Digital Signal Processing (in 2700 seconds)

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Advanced Photon Source

Some Applications of Digital Signal Processing

Consumer Applications

- Communications
  - Digital cellular phones
  - Echo cancellation
- Data compression
  - HDTV
  - MP3 digital audio
- Video games
- Automobiles
  - Engine management
  - Adaptive suspension

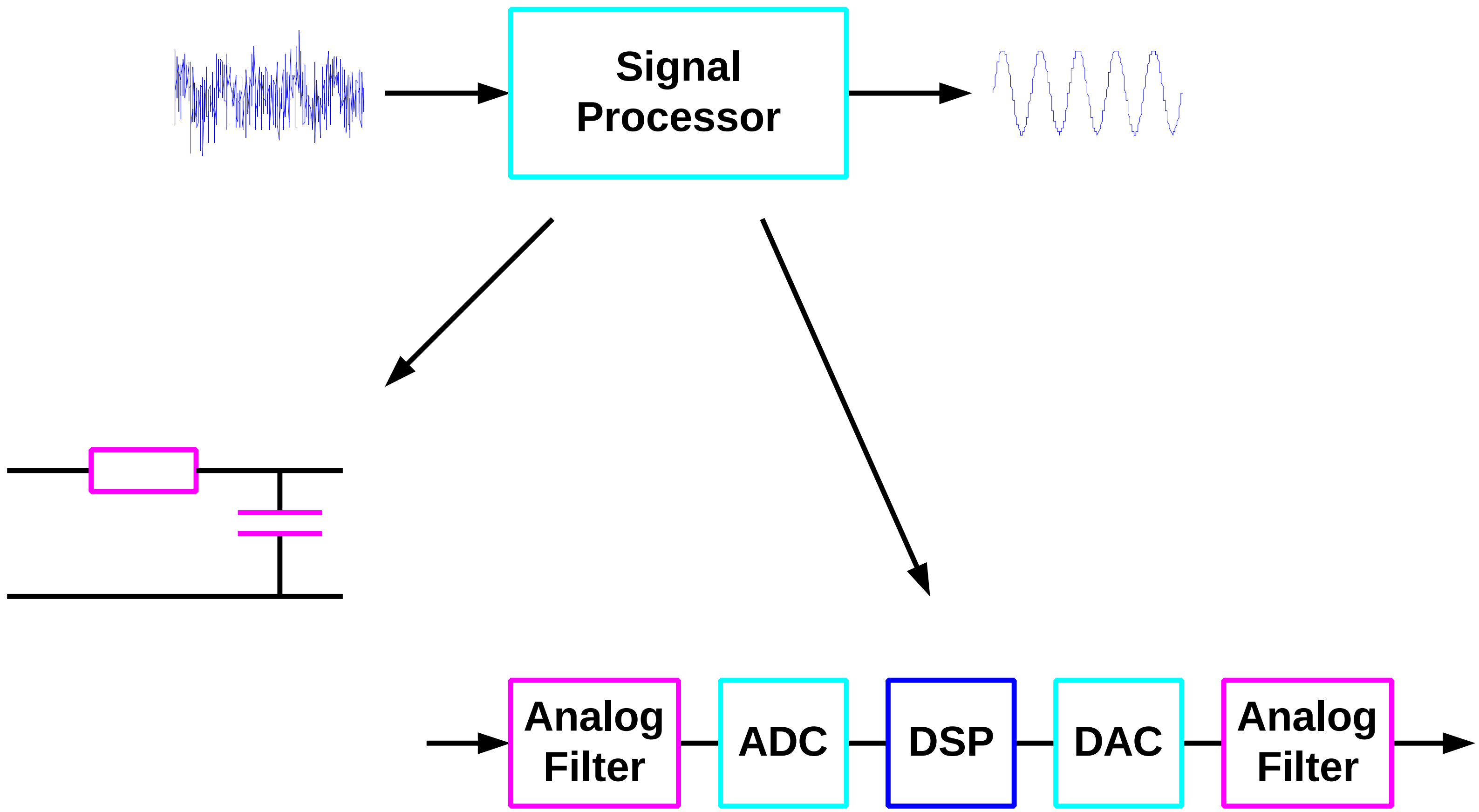
Accelerator Applications

- BPM Processing
- Feedback control
  - orbit control
  - multi-bunch feedback
- RF applications
  - direct digital down-conversion
  - digital I/Q sampling
- Accelerator tuning
  - spectral estimation
- High precision power supplies

Overview

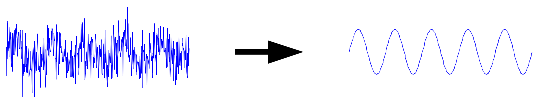
- DSP Essentials
- Filtering BPM data
- Direct digital down-conversion
- Optimal & Adaptive Filters
- DSP Hardware

