EE 230
Lecture 39

Data Converters

Review from Last Time:

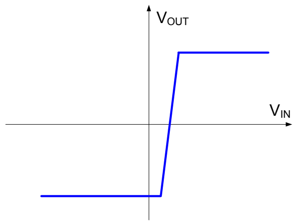
Types of ADCs

- Flash
- Pipelined
- Folded
- Serial
  - Single-slope
  - Dual-slope
- Interpolating
- Iterative (Algorithmic, Cyclic)
- Successive Approximation (SAR)
- Oversampled (Delta-Sigma)
- Charge Redistribution
- Several others

Review from Last Time:

Metastability

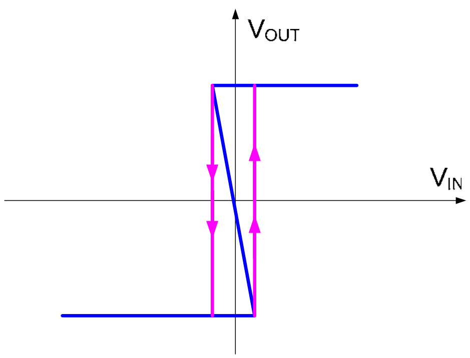
High-gain amplifier



for some input values, output may not be at a level that is predictably interpreted by subsequent logic circuits

this range can be very small if the gain is large enough

Bistable amplifier



Bistable amplifier will always make a decision

For any fixed finite time T, there is always a small nonzero probability that the decision will not be made in time T

This probability can be made very low through proper circuit design techniques but never made to be zero

Review from Last Time:

Metastability

Metastability in ADCs caused by comparators

A comparator is said to be in a metastable state if the output of the comparator can not be interpreted by subsequent digital logic

For any finite time T, any comparator that has been “asked” to make a binary decision has a finite nonzero probability P that subsequent logic will not correctly interpret the output In the interval of length T

This probability can be made very low through proper circuit design techniques but never made to be zero

Metastability in ADCs caused by transient conditions in logic circuits

Due to asynchronous operation of the ADC

Can be eliminated by circuit modifications that make operation synchronous or by appropriate timing of asynshronous operation

Review from Last Time:

Metastability

Flash ADC

Interpolating

Pipelined

Successive Approximation (SAR)

Iterative (Algorithmic, Cyclic)

Serial

Folded

Oversampled (Delta-Sigma)

Dual-slope

Single-slope

Charge Redistribution

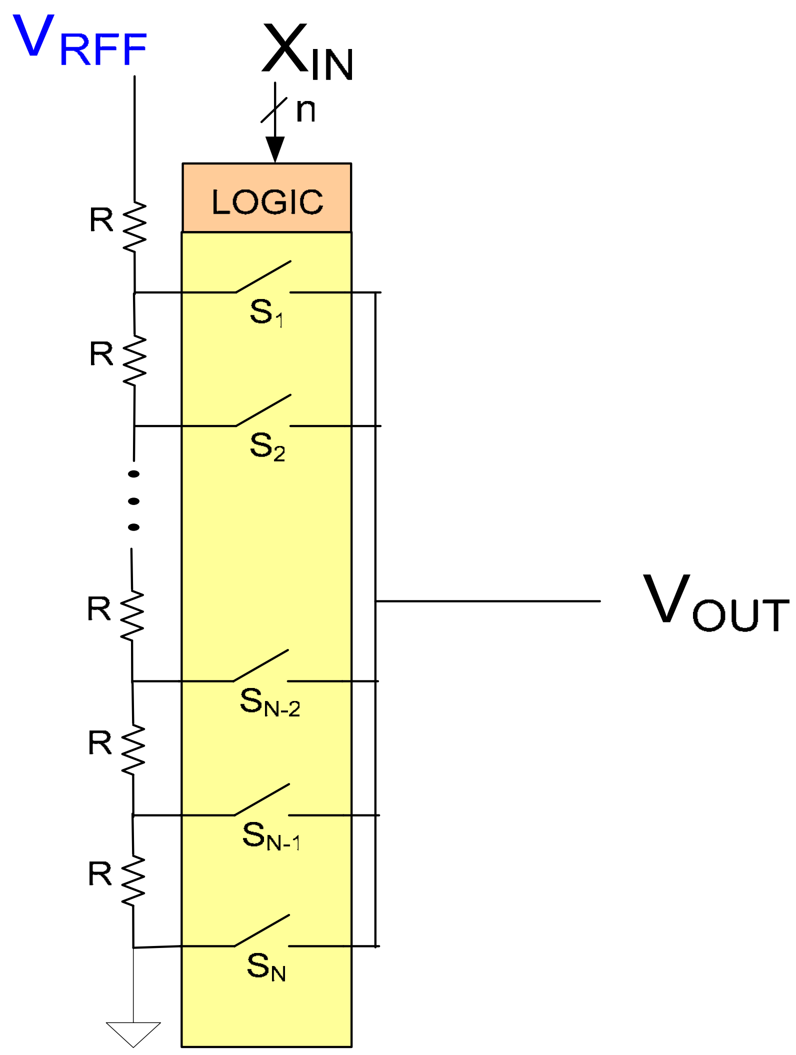
Metastability <u>can never be eliminated in an ADC</u>, its effects can just be reduced to a level that results in an acceptably low probability of causing an unacceptable outcome

Types of DACs

- Current steering
- R-String
- Ladder (R-2R)
- Parallel
- Pipelined
- Subranging
- Charge Redistribution
- Algorithmic
- Serial
- Subranging
- Oversampled (Delta-Sigma)
- Several others

