Fundamentals of telecommunications

Training materials for wireless trainers

The Abdus Salam
International Centre
for Theoretical Physics

Goals

To present the basics concepts of telecommunication systems with focus on digital and wireless

Basic Concepts

- Signal
  - Analog, Digital, Random
- Bandwidth
- Spectrum, Fourier transform
- Impulse response and transfer function
- Frequency translation
- Ideal channel, attenuation, delay
- Filters
- Sampling
- Quantization and coding
- Channel capacity, Noise, Interference, Information
- BER
- Modulation
- Multiplexing
- Duplexing

Telecommunication Signals

Telecommunication signals are variation over ***time*** of voltages, currents or light levels that carry information.

For analog signals, these variations are directly proportional to some physical variable like sound, light, temperature, wind speed, etc.

The information can also be transmitted by digital binary signals, that will have only two values, a digital ***one*** and a digital ***zero***.

Telecommunication Signals

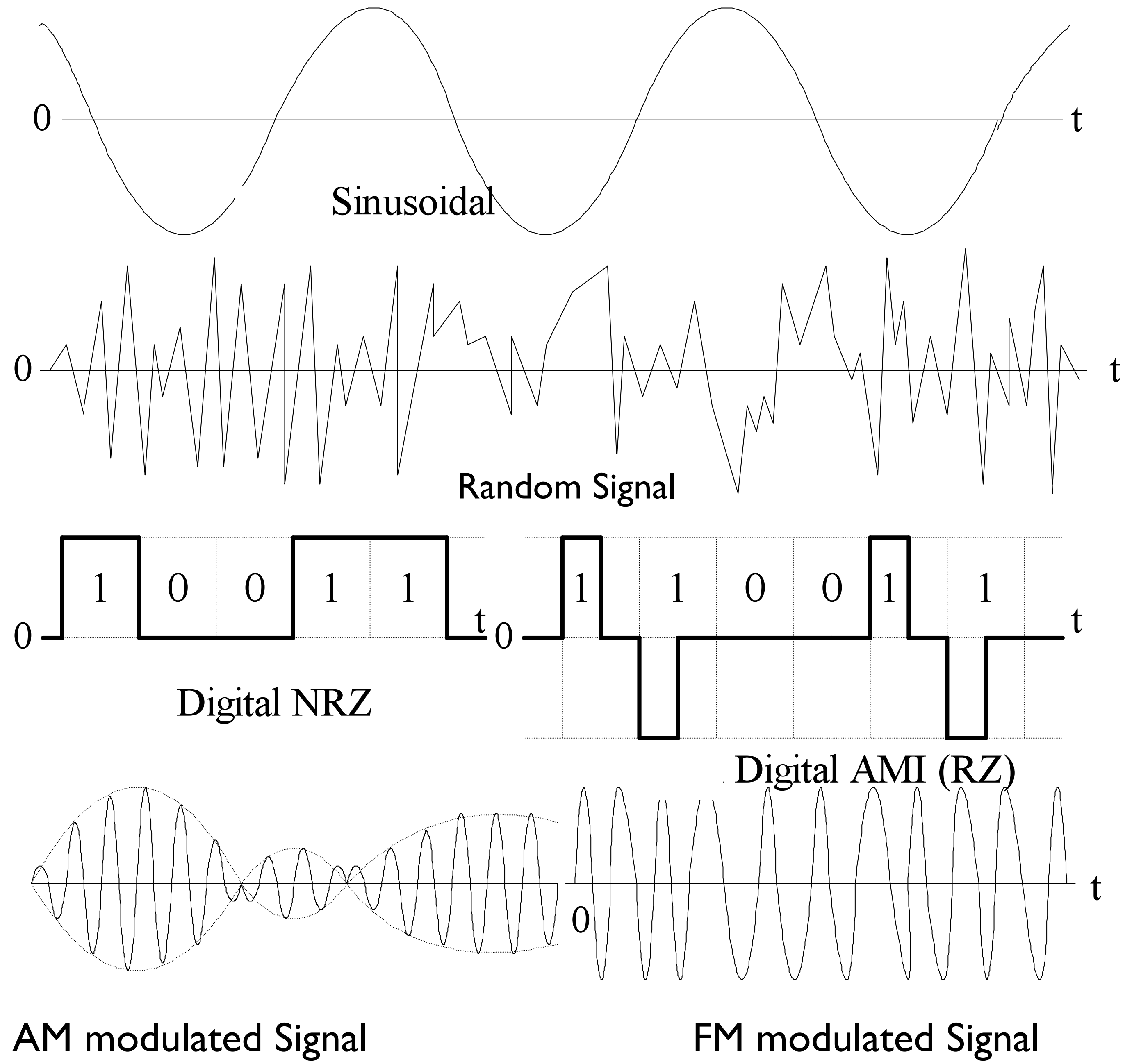
Any analog signal can be converted into a digital signal by appropriately ***sampling*** it.

The sampling frequency must be at least twice the maximum frequency present in the signal in order to carry ***all*** the information contained in it.

Random signal are the ones that are unpredictable and can be described only by statistical means.

