

Oversampling (3B)

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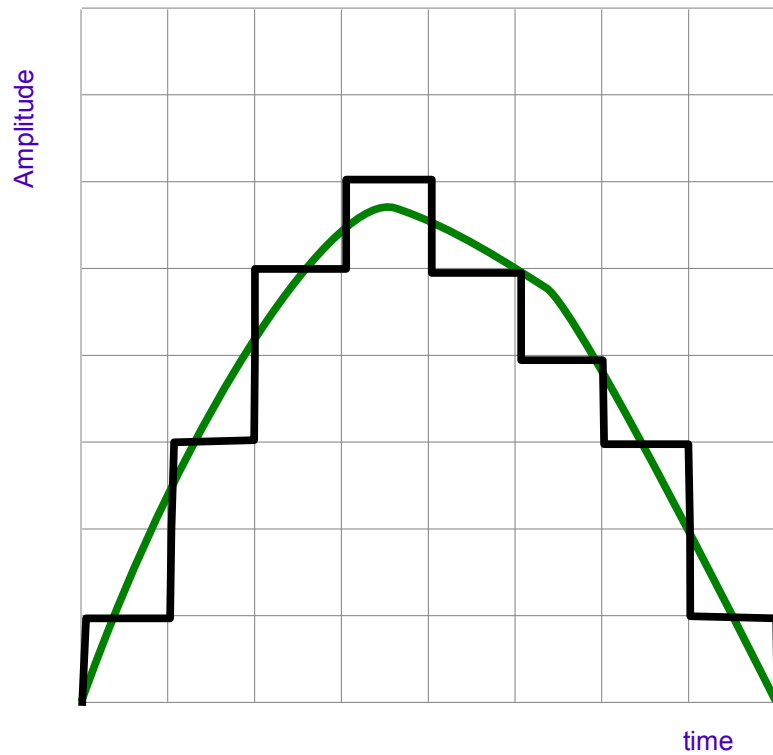
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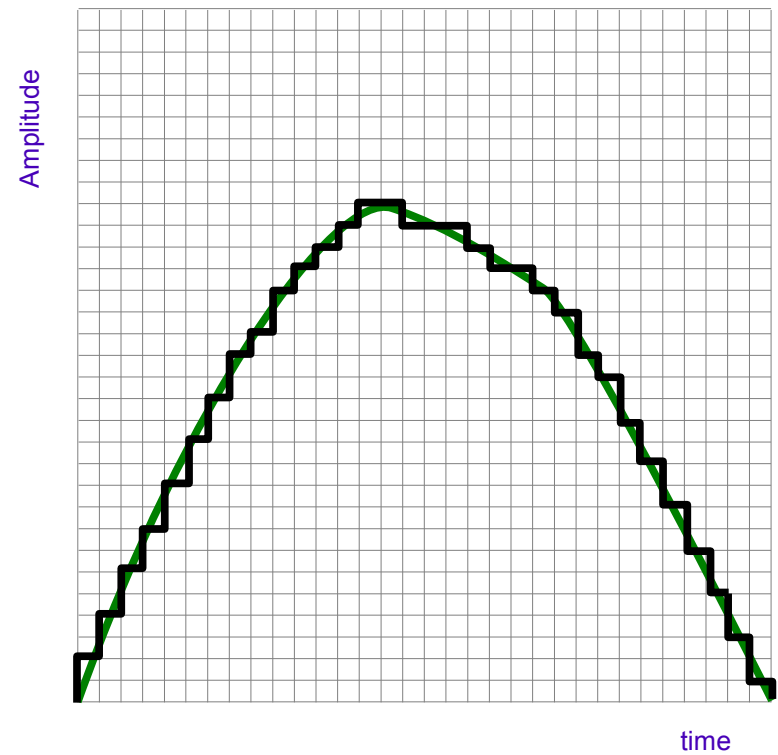
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Bits and Samples