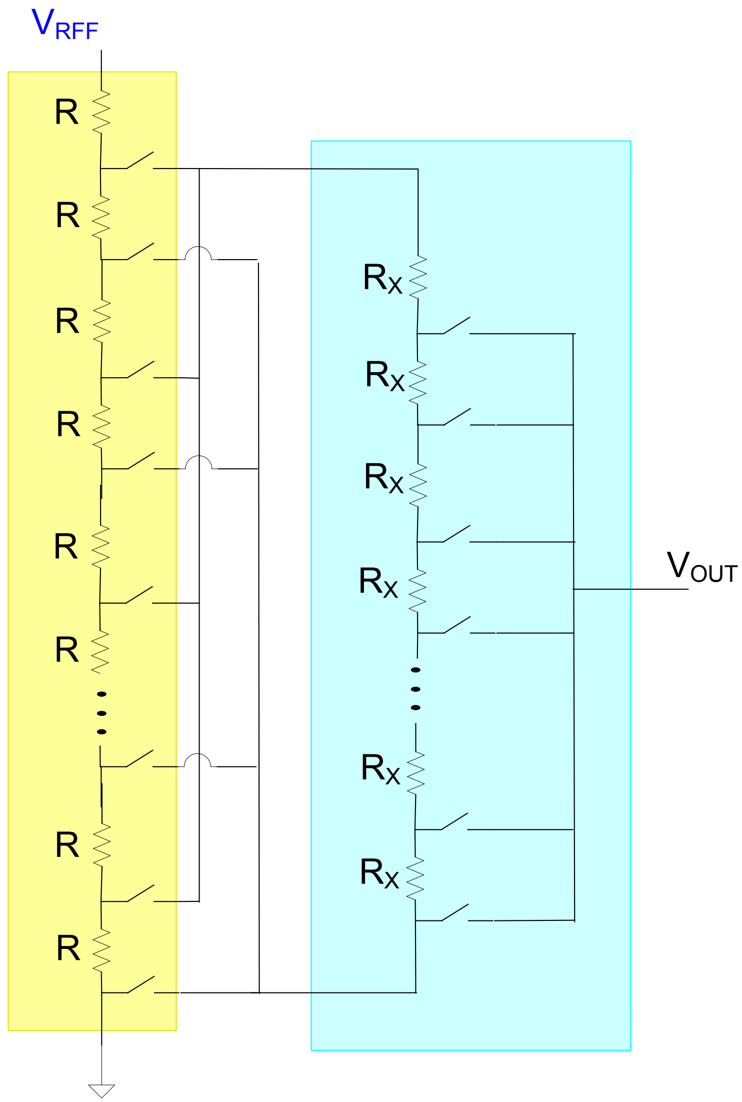
Types of DACs

R-string DAC



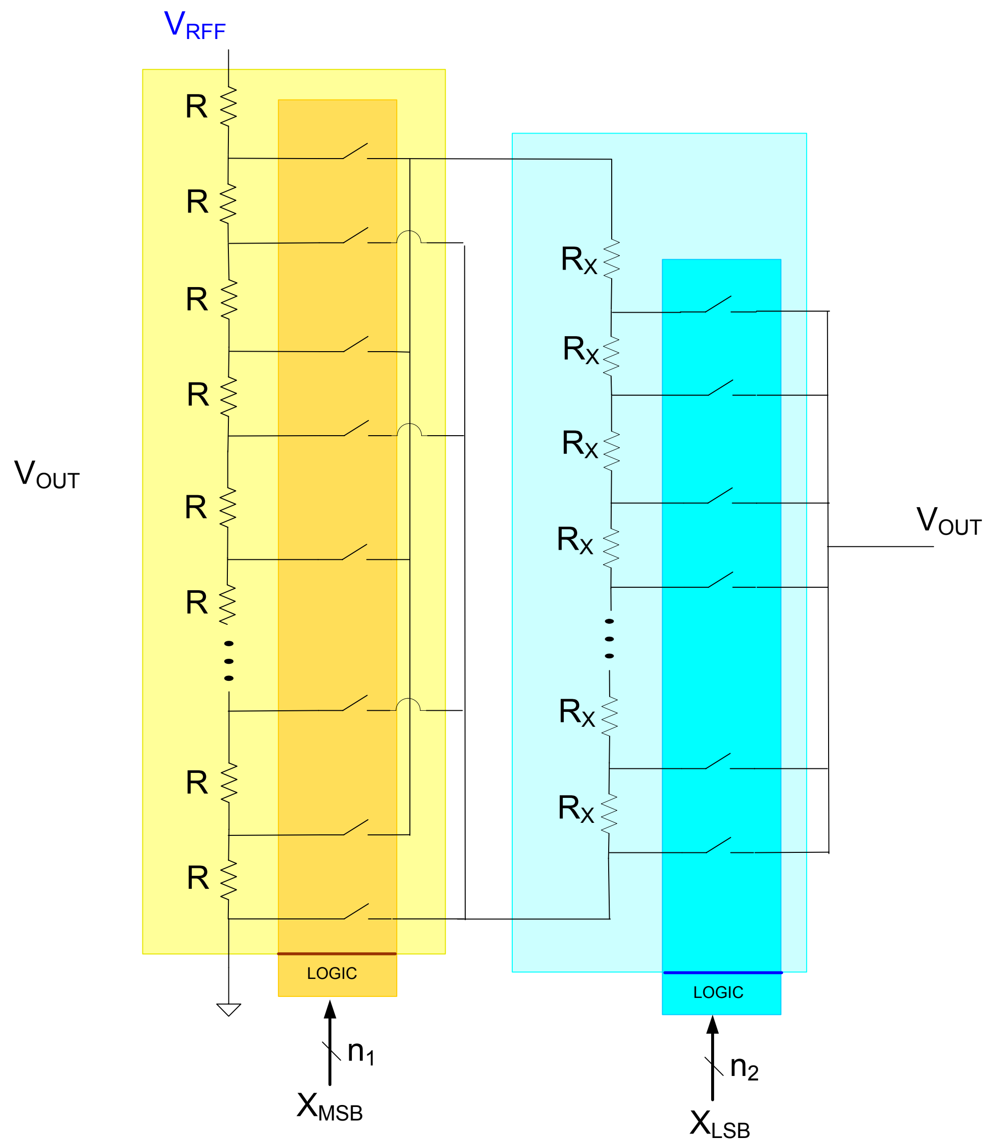
Types of DACs

Interpolating DAC



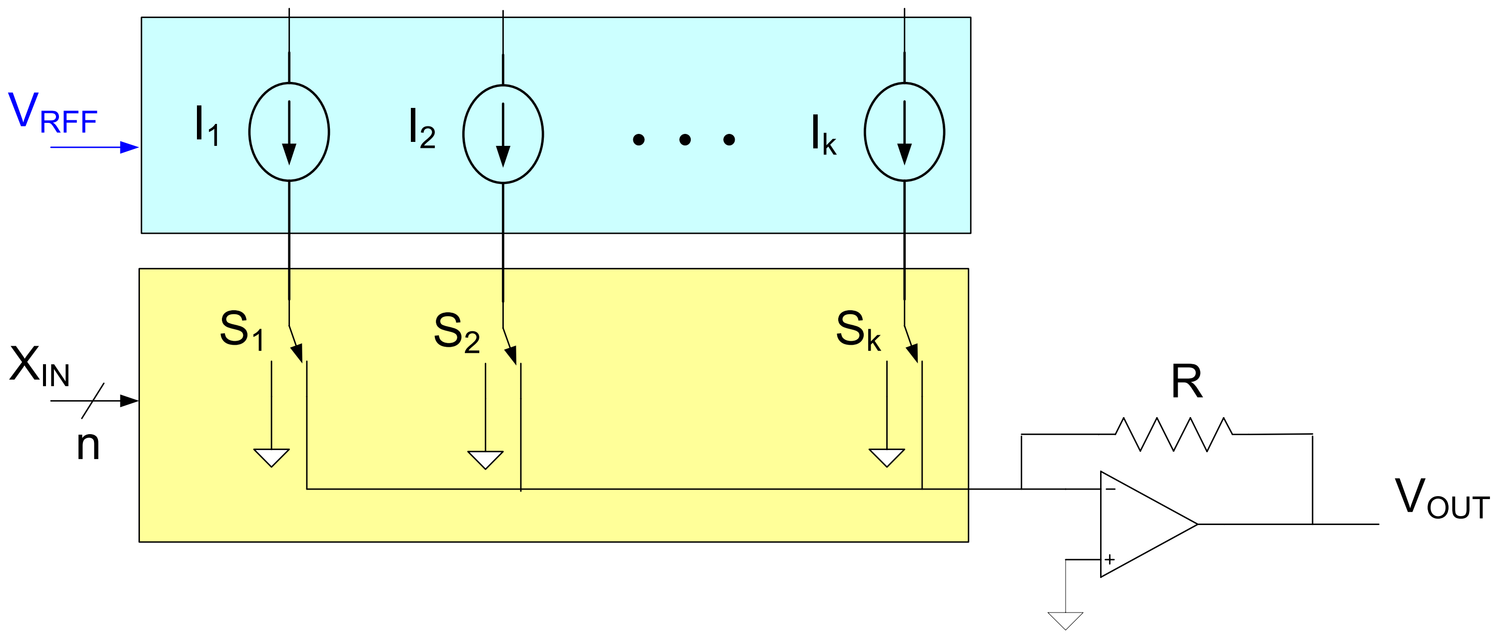
Types of DACs

Interpolating DAC



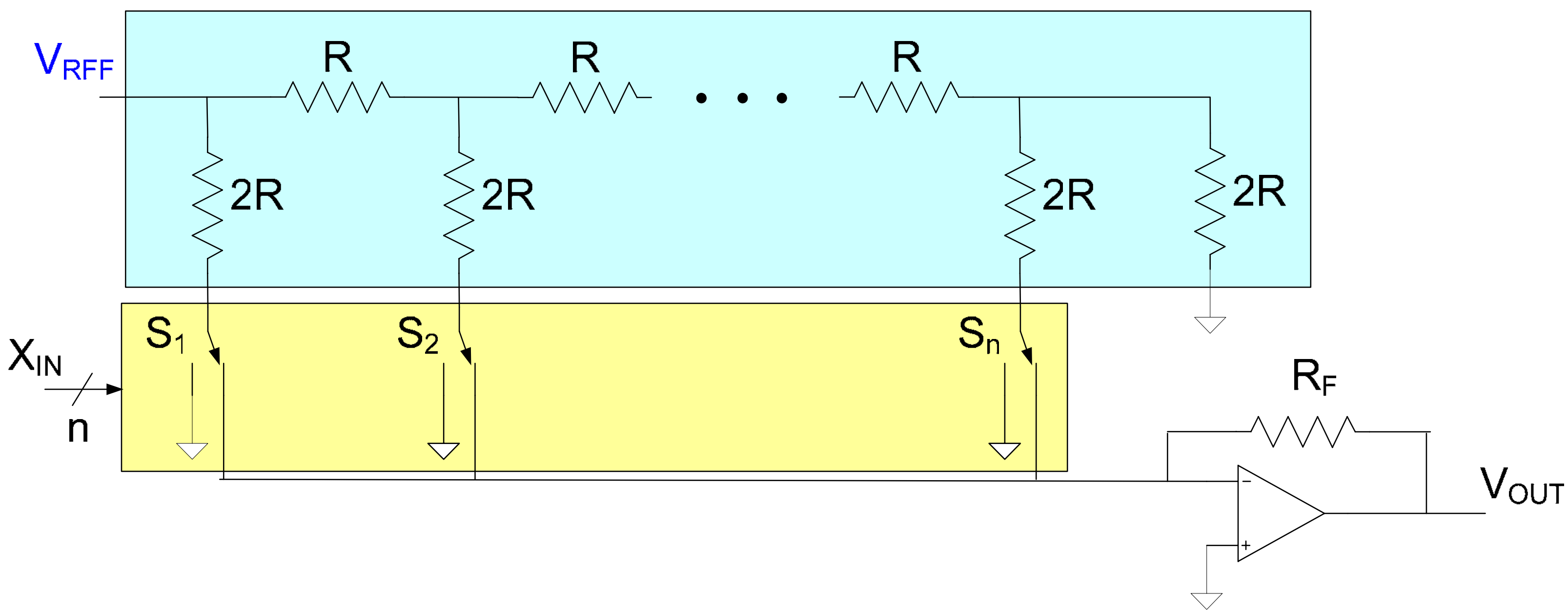
Types of DACs

Current-steering DAC



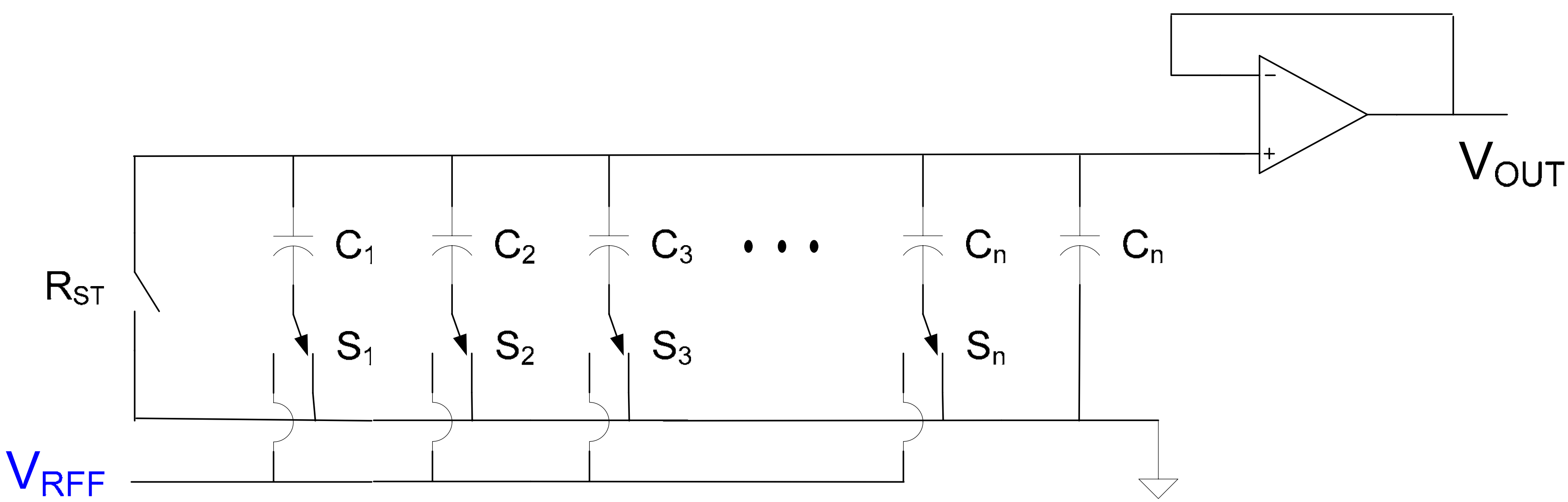
Types of DACs

Ladder DAC (R-2R)