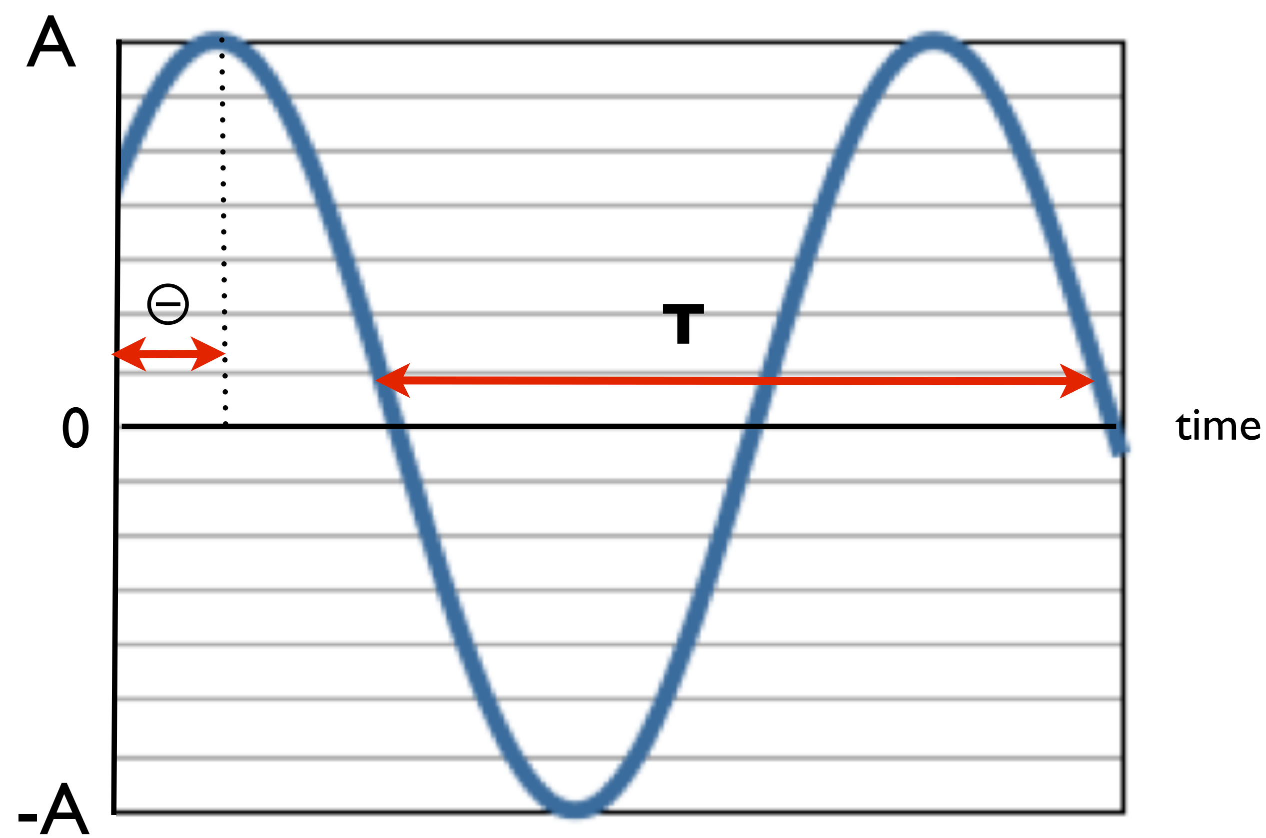
Noise is a typical random signal, described by its mean power and frequency distribution.

Examples of Signals



Sinusoidal Signal

$$v(t) = A\cos(\omega_o t - \Theta)$$



A = Amplitude, volts

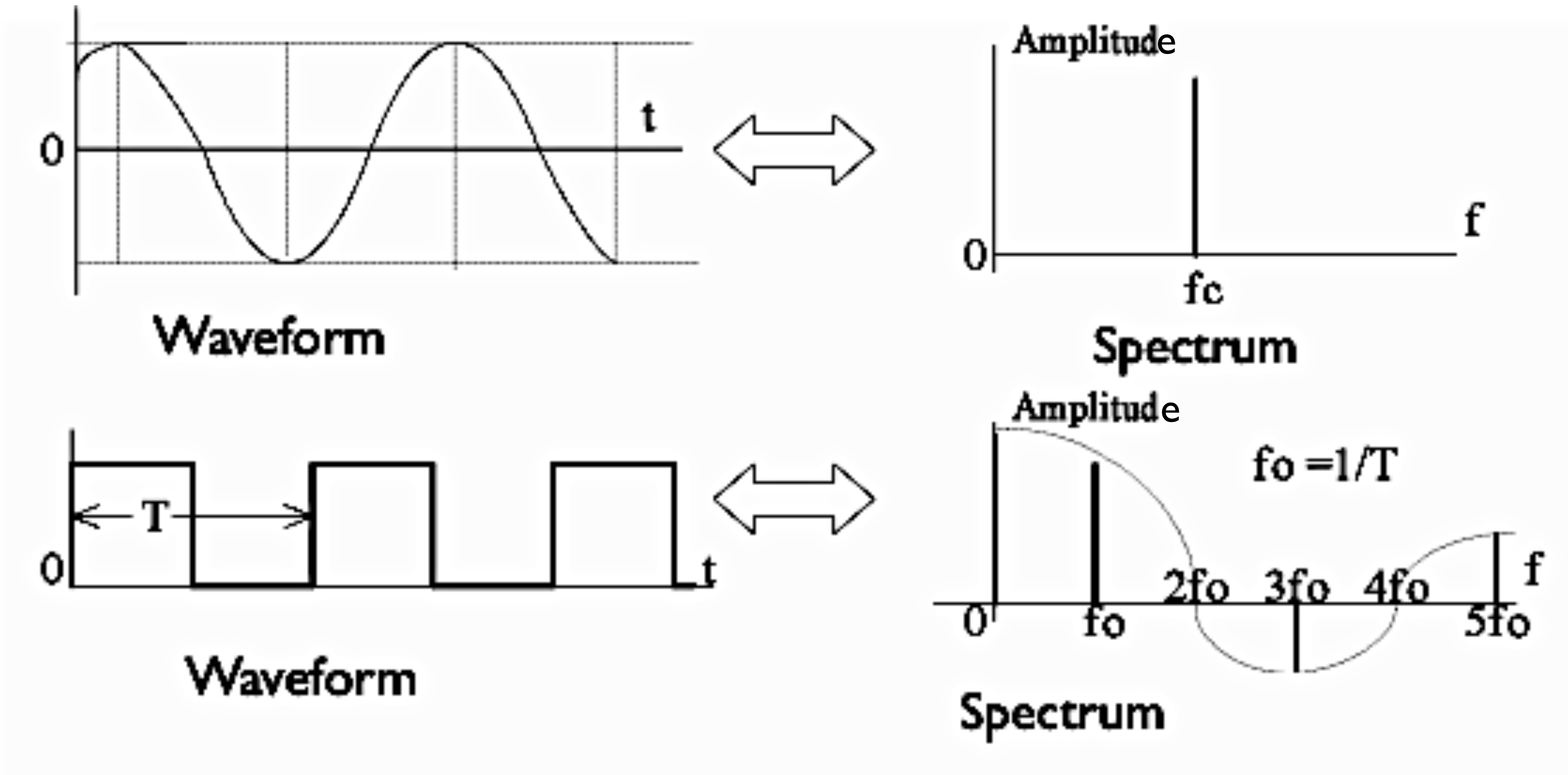
$\omega_o = 2\pi f_o$, angular frequency in radians

