

Analog Integrated Systems Design

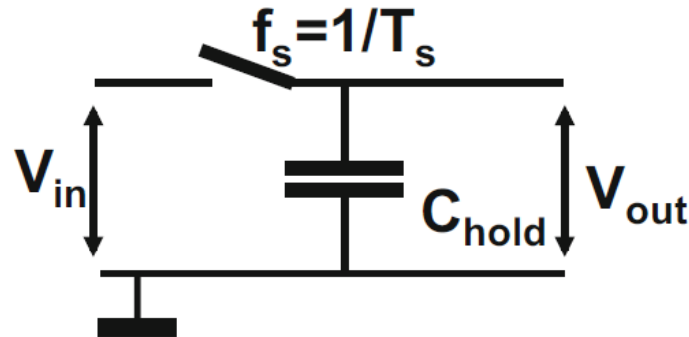
Lecture 02 Sampling

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Sampling

- ❑ Sampling is time discretization
 - Converts a continuous time (CT) signal to a discrete time (DT) signal
 - The result is a sequence of samples
- ❑ The sampling instants are defined by a clock signal ($T_s = 1/f_s$)
 - The clock signal controls an electronic switch (e.g., MOS transistor)
- ❑ The sampled signal is stored as a voltage on a capacitor
- ❑ The circuit is called sample-and-hold (S/H) circuit



$$t = \frac{n}{f_s} = nT_s, \quad n = -\infty, \dots, -3, -2, -1, 0, 1, 2, 3, \dots, \infty.$$

Time and Frequency Domains

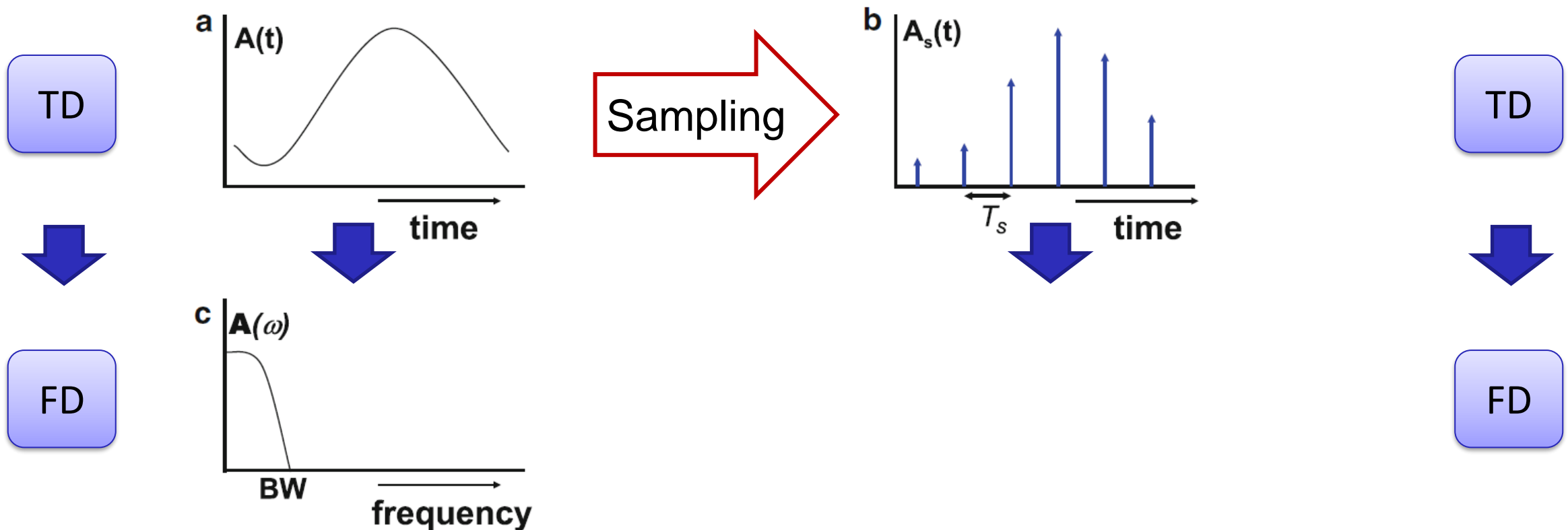
$$TD \text{ step} = \Delta t = \frac{1}{f_s} = \frac{1}{FD \text{ period}}$$

$$FD \text{ step} = \Delta f = \frac{1}{T_o} = \frac{1}{TD \text{ period}}$$

Time domain		Technique	Frequency domain		Where in the chain?
CT/DT	Periodic	\leftrightarrow	C/D	Periodic	
CT	Yes	CT Fourier series (CTFS)	Discrete	No	-
CT	No	CT Fourier transform (CTFT)	Continuous	No	Before S/H
DT	Yes	DT Fourier series (DTFS) \rightarrow FFT	Discrete	Yes	After ADC
DT	No	DT Fourier transform (DTFT)	Continuous	Yes	After S/H

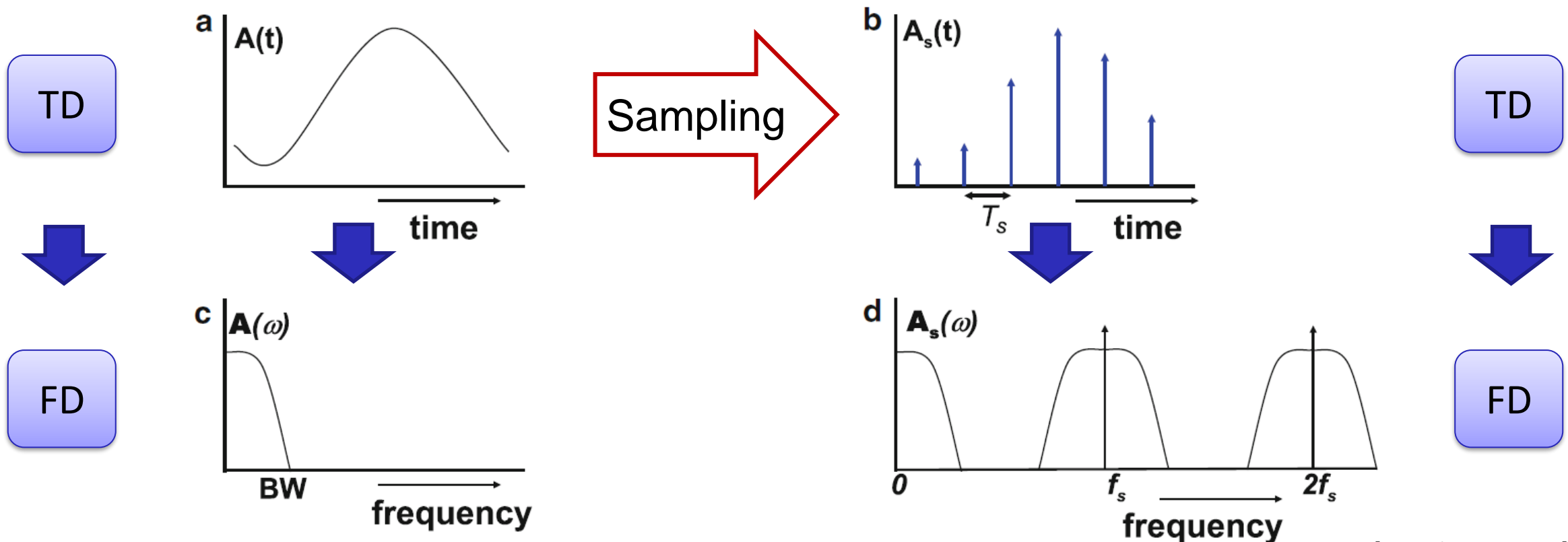
Discrete \leftrightarrow Periodic

- ❑ Sampling causes “images” in the frequency domain
 - The sampled signal is folded around f_s and its multiples
 - The part from 0 to $f_s/2$ is the only part that has physical meaning



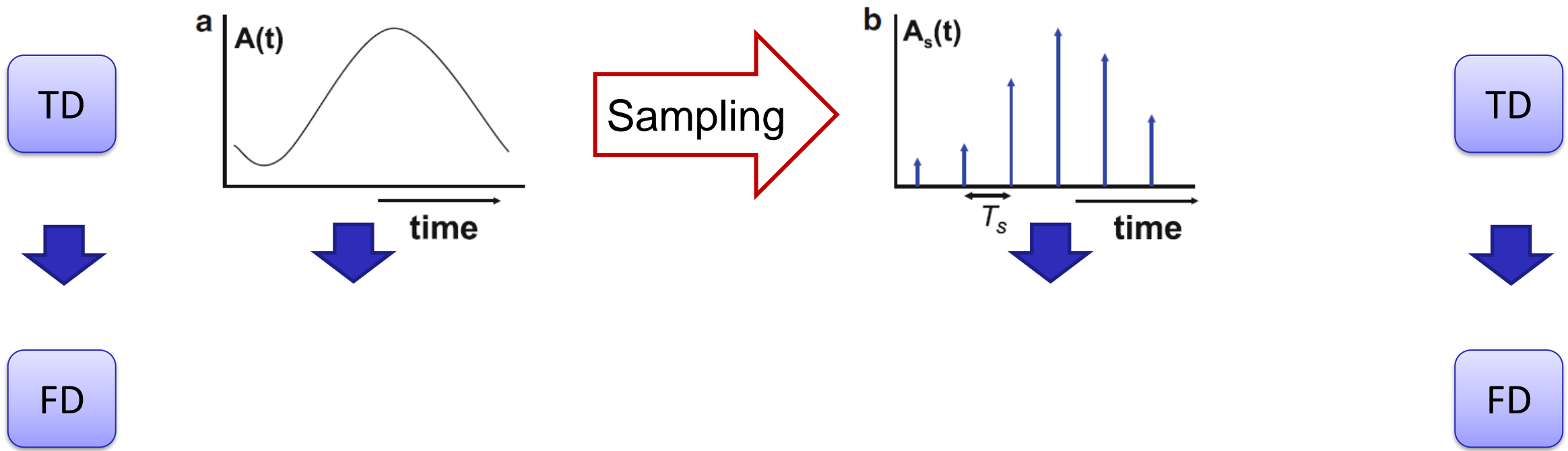
Discrete \leftrightarrow Periodic

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What about Bandpass Signals?