Types of DACs

Charge-Redistribution DAC

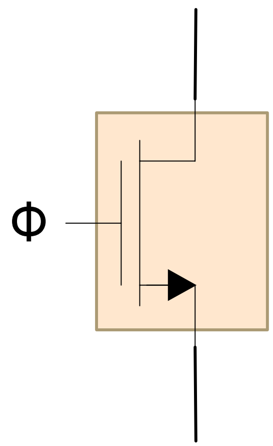


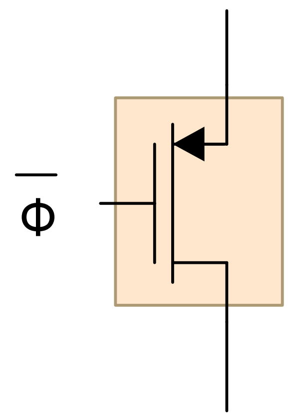
$$C_k = \frac{C}{2^{k-1}}$$

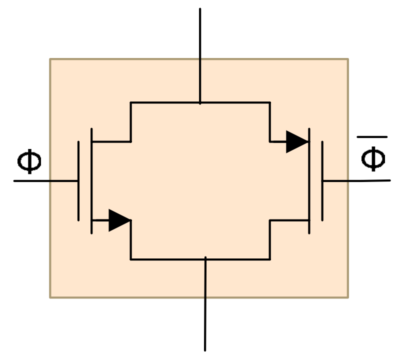
Observation: Most of the ADCs and DACs use switches

Switches used in DACs and ADCs DAC

Usually switches are simple a single MOS transistor or two MOS transistors







Engineering Issues for Using Data Converters

1. Inherent with Data Conversion Process

- Amplitude Quantization
- Time Quantization

(Present even with Ideal Data Converters)

2. Nonideal Components

- Uneven steps
- Offsets
- Gain errors
- Response Time
- Noise

(Present to some degree in all physical Data Converters)

How do these issues ultimately impact performance ?

Engineering Issues for Using Data Converters

Inherent with Data Conversion Process

- Time Quantization
- Amplitude Quantization

How do these issues ultimately impact performance ?

