Digital Downconversion Processing Algorithm

nanoBPM Project
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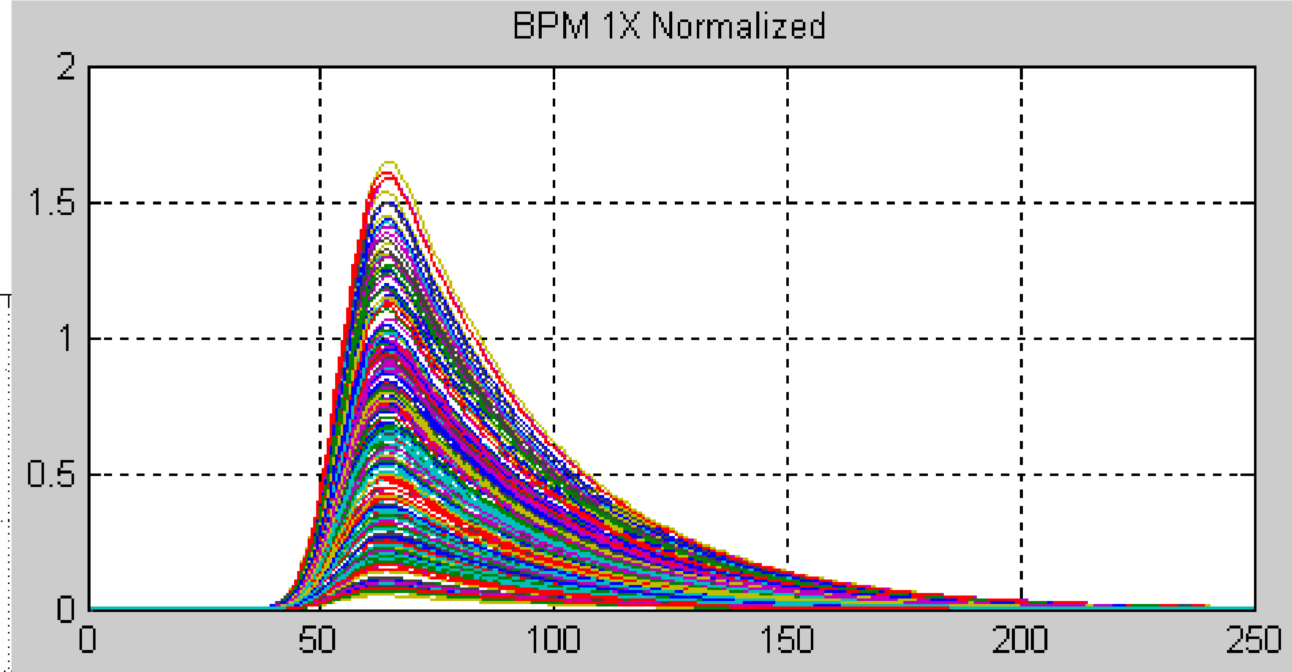
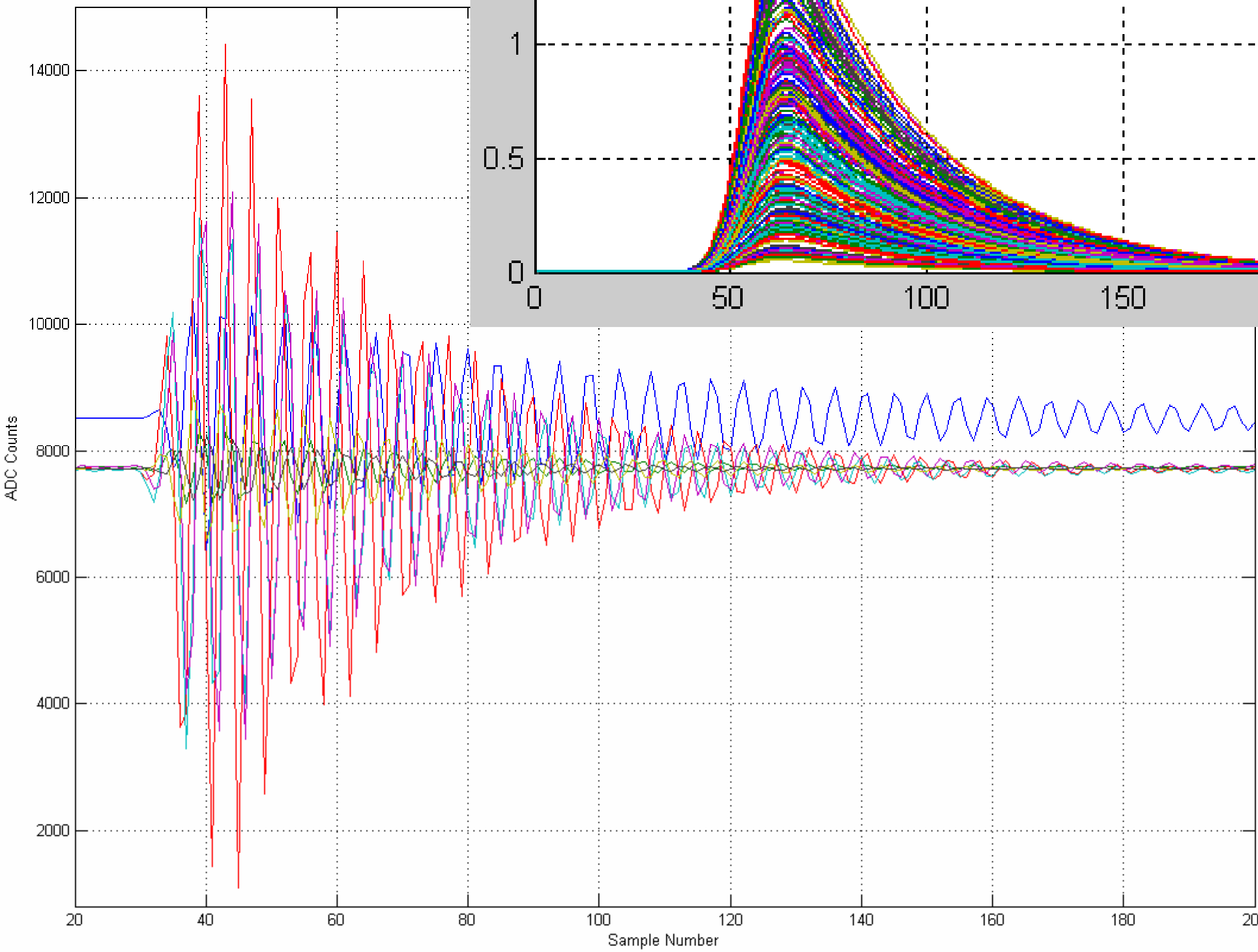
Cavity BPMs