f_o = frequency in Hz

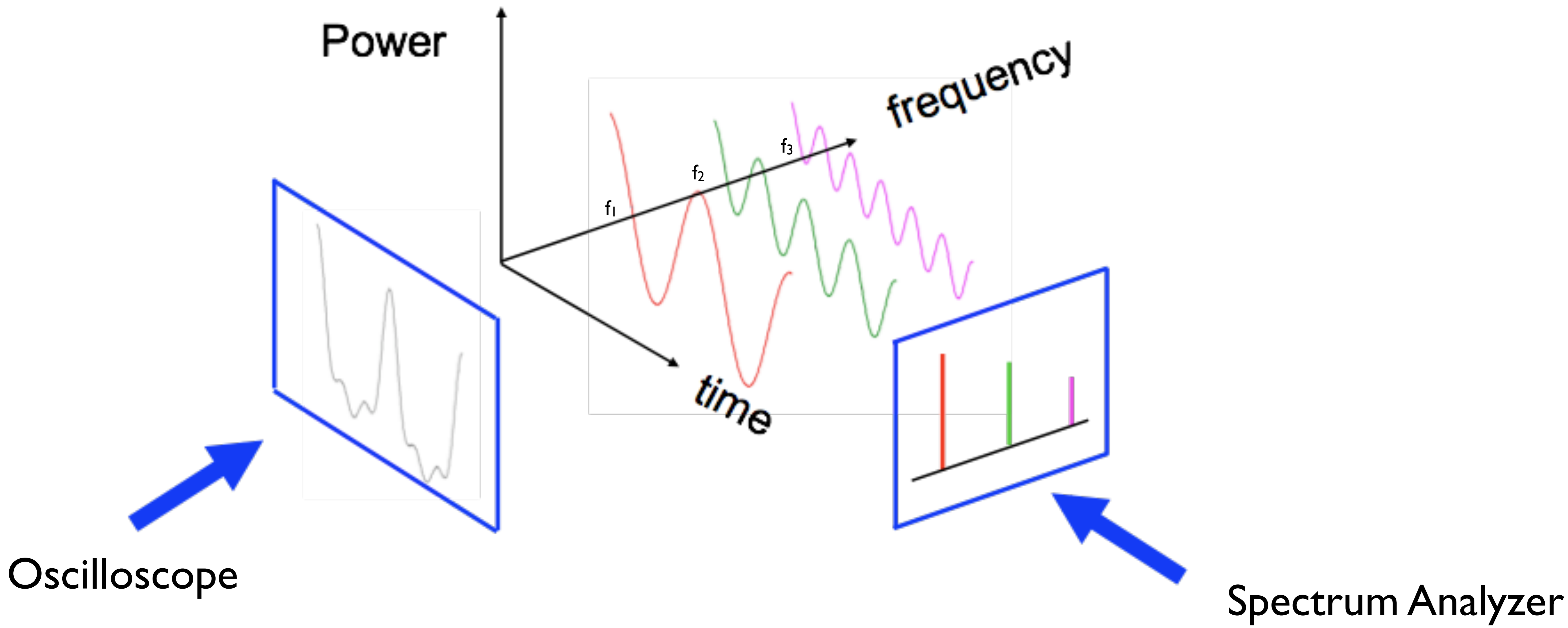
T = period in seconds, $T = 1/f_o$

Θ = Phase

Signals and Spectra



Spectral analysis and filters



Signals and Spectra

Given the time domain description of a signal, we can obtain its spectrum by performing the mathematical operation known as *Fourier Transform*.

The Fourier transform it is very often calculated digitally, and a well known algorithm to expedite this calculation is the *Fast Fourier Transform, FFT*.

The signal can be obtained from its spectrum by means of the *Inverse Fourier Transform*.

Signals and Spectra math

given a signal x(t) its Fourier transform is

$$X(f) = \int_{-\infty}^{\infty} x(t)\, e^{-i\omega t}\, dt$$

conversely, if we know the spectrum, we can find the signal by performing the inverse Fourier transformation

$$x(t) = \int_{-\infty}^{\infty} X(f)\, e^{-i\omega t}\, df$$

$\omega = 2\pi f$

Fast Fourier Transform

Similarly, the fast Fourier transform of a sequence of N equally spaced samples x_n is given by