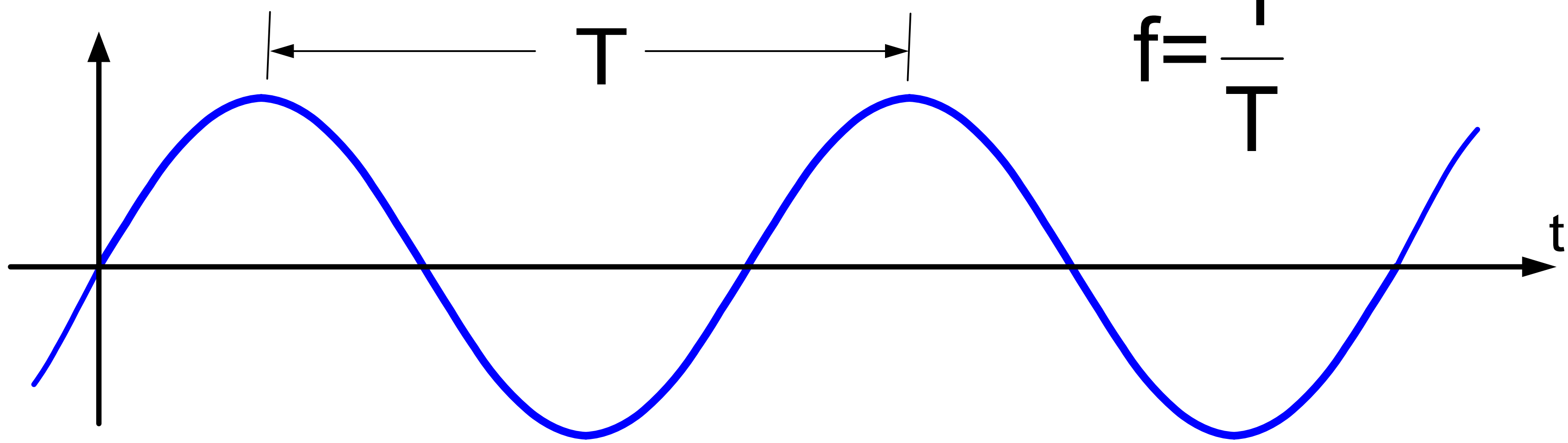
Time Quantization

Sampling Theorem

- Aliasing
- Anti-aliasing Filters
- Analog Signal Reconstruction

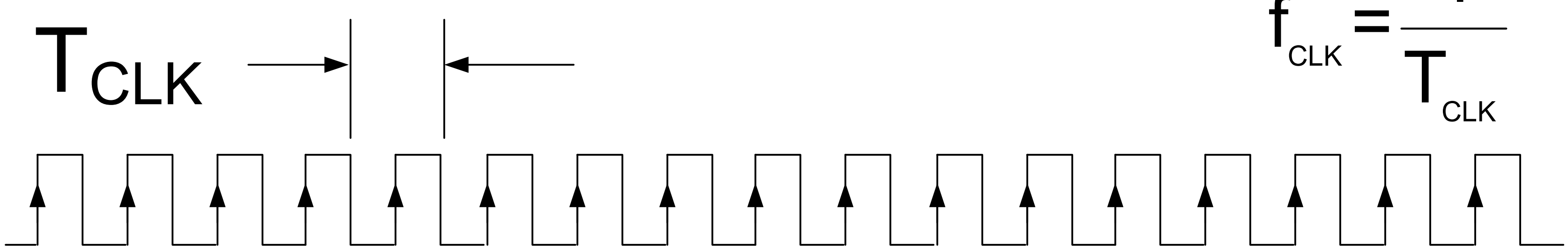
Time Quantization

For convenience, consider a sinusoidal signal



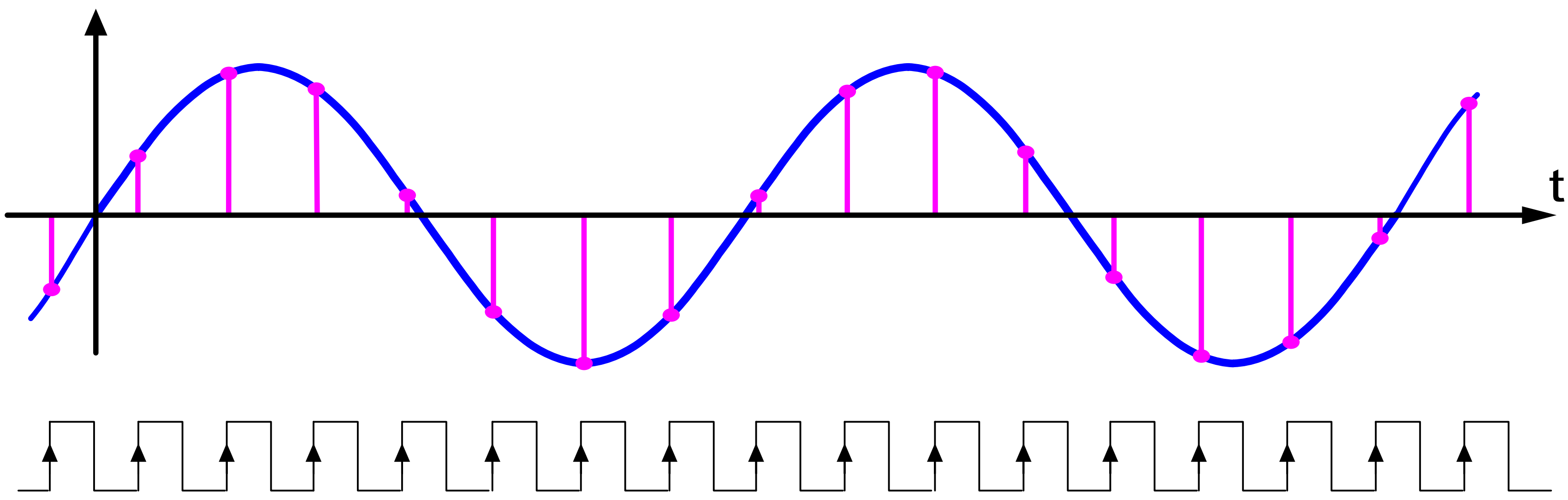
$$f = \frac{1}{T}$$

Consider a positive-edge triggered sampling clock signal

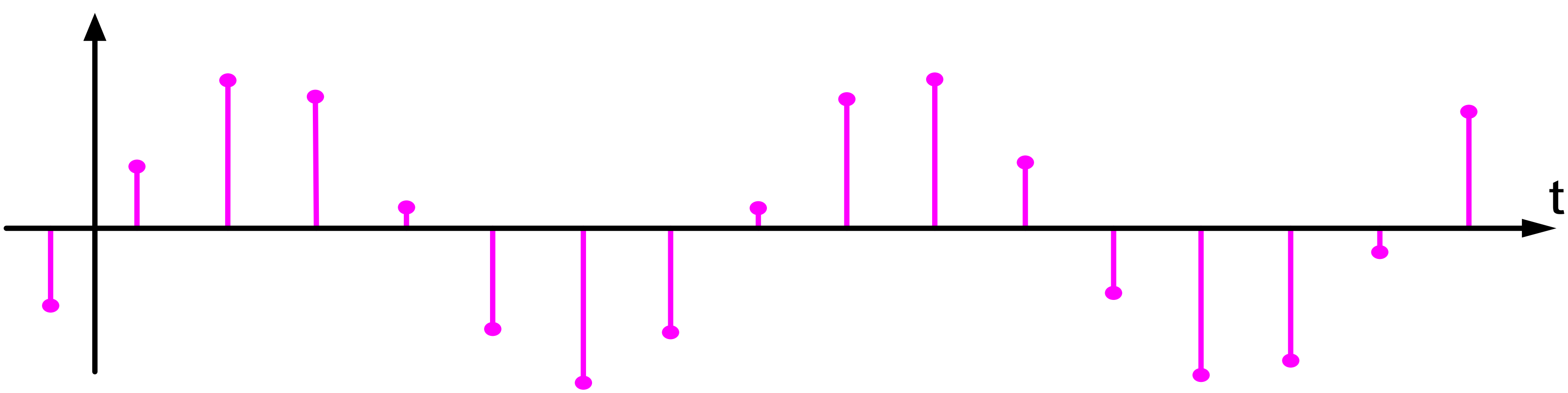


$$f_{CLK} = \frac{1}{T_{CLK}}$$

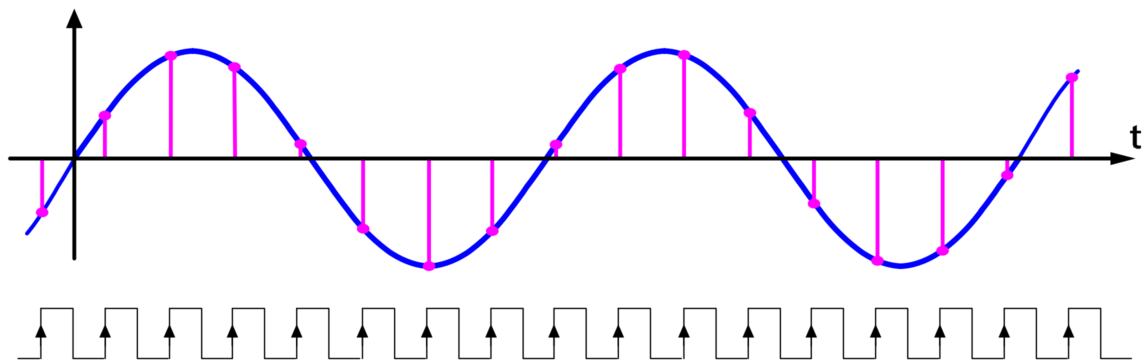
Time Quantization



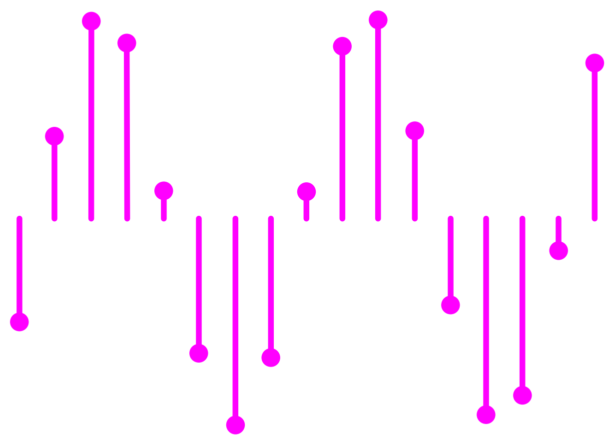
Time-quantized samples of signal



Time Quantization



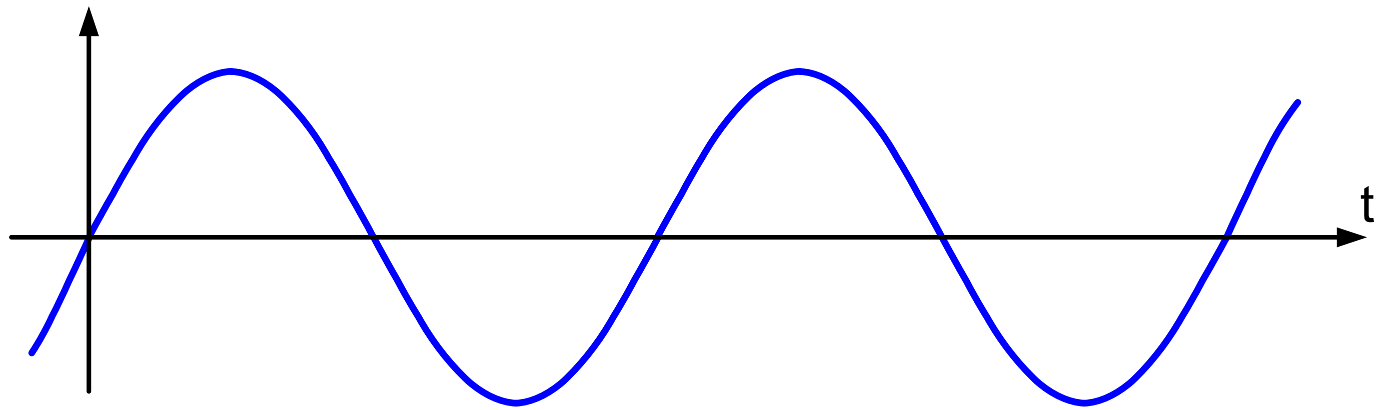
Time-quantized samples of signal



$[V_1, V_2, \ldots\ldots V_m]$

Once time-quantized, the samples become a sequence of real numbers and the time axis need no longer be specified (the time where the first sample was taken and the clock period may be recorded as real numbers as well)

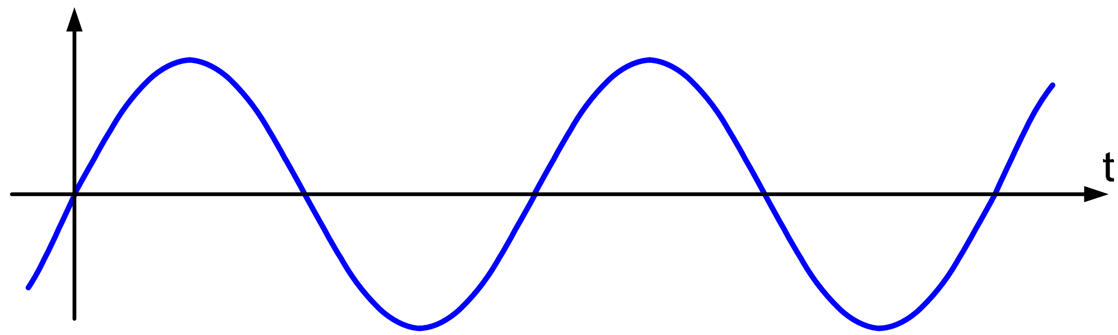
Time Quantization



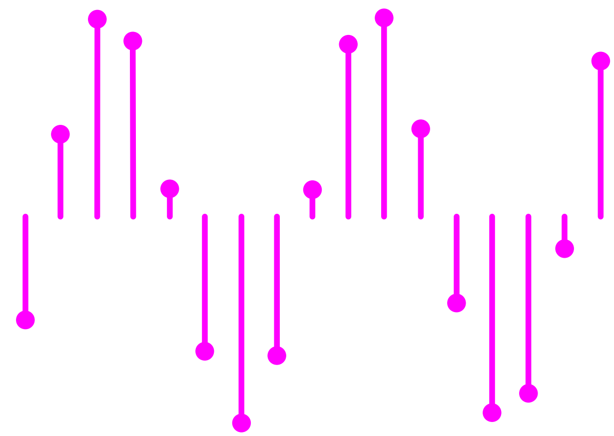
Time-quantized samples of signal

All information about original signal between the sample points is lost when the signal is sampled

Time Quantization



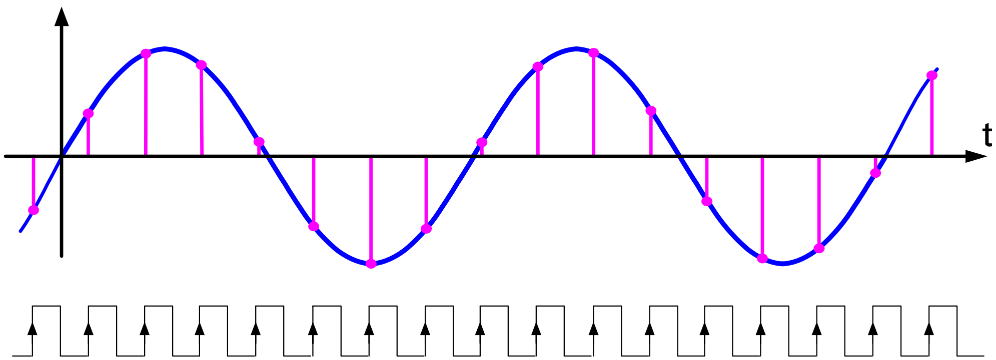
Time-quantized samples of signal

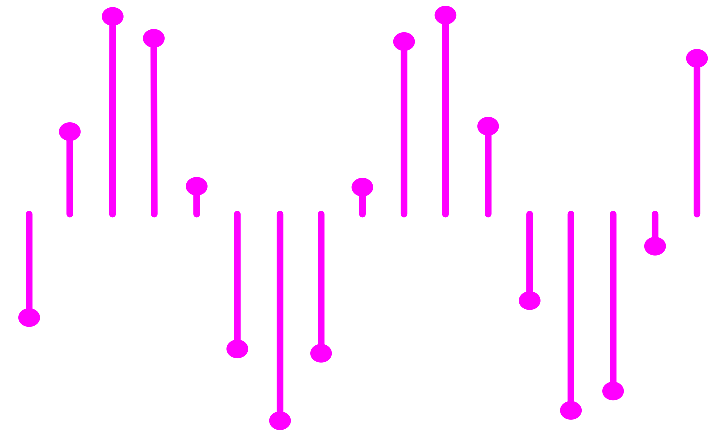


All information about original signal between the sample points is lost when the signal is sampled

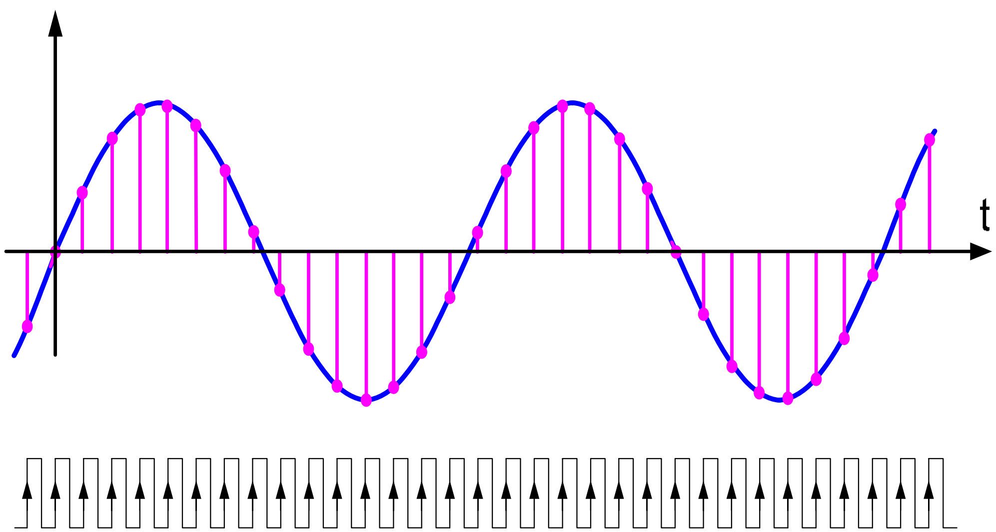
How often must a signal be sampled so that enough information about the original signal is available in the samples so that the samples can be used to represent the original signal ?