“Black Box” View

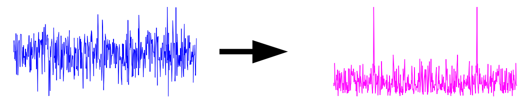


Key DSP Operations

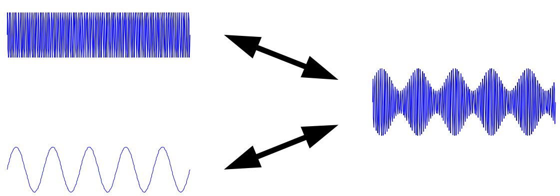
- Filtering



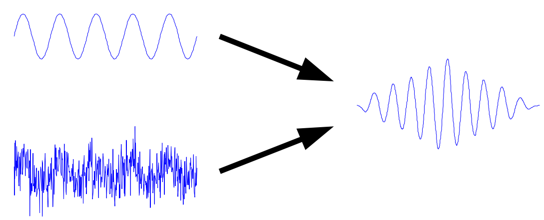
- Transformation into another domain



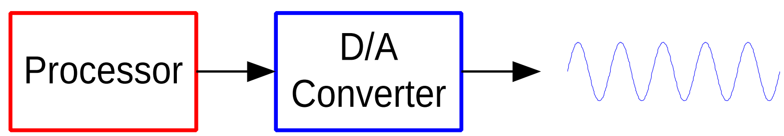
- Modulation and demodulation



- Correlation of two signals

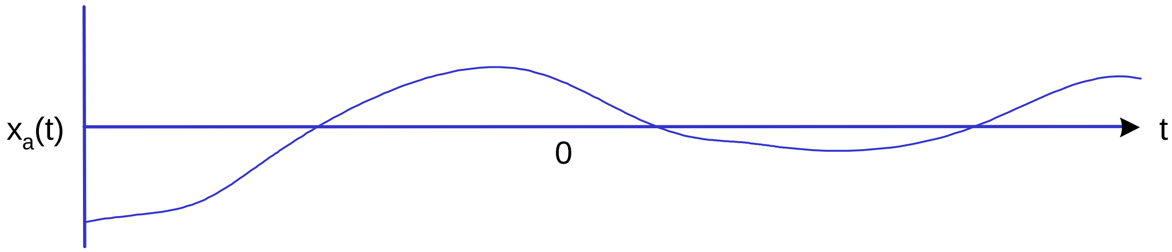


- Signal generation, frequency synthesis

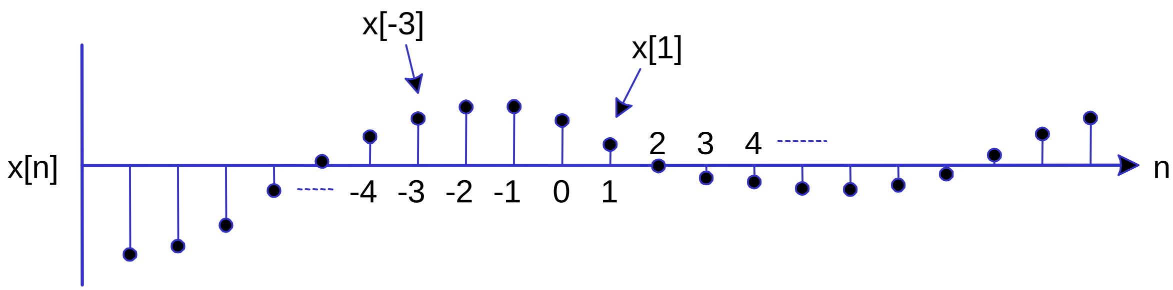


Discrete-Time vs Continuous-Time

- Continuous-time signals are functions of a continuous-valued independent variable *t*, and they exist at all values of *t*.

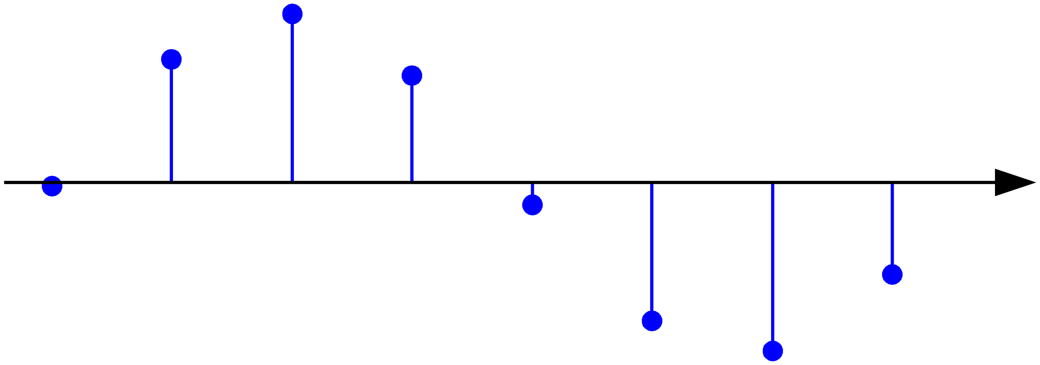


- Discrete-time signals are functions of an integer-valued index (eg *n*, *m*, *k*), and the signals have no meaning for non-integer values of the independent variable.



Ambiguity of Sampled-Data Signals

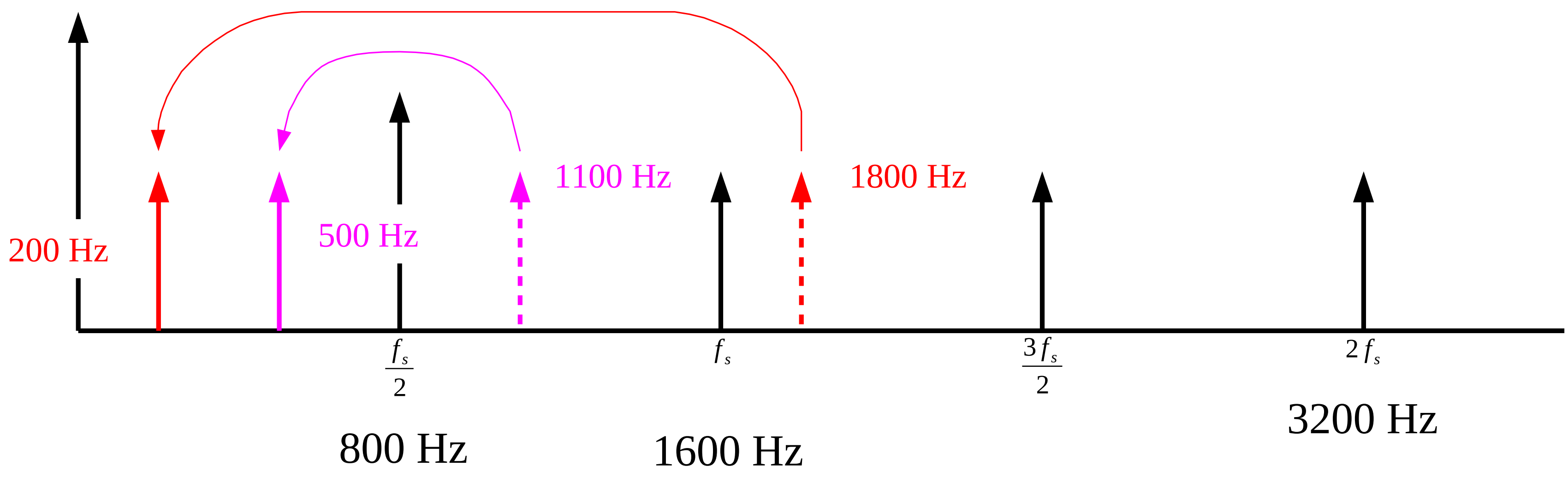
- Which continuous-time signal does this discrete-time sequence represent?



- Knowing the sampling rate, is not enough to uniquely reconstruct a continuous-time signal from a discrete-time sequence.
- The uncertainty is a result of *aliasing*.

Aliasing of Tones

- Single Frequency Tones greater than $f_s/2$ appear as aliases.
- Consider the following spectrum that is sampled at 1600Hz.



How to Avoid Aliasing

- Digitize the analog signal at least 2x the highest frequency component (Shannon's Sampling Theory).
- Use analog *anti-aliasing* filter to get rid of high frequency components <u>before</u> the digitizer.

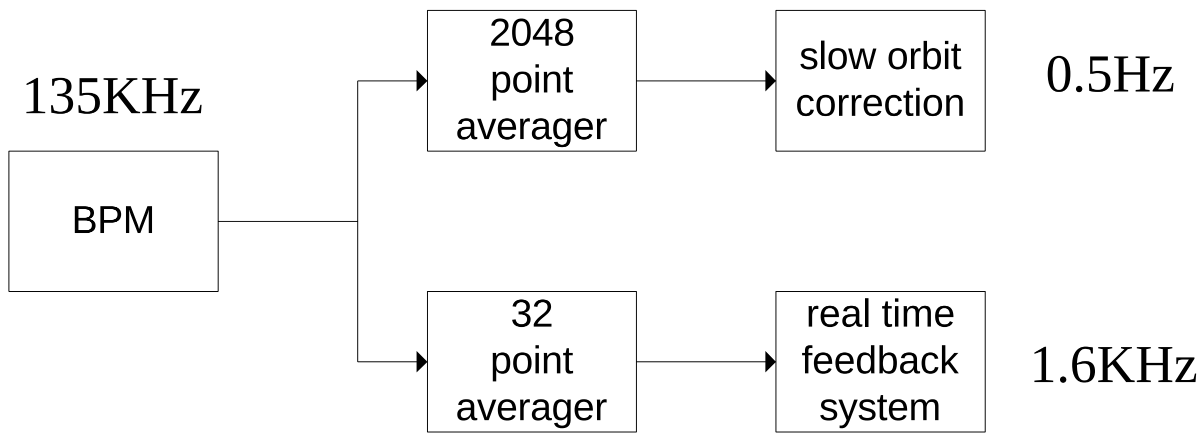
- Realize there will *always* be aliasing to some degree, the question is how much can be tolerated...