More Samples

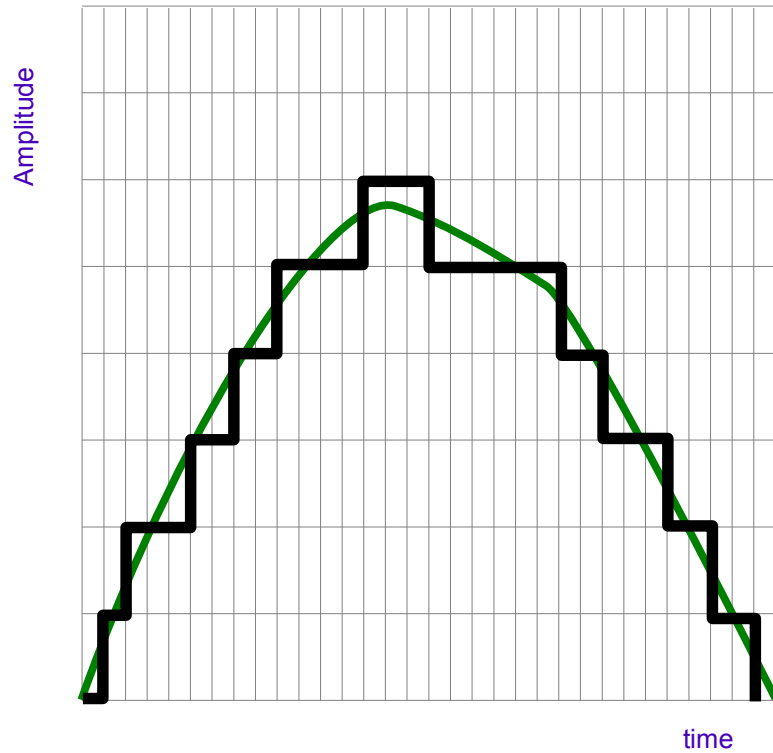


More Bits

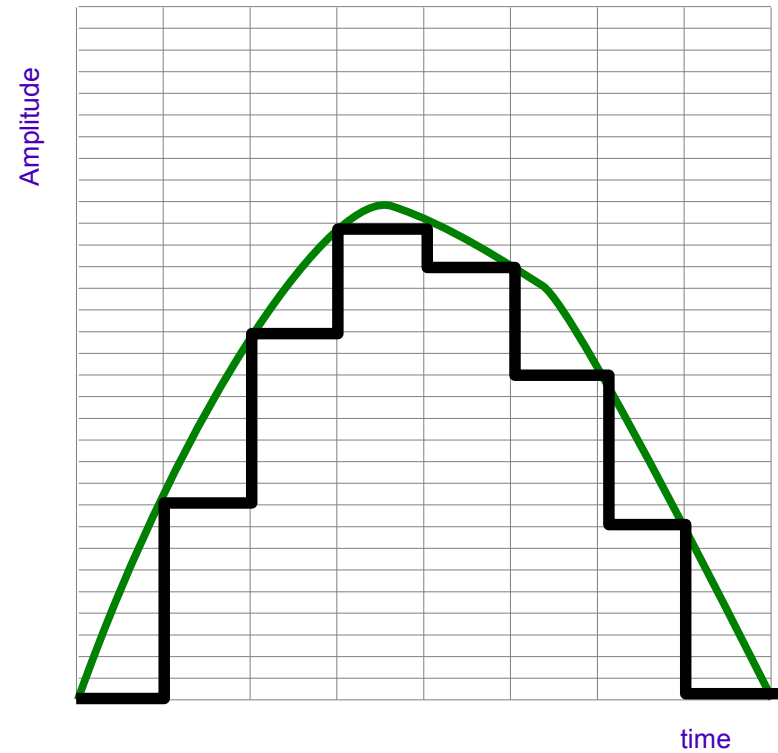


Bits and Samples

More Samples

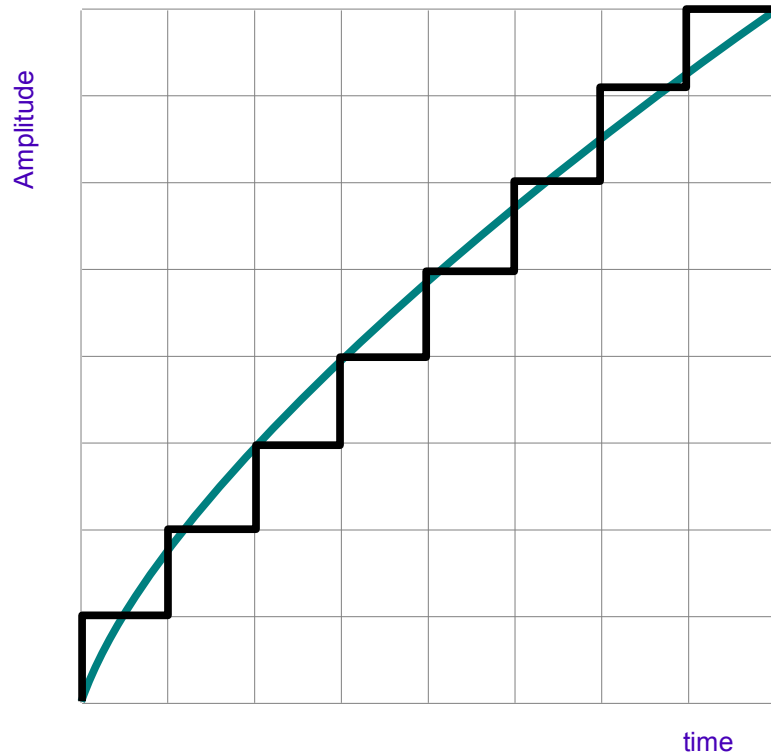


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