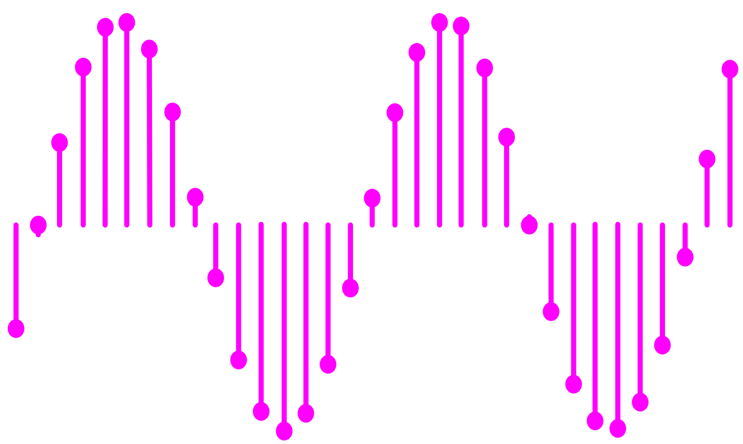
Time Quantization



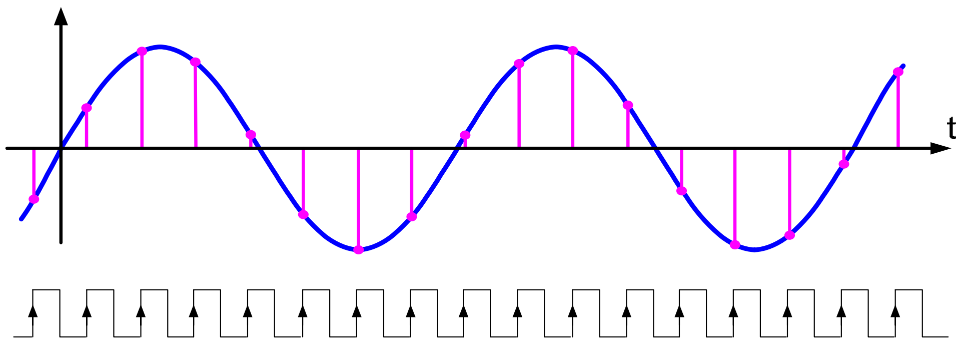


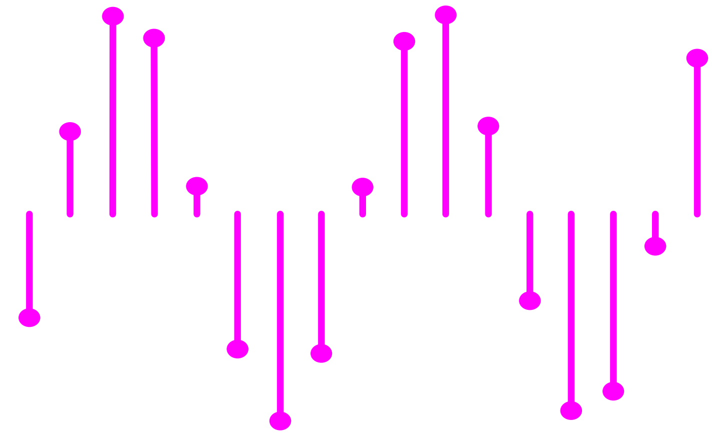
more samples:



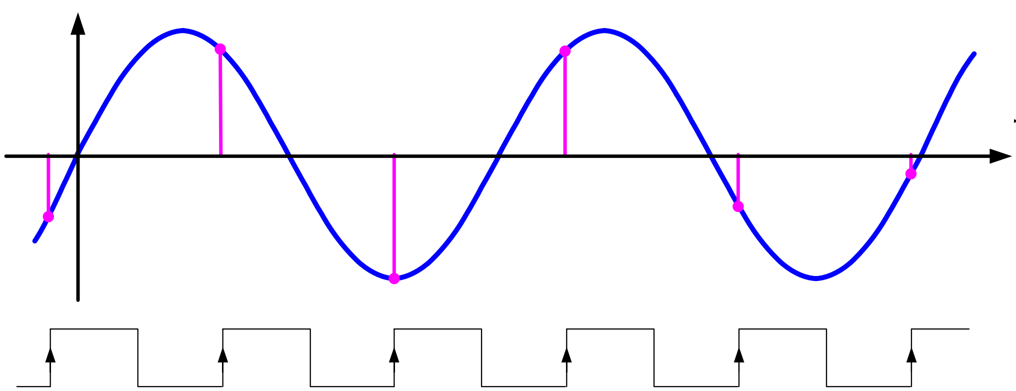


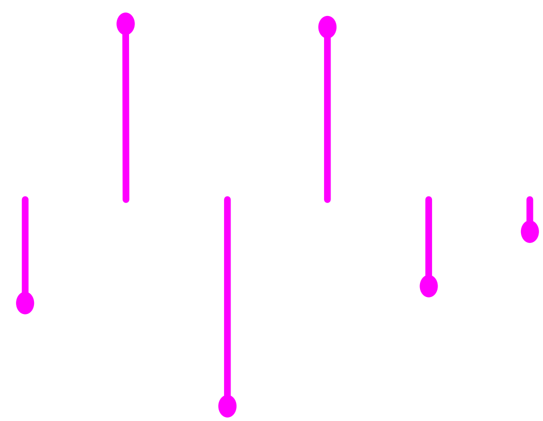
Time Quantization



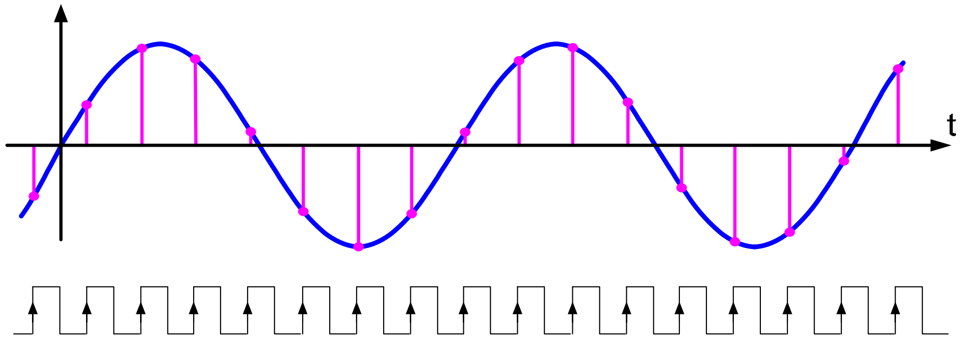


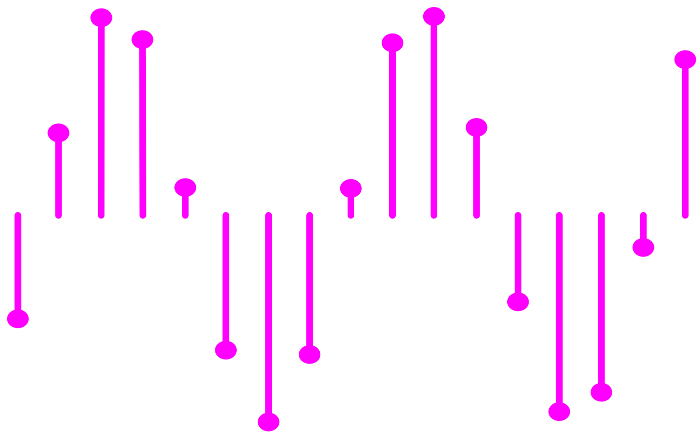
less samples:



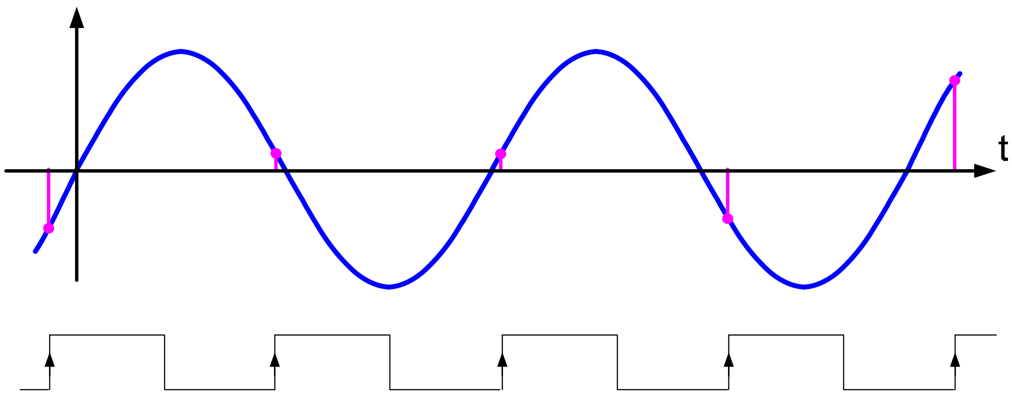


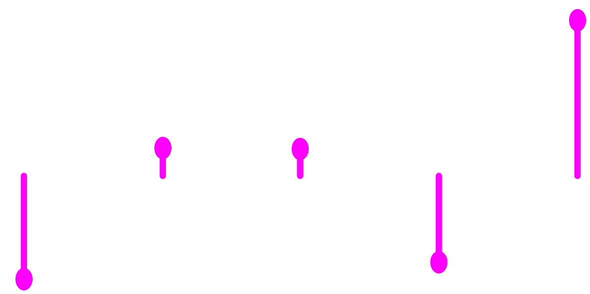
Time Quantization



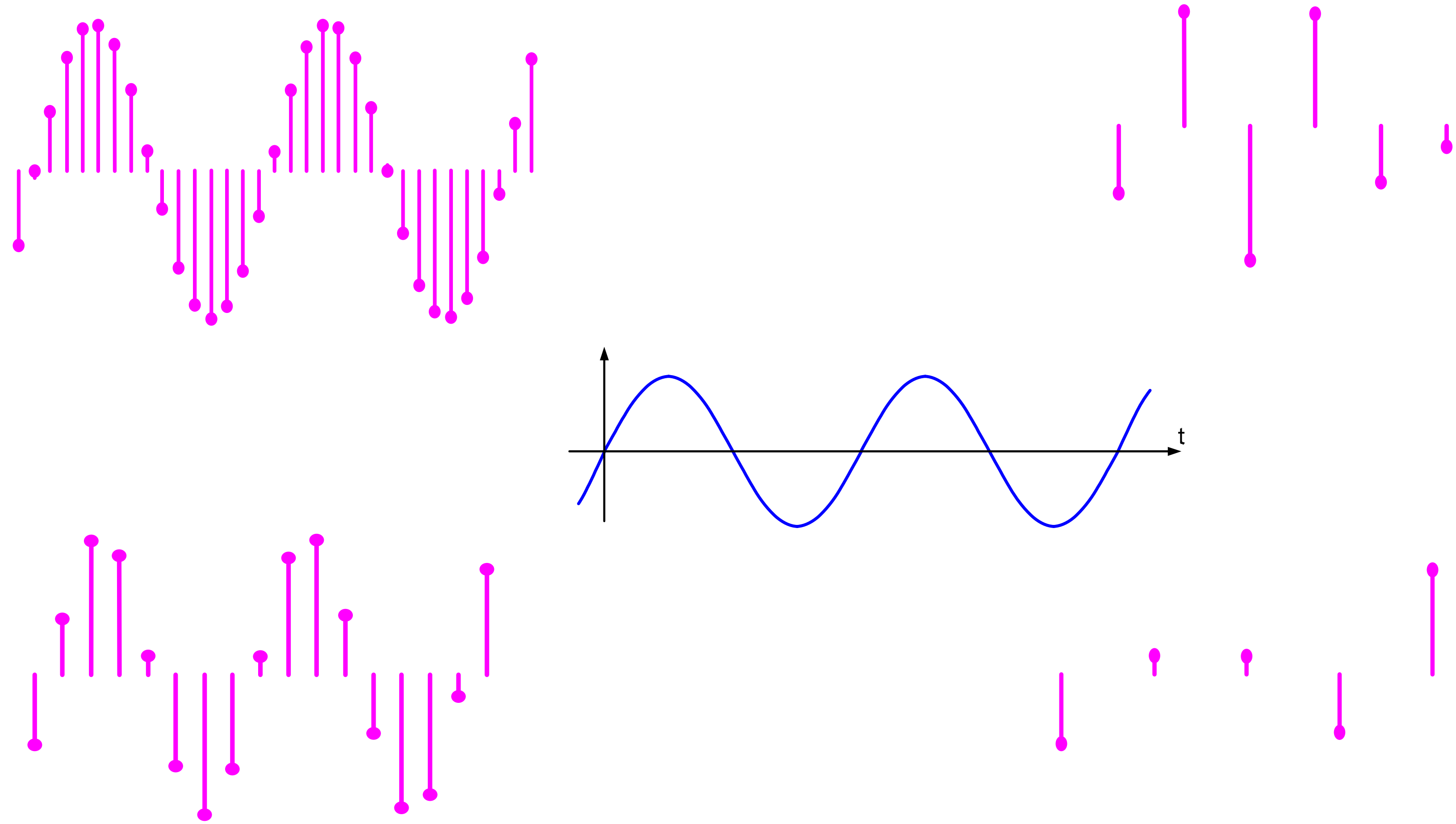


even less samples:



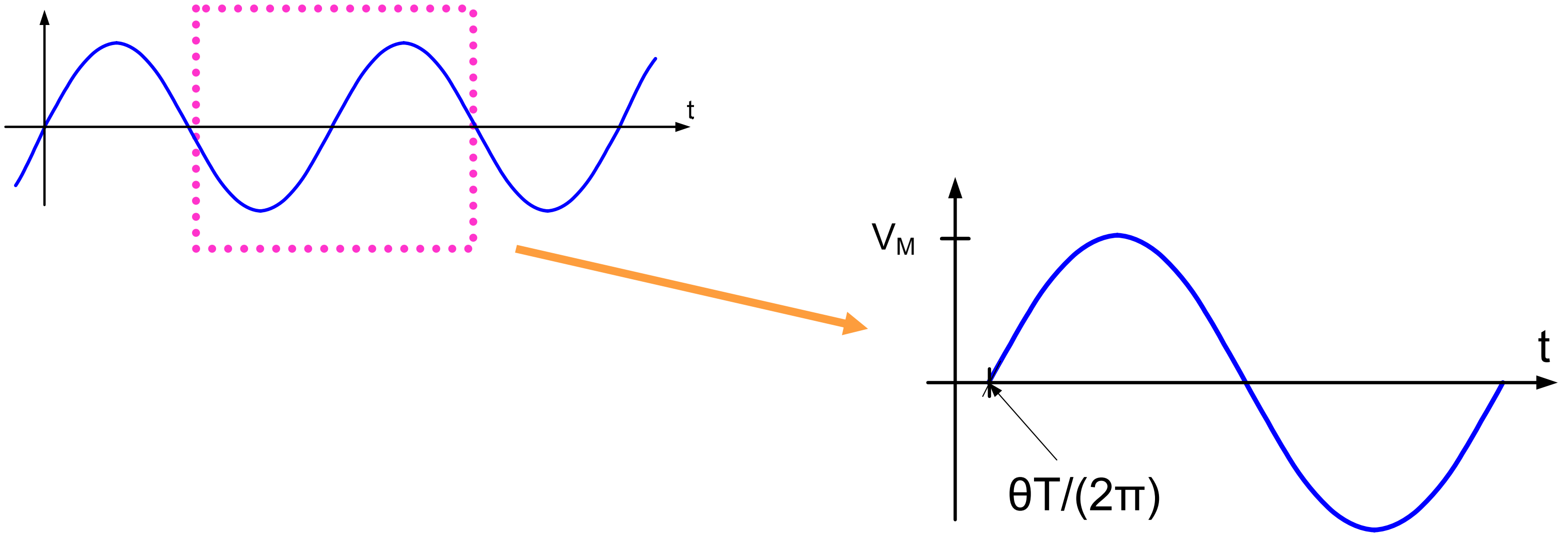


Time Quantization



How often must a signal be sampled so that enough information about the original signal is available in the samples so that the samples can be used to represent the original signal ?

Time Quantization



How often must a signal be sampled so that enough information about the original signal is available in the samples so that the samples can be used to represent the original signal ?

$$f(t)=V_M \sin(\omega t-\theta)$$

If the sampling times are known, there are two unknowns in this equation, V_M and θ.

So two samples during this period that provide two non-zero values of f(t) will provide sufficient information to completely recreate the signal f(t)!

Time Quantization

How often must a signal be sampled so that enough information about the original signal is available in the samples so that the samples can be used to represent the original signal ?

The Sampling Theorem

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency is greater than twice the signal bandwidth.

Sometimes termed Shannon's sampling theorem or the Nyquist-Shannon sampling theorem

Time Quantization

How often must a signal be sampled so that enough information about the original signal is available in the samples so that the samples can be used to represent the original signal ?

The Sampling Theorem

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency is greater than twice the signal bandwidth.

This is a key theorem and many existing communication standards and communication systems depend heavily on this property

This theorem often provides a lower bound for clock frequency of ADCs

Sometimes termed Shannon's sampling theorem or the Nyquist-Shannon sampling theorem

Time Quantization

The terms "band limited" and "signal bandwidth" require considerable mathematical rigor to be precise but an intuitive feel for the sampling theorem and the ability to effectively use the sampling theorem can be developed without all of that rigor

The rigorous part:

The Fourier Transform, Y(ω) of a function y(t) is defined as

$$Y(\omega) = \frac{1}{\sqrt{2\pi}} \int_{t=-\infty}^{\infty} y(t)\, e^{-j\omega t}\, dt$$

If y(t) is well-behaved (and most functions of interest are), then y(t) can be obtained from Y(ω) from the expression

$$y(t) = \frac{1}{\sqrt{2\pi}} \int_{\omega=-\infty}^{\infty} Y(\omega)\, e^{j\omega t}\, d\omega$$

Sometimes termed Shannon's sampling theorem or the Nyquist-Shannon sampling theorem

Time Quantization

The rigorous part:

$$\mathrm{Y}(\omega)=\frac{1}{\sqrt{2\pi}}\int_{t=-\infty}^{\infty}\mathrm{y}(\mathrm{t})\mathrm{e}^{-\mathrm{j}\omega\mathrm{t}}\mathrm{dt}$$

Observe the Fourier Transform is very closely related to the Laplace Transform for many (almost all where data converters are used) functions of interest, and they are related by the expression

$$\mathrm{Y}(\omega)=Y(s)\Big|_{s=j\omega}$$

Y(ω) is generally a complex quantity

Time Quantization

The rigorous part:

Signal Bandwidth Definition

If the Fourier Transform of the function $y(t)$ exists and if B is the smallest finite real number for which $Y(\omega)=0$ for all $\omega > B$, then B is the Signal Bandwidth of $y(t)$

Band-limited Definition

If the Fourier Transform of a function $y(t)$ exists, then $y(t)$ is band-limited if there exists a finite real number H such that $Y(\omega)=0$ for all $\omega>H$.

If the signal $y(t)$ is periodic, the sampling theorem can also be given and the concepts of band-limits and signal bandwidth may be more intuitive. This will be discussed later.