APS Turn-by-Turn Beam Position Monitors

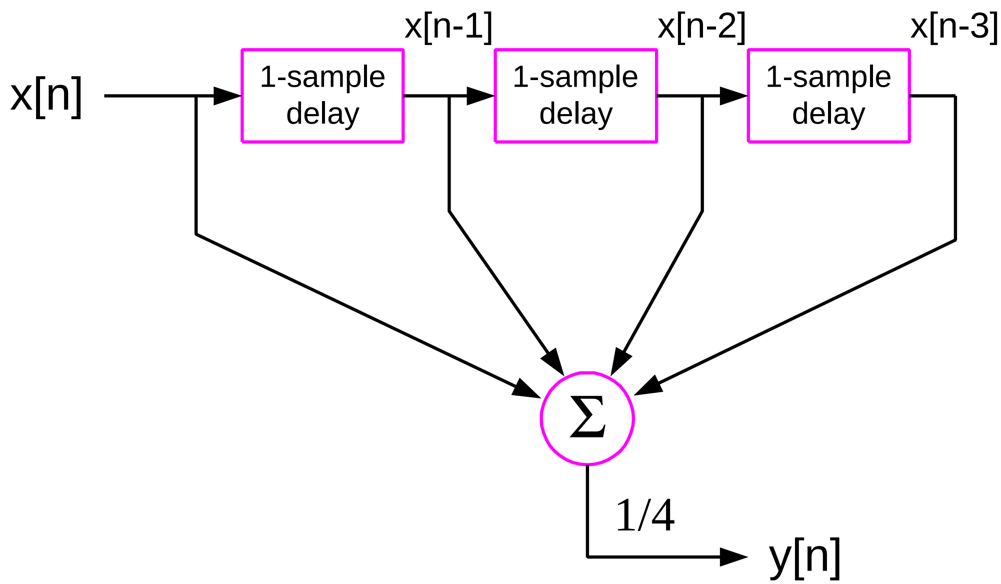
- 360 beam position monitors in each plane.
- BPMs are digitized every 7.4μS
- Data is averaged to get rid of high frequency noise so it can be used for orbit control
  - RT orbit feedback, running at 1.6KHz
  - Orbit Correction, running at 0.5Hz

BPM Memory Scanner

- Boxcar averaging is used to lowpass filter turn-by-turn data for the orbit correction systems.



Averager Block Diagram (DSP Viewpoint)



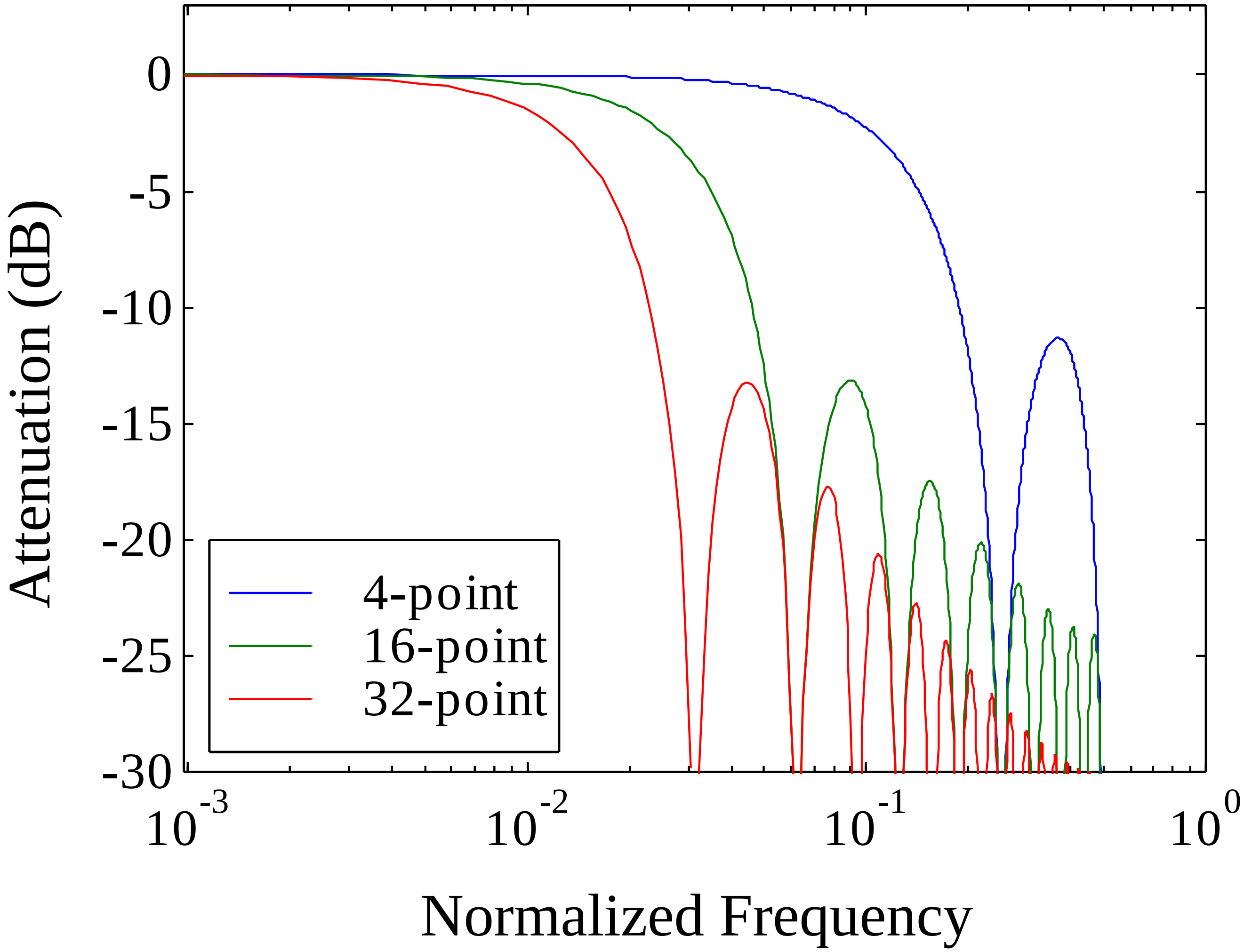
- This can be described with the following *difference equation*