- Position measurement is cavity dipole signal amplitude & phase.
- C-Band Cavities
  - First dipole mode frequencies are ~ 6426 MHz
- Dual Downconversion:
  - First IF at 476 MHz
  - Second IF at 25 MHz
- Digitize at 100 MSamples/sec
  - ~ 4 samples/ IF period
- Why don't we downconvert to baseband in analog to begin with?
- Digitizing at IF removes sensitivity to
  - Offsets
  - Low frequency noise
  - Gain differences (if (I,Q) down conversion is analog)

Data: Raw & Demodulated

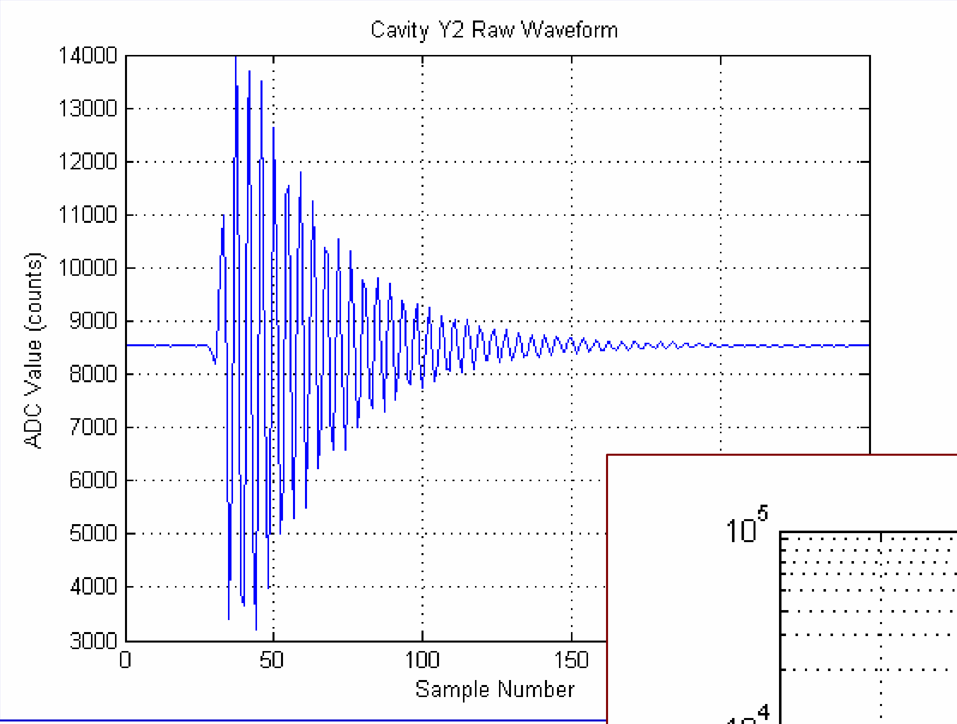


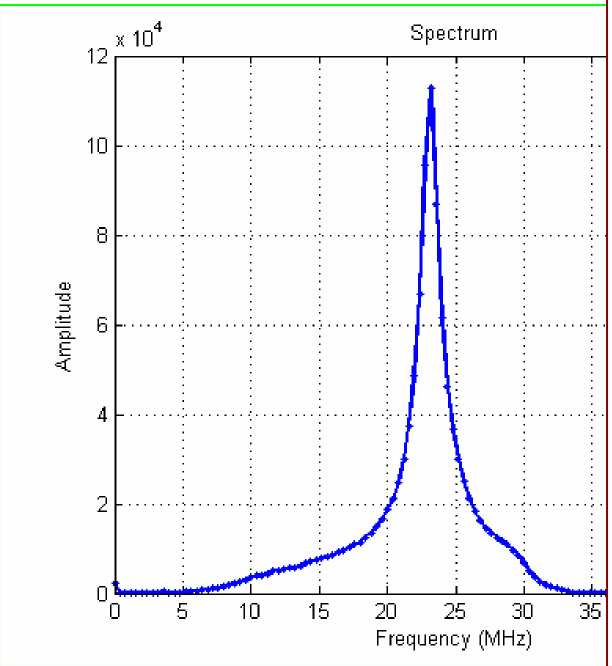


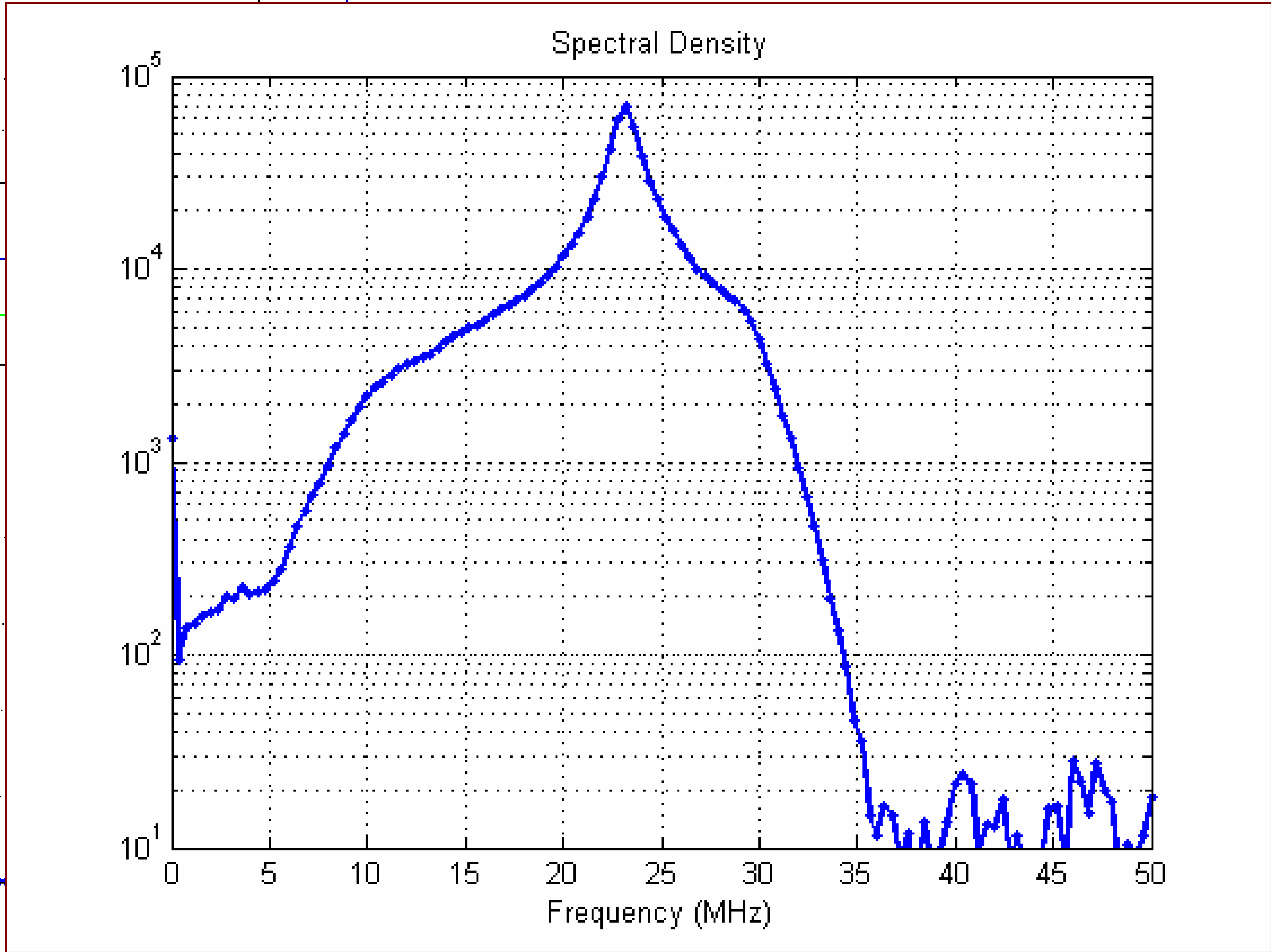
Algorithm Overview

- Digitized IF