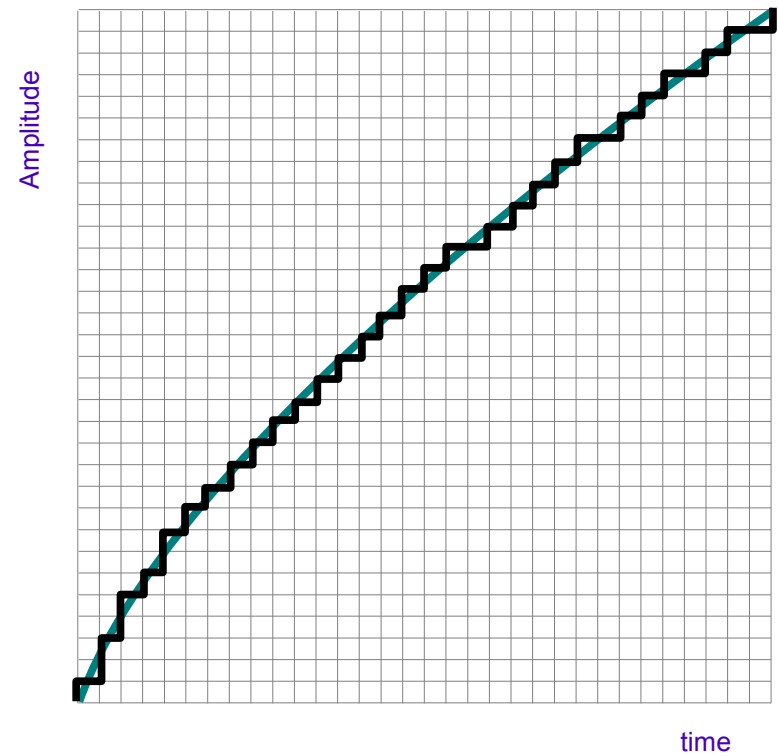


Bits and Samples

More Samples

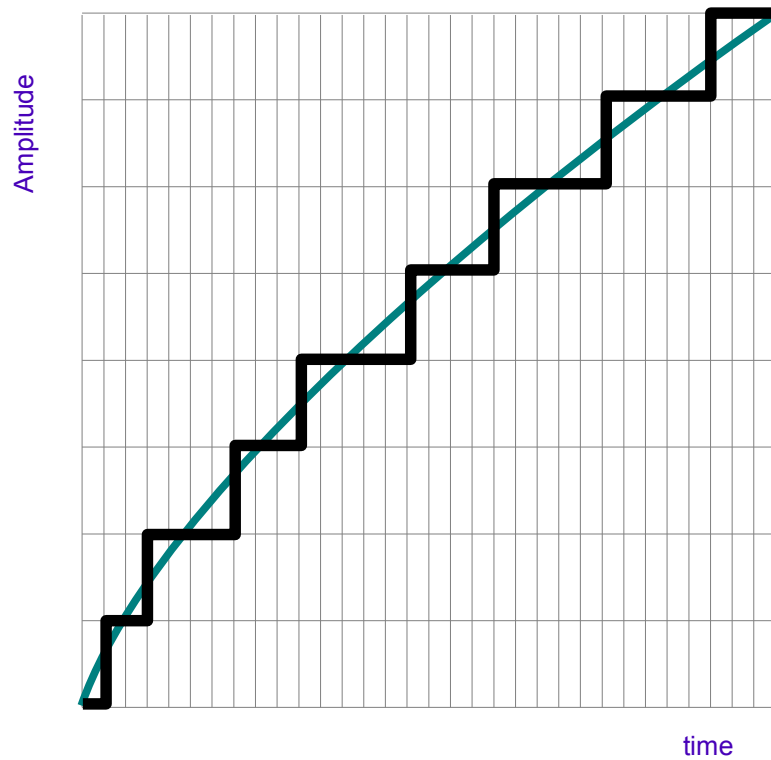


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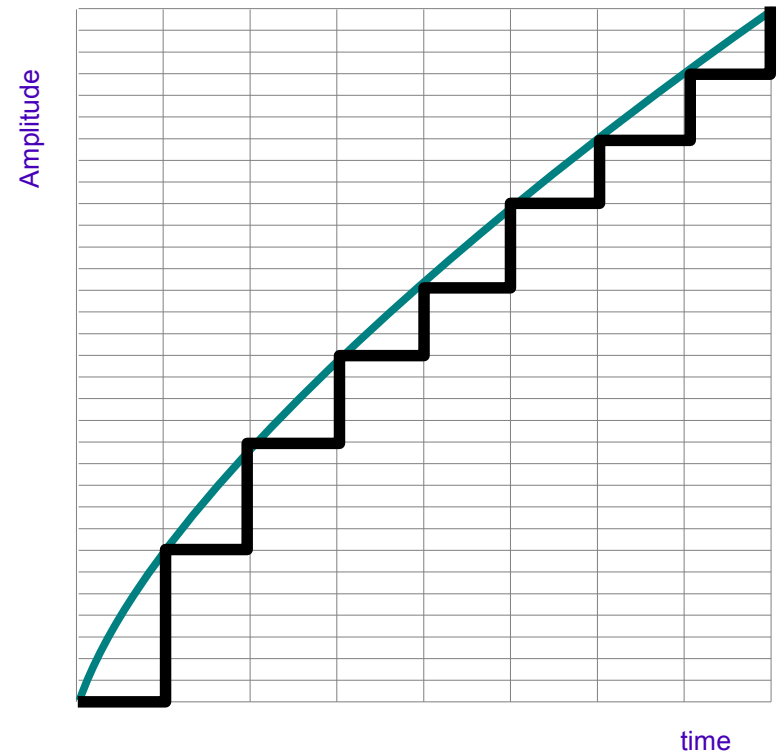