Time Quantization

How often must a signal be sampled so that enough information about the original signal is available in the samples so that the samples can be used to represent the original signal ?

The Sampling Theorem

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency is greater than twice the signal bandwidth.

- the term "band limited" is closely related to term "signal bandwidth"

- the term "Nyquist Rate" in reference to a bandlimited signal is the minimum sampling frequency that can be used if the entire signal can be reconstructed from the samples

$$f_{NYQ}=2B$$

Sometimes termed Shannon's sampling theorem or the Nyquist-Shannon sampling theorem

Time Quantization

The Sampling Theorem

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency is greater than twice the signal bandwidth.

Alternatively

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency exceeds the Nyquist Rate.

Time Quantization

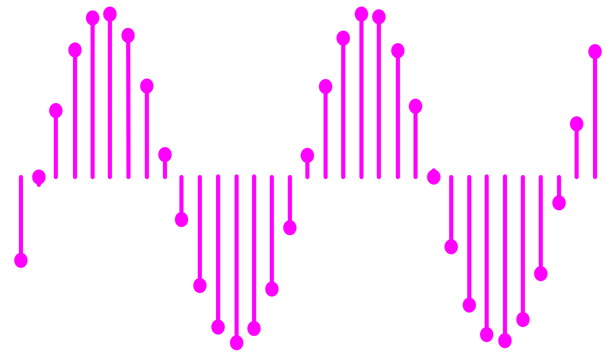
The Sampling Theorem

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency exceeds the Nyquist Rate.

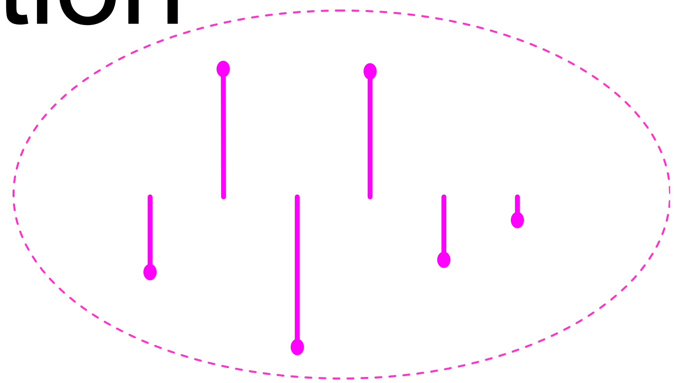
Practically, signals are often sampled at frequency that is just a little bit higher than the Nyquist rate though there are some applications where the sampling is done at a much higher frequency (maybe with minimal benefit)

The theorem as stated only indicates sufficient information is available in the samples if the criteria are met to reconstruct the original continuous-time signal, nothing is said about how this can be practically accomplished.

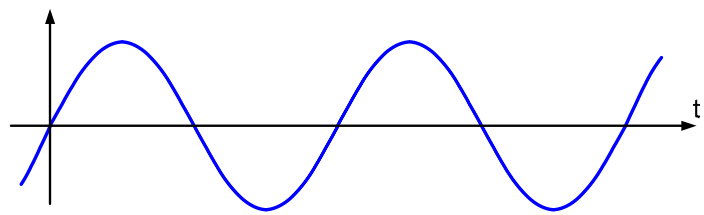
Time Quantization

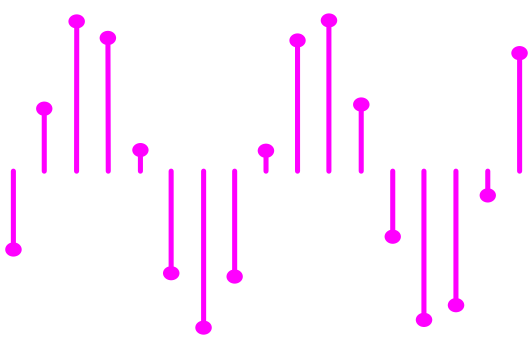


Approximately 6 times f_{NYQ}

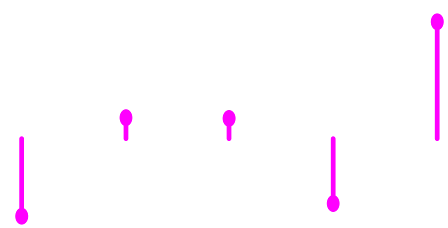


Slightly above f_{NYQ}





Approximately 3 times f_{NYQ}



Below f_{NYQ}

Time Quantization

The Sampling Theorem

An exact reconstruction of a continuous-time signal from its samples can be obtained if the signal is band limited and the sampling frequency exceeds the the Nyquist Rate.

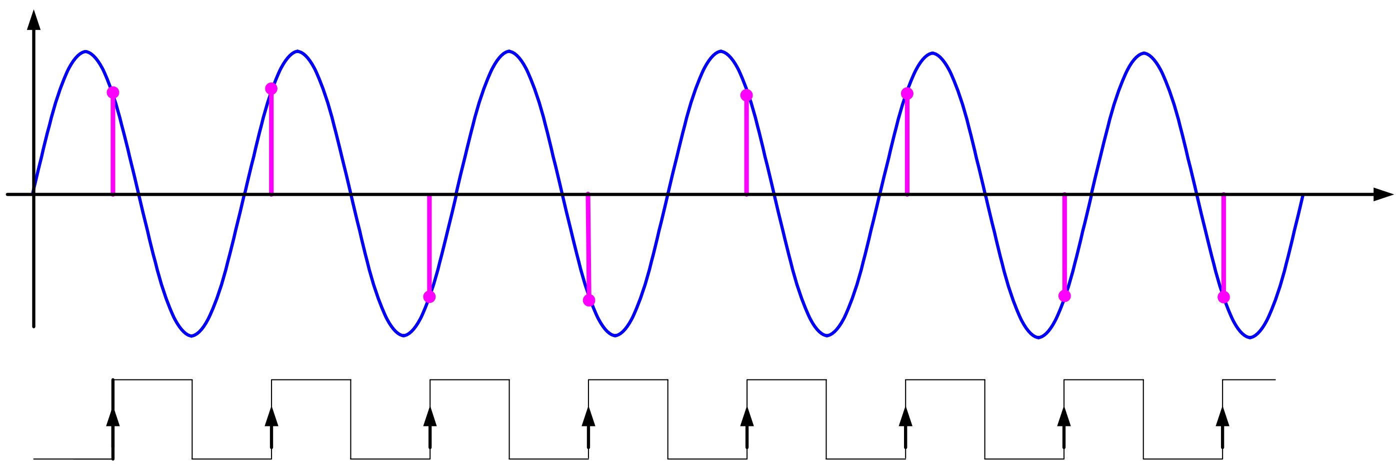
What happens if the requirements for the sampling theorem are not met?

How can a continuous-time signal be practically reconstructed from the samples if the hypothesis of the sampling theorem was satisfied when the samples were taken?

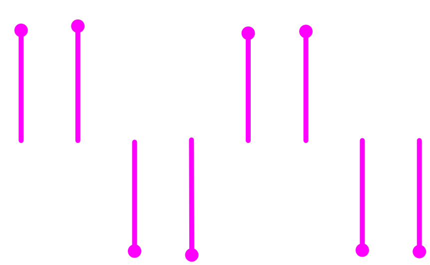
Time Quantization

What happens if the requirements for the sampling theorem are not met?

Example: Consider a signal that is of frequency 3/4 f_{CLK}
Signal violates the hypothesis of the sampling theorem, it is higher in frequency than ½ f_{CLK}



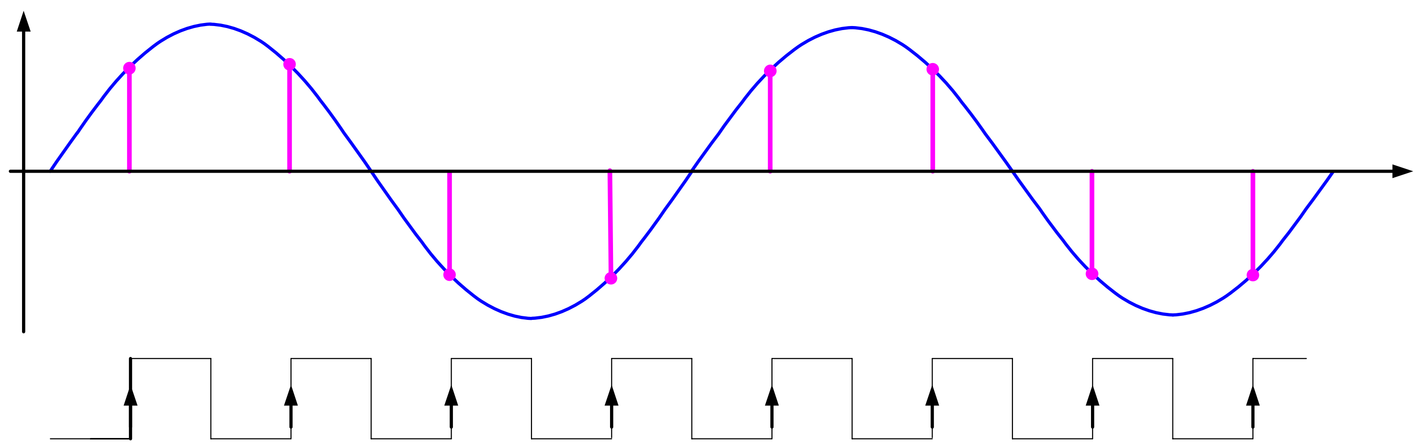
Sampled output sequence:



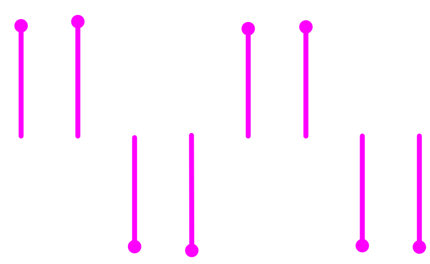
Time Quantization

What happens if the requirements for the sampling theorem are not met?