  - 14 bit ADC resolution
  - ~4 samples/ IF period
- Digital Downconversion:
  - Subtract approximate ADC zero
    - value not critical
    - Can use 1st sample or mean of first 20 samples, *et cetera*
  - Multiply digital waveform by complex "local oscillator" $e^{i\omega t}$
  - Low-pass filter (currently 2.5 MHz B/W)
- Sample complex amplitude of position cavity at "peak"
- Divide by complex amplitude from reference cavity
- Shift/Rotate/Scale by calibration constants

Cavity Signal Spectrum





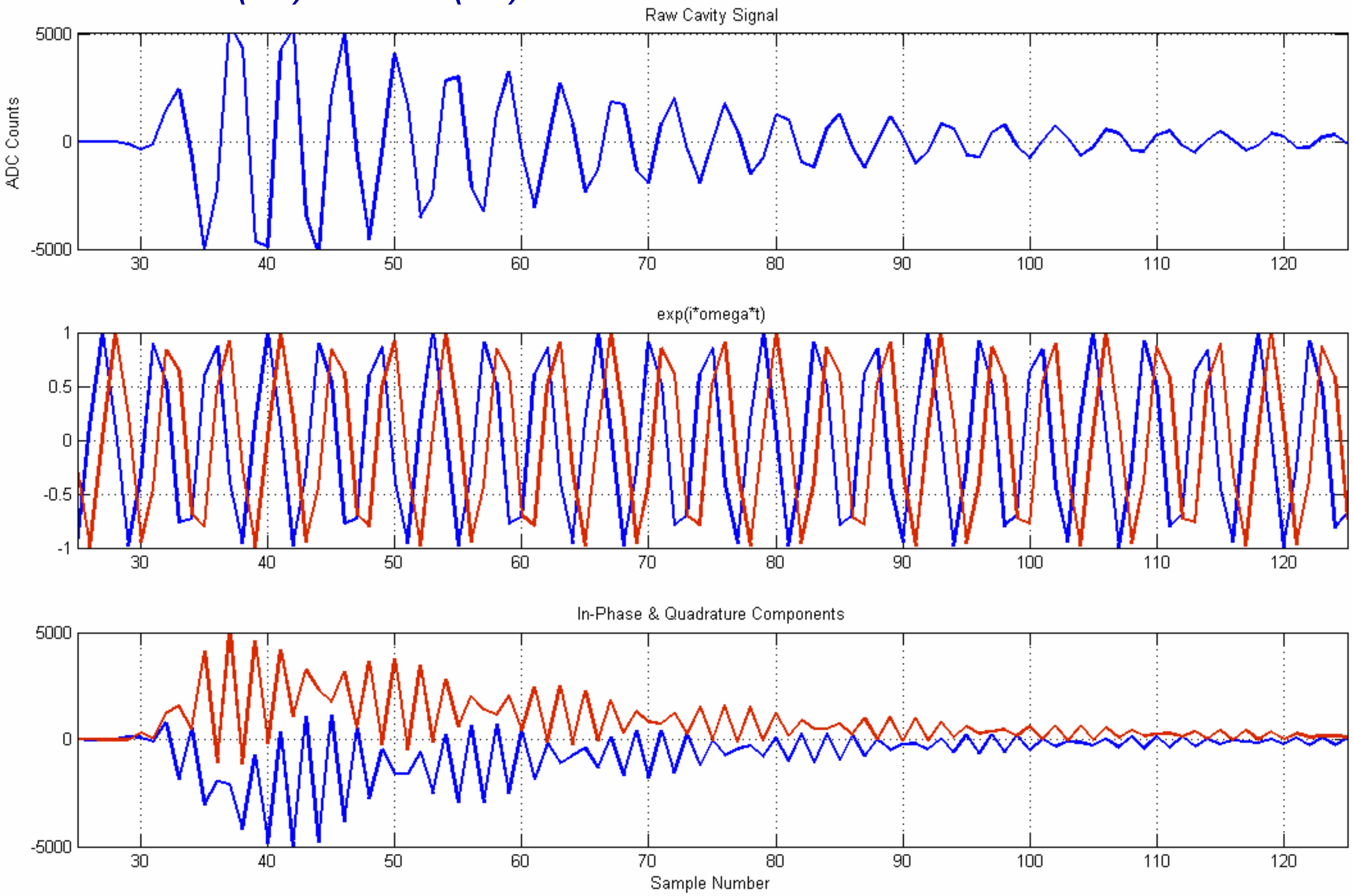


Digital Mixing

- Modulating by sine-like signal produces sum & difference frequencies:

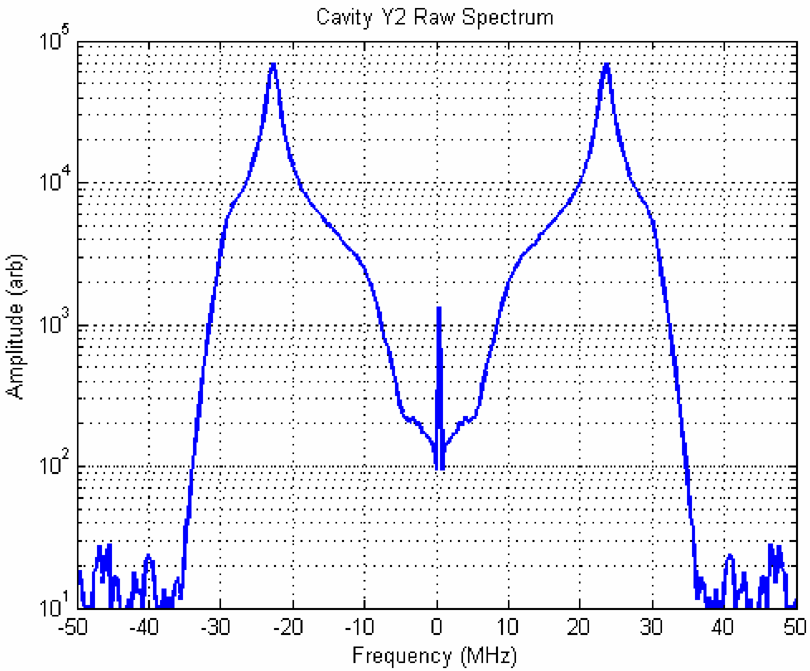
$cos(\omega_1 t+\phi_1)*cos(\omega_2 t+\phi_2) = (cos((\omega_1+\omega_2)t+(\phi_1+\phi_2))+ cos((\omega_1-\omega_2)t+(\phi_1-\phi_2))/2$

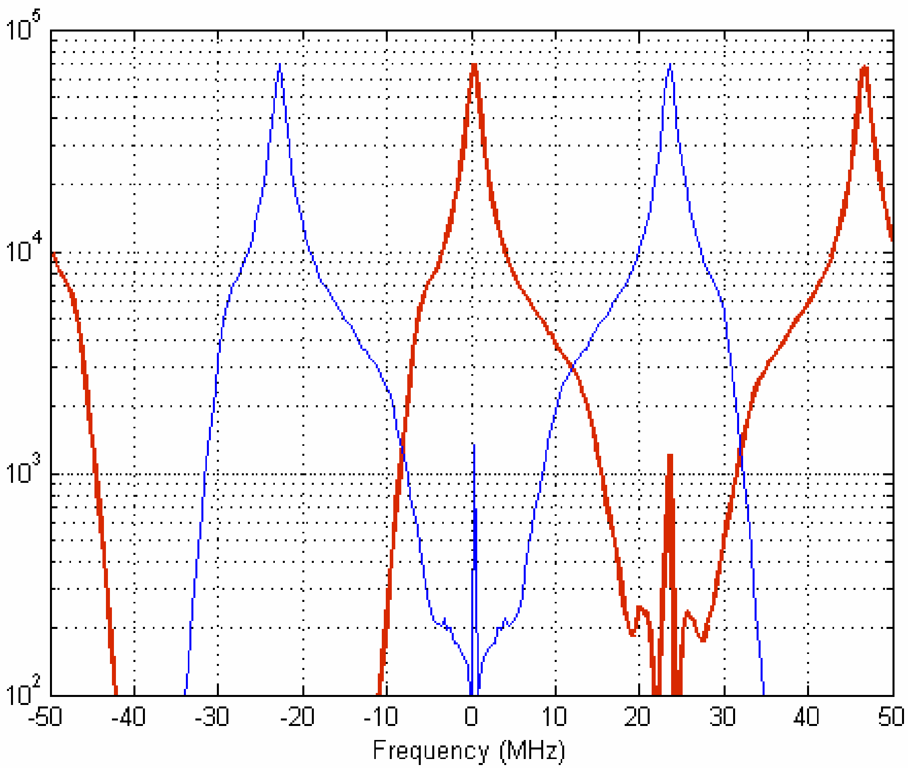
- Don't pick out a particular phase yet,
  - keep phase information by multiplying by
  - $LO = cos(\omega t) + i*sin(\omega t) = e^{i\omega t}$



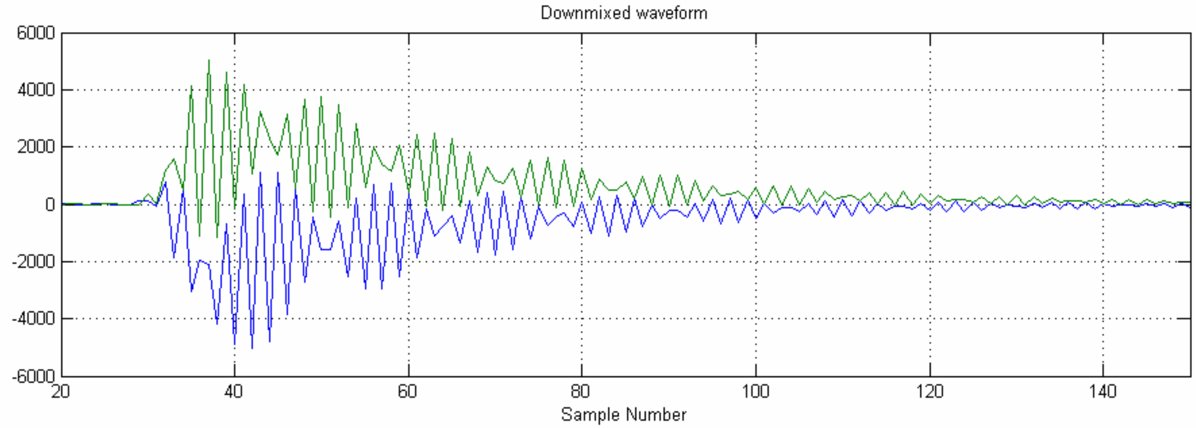
Digital Downconversion as Frequency Shift:

- $LO = \cos(\omega t) + i*\sin(\omega t) = e^{i\omega t}$
- Think of this in either of two ways:
  - A convenient way of keeping track of the sine-like and cosine-like parts of the (real) signal
  - OR a frequency shift operator acting on the +/- frequencies of which the original signal was composed.
- Spectra
  - Raw signal (left)
  - multiplied by $e^{i\omega t}$ (right)