Bits and Samples

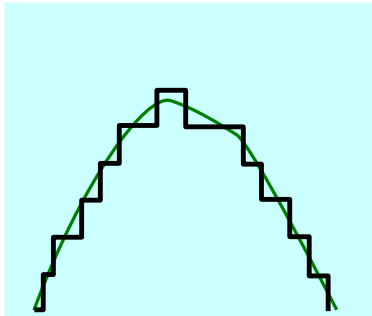
More Samples



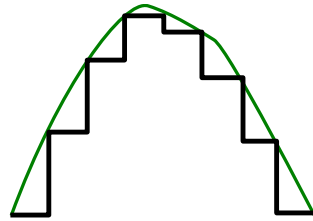
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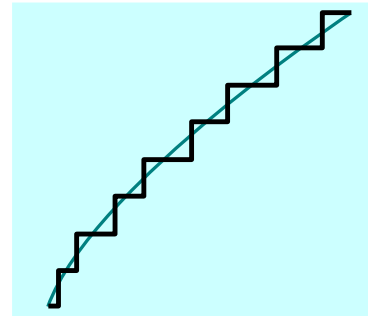
Over-Sampling



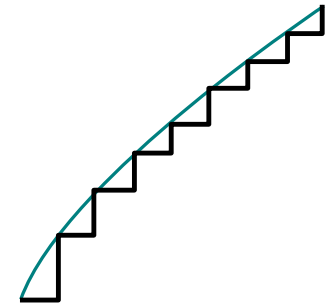
More Samples



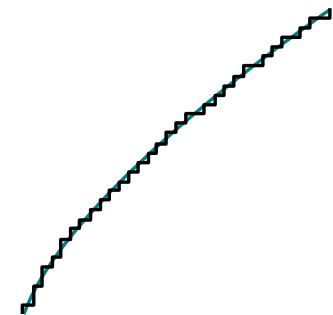
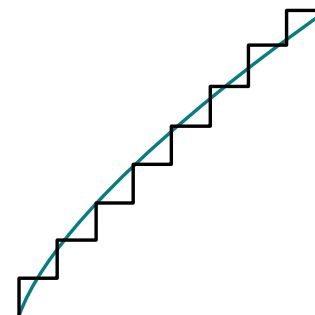
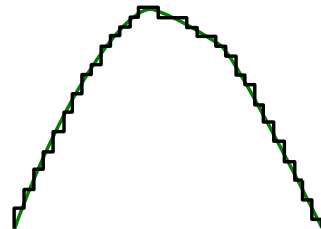
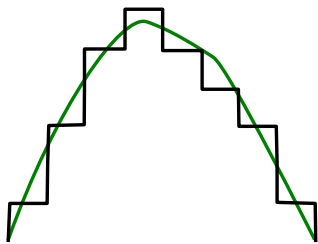
More Bits



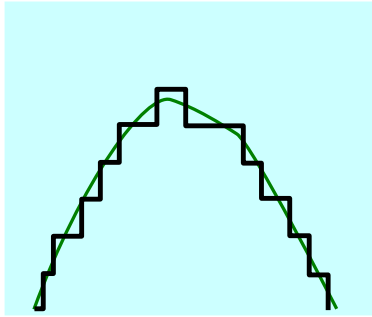
More Samples



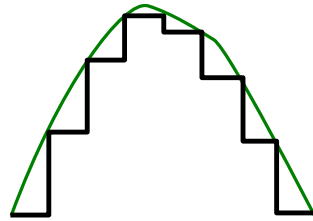
More Bits



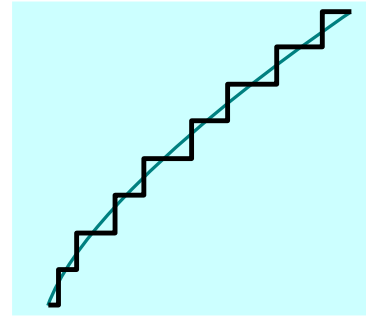
Over-Sampling



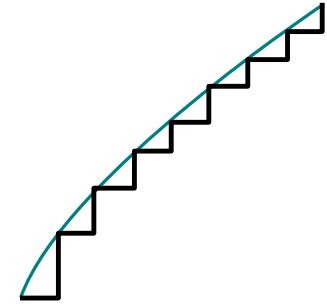
More Samples



More Bits



More Samples



More Bits

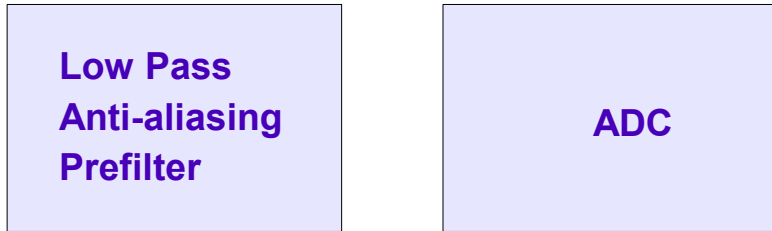
{ Sampling → Quantization in time axis
Quantization → Quantization in amplitude axis