$$y[n] = 0.25 \cdot \left(x[n] + x[n-1] + x[n-2] + x[n-3] \right)$$

- Or with the following z-transform transfer function

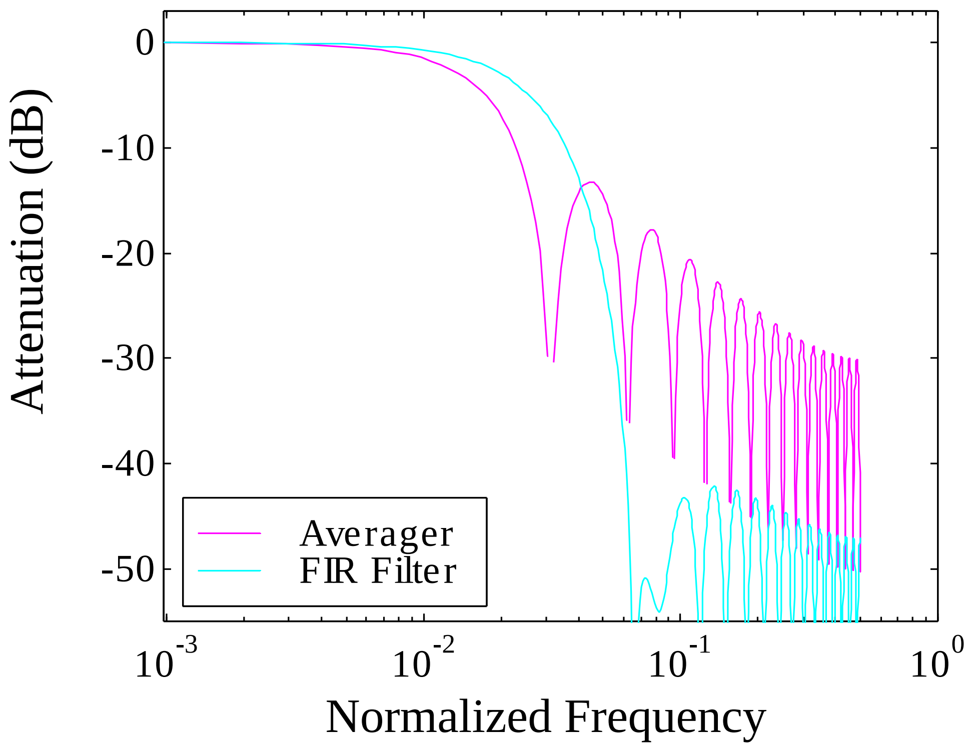
$$H_{lp}(z) = \frac{Y(z)}{X(z)} = \frac{1}{4}(1 + z^{-1} + z^{-2} + z^{-3})$$

Averagers with Different Number of Points

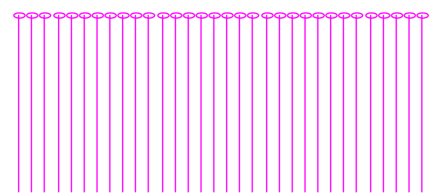


32-Tap Averager vs 32-Tap FIR Filter

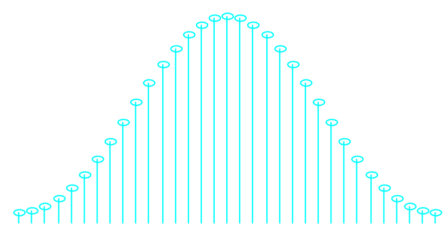
- A boxcar averager is simple to implement, but does not provide the optimum level of filtering



Averager Coefficients

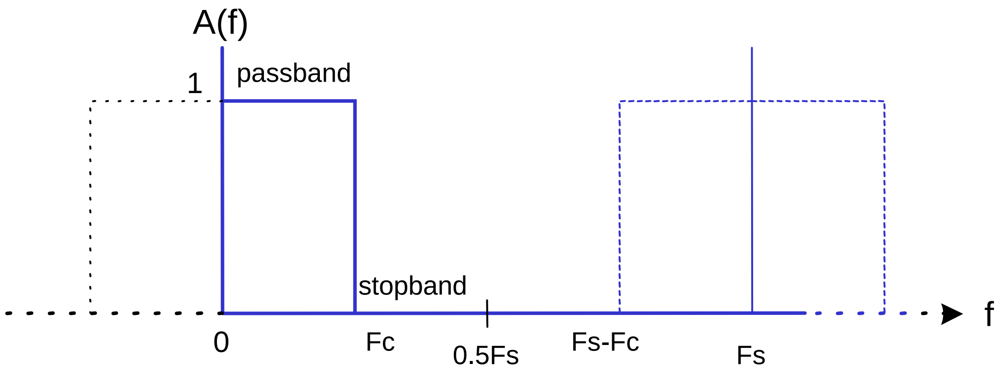


FIR Filter Coefficients

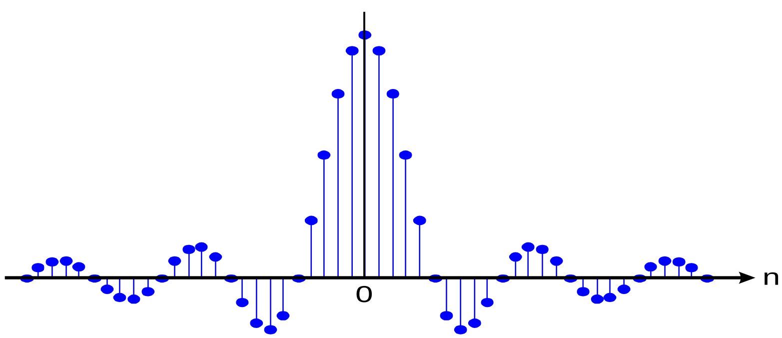


Ideal Frequency-Selective Digital Filters

- An ideal frequency-selective lowpass filter has a passband with constant magnitude, an infinitely sharp transition between passband and stopband, and infinite attenuation in the stopband. The phase delay is zero for all frequencies.



- The impulse response (coefficient weights) of this ideal filter follow a doubly-infinite sin(*x*)/*x* function



Radio Frequency Applications of DSP

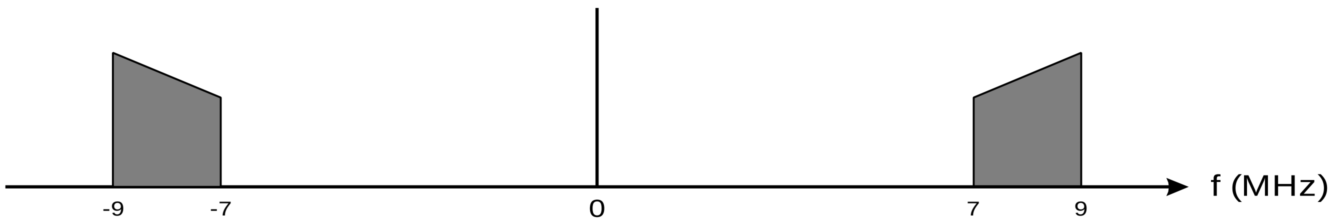
- Two particular methods of sampling RF signals are gaining attention because of the advent of high-speed A/D converters.
- Both are associated with sampling band-limited signals that ride on a high-frequency carrier, eg
  - sampling RF probes for cavity field control.
  - accelerator tune measurement.

- Direct digital down-conversion (software radio)
  - eliminates the need for a conventional analog RF mixer.