- ❑ Sampling causes “images” in the frequency domain
 - The sampled signal is folded around f_s and its multiples
 - The part from 0 to $f_s/2$ is the only part that has physical meaning



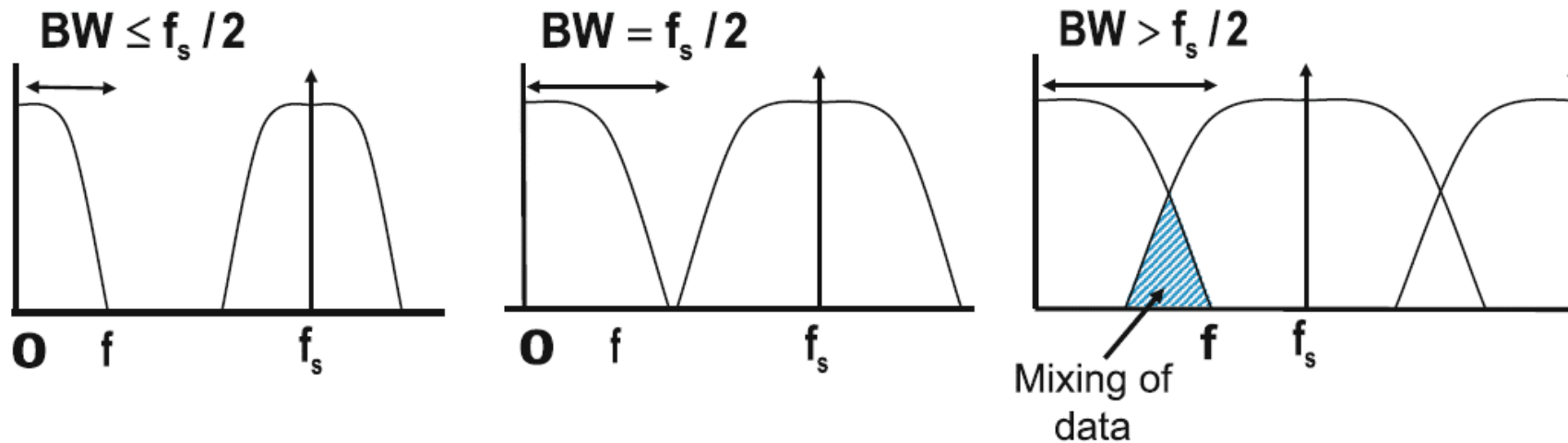
Aliasing and Nyquist Criterion

- ❑ Aliasing is an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled.
- ❑ Nyquist Criterion:

$$f_s > f_{Nyq} = 2 \times BW$$

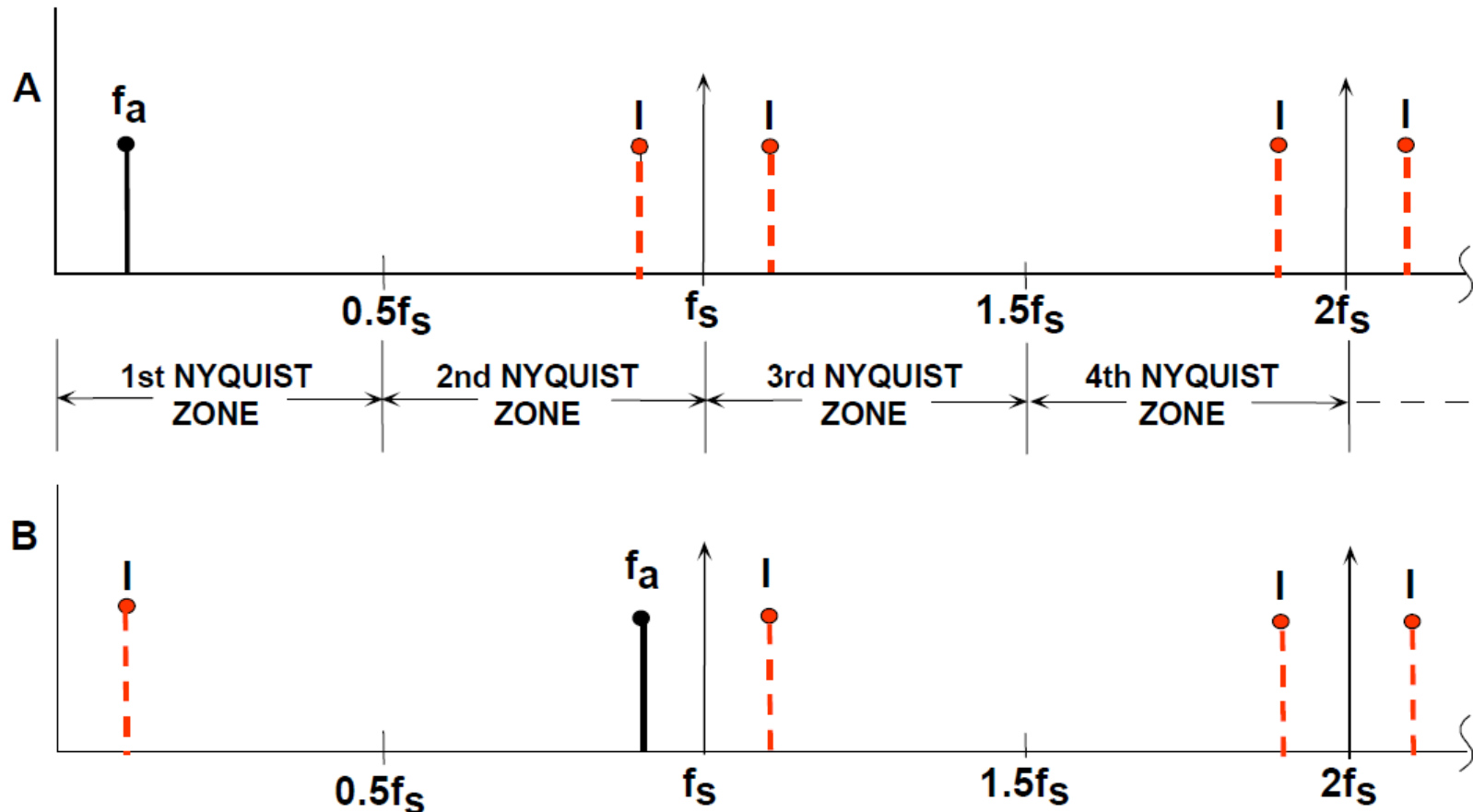
NOT

$$f_s > 2 \times f_{max}$$

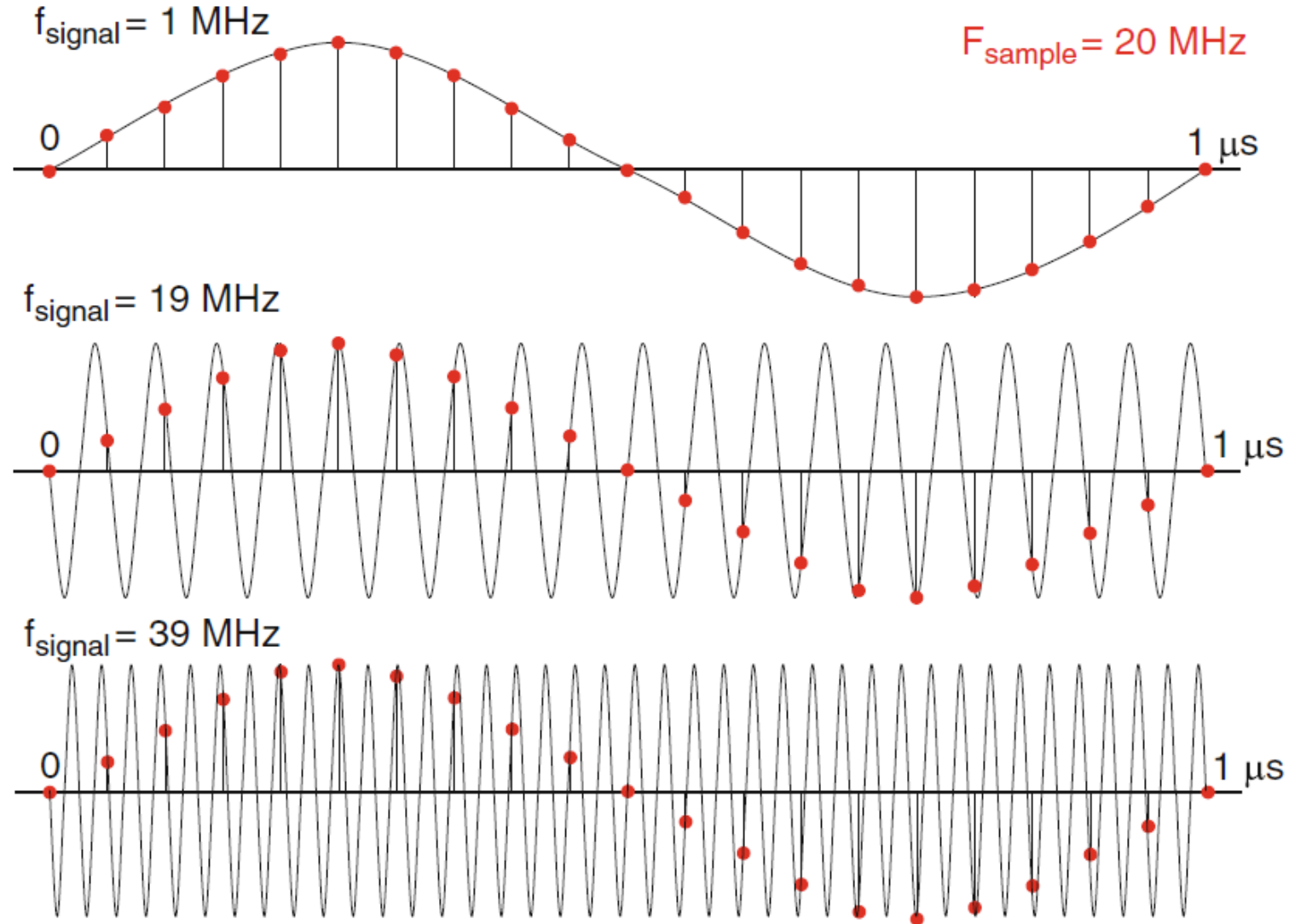


Aliasing/Subsampling in Frequency Domain

- ❑ The sampled signal does not have to be a baseband signal
 - Images appear at $|\pm k f_s \pm f_a|$, where $k = 0, 1, 2, \dots$
- ❑ A.k.a. Under-sampling, Harmonic Sampling, Bandpass Sampling, IF Sampling, Direct IF-to-Digital Conversion