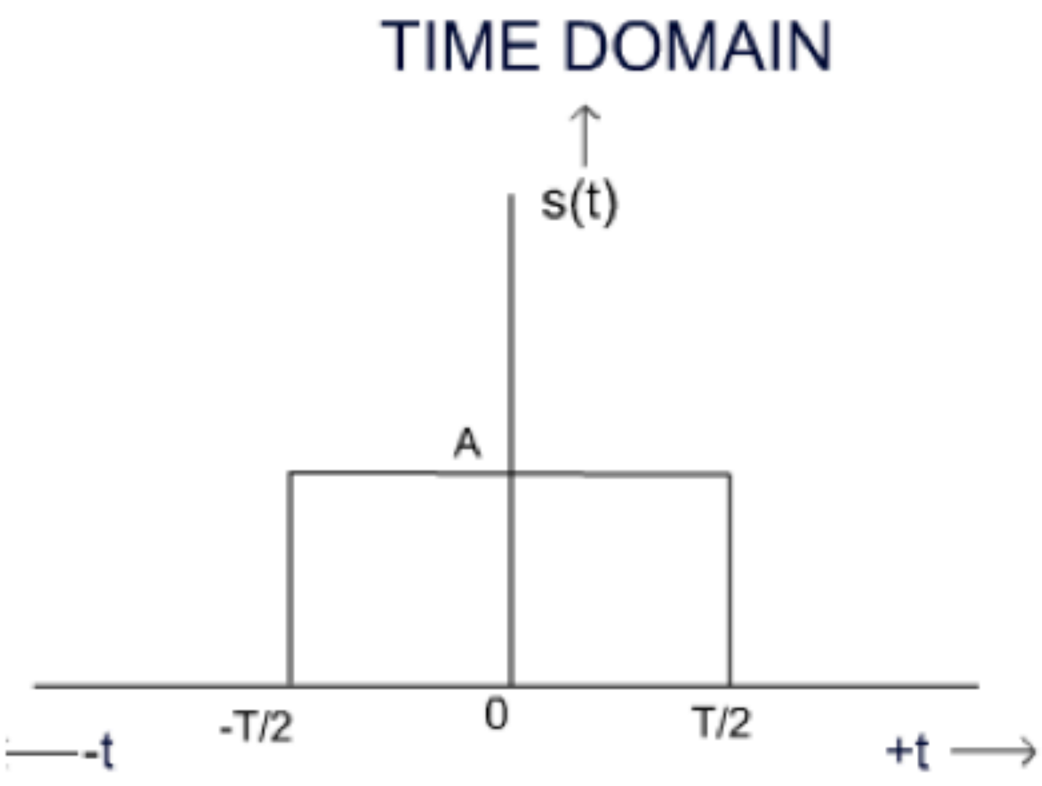
$$X_k = \sum_{n=0}^{N-1} x_n e^{-i2\pi k \frac{n}{N}} \qquad k = 0, \dots, N-1.$$

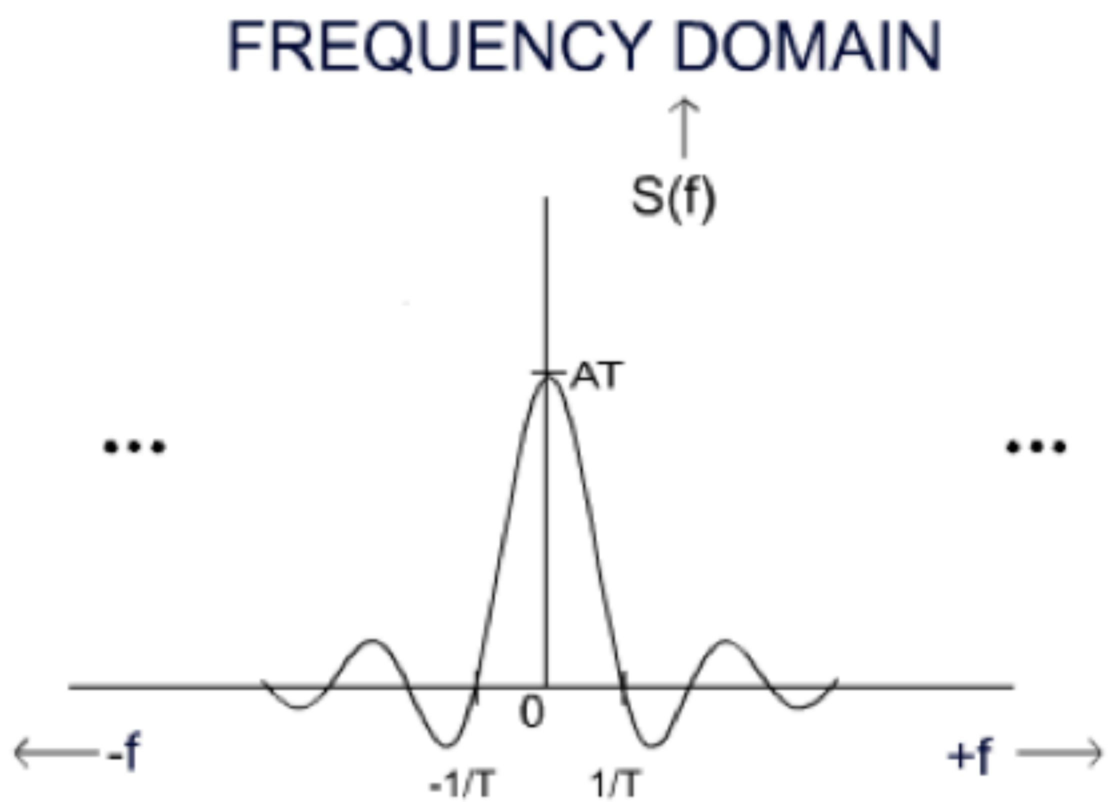
http://www.westga.edu/~jhasbun/osp/Fourier.htm
Applet for Fourier transform, sampling and quantifying

http://www.dspdimension.com/admin/dft-a-pied/
Mastering the Fourier Transform in one day
http://www.fourier-series.com/
http://www.fourier-series.com/IQMod/flashprograms/IQMod.html