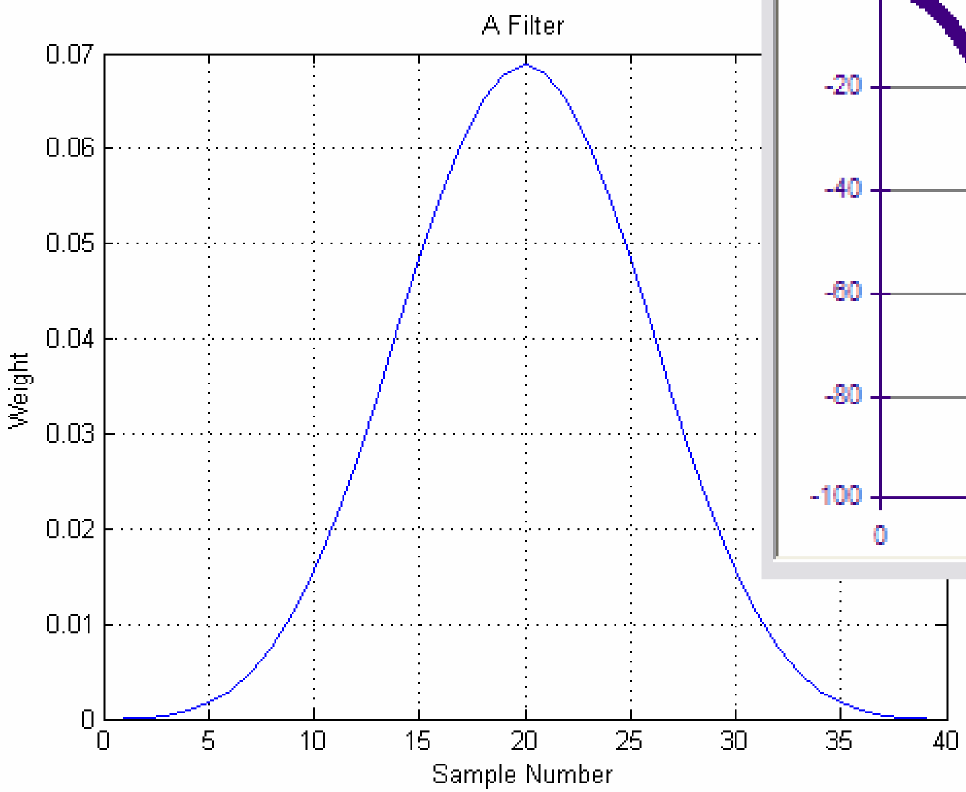
Filter

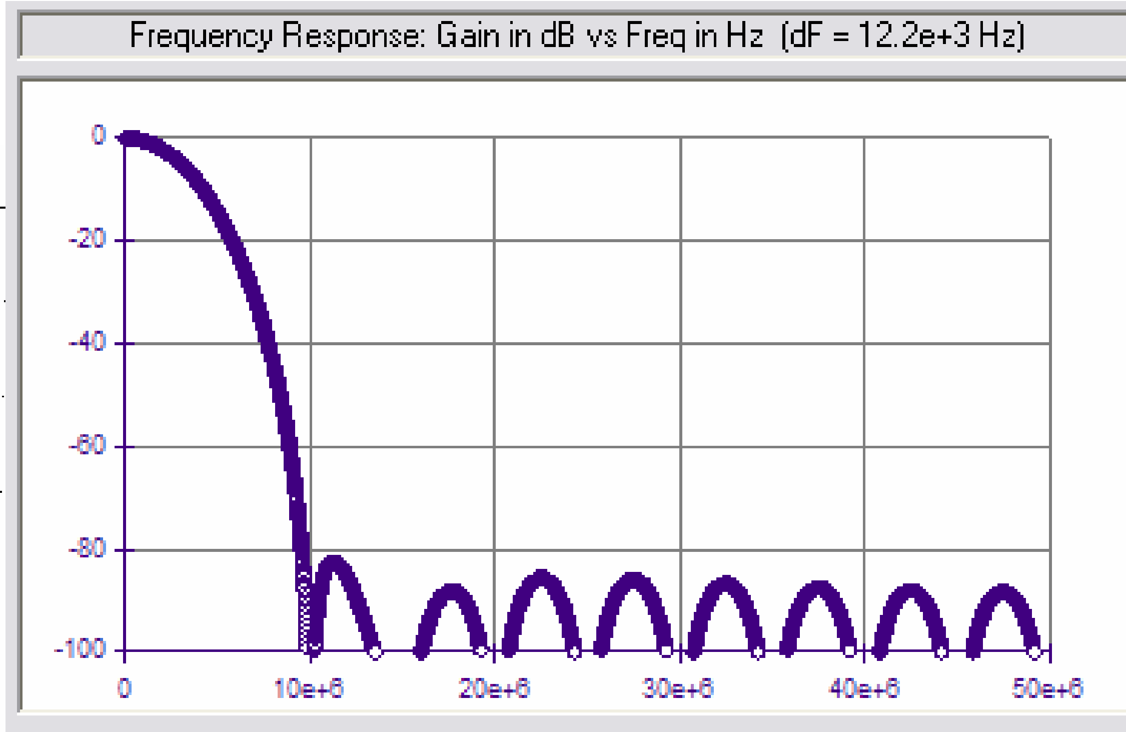


- Product is dominated by sum & difference frequencies
- Need a digital filter to remove
  - 2ω component (sum frequency)
  - (and ω component due to DC offset, low frequency noise)
  - other out-of-band noise
- Require:
  - lots of suppression at 2 ω
  - Finite width in time domain
- Prefer
  - Quasi-Gaussian
  - Flat group delay (don't spread signal energy around in time)