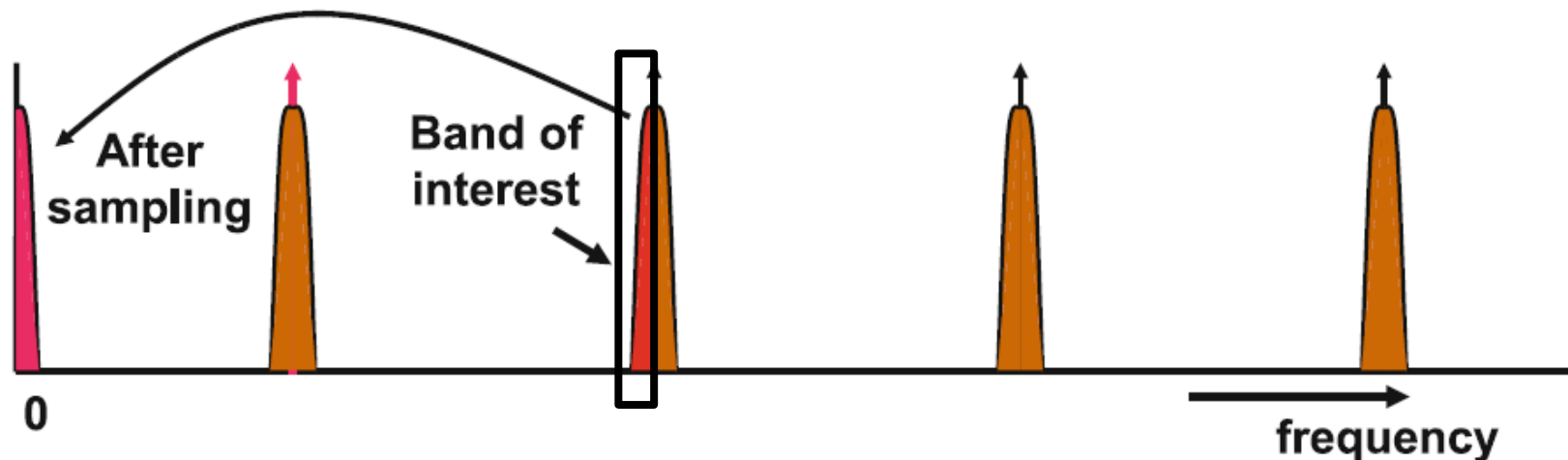


Aliasing/Subsampling in Time Domain



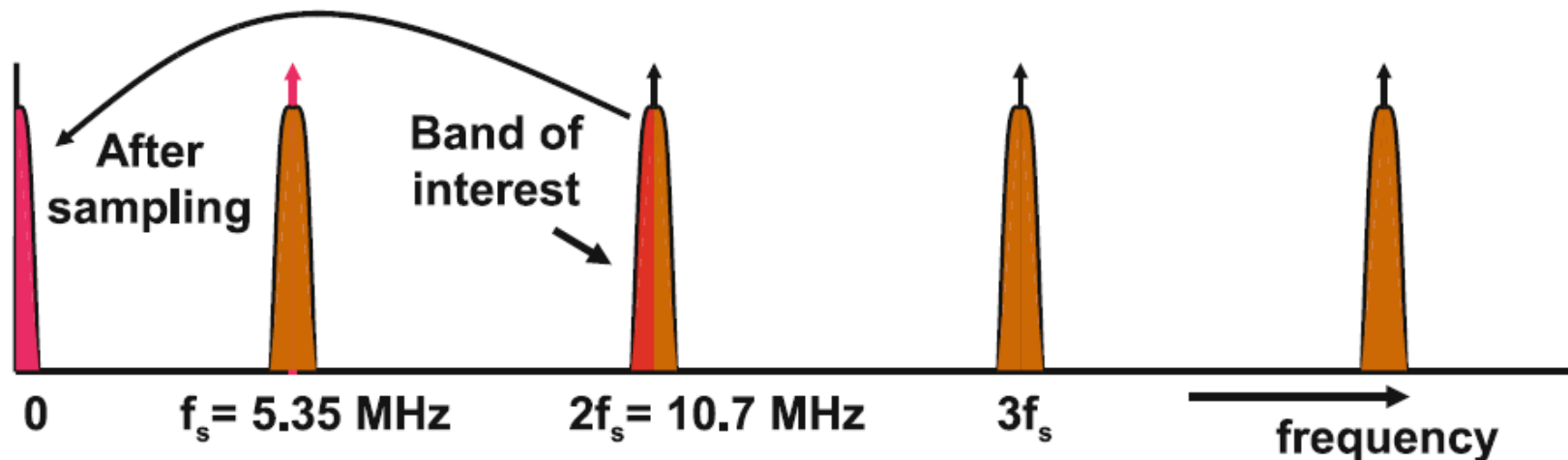
Subsampling

- ❑ The sampled signal does not have to be a baseband signal
- ❑ Subsampling can be used to demodulate (down-convert) an RF signal
- ❑ A.k.a. Under-sampling, Harmonic Sampling, Bandpass Sampling, IF Sampling, Direct IF-to-Digital Conversion
- ❑ Example: FM signal of 100kHz bandwidth at 10.7MHz
 - The signal is down-converted by a ? MHz sampling clock



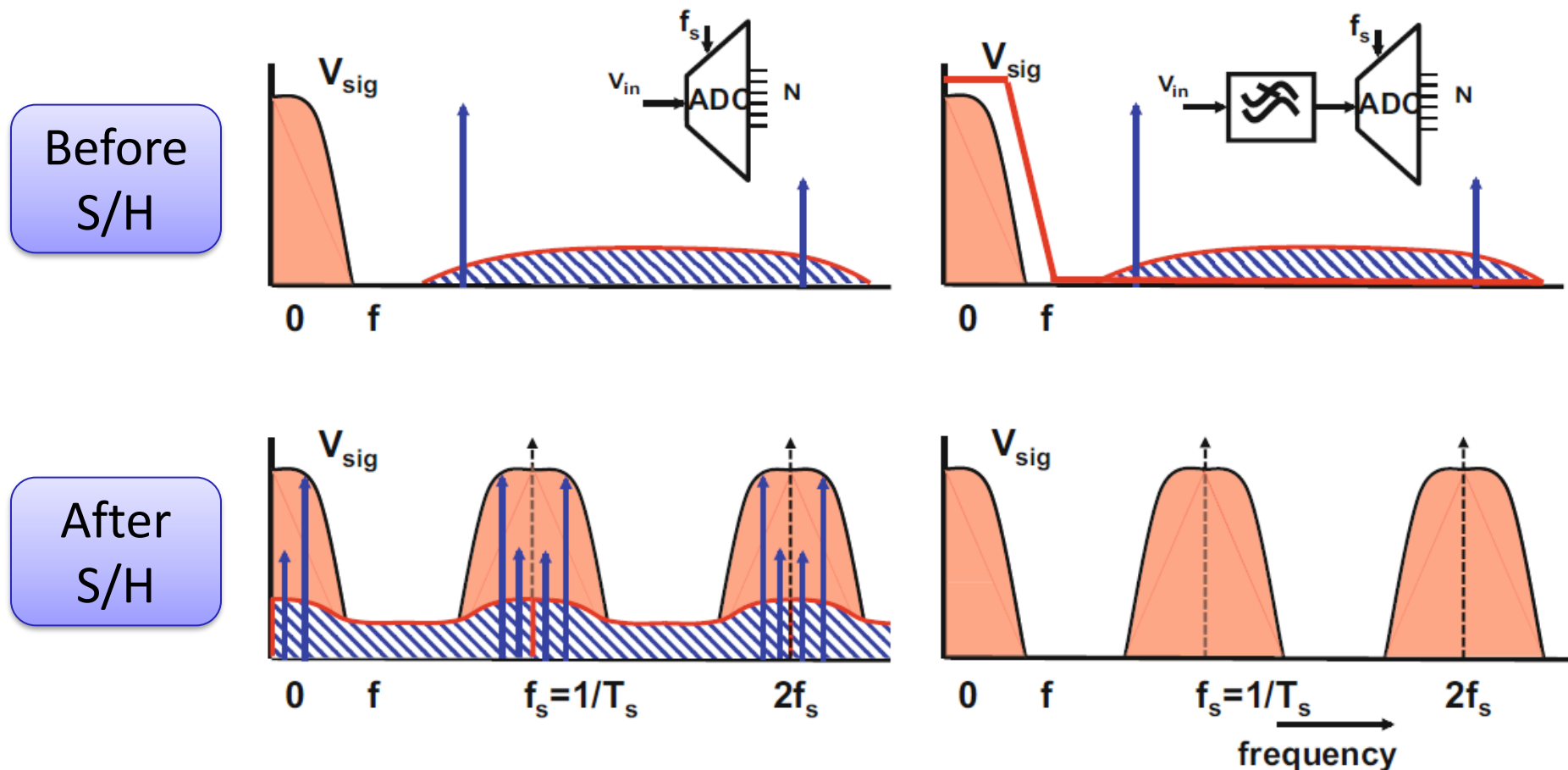
Subsampling

- ❑ Subsampling can be used to demodulate (down-convert) an RF signal
- ❑ A.k.a. Under-sampling, Harmonic Sampling, Bandpass Sampling, IF Sampling, Direct IF-to-Digital Conversion
- ❑ Example: FM signal of 100kHz bandwidth at 10.7MHz
 - The signal is down-converted by a 5.35MHz sampling clock



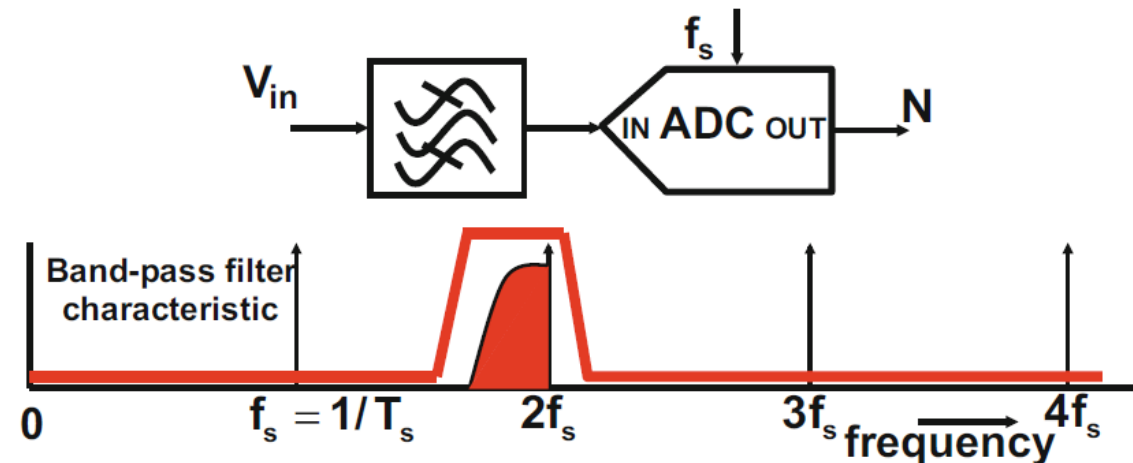
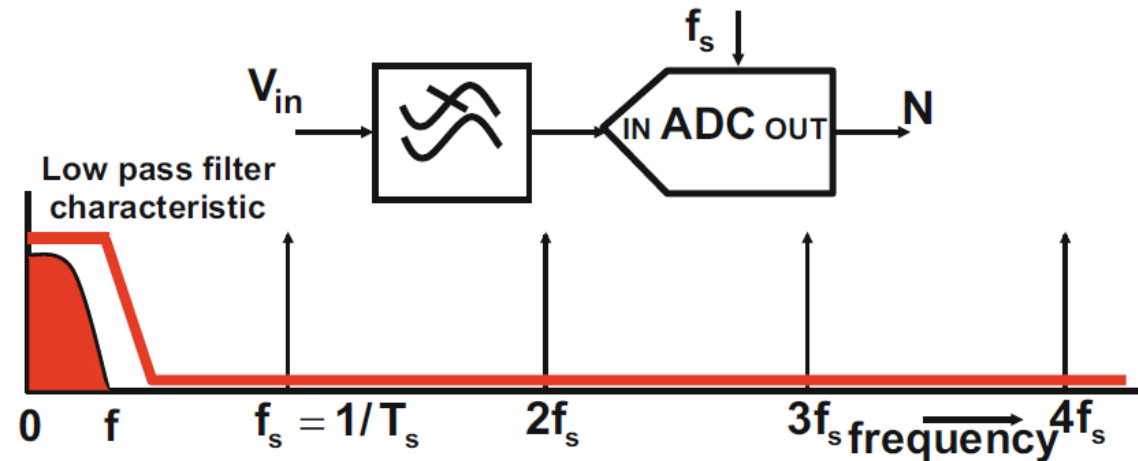
Anti-Aliasing Filter (AAF)

- ❑ Anti-aliasing filters are active or passive (CT or DT?) filters.
 - The signal must be filtered before sampling (time discretization).