http://www-rohan.sdsu.edu/~jiracek/DAGSAW/3.4.html interactive graphics

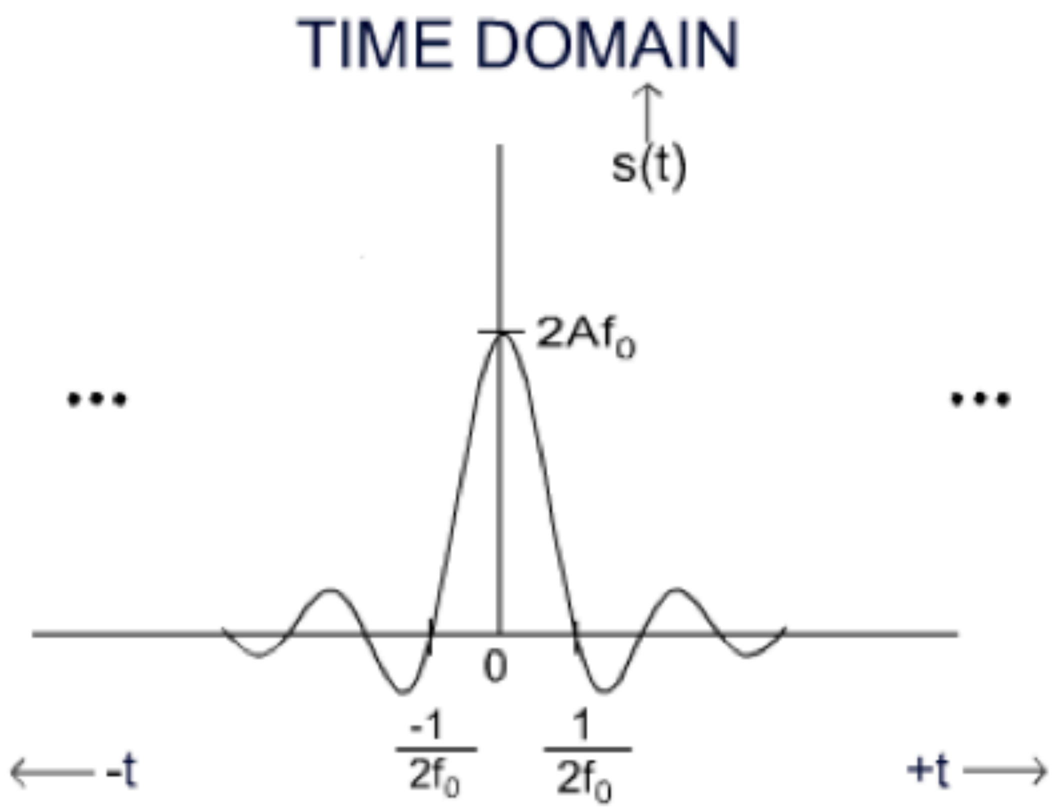
Common signals and spectra

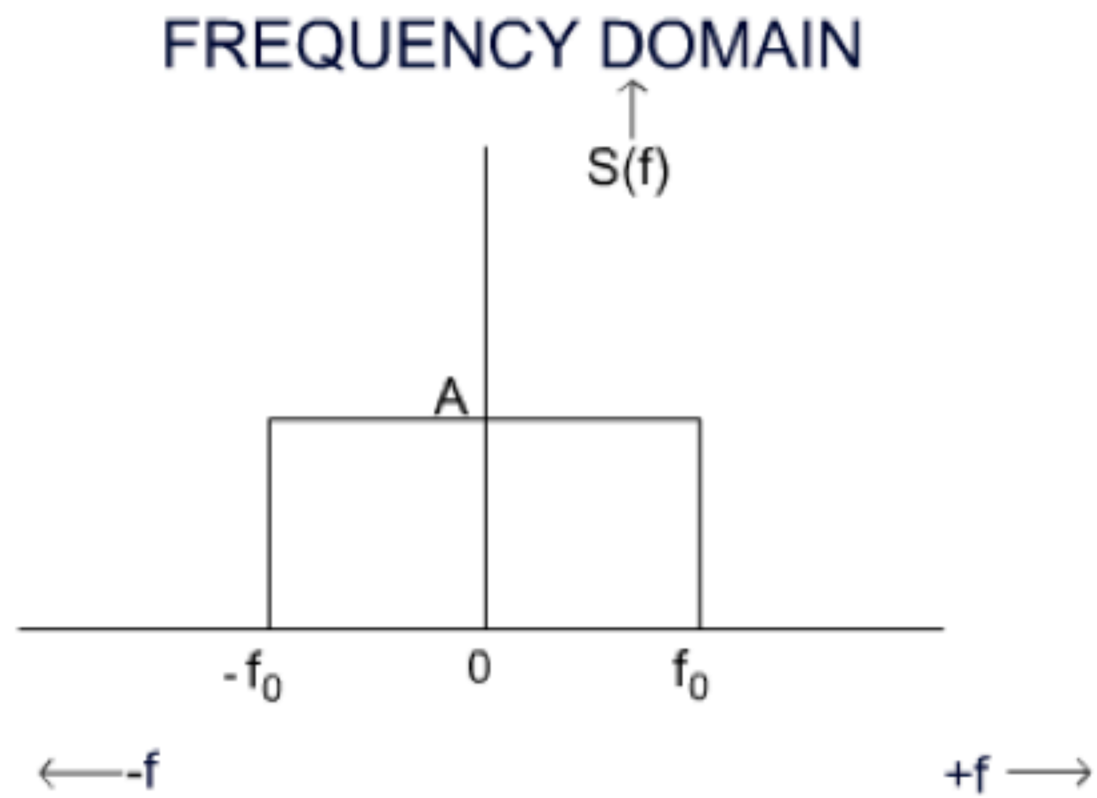




$$s(t) = \begin{cases} A & |t| < T/2 \\ A/2 & |t| = T/2 \\ 0 & |t| > T/2 \end{cases} \quad \longleftrightarrow \quad S(f) = AT\,\frac{\sin(\pi T f)}{\pi T f}$$