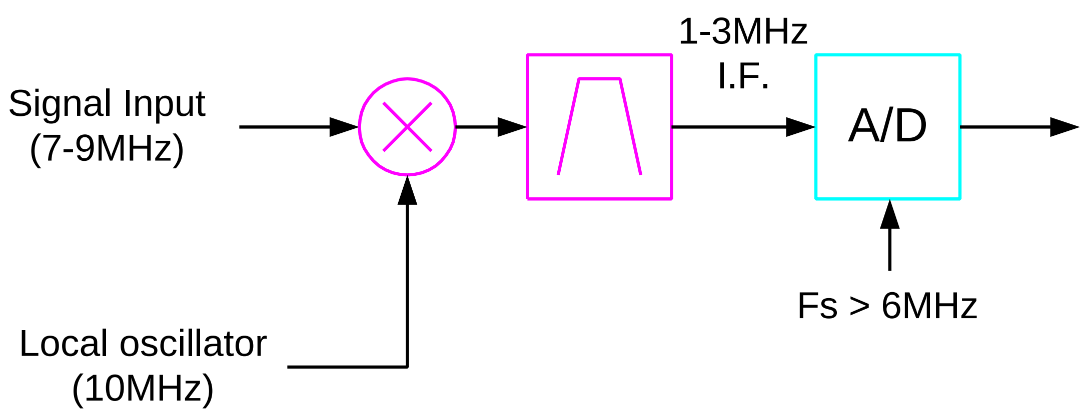
- Digital I/Q sampling
  - eliminates difficulties associated with detecting in-phase and quadrature components of an RF signal.

Sampling Band-limited Signals

- Consider a 2MHz band-limited signal riding on an 8MHz carrier.

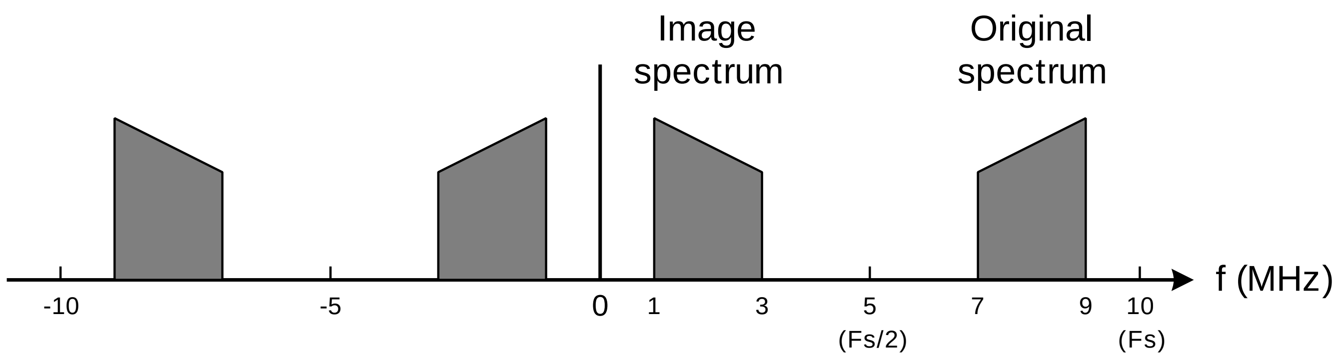


- The IF could be extracted by mixing with a local oscillator at 10MHz and sampled at 6MHz, or could be directly sampled at > 18MHz.



Bandpass Sampling Example

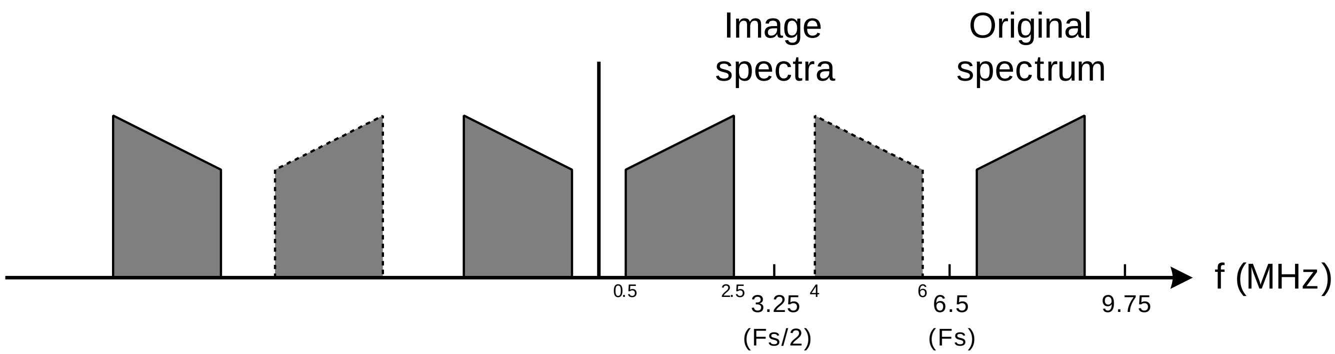
- Instead, let's directly sample the signal at only 10M samples/second.



- In this case the Nyquist frequency would be 5MHz, and the original spectrum is in the range of *Fs/2* to *Fs*, instead of the range *DC-Fs/2* (as we are used to seeing).
- The original spectrum is aliased into the lower half of the frequency band, reflected about the Nyquist rate of 5MHz, appearing in the frequency range 3Mhz - 1MHz.
- So, we have successfully sampled the signal using a sampling rate almost half the 'officially' required rate

Bandpass Sampling Example (cont)

- What if we sample at only 6.5M samples/second??



- This time the original spectrum lies between *Fs* and 1.5*Fs*.
- Here, the spectrum is reflected about the sampling rate, to appear in the range from *Fs/2* to *Fs*, spanning 6MHz - 4MHz.
- It is then reflected a second time about *Fs/2*, finally appearing in the lower half of the sampled frequency range between 0.5MHz and 2.5MHz.

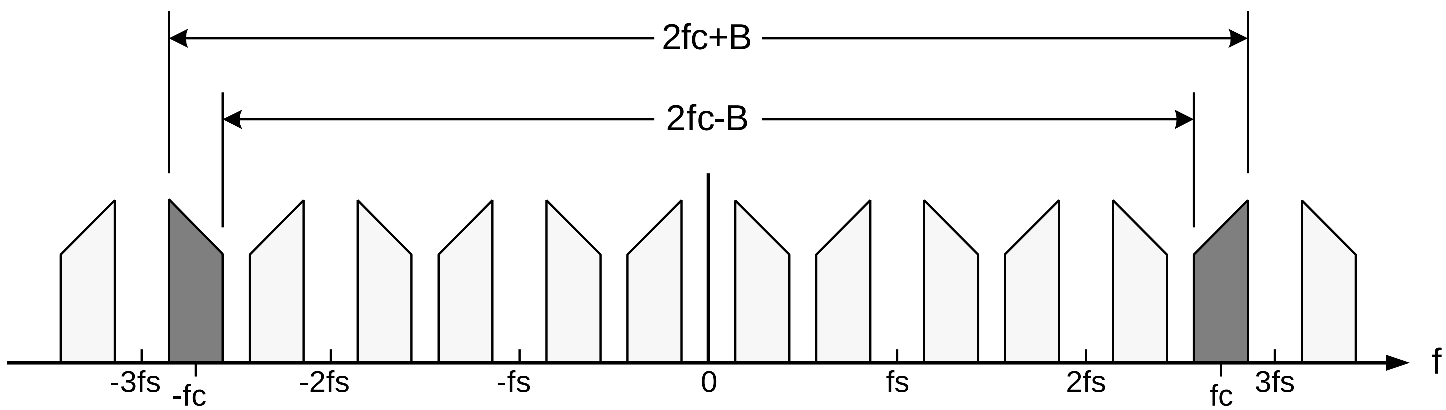
Can we sample at an even lower rate and still get a unique spectrum??