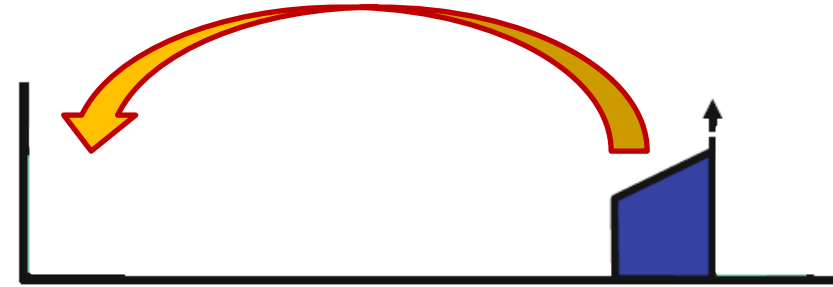
Baseband vs Passband Signals

- ❑ Sampling of baseband signal: AAF is a (LPF,BPF,HPF)?
- ❑ Subsampling of RF (passband) signal: AAF is a (LPF,BPF,HPF)?



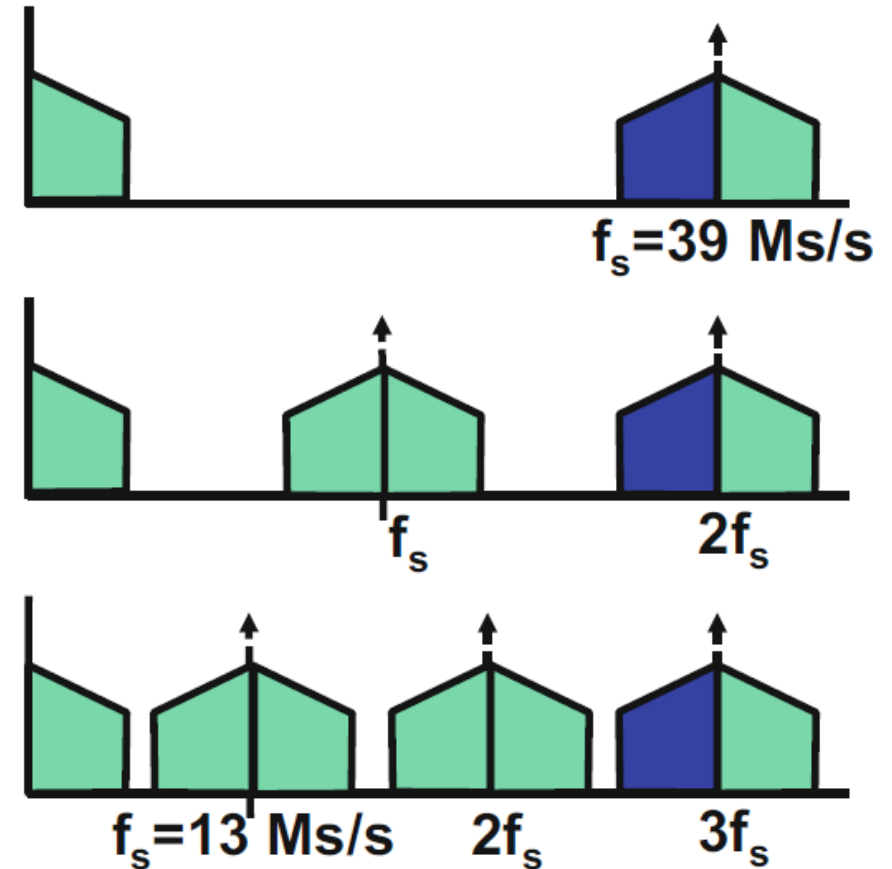
Subsampling Example

- ❑ Signal band: 33 MHz to 39MHz
- ❑ Which sampling rate to choose?
 - 78 MS/s
 - 39 MS/s
 - 19.5 MS/s
 - 13 MS/s



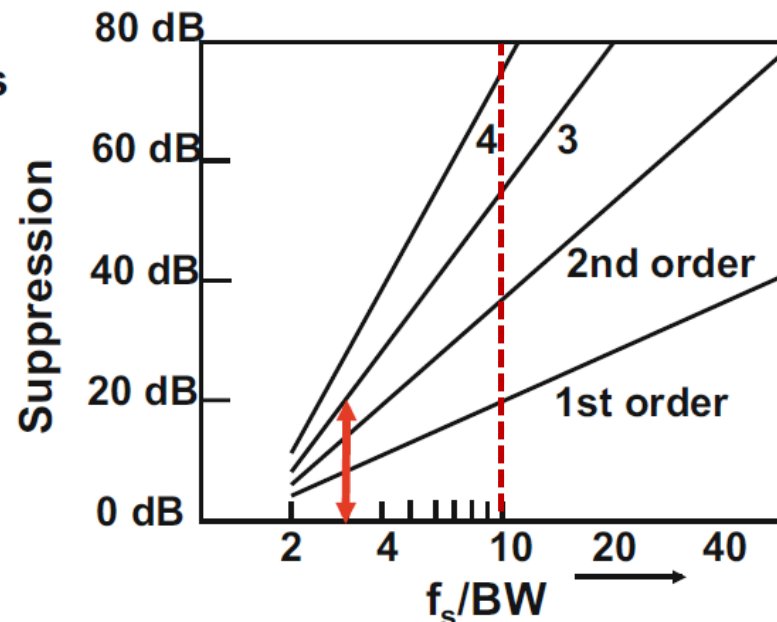
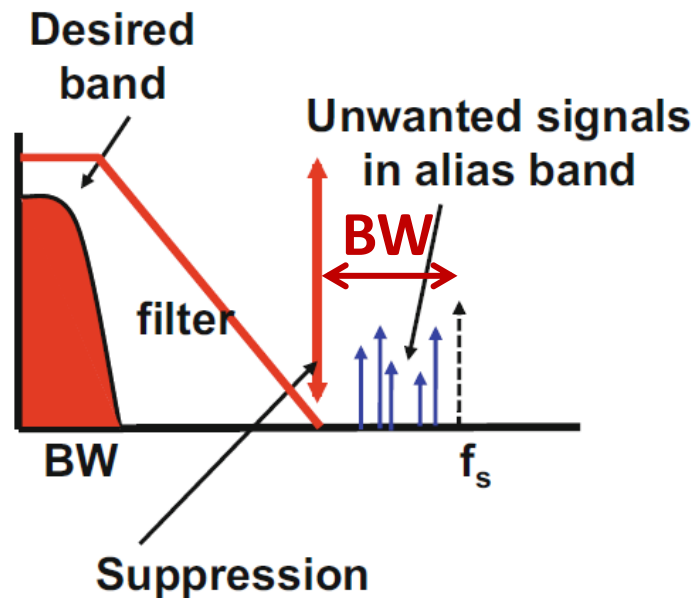
Subsampling Example

- ❑ Signal band: 33 MHz to 39MHz
- ❑ Which sampling rate to choose?
 - 78 MS/s
 - 39 MS/s
 - 19.5 MS/s
 - 13 MS/s
- ❑ What is the advantage/disadvantage of higher sampling frequency?



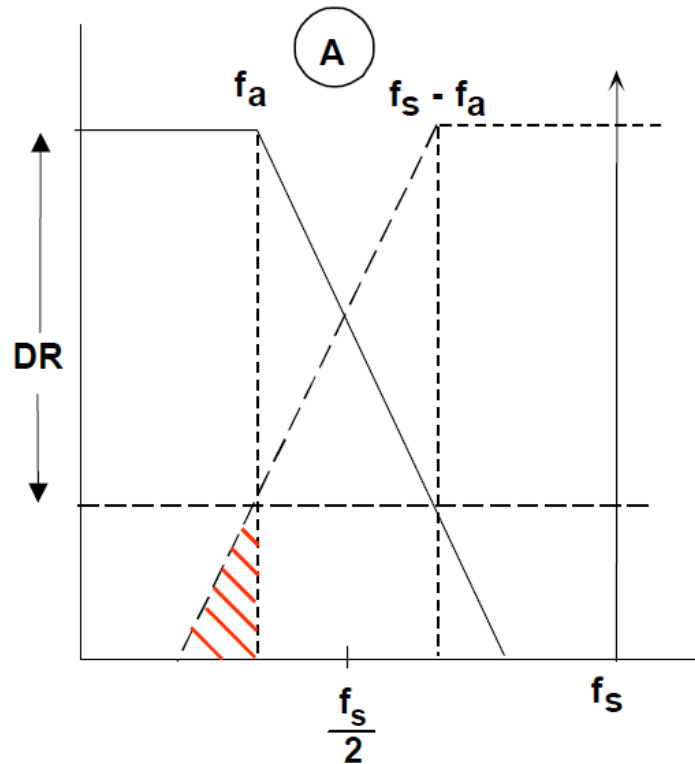
Alias Band Suppression

- ❑ Working at the limit of Nyquist criterion requires an ideal filter that does not exist.
- ❑ Signals in the alias band ($f_s - BW$ to f_s) will alias in the desired signal band after sampling.
 - Must be suppressed by AAF.
- ❑ Each pole gives a roll-off slope of 20 dB/decade = 6 dB/octave
- ❑ How much suppression @ ($f_s - BW$) if $f_s/BW = 2$ for 4th order filter?