How to Describe a Filter

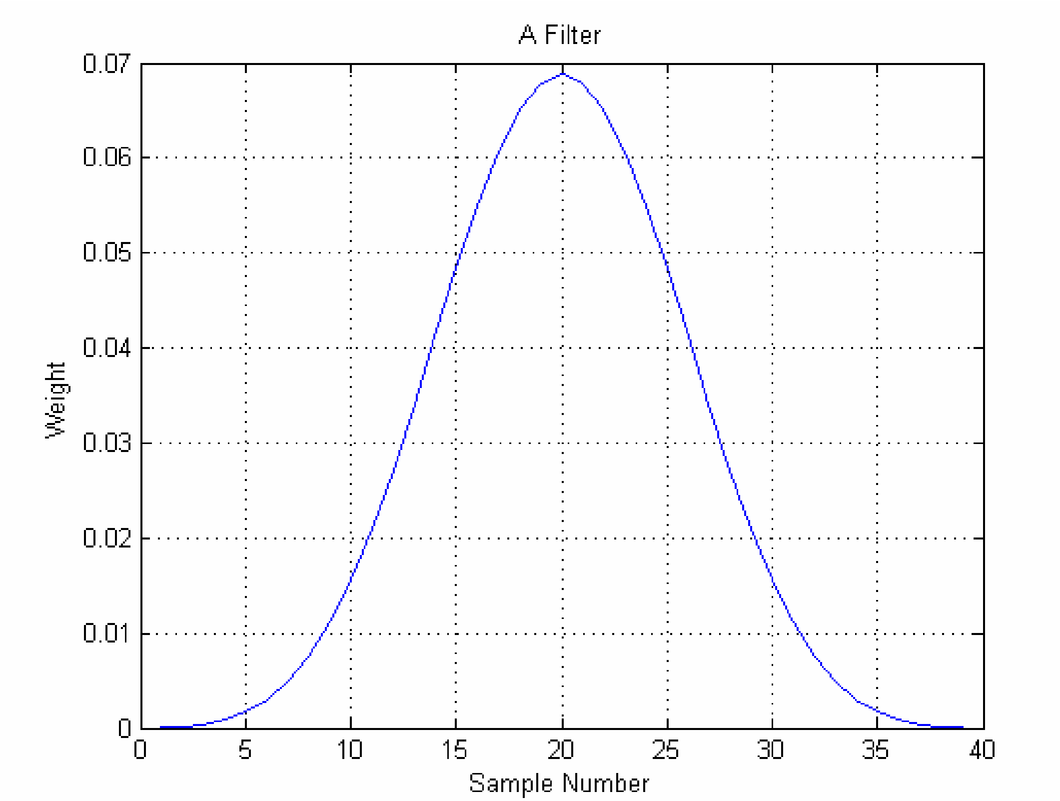
- Time domain
  - Vector of weights
  - Convolution with digitized waveform
- Frequency domain
  - Frequency response
  - Multiply with Fourier Transform of waveform
- Example:
  - 2.5 MHz - 39 Coeficients





FIR Filter Coefficients

- Choose Finite Impulse Response (FIR) filter
  - Symmetric coefficents -> flat group delay



5.25685969697302E-05
1.6454424343439E-04
4.07154578172953E-04
8.72385025321259E-04
1.6766004340434E-03
2.95726074784519E-03
4.86225869836532E-03
7.53264980357207E-03
1.10807385993617E-02
1.55664596752274E-02
2.09756014054222E-02
2.72035613167032E-02
3.40479495420194E-02
4.12124906972027E-02
4.83233995659566E-02
5.49578635296309E-02
6.06826371814424E-02
6.50992446362904E-02
6.78910913982464E-02
6.88670806495452E-02
6.78910913982464E-02
6.50992446362904E-02
6.06826371814424E-02
5.49578635296309E-02
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1.55664596752274E-02
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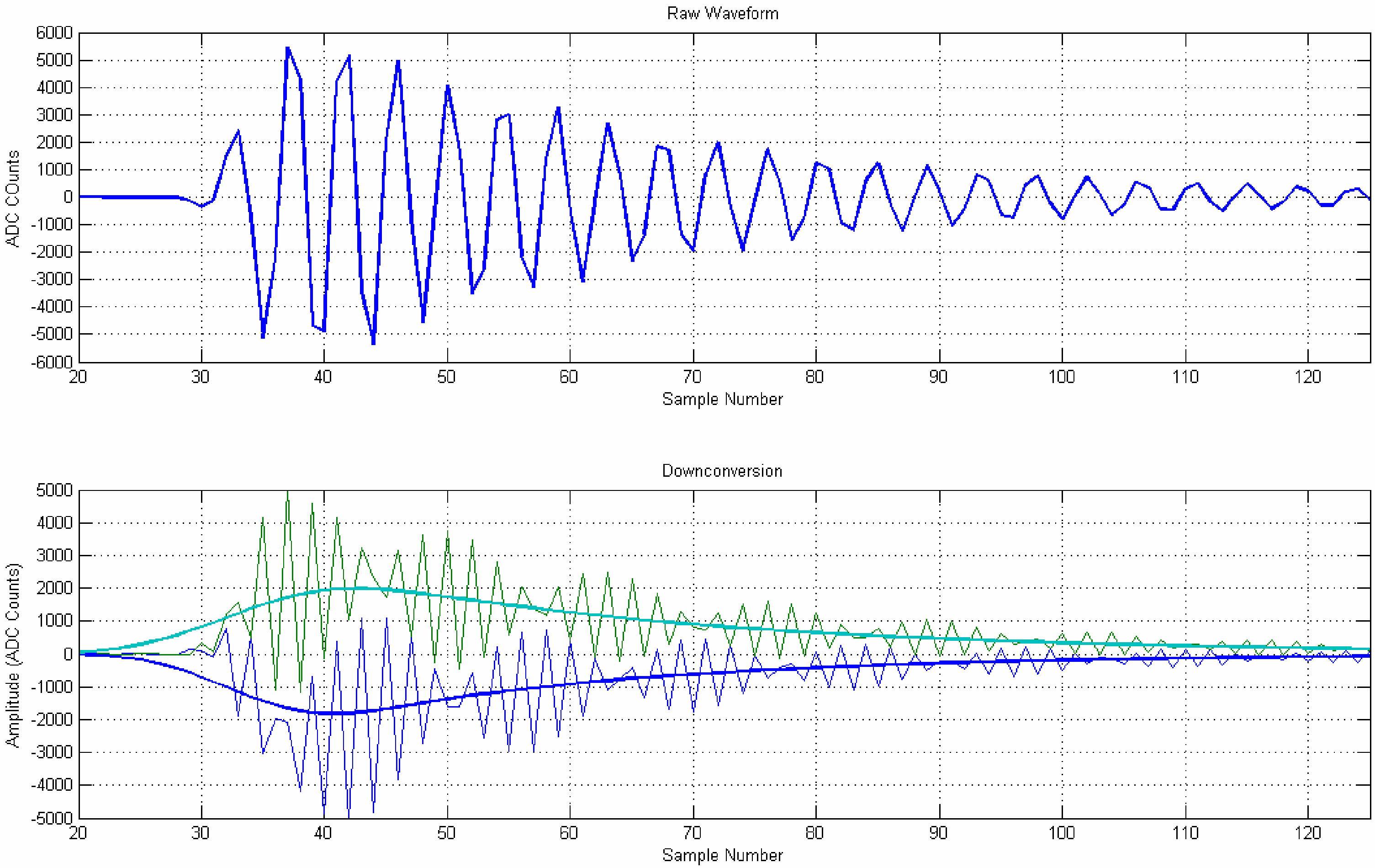
1.6766004340434E-03
8.72385025321259E-04
4.07154578172953E-04
1.6454424343439E-04
5.25685969697302E-05