General Case of Bandpass Sampling

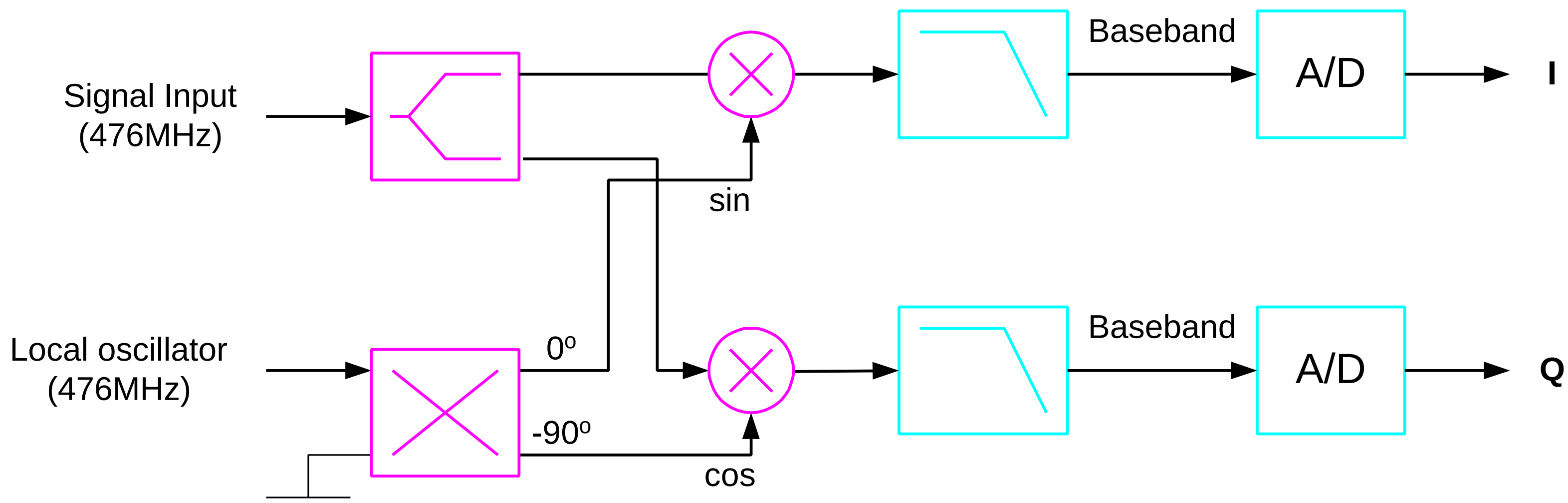
- In general, it can be shown that if there are *m* image spectra between the original and its negative image, the range of possible sampling frequencies is given by the expression

$$\frac{2fc - B}{m} \geq fs \geq \frac{2fc + B}{m+1}$$

- Example with $m = 5$



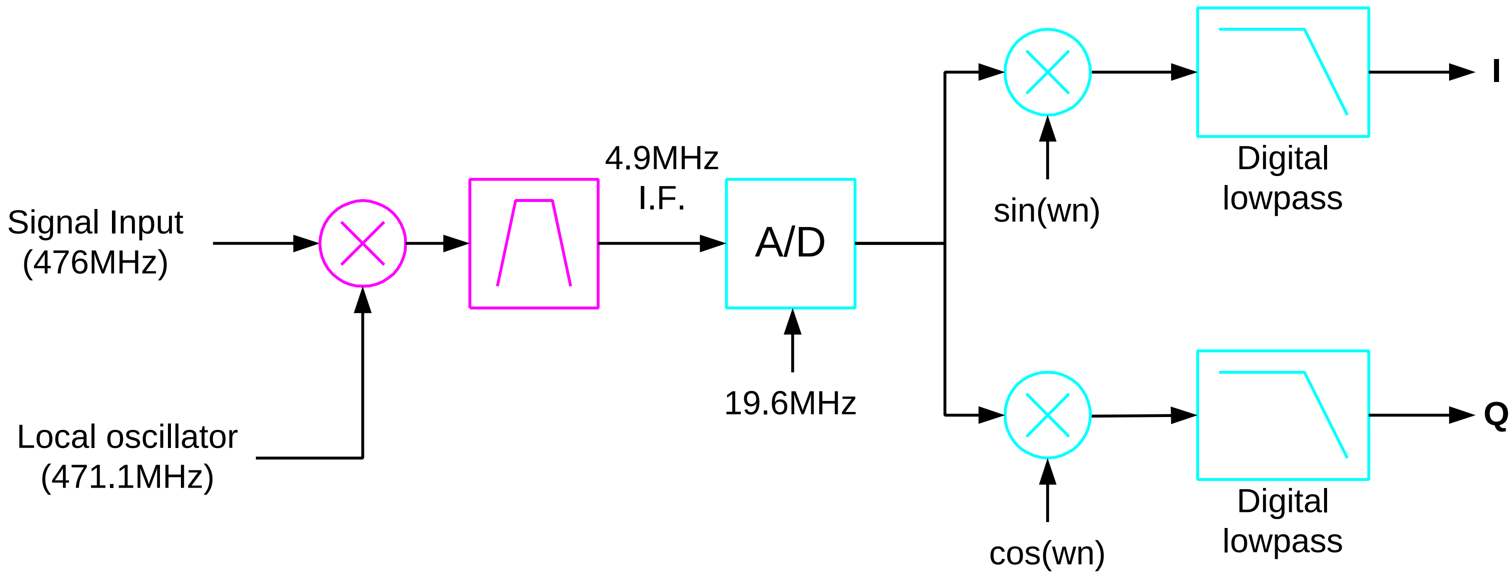
Analog I/Q Detector



- Issues: DC offsets in mixer, quadrature phase errors, impedance matching, ...

Quadrature Sampling with Digital Mixing

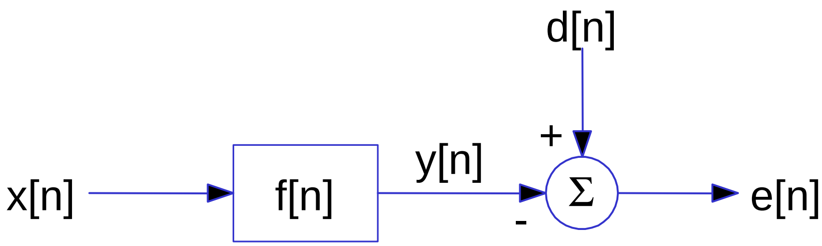
- Digital technology now offers a completely digital approach to this problem.



- The continuous-time signal is sampled at exactly 4 times the IF frequency.
- Digital sine and cosine signals are multiplied with the incoming discrete-time sequence to generate the real and imaginary part of the signal.