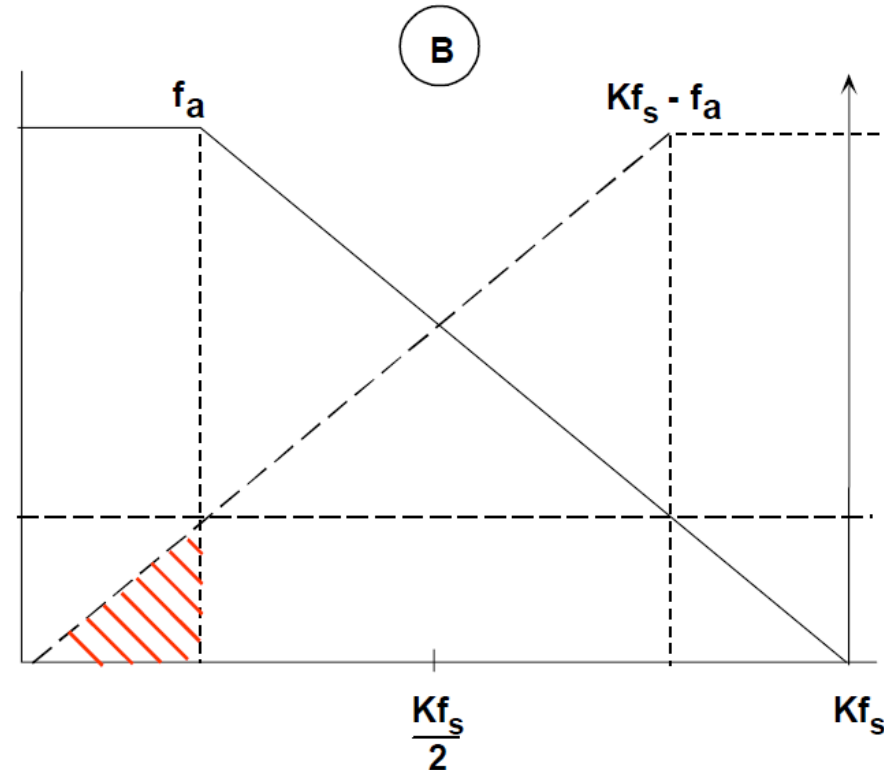


Oversampling

- ❑ Oversampling relaxes requirements on baseband antialiasing filter.



STOPBAND ATTENUATION = DR
TRANSITION BAND: f_a to $f_s - f_a$
CORNER FREQUENCY: f_a

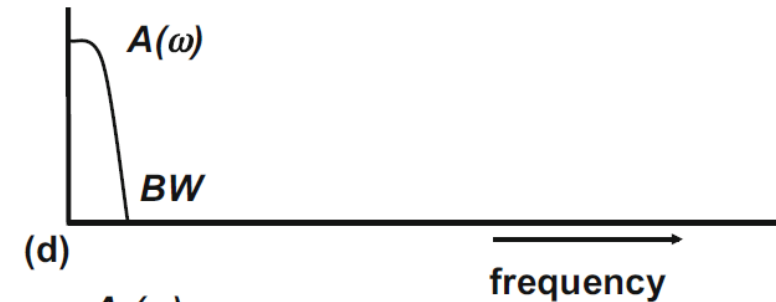
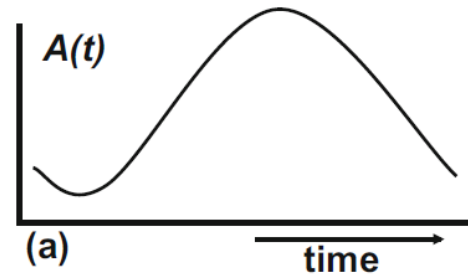


STOPBAND ATTENUATION = DR
TRANSITION BAND: f_a to $Kf_s - f_a$
CORNER FREQUENCY: f_a

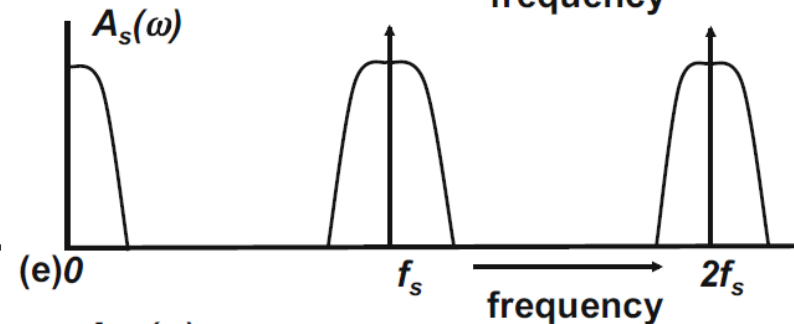
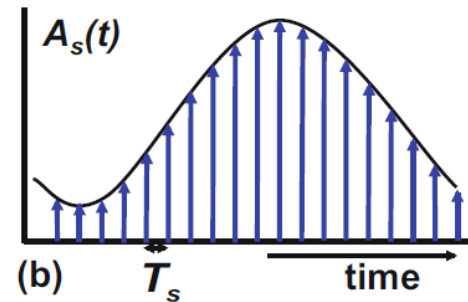
Decimation

- ❑ Decimation is the process of reducing the sample rate of a signal.
- ❑ Unless the signal is already filtered and oversampled, digital filtering is necessary.

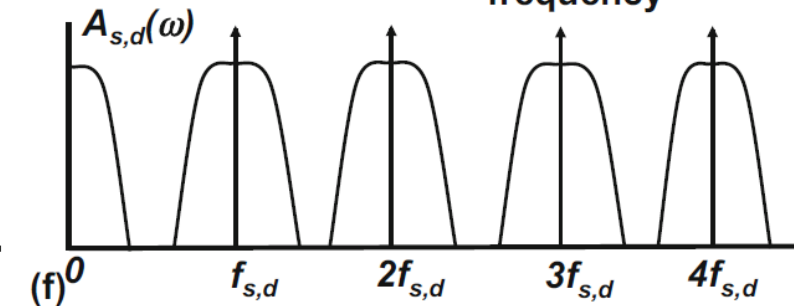
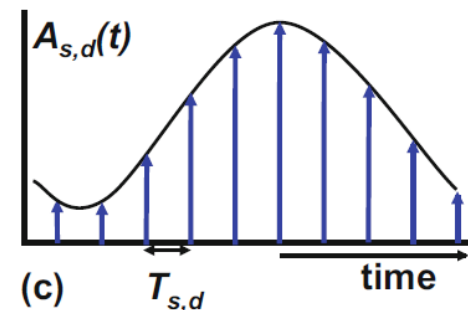
Before
sampling



After
sampling

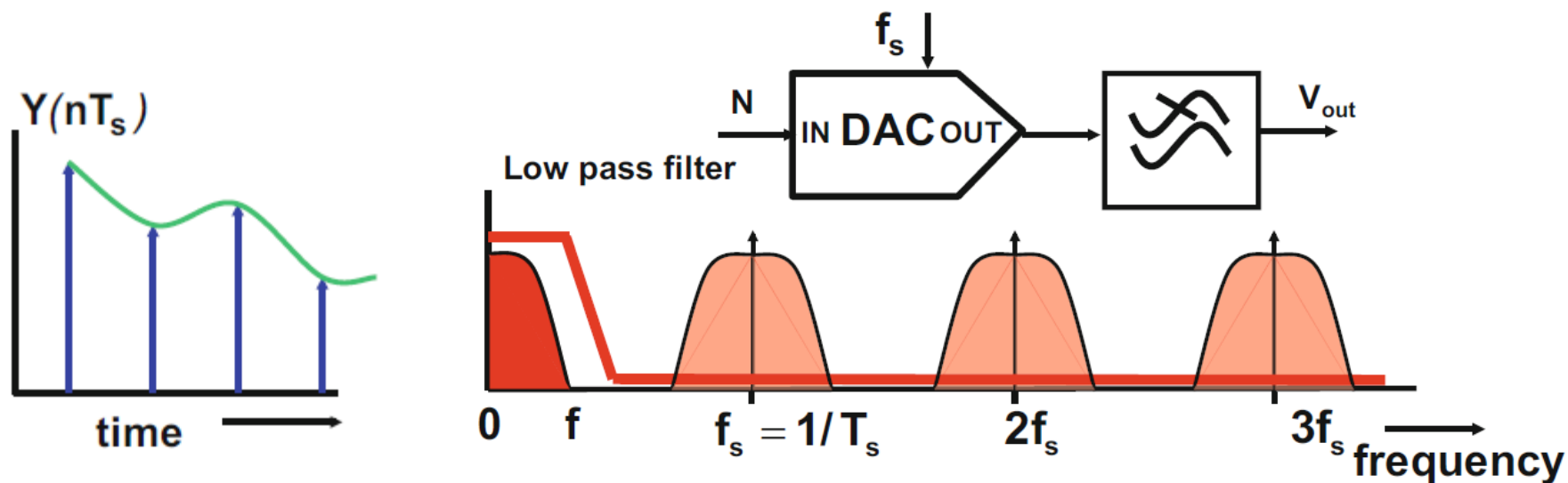


After
decimation



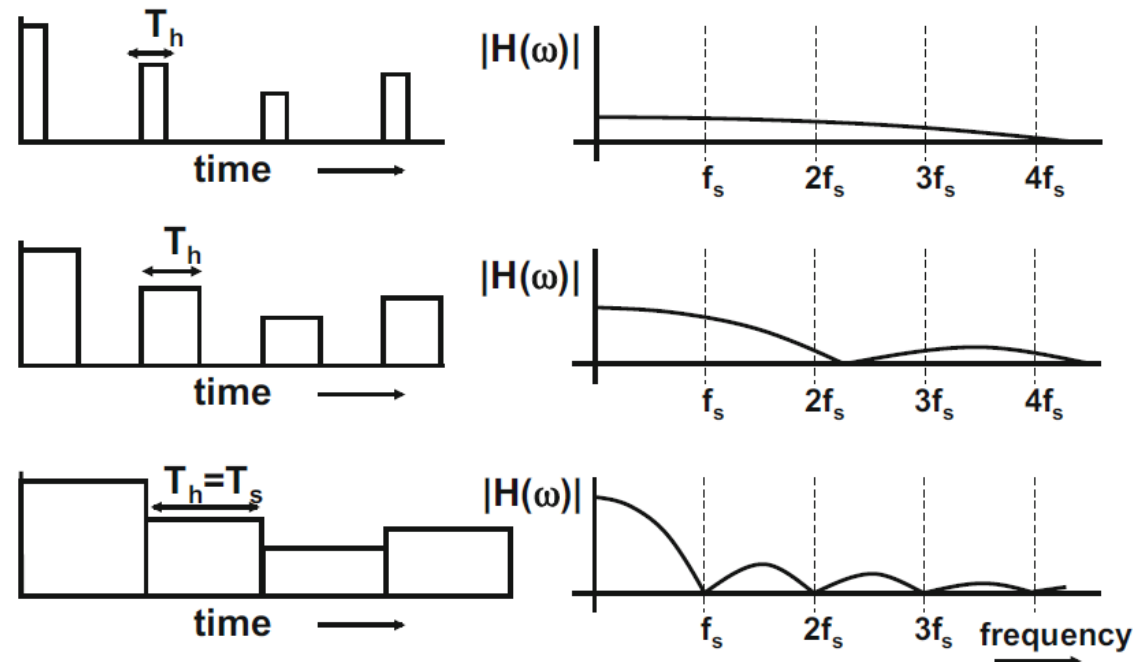
Reconstruction Filter

- ❑ ADC needs an anti-aliasing filter.
- ❑ DAC needs a reconstruction (smoothing) filter.
 - TD: The reconstruction filter “interpolates/restores/reconstructs” the signal.
 - FD: The reconstruction filter suppresses the “images”.