OverSampling → trade off bits for samples
→ alleviate the need for high quality prefilters and postfilters

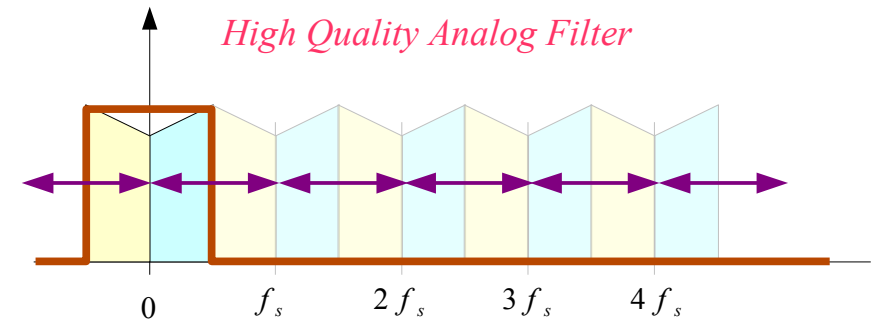
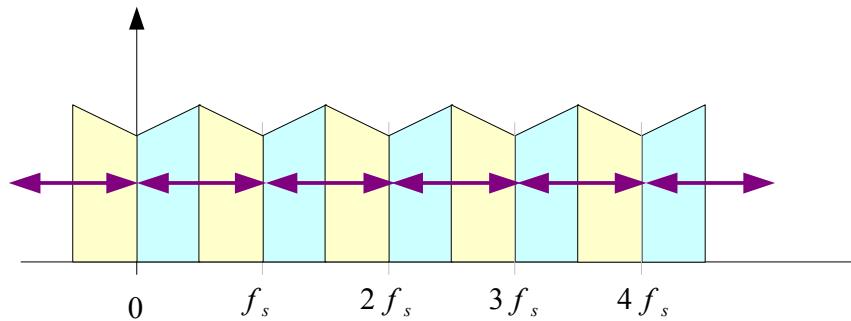
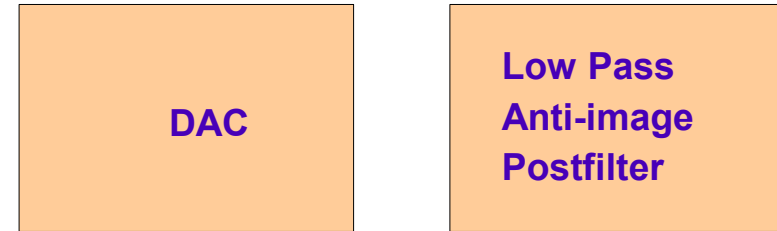
Band-limited Signal

Band-limited Signal

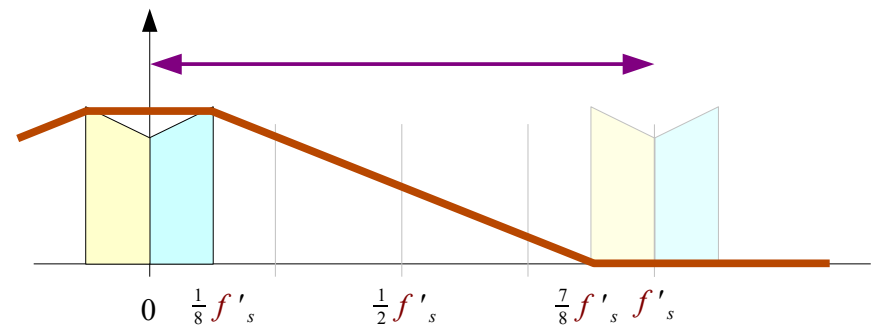
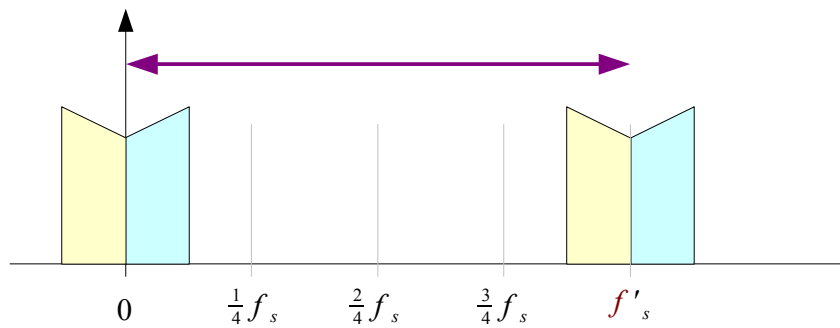
Analog Filter