Filter Downconverted Waveform

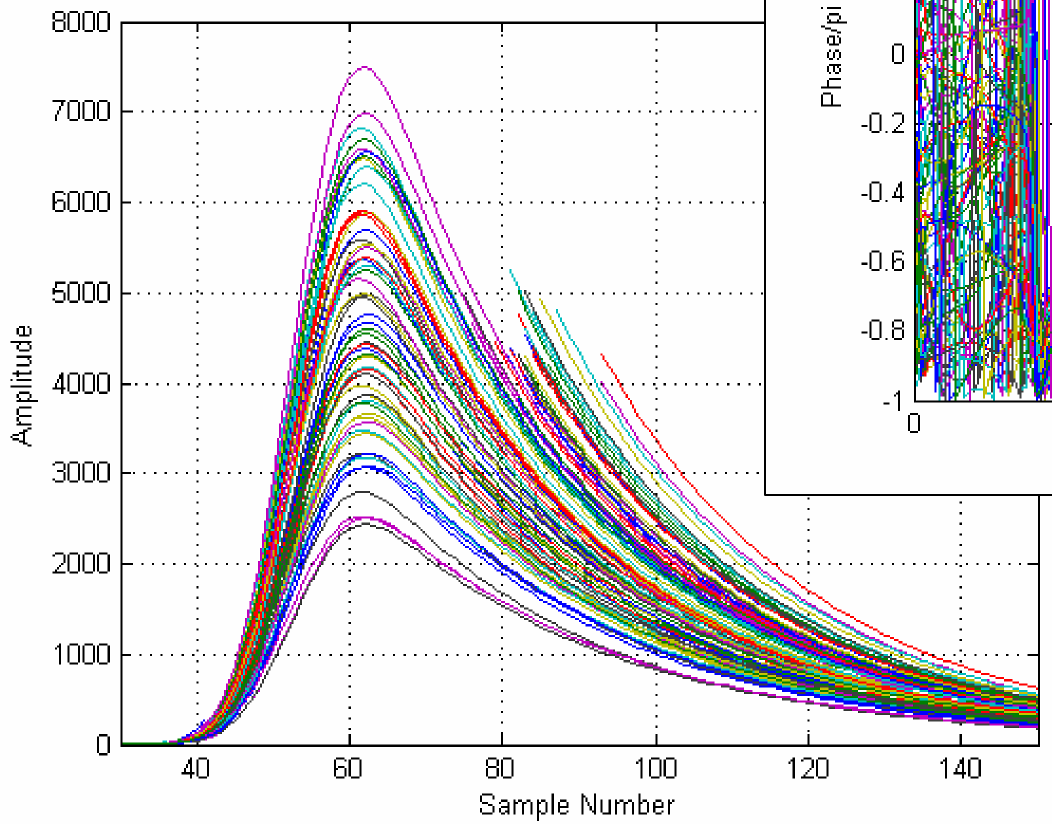


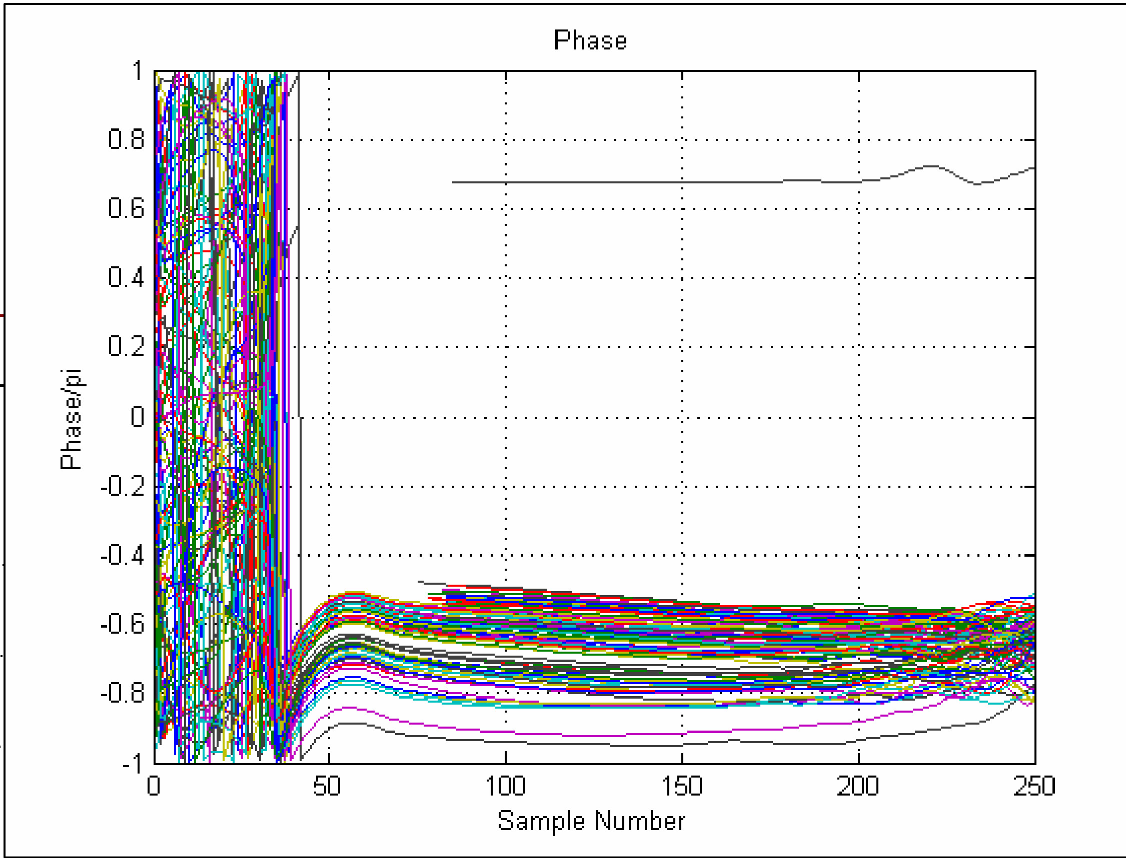
- Upper trace is Raw waveform
- Lower is demodulated waveform