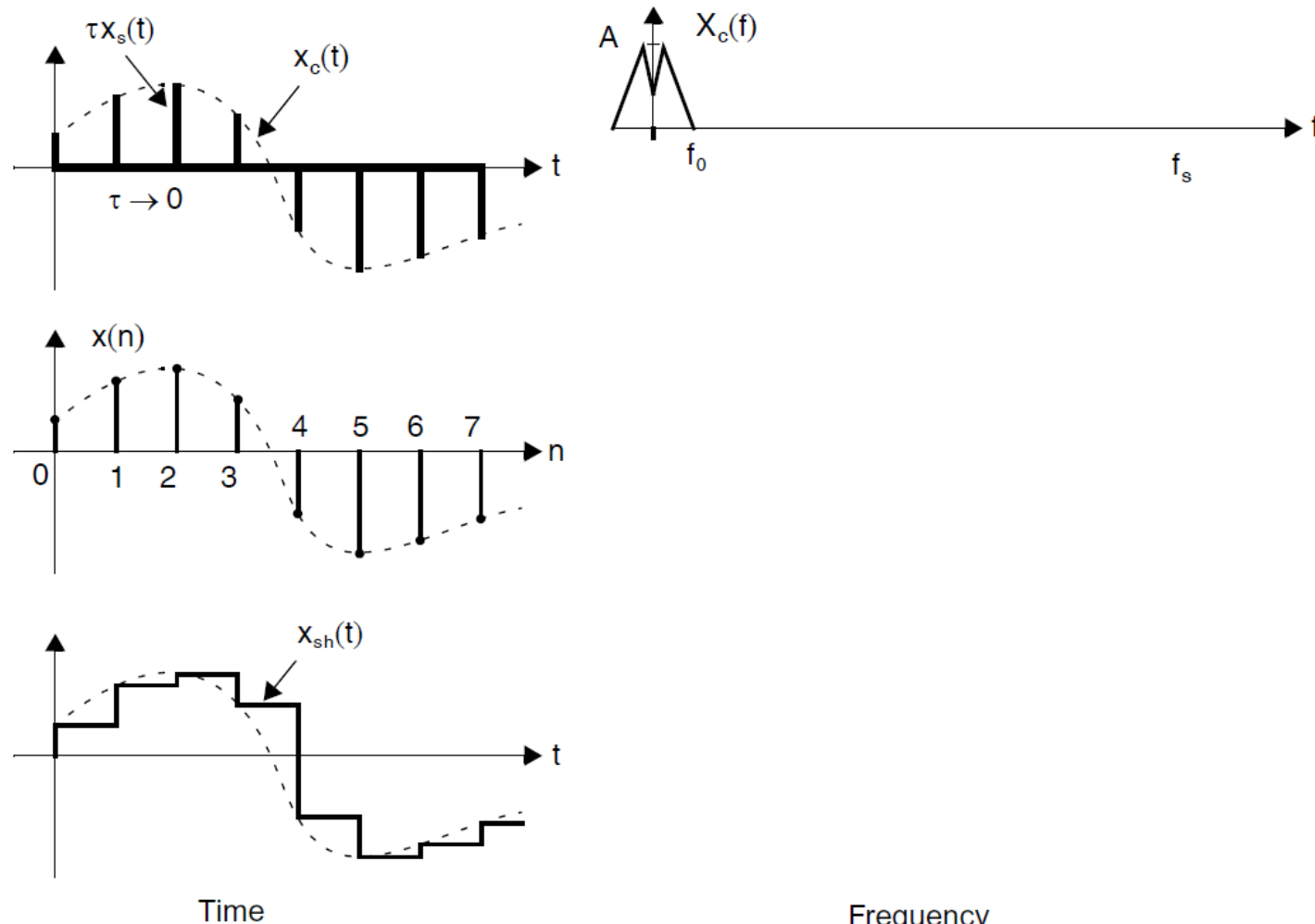
Zero-Order Hold (ZOH)

- ❑ Zero-order hold (ZOH) keeps the value of the signal at the sample moment.
- ❑ The Fourier transform of ZOH is a sinc function: $\text{sinc}(x) = \sin(x)/x$
 - Nulls of $\text{sinc}(x)$ at the inverse of hold time (pulse width)
- ❑ The zero-order hold (ZOH) performs inherent reconstruction (filtering out images).



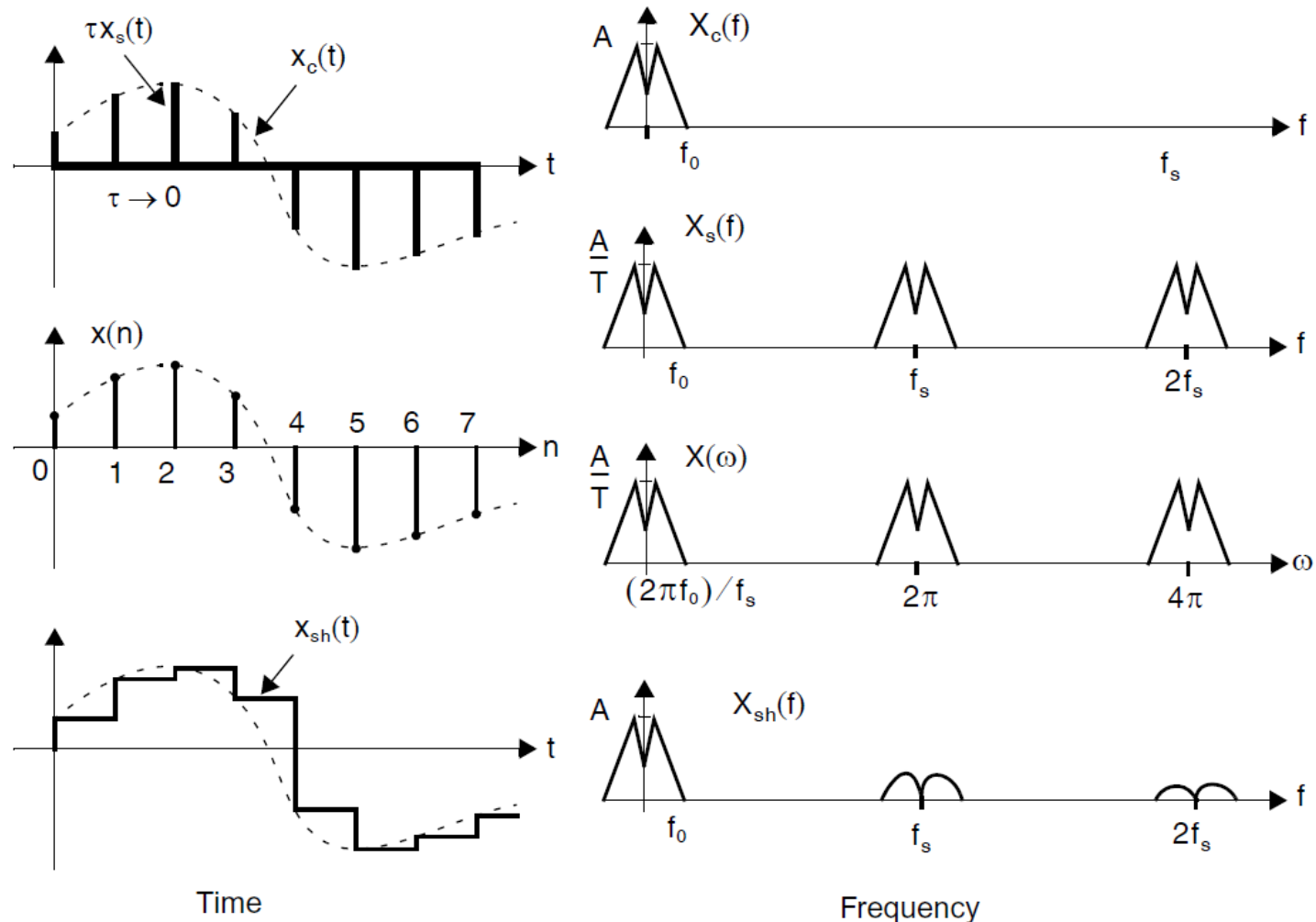
Zero-Order Hold (ZOH)

- The zero-order hold (ZOH) performs inherent reconstruction (filtering out images).
 - Succeeding reconstruction filter performs further interpolation (image suppression).