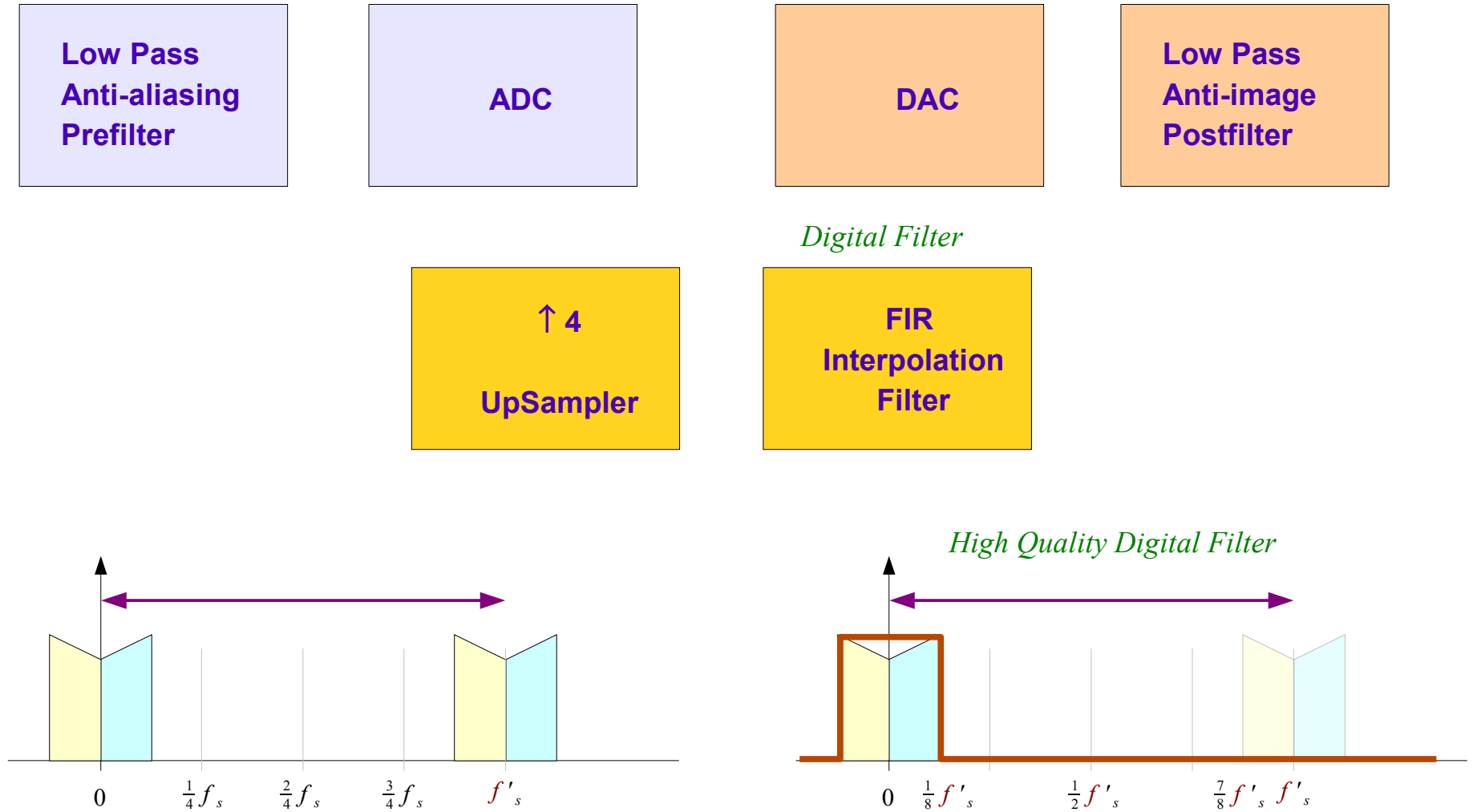
Analog Filter



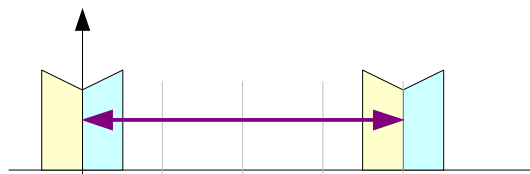
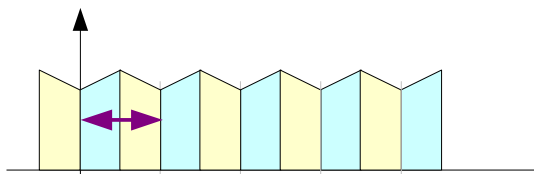
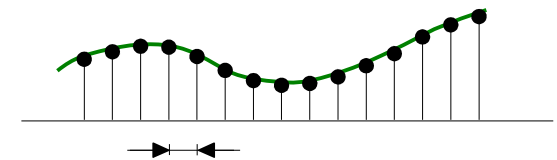
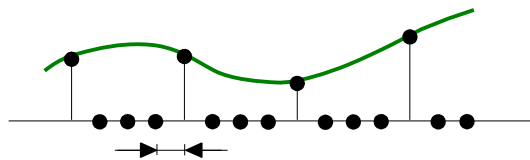
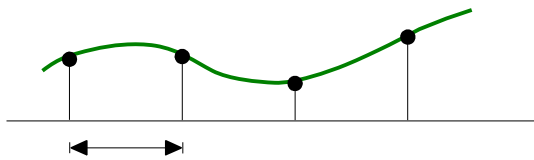
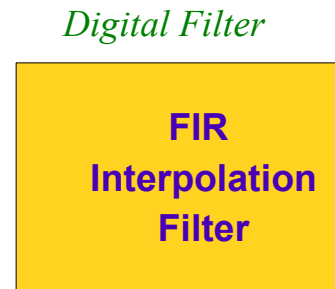
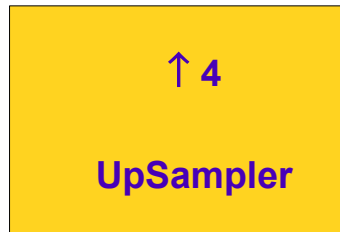
High Quality Analog Filter