Example: Consider a signal that is of frequency 1/4 f_{CLK} - assume f_{CLK} same as before
Signal violates the hypothesis of the sampling theorem, it is higher in frequency than ½ f_{CLK}



Sampled output sequence:

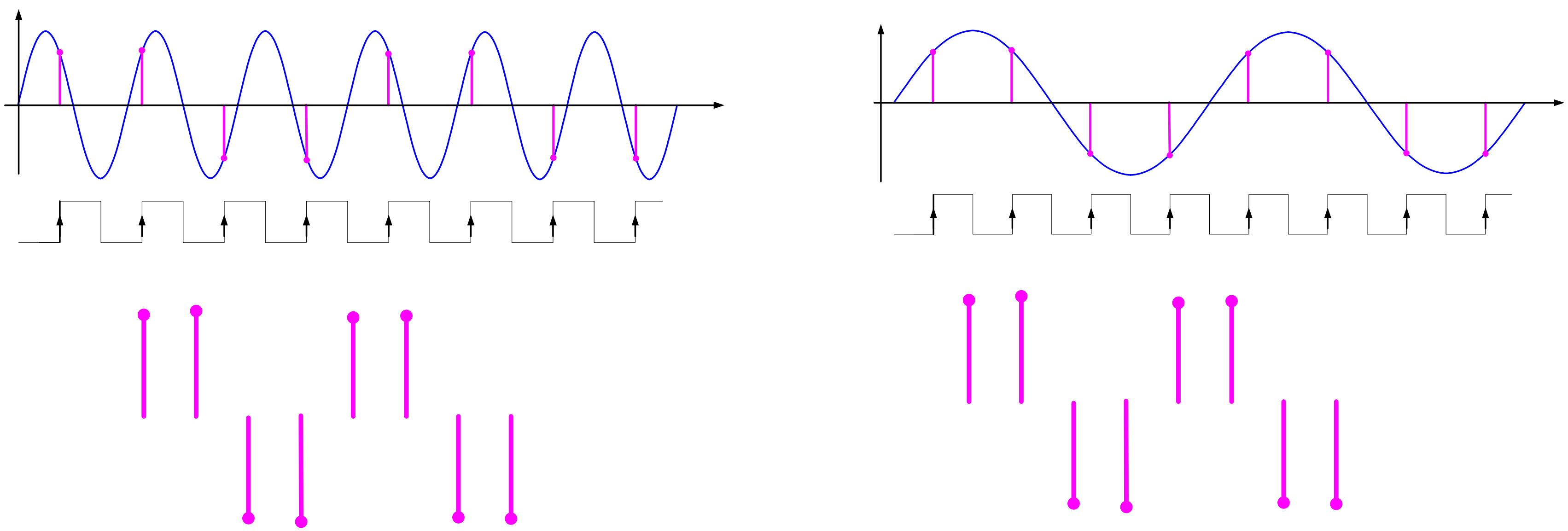


Time Quantization

What happens if the requirements for the sampling theorem are not met?

Example:

$f_{SIG}=3/4\ f_{CLK}$ $f_{SIG}=1/4\ f_{CLK}$

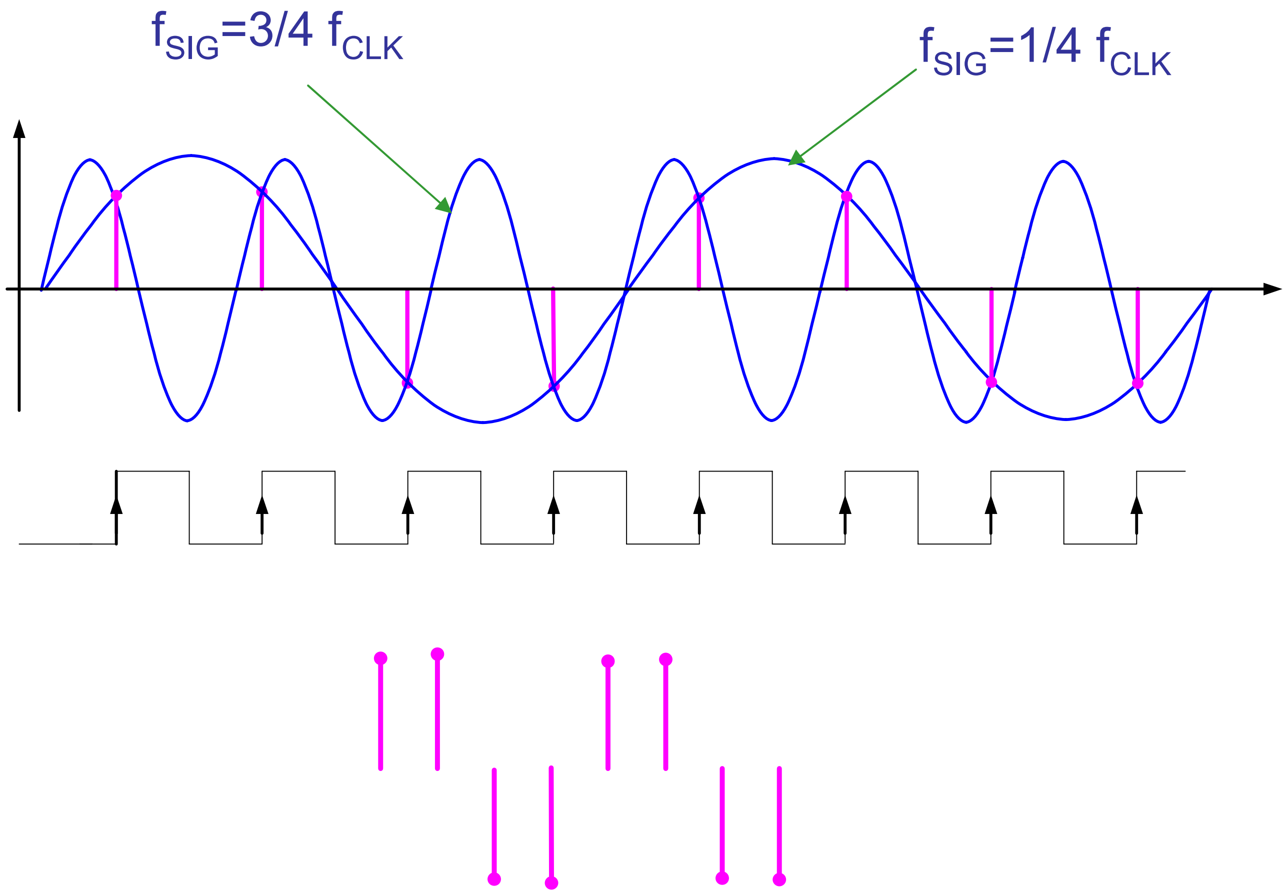


Output sampled sequences are identical!

Time Quantization

What happens if the requirements for the sampling theorem are not met?

Example:

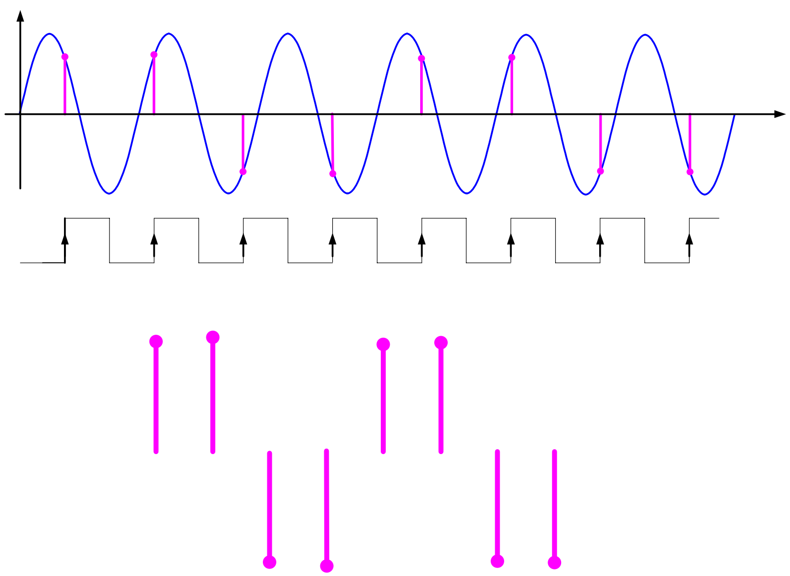


Time Quantization

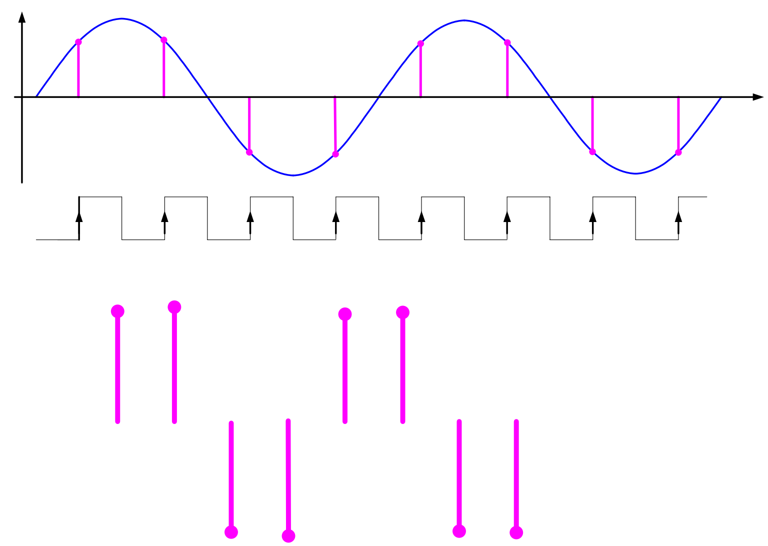
What happens if the requirements for the sampling theorem are not met?

Example:

f_{SIG}=3/4 f_{CLK}



f_{SIG}=1/4 f_{CLK}



Since two different signals have same sampled sequence, can not uniquely reconstruct the signal from the samples

Time Quantization

What happens if the requirements for the sampling theorem are not met?

Since two different signals have same sampled sequence, can not uniquely reconstruct the signal from the samples

This makes the samples of a signal that was at a frequency above the Nyquist Rate look like those of a signal that meets the Nyquist Rate requirements

The creation of samples that appear to be of a lower frequency is termed aliasing.

Aliasing will occur if signals are sampled with a clock of frequency less than the Nyquist Rate for the signal.