Common signals and spectra

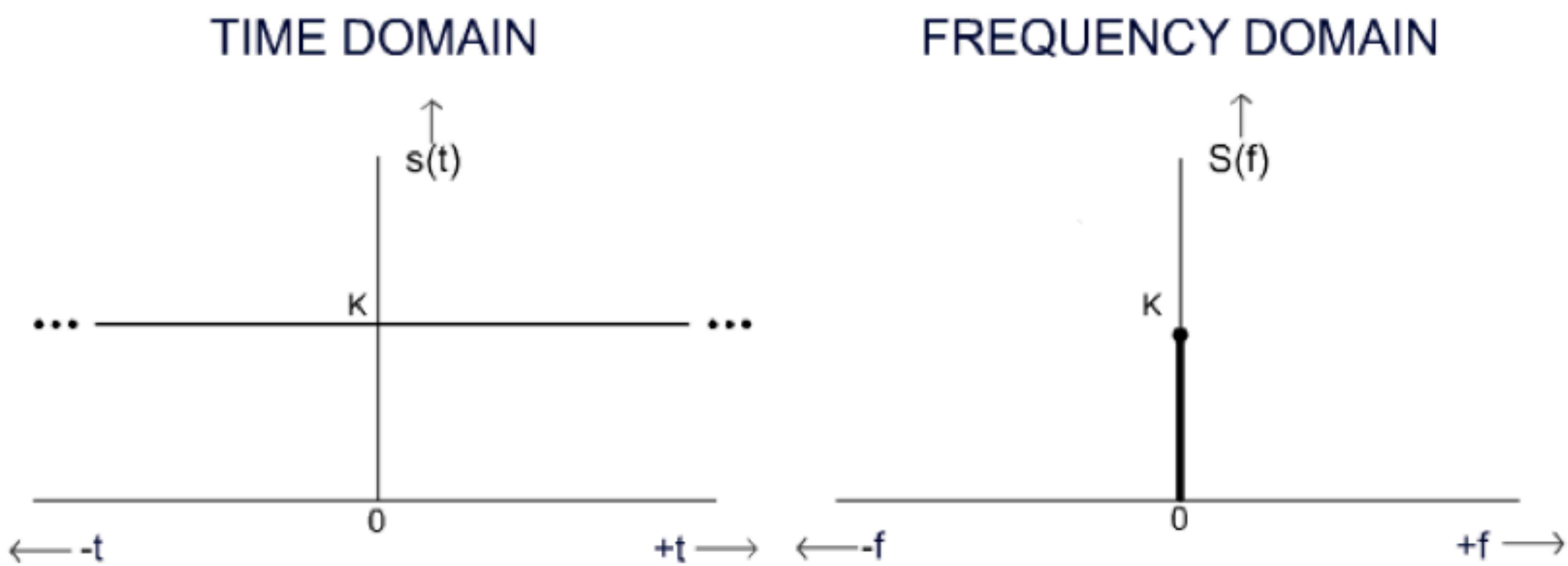




$$s(t) = 2Af_0 \frac{\sin(2\pi f_0 t)}{(2\pi f_0 t)}$$

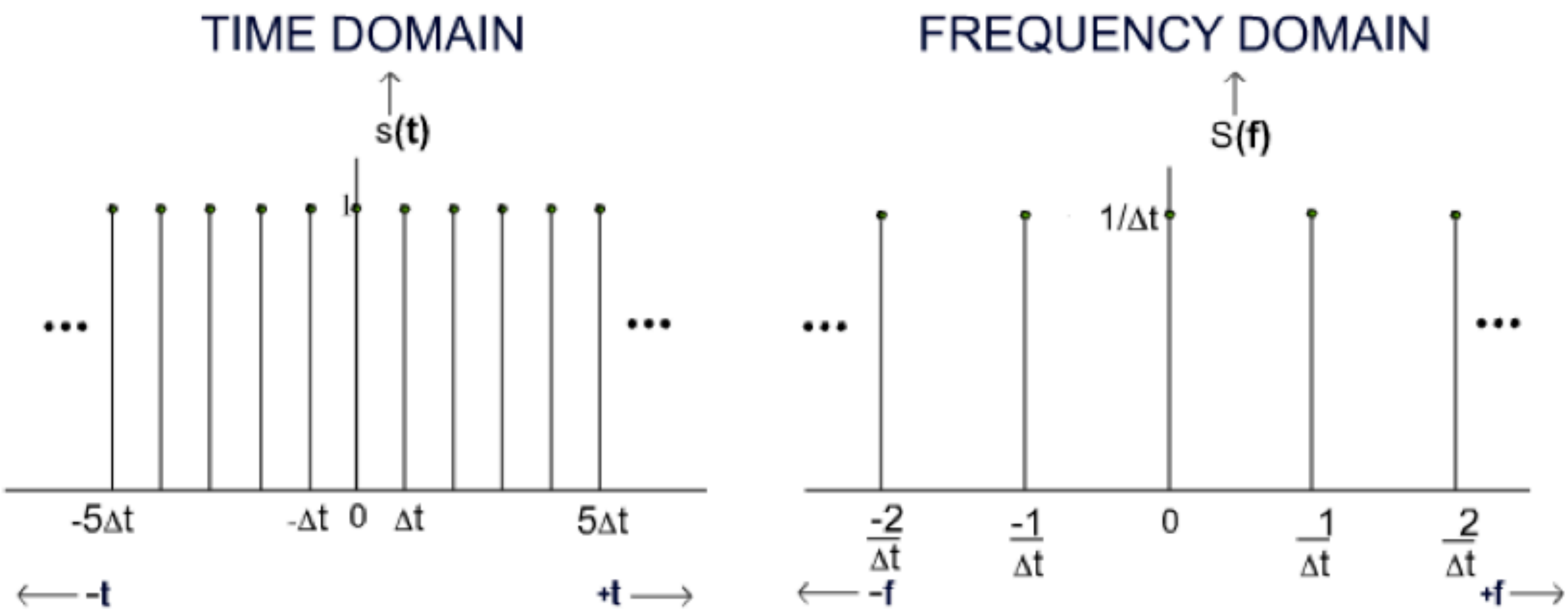
$$S(f) = \begin{cases} A & |f| < f_0 \\ A/2 & |f| = f_0 \\ 0 & |f| > f_0 \end{cases}$$

