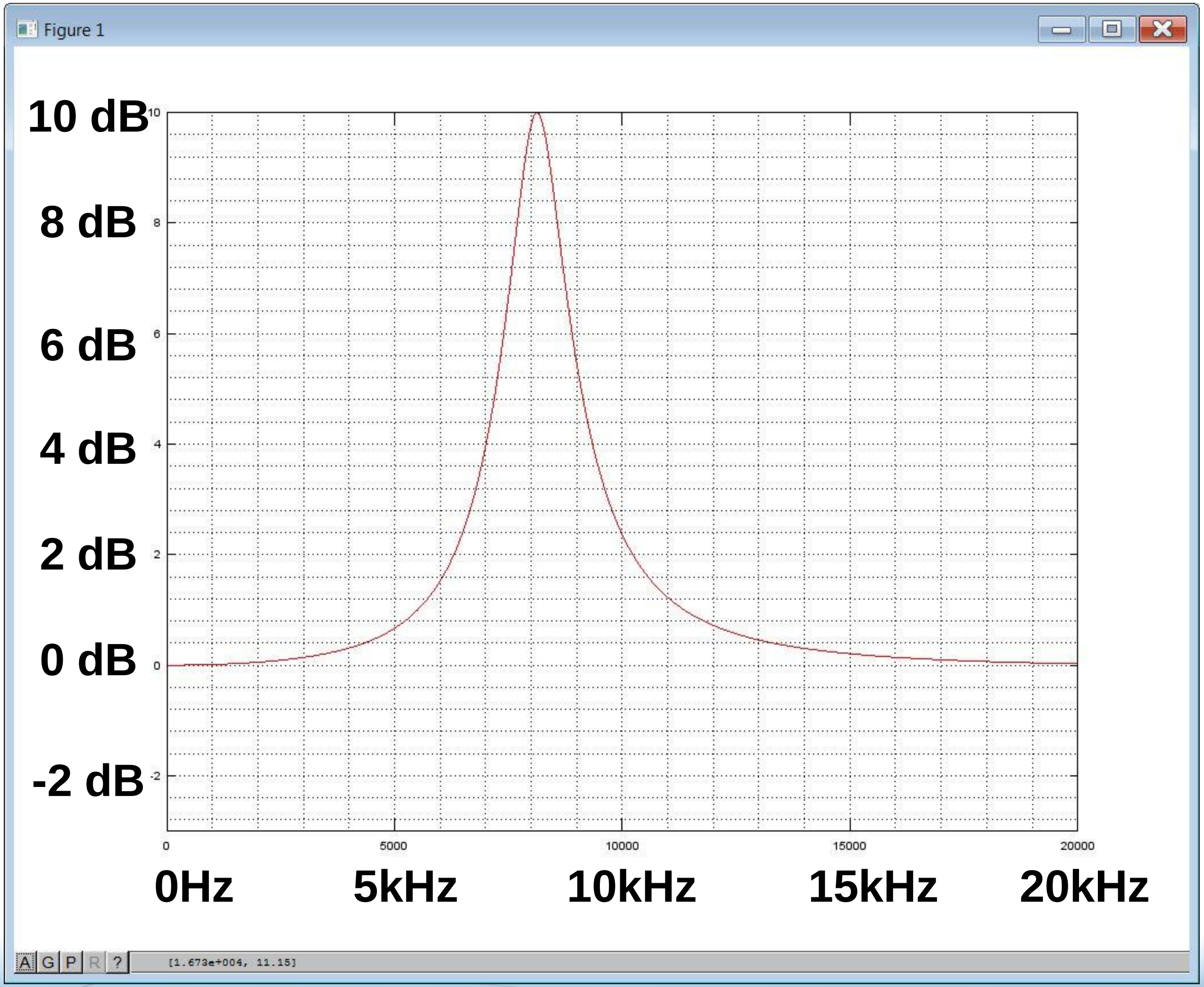
Not bad, except…

- Goal for speaker design is +/-3dB response
- Response curve droops and peaks above 8kHz
- Ideal response is 20Hz to 20kHz

EQ filter to the rescue

- Use EQ filters to correct the response
- Again, use IIR biquads to create bandpass filters of the same width with gain
- 1/3 octave wide and +10dB at 8kHz

1/3 octave EQ @ 8kHz



Lab 3: Implement a Correction Filter

Lab 3 Objectives

- Implement a correction filter for the speaker response curve
- Understand the placement of this filter in the signal chain
- Understand the impact on the system of this filter

Lab 3 Summary

- Implemented a correction filter for the speaker response curve
- Understand the placement of this filter in the signal chain
- Understand the impact on the system of this filter

32Bit Middleware and Software Ecosystem

- **Software Platform Key Features and Benefits:**
  - Modular peripheral drivers and Middleware layers
  - Dynamic Multi-client driver support
  - Designed to be RTOS friendly
  - Designed for Interoperability and 32Bit MCU scalability
  - Integrated, Verified & Supported 3rd Party Partner solutions
- **Interested in learning more? Attend "17033 MPLAB® Harmony" for the Next Generation Middleware/Ecosystem or stop by the "Ask-the-Experts" booth for more info**

Demo

- **We've spent the past 3.5 hours working and testing the pieces of a 2.1 speaker system**
- **It is only appropriate to try this out and see if it actually works**

Summary

Summary

- **Today we covered:**
  - Creating a DSP application for the PIC32
  - Designing and simulating DSP elements with free, multi-platform tools
  - Measuring performance and understand maximum capacity on the PIC32
  - Implementing a DSP based system on the PIC32

Additional Resources

- **Internet**
  - Introduction to Signal Processing
    - **http://www.ece.rutgers.edu/~orfanidi/intro2sp/**
  - RaneNote 160 by Dennis Bohn
    - **http://www.rane.com/note160.htm**
  - Cookbook formulae for audio EQ biquad filter coefficients
    - **http://www.musicdsp.org/files/Audio-EQ-Cookbook.txt**

Parts Used in Demo

- **Microchip**
  - DM320014 - PIC32 USB Digital Audio Accessory Board (Qty. 2)
- **Parts Express**
  - 320-330 – 2x15W Class D Amp (Qty. 2)
  - 120-056 – 12Vdc, 5A Power supply
  - 300-7064 – 0.56cu ft MDF cabinet
  - 285-111 – Dayton Audio 2” speaker (Qty. 2)
  - 290-352 – Goldwood 6.5” DVC Subwoofer

Dev Tools For This Class

- **DM320014 - PIC32 USB Digital Audio Accessory Board**

- **DV244005 - MPLAB® REAL ICE™ In-Circuit Emulator**

- **AC164110 - RJ-11 to ICSP Adaptor**

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