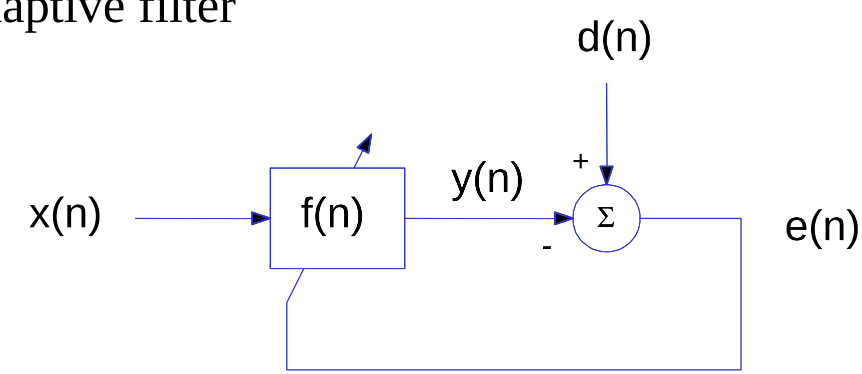
Optimal and Adaptive Filters

- Consider a situation where a signal *x[n]* is to be filtered so that the output sequence is as close as possible to a desired signal *d[n]*



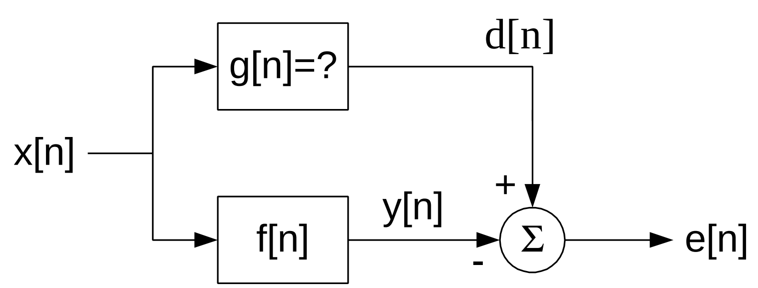
$$e[n] = d[n] - \sum_k f[k] \cdot x[n-k]$$

- If the statistics of the input process are known and stationary, the optimum filter coefficients can be determined using a set of *Normal Equations.*
- If we don't know the statistics exactly (or if they are time-varying), we need an adaptive filter