Common signals and spectra



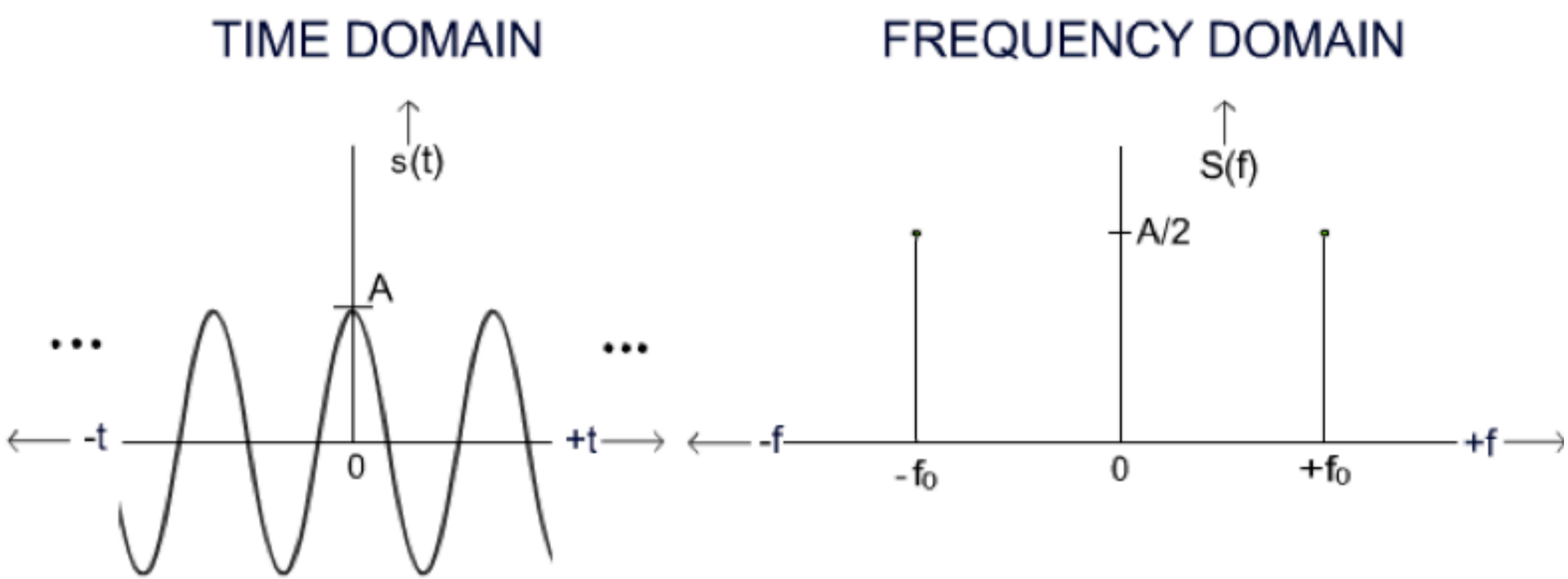
$s(t) = K \longleftrightarrow S(f) = K\,\delta(f)$