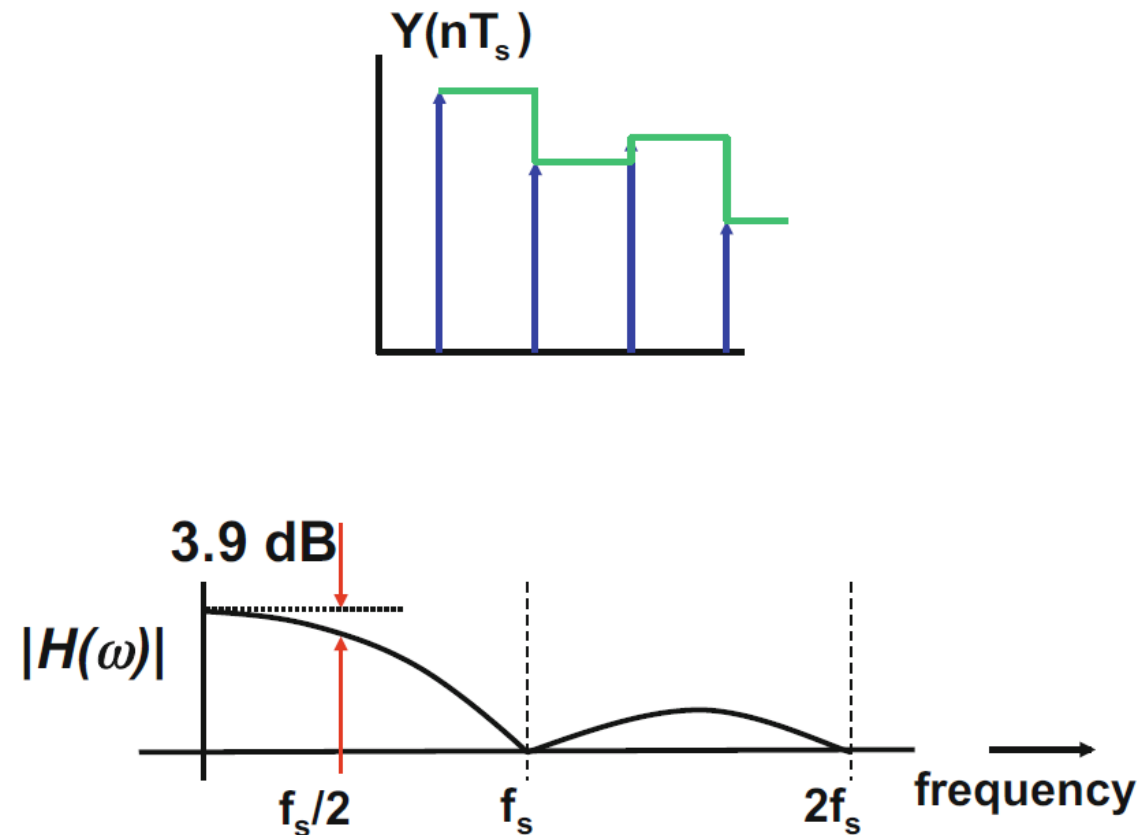
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Passband Droop

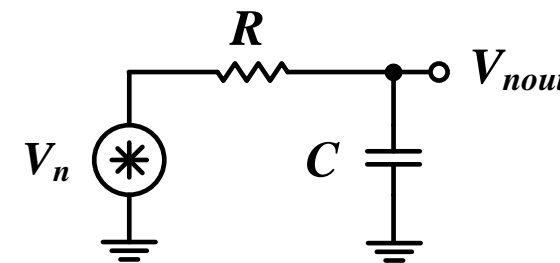
- ❑ ZOH suppresses images but introduces amplitude distortion.
 - The passband distortion may be compensated by inverse-sinc response in the digital or analog domains.



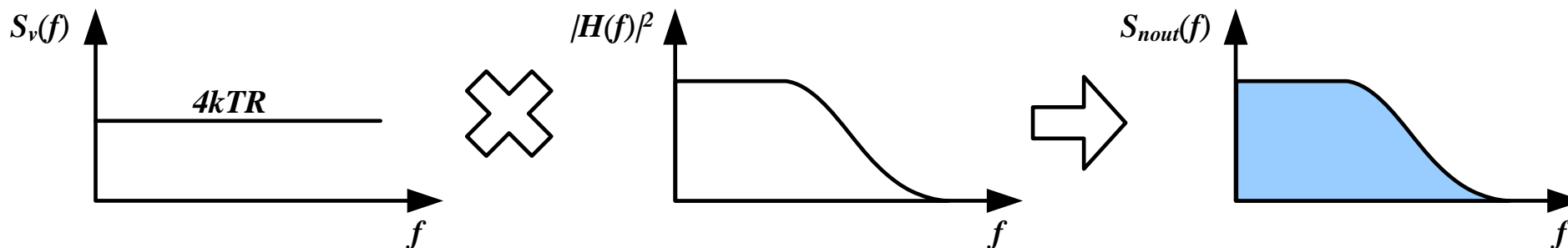
Noise in RC Circuit

- ❑ Resistors generate white thermal noise.
 - But the BW is always limited by a cap.

$$S_{nout}(f) = S_v(f) \left| \frac{V_{nout}(j\omega)}{V_n(j\omega)} \right|^2$$
$$\overline{V_{nout}^2} = V_{noutrms}^2 = \int_{-\infty}^{\infty} S_{nout}(f) df$$



$$\overline{V_{nout}^2} = \frac{kT}{C}$$

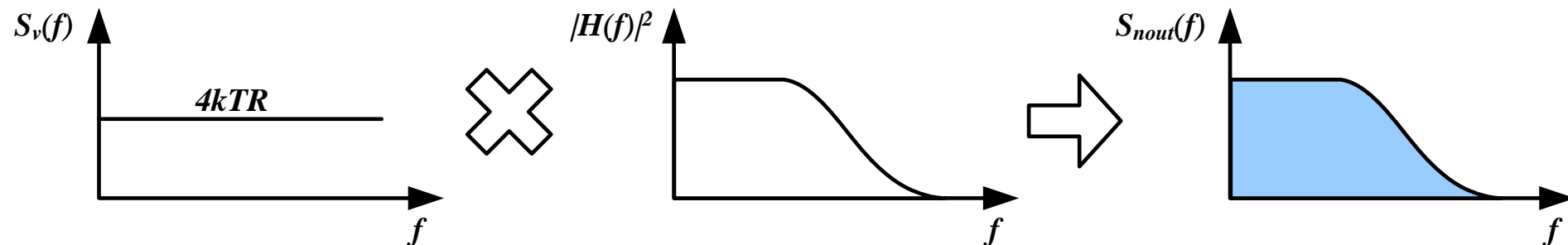


Noise in RC Circuit

$$\overline{V_{nout}^2} = \frac{kT}{C}$$

- ❑ RMS noise is independent of R ! (why?)
- ❑ For $C = 1 \text{ pF} \rightarrow V_{nrms} \approx 64 \mu\text{Vrms}$

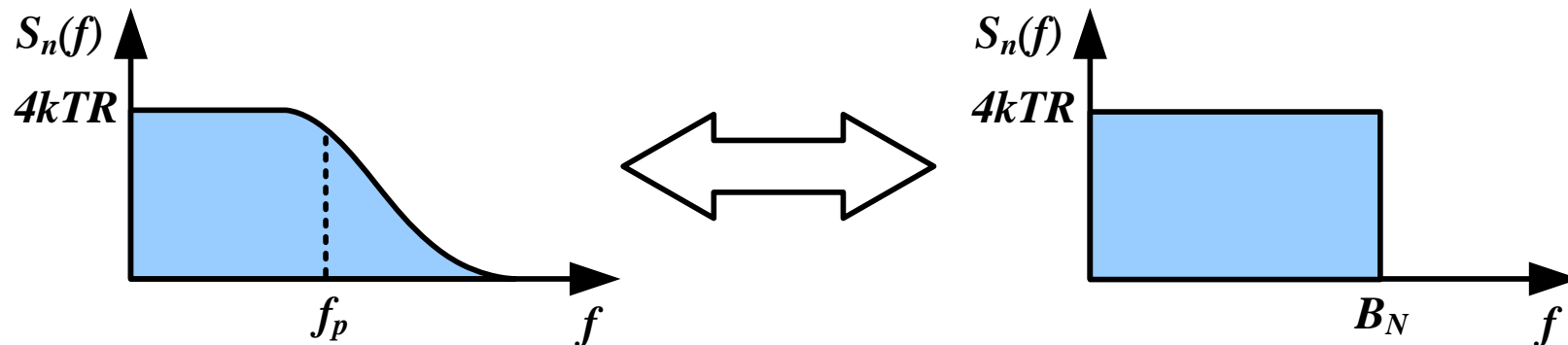
$$V_{nrms} \approx \sqrt{\frac{1 \text{ p}}{C}} \times 64 \mu\text{Vrms}$$



Equivalent Noise Bandwidth

- Define an equivalent noise BW (B_N) such that the area under a brick-wall response is the same area under the actual spectral density curve
- For a first order system

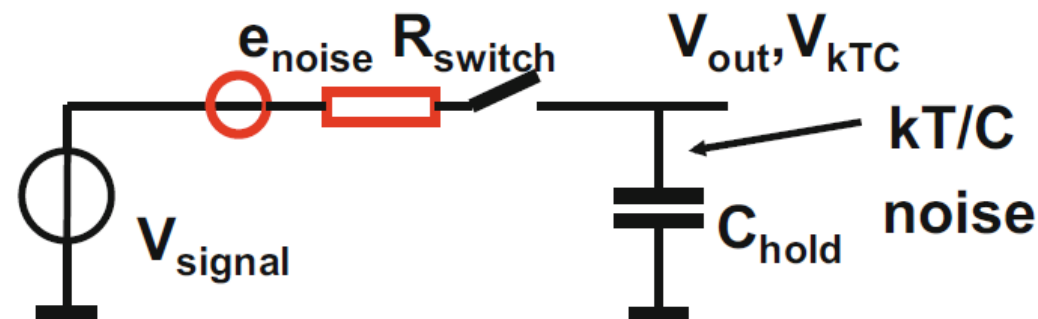
$$V_{nrms}^2 = \int_{-\infty}^{\infty} S_n(f) df = 4kTR \times B_N = \frac{kT}{C}$$
$$B_N = \frac{1}{4RC} = \frac{\pi}{2} f_p$$



Sampling Noise

□ The sampling capacitor determines noise power → SNR → No. of ADC bits.

C_{hold}	$V_{nrms} = \sqrt{\frac{kT}{C}}$ at $T = 300\text{ K}$	SNR (assume $V_{sigrms} = 1\text{ Vrms}$)	No. of bits (see next lecture)
100 fF	203 μVrms	74 dB	12-bit
1 pF	64 μVrms	84 dB	13.7-bit
10 pF	20.3 μVrms	94 dB	15.4-bit

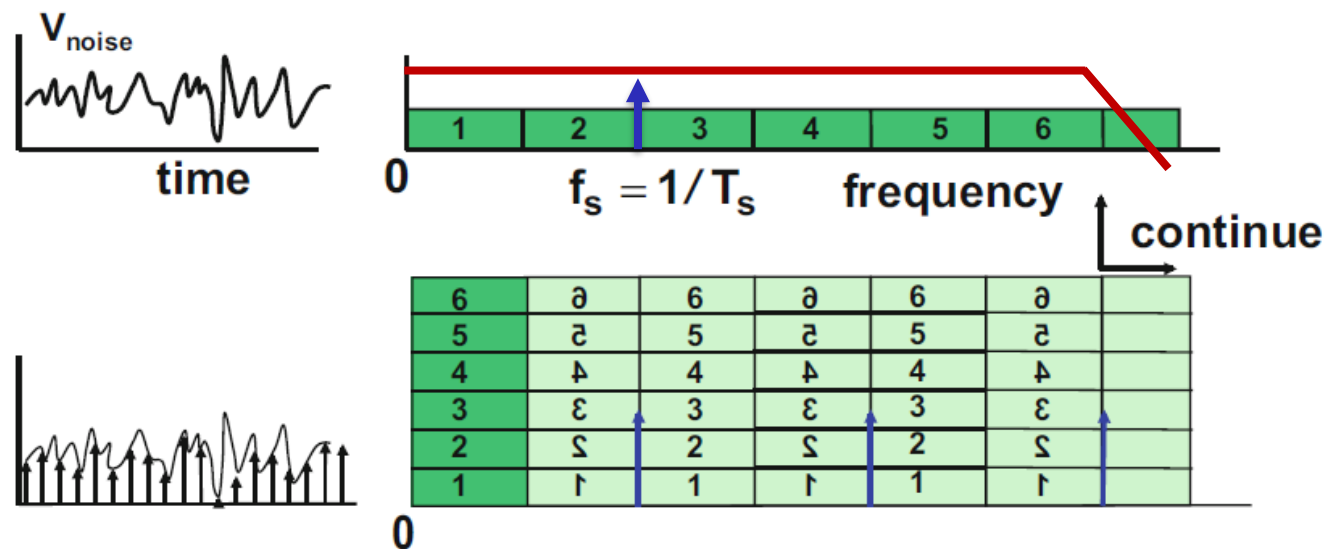


Noise Folding

- Before sampling: $P_n = kT/C = S_n(f) \times B_N$
- After sampling P_n is unchanged: $P_n = kT/C = S_{n,sampled}(f) \cdot \frac{f_s}{2}$

$$S_{n,sampled}(f) = \frac{kT}{C} \times \frac{2}{f_s} = S_n(f) \times B_N \times \frac{2}{f_s}$$

$$S_{n,sampled}(f) = S_n(f) \times \frac{2B_N}{f_s} = S_n(f) \times \frac{\pi BW}{f_s}$$
- Noise power is unchanged, but noise density increases (noise folding).