Band-limited Signal



Band-limited Signal



Band-limited Signal

Low Pass
Anti-aliasing
Prefilter

ADC

DAC

Low Pass
Anti-image
Postfilter



↑ 4

$$f_{oversampling} = 4^n \cdot f_s$$

UpSampler

FIR
Interpolation
Filter

$$f_s > 2 \cdot f_H$$

Band-limited Signal

$$f_{\text{oversampling}} = 4^n \cdot f_s$$

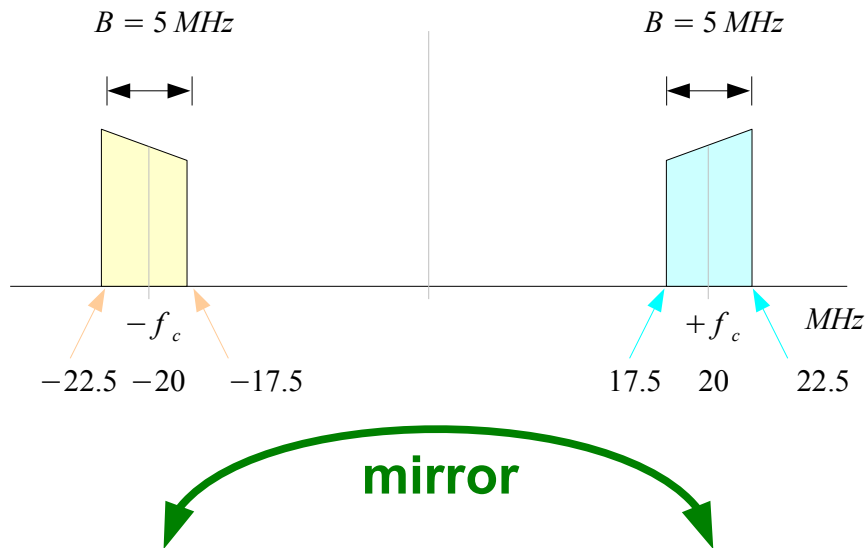
Oversampling and Decimation
Oversample and Lowpass Filter