Waveform Amplitude & Phase

Pick a time (sample number) at which to evaluate waveform amplitude & phase

Fixed sample time required to preserve linearity





Normalization

- Position cavity phase and amplitude are normalized to that of the reference cavity
- Divide (complex) position cavity amplitude by
  (complex) reference cavity amplitude
- Takes out charge and arrival phase of beam

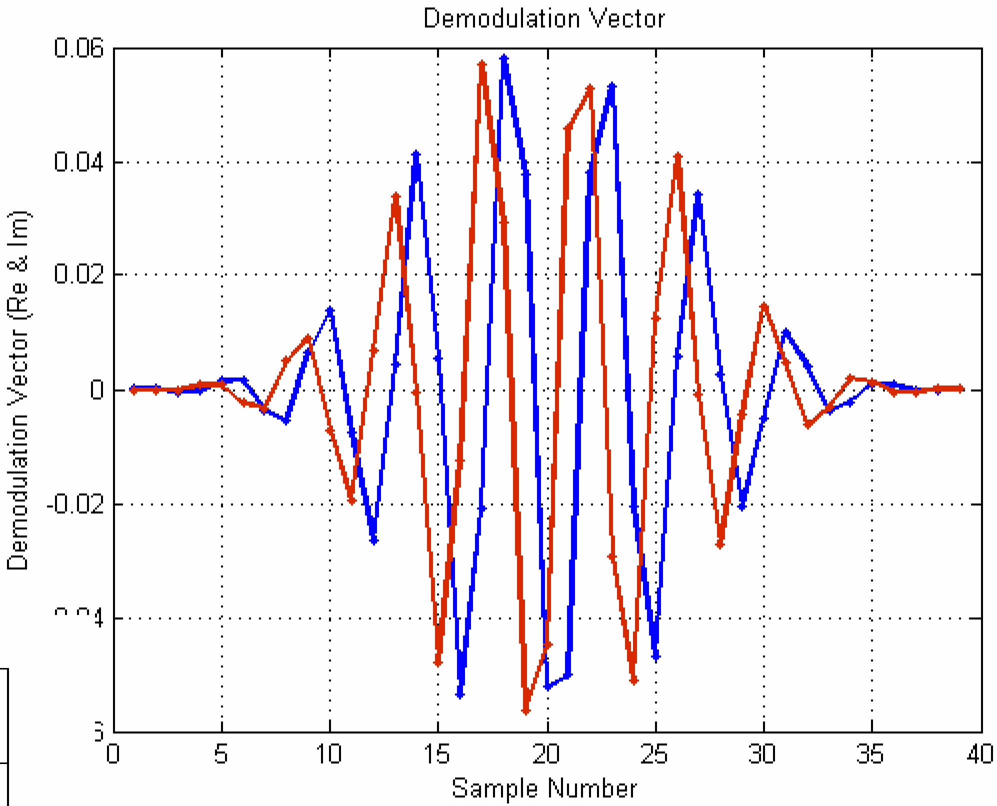
Summary

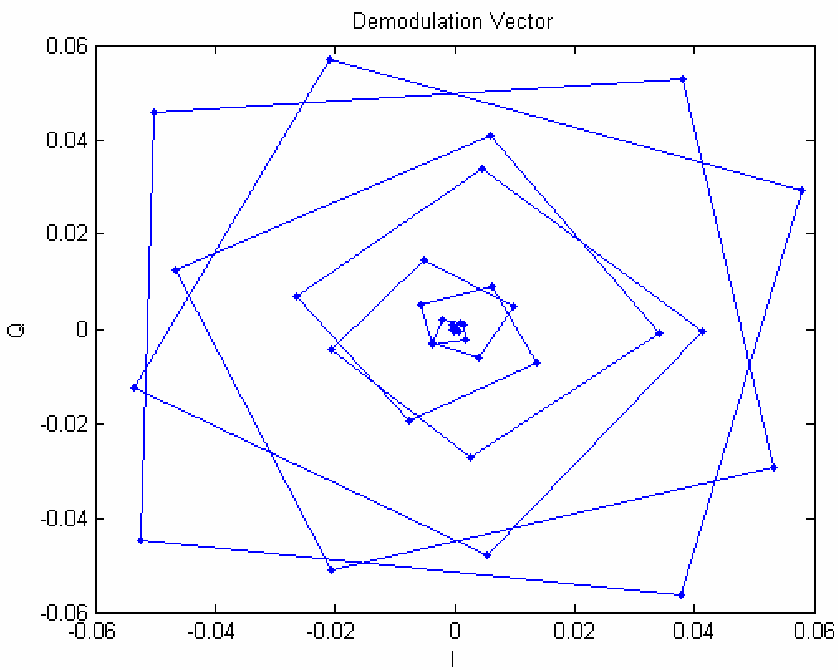
1. Mix waveform with LO
   i.e. frequency shift
2. Lowpass filter
3. Evaluate amplitude & phase at some fixed time
4. Normalize to reference cavity amplitude & phase
5. Shift, project out (position, tilt) components, & scale.

- Note that steps 1-3 above is numerically equivalent to taking the dot product of a vector of N_f elements with N_f digitized values of the cavity waveform where N_f is the length of the filter vector.

Demodulation Vector

- To get complex amplitude
- Dot this into equal length vector from cavity waveform





Saturation

- Many runs have few or no saturated pulses
  - (just ignore them?)
- My saturation algorithm:
  - Use first demodulated sample at or after nominal sample time uncontaminated by saturated ADC data
  - Correct phase using cavity frequency to roll back phase
  - Correct amplitude using decay constant to extrapolate amplitude back to nominal time.
- Not much study of how well this works.

Interfering Signals

- Sources:
  - Monopole Mode(s)
  - X-Y coupling
  - Pink (1/f) noise
  - Quadrupole modes
- What do they look like?
  - Time domain
  - Frequency domain
- How do they affect position measurement?
  - Quick answers;
  - Monopole mode => Fixed offset
  - X-Y coupling => linear transformation in (I,Q) space
  - Pink Noise => Suppressed by DDC
  - Quadrupole modes => Nonlinearity in (x,y) transfer function for large excursions

To Do

- Install Digital Downconversion in ROOT
- Approximately optimize
  - Filtering
  - Sample time
- Investigate handling of saturation
- Understand Calibration
- Should we establish DST files?
  - what format?