References

- ❑ M. Pelgrom, Analog-to-Digital Conversion, Springer, 3rd ed., 2017.
- ❑ W. Kester, The Data Conversion Handbook, ADI, Newnes, 2005.
- ❑ B. Boser and H. Khorramabadi, EECS 247 (previously EECS 240), Berkeley.
- ❑ B. Murmann, EE 315, Stanford.
- ❑ Y. Chiu, EECT 7327, UTD.

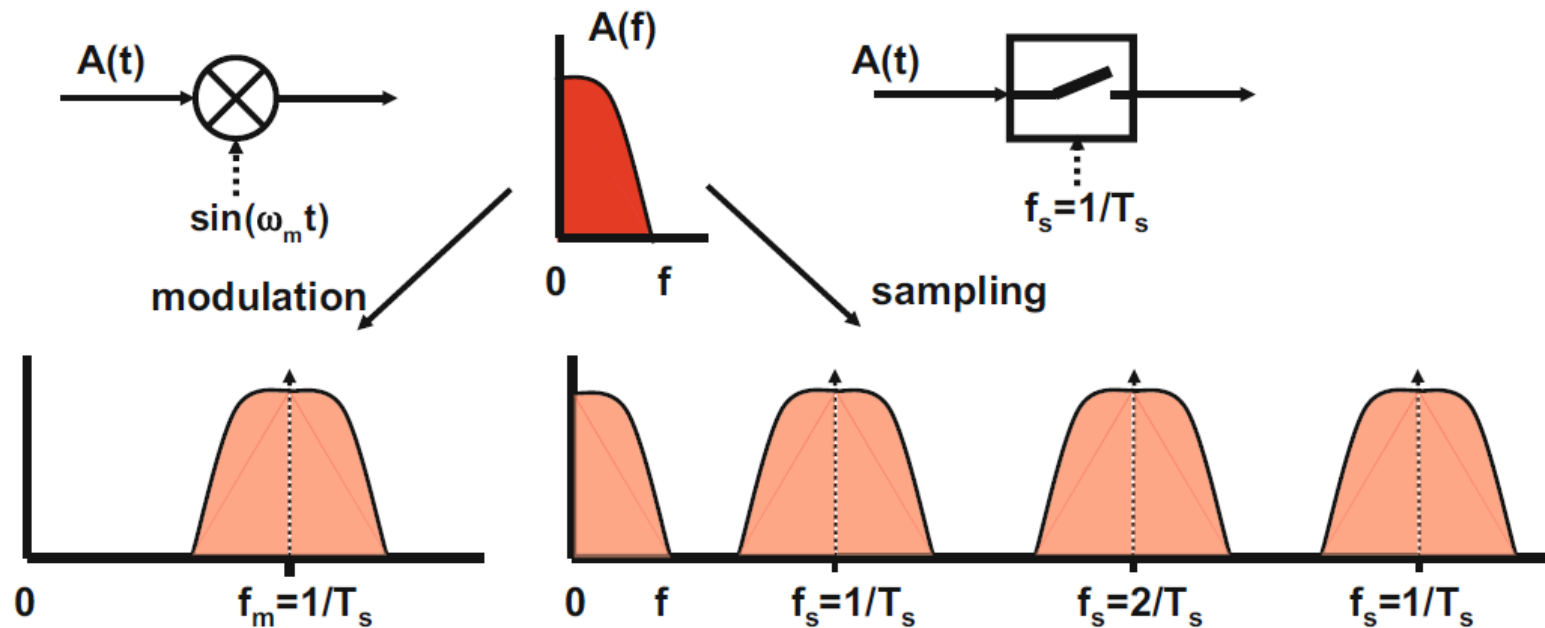
Thank you!

Sampling vs Modulation

- Two important dualities between TD and FD

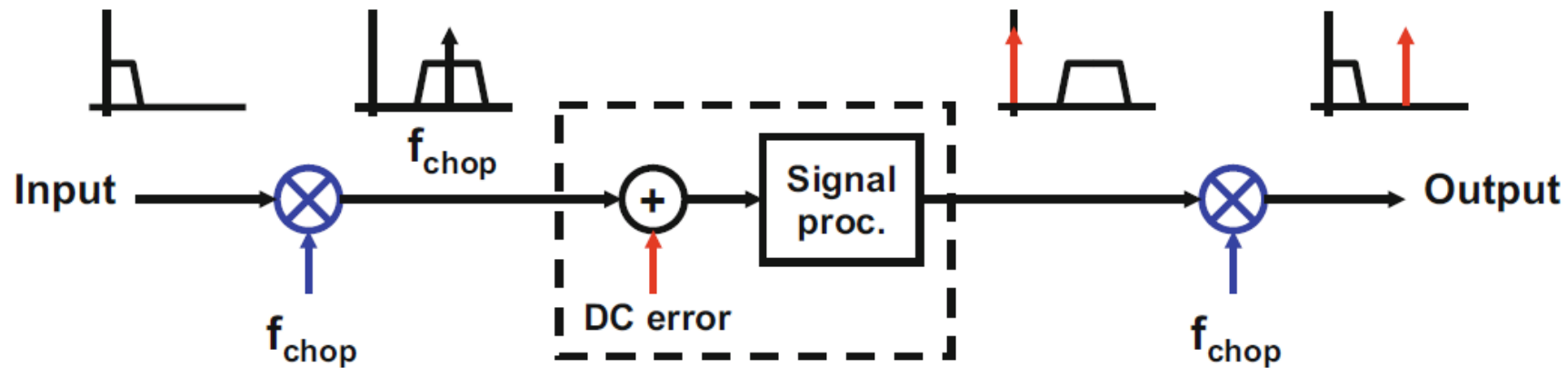
Discrete \leftrightarrow Periodic & **Multiplication \leftrightarrow Convolution**

- A stream of impulses with period T_s in TD is a stream of impulses with period f_s in FD
 - Sampling can be viewed as a summation of modulations

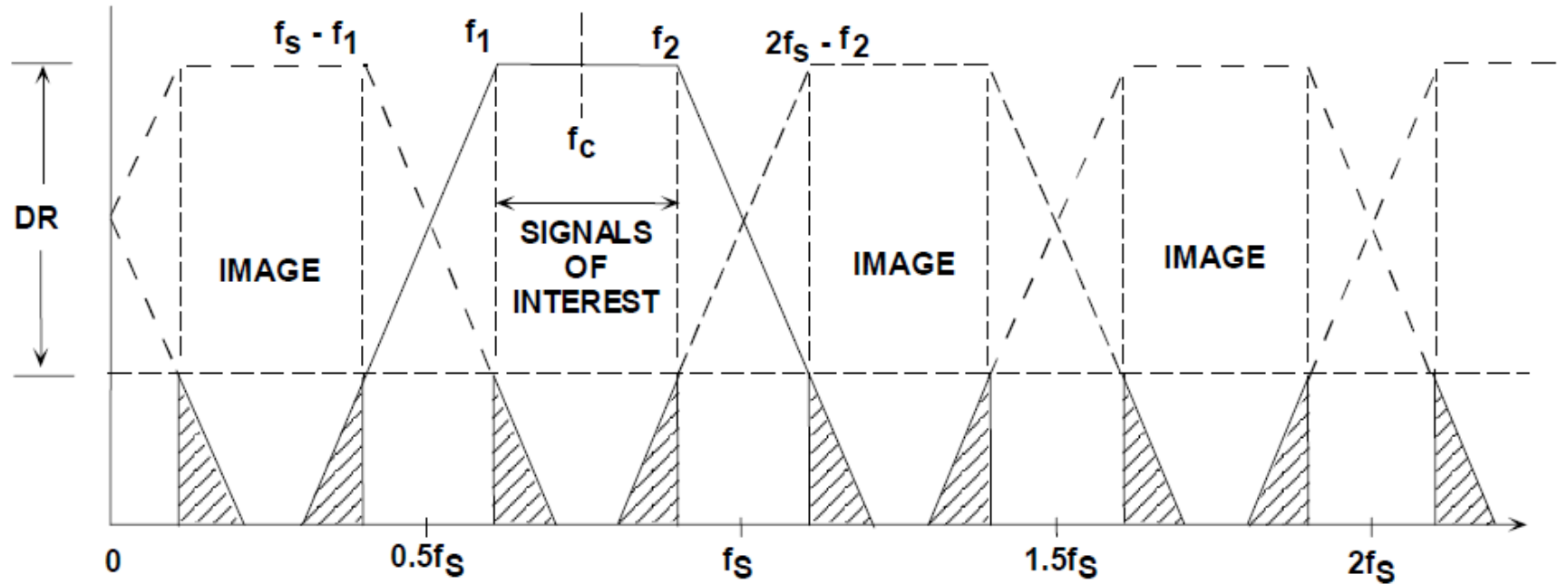


Chopping

- ❑ Chopping is a technique used for improving accuracy.
 - Sensitive signals are modulated to frequency bands where the signal processing is free of errors.
 - Mitigates the effect of DC offsets, flicker noise, etc.
- ❑ In differential circuits, chopping is implemented easily by alternating between the differential branches.



Subsampling Anti-Aliasing Filter



BANDPASS FILTER SPECIFICATIONS:

STOPBAND ATTENUATION = DR
TRANSITION BAND: f_2 TO $2f_s - f_2$
 f_1 TO $f_s - f_1$
CORNER FREQUENCIES: f_1, f_2