- **Normal Averaging**
- **Decimation / Interpolation**

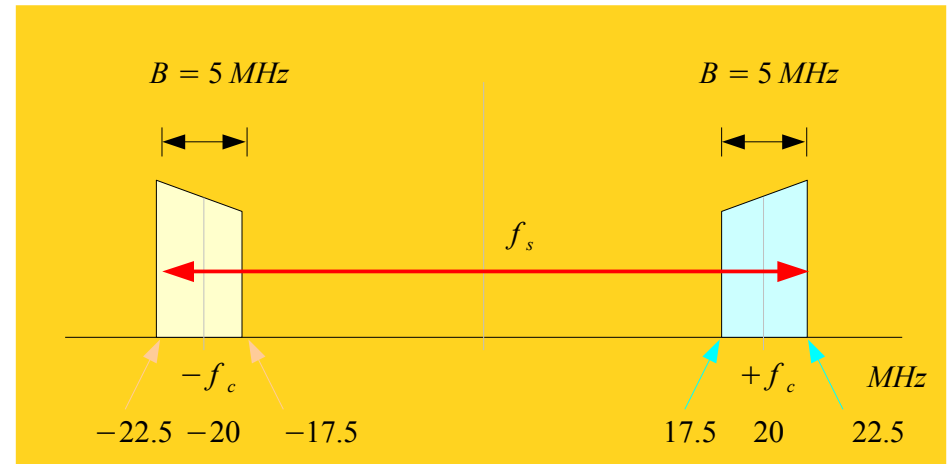


$$f_s > 2 \cdot f_H$$

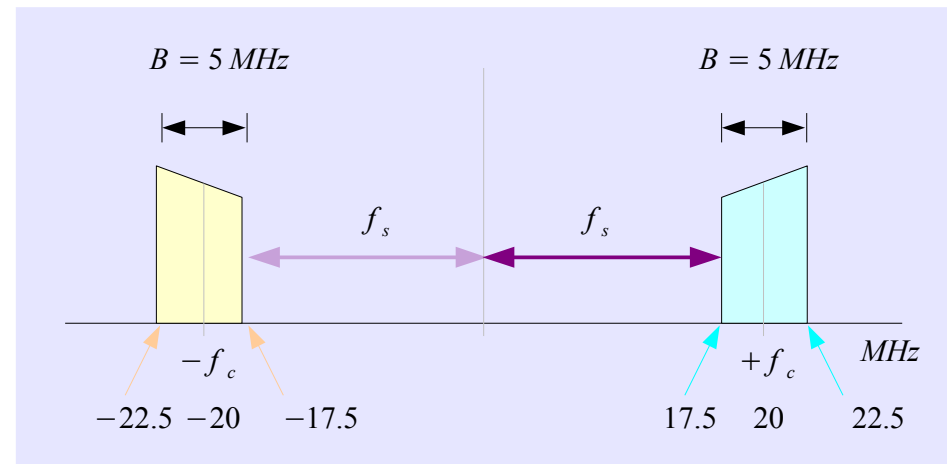
Band-limited Signal



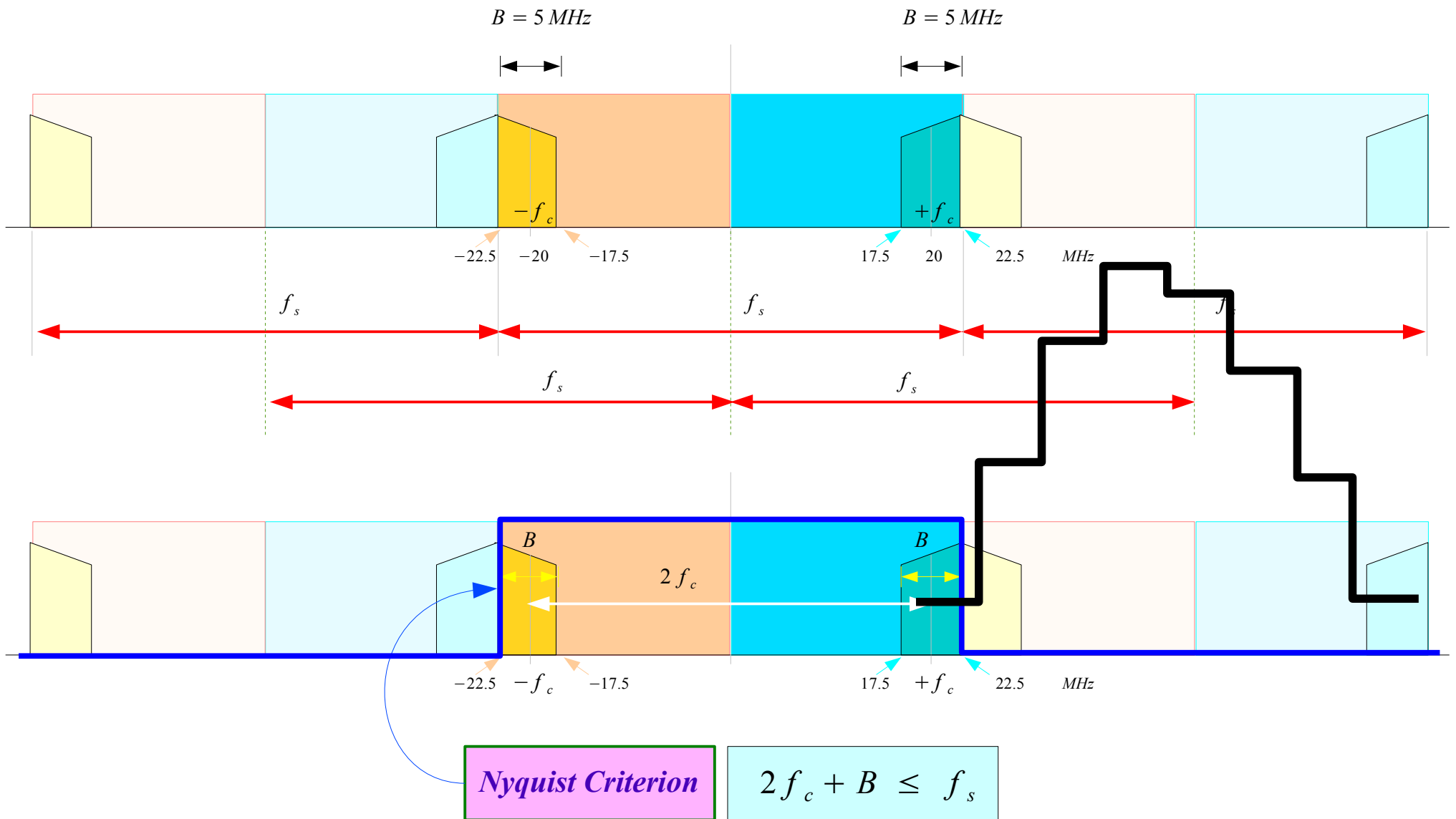
- Bandpass Sampling
- IF filtering
- Harmonic Sampling
- Sub-Nyquist Sampling



- Lowpass Sampling



Low-pass Signal Sampling



References

- [1] <http://en.wikipedia.org/>
- [2] J.H. McClellan, et al., Signal Processing First, Pearson Prentice Hall, 2003
- [3] A “graphical interpretation” of the DFT and FFT, by Steve Mann
- [4] R. G. Lyons, Understanding Digital Signal Processing, 1997
- [5] AVR121: Enhancing ADC resolution by oversampling
- [6] S.J. Orfanidis, Introduction to Signal Processing
www.ece.rutgers.edu/~orfanidi/intro2sp