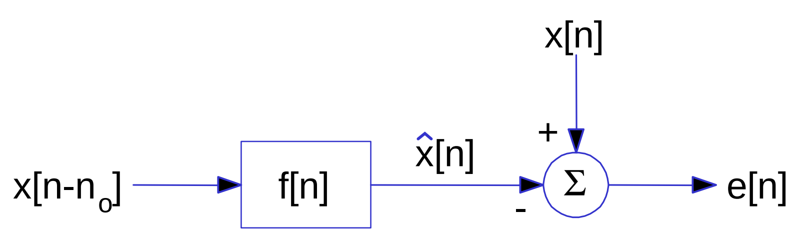


Applications of Optimal Filters

- System identification - *generate linear model of unknown system*

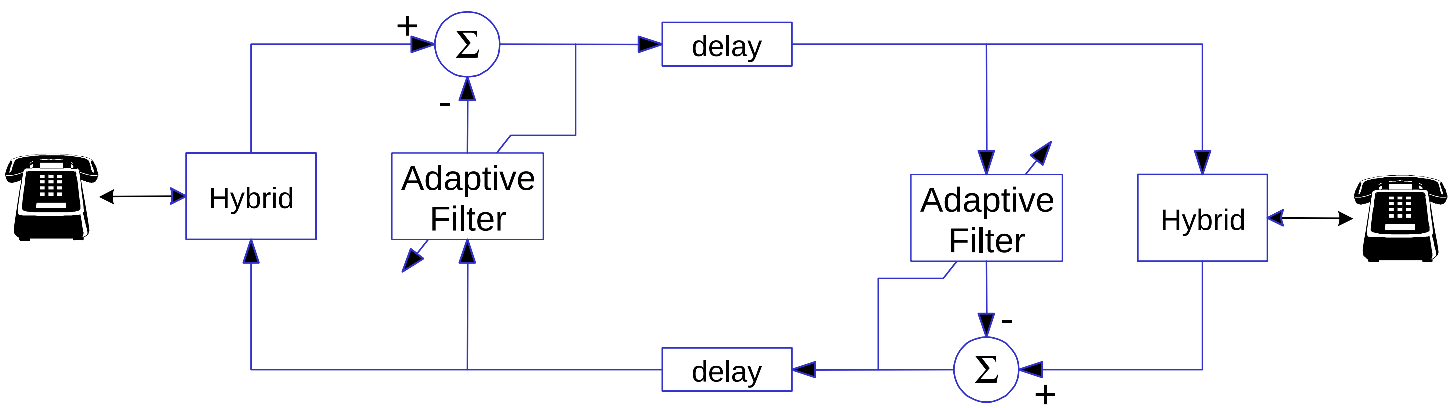


- Linear prediction - *estimate the <u>future</u> value of a signal*

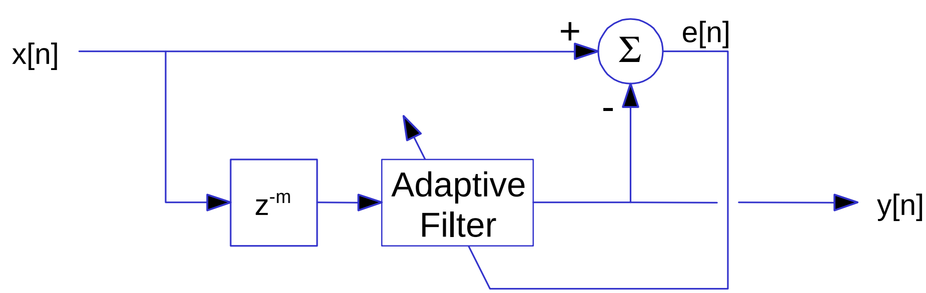


Adaptive Filter Application Examples

- Adaptive echo cancellation

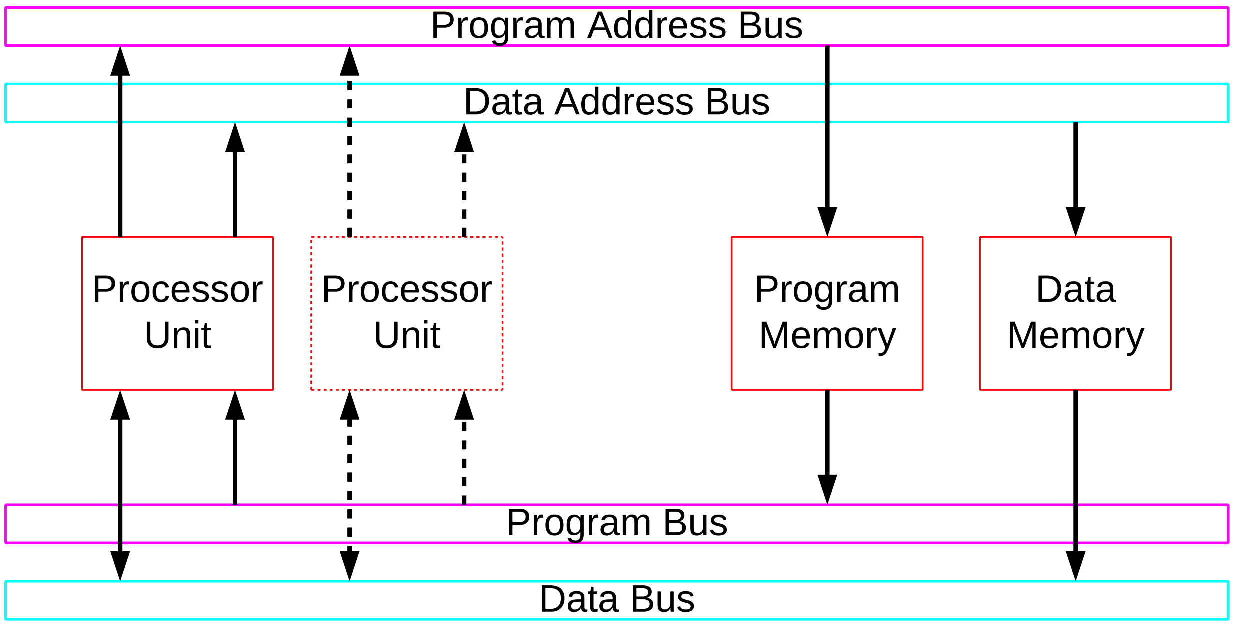


- Adaptive line enhancement (detect small periodic signals buried in noise)



DSP Processors

- DSP processors are optimized for *Multiply/Accumulate* (MAC) operations.
- Multiple data/program busses inside the chip allow simultaneous access to program and data memory (Harvard Architecture).
- Modern DSP chips can implement up to 8 instructions in a single clock cycle.



DSP Processor Performance

- Digital Signal Processor chips are amazingly fast!
  - TI C67 ($200): 32-bit floating-point, 1GFLOP
  - ADI SHARC ($60): 32-bit floating-point, 150MFLOP

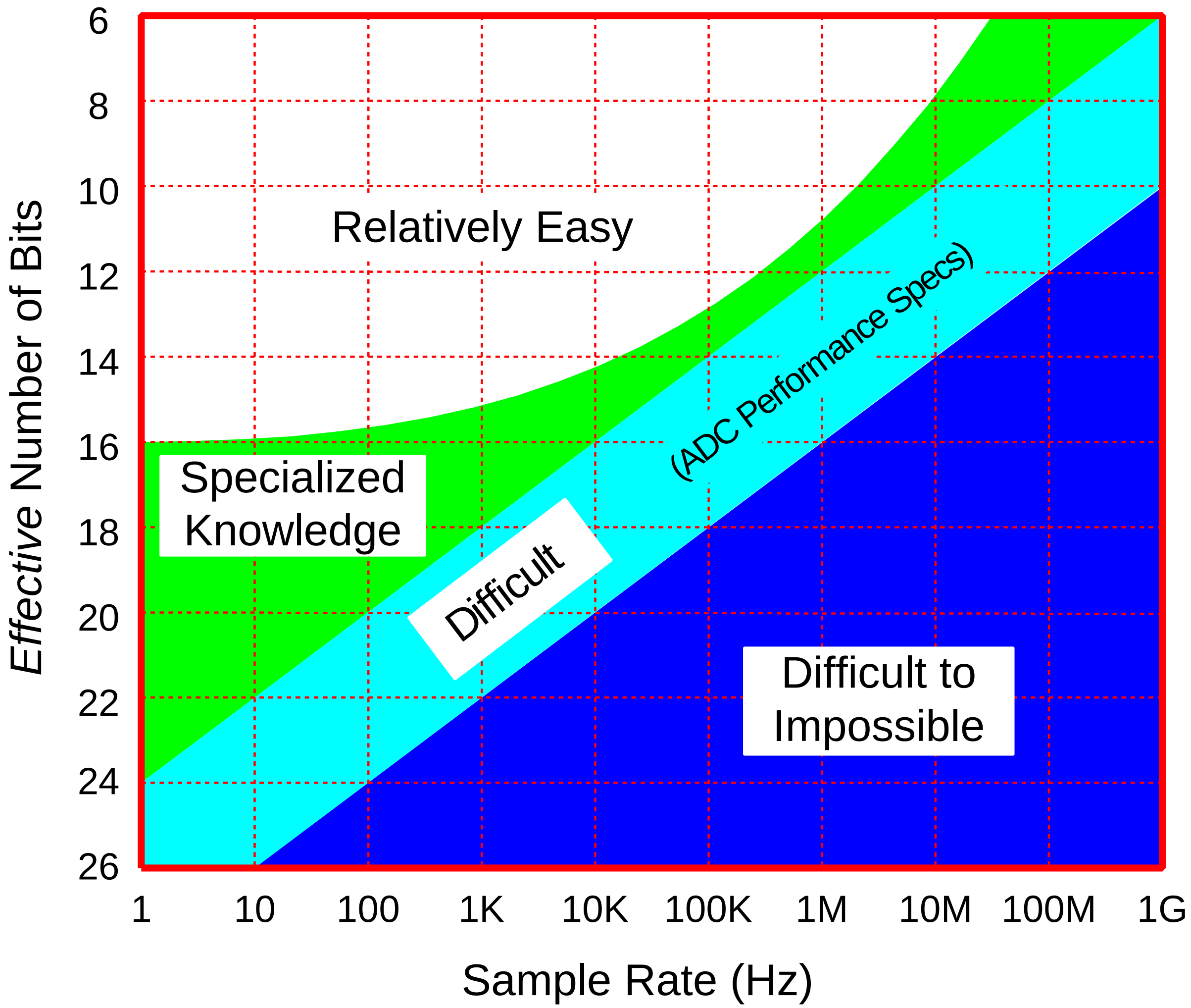
1024 Point Complex Radix 2 FFT with bit reversal	
Processor	**Speed (usec)**
ADI TigerSHARC @ 150MHz	69
TI C67 @ 167MHz	124
TI C40 @ 50 MHz	1435
Power PC 604e @ 333 MHz	230
Intel Pentium @ 200 MHz	750
Vax 8600	21700

Other DSP operations (Based on TI C67 @ 167 MHz)	
Operation	**Speed (usec)**
8 Cascaded Biquad filters	0.366
Matrix-Vector Multiply 38x160 * 160x1	41.2
Autocorrelation, 18 x 8	0.606

DSP Performance

- *BUT!*
  - *To achieve the benchmark performance:*
    - Algorithm must run in a tight loop
    - Processor pipeline must be kept full
    - Code for Algorithm must fit in on chip cache
    - Data arrays must be within on chip cache
    - Parallel execution must be maximized
  - Scatter/Gather operations will suffer performance degradation.

Effort Required to Achieve A/D Performance



Ref: "Practical Limits of Analog-to-Digital Conversion" (Jerry Horn)

Conclusions

- Applications for digital signal processing are exploding, largely fueled by the availability of inexpensive high performance processors.

- There are certain applications where digital is clearly better than analog.

- The accelerator community is starting to tap the capabilities of DSP technology, but this is just the beginning...