Common signals and spectra



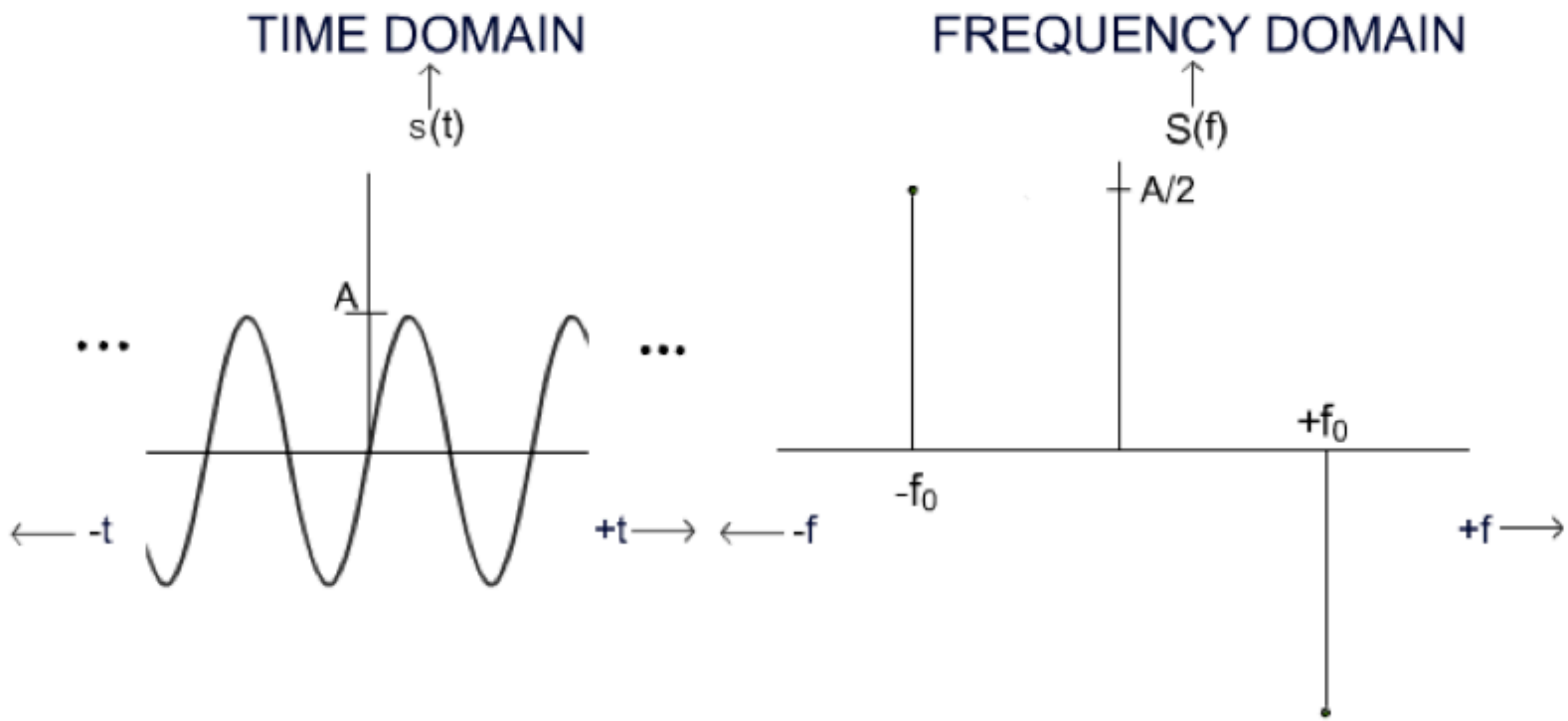
$$s(t) = \sum_{k=-\infty}^{\infty} \delta(t-k\Delta t) \longleftrightarrow S(f) = 1/\Delta t \sum_{k=-\infty}^{\infty} \delta(f-k/\Delta t)$$

Common signals and spectra



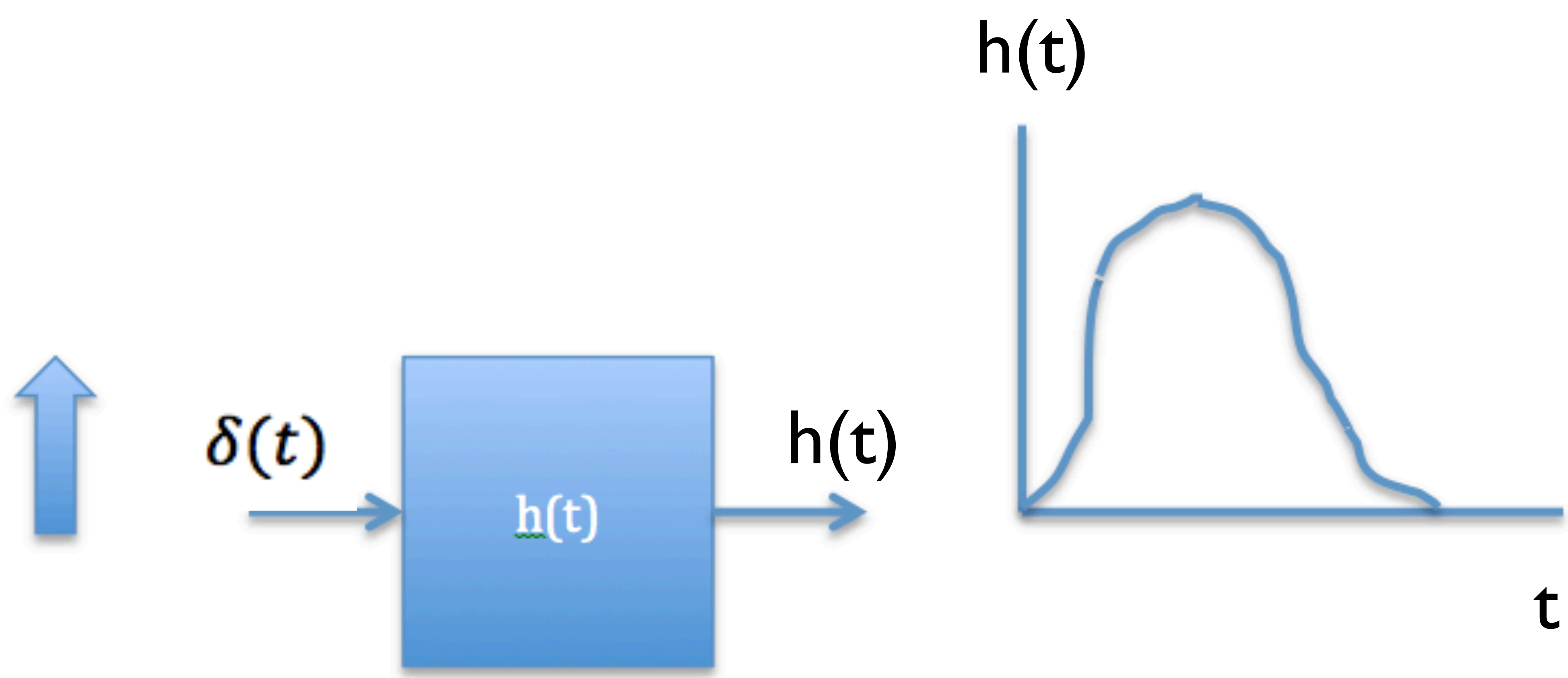
$$s(t) = A\cos(2\pi f_0 t) \longleftrightarrow S(f) = A/2\ \delta(f - f_0) + A/2\ \delta(f + f_0)$$

Common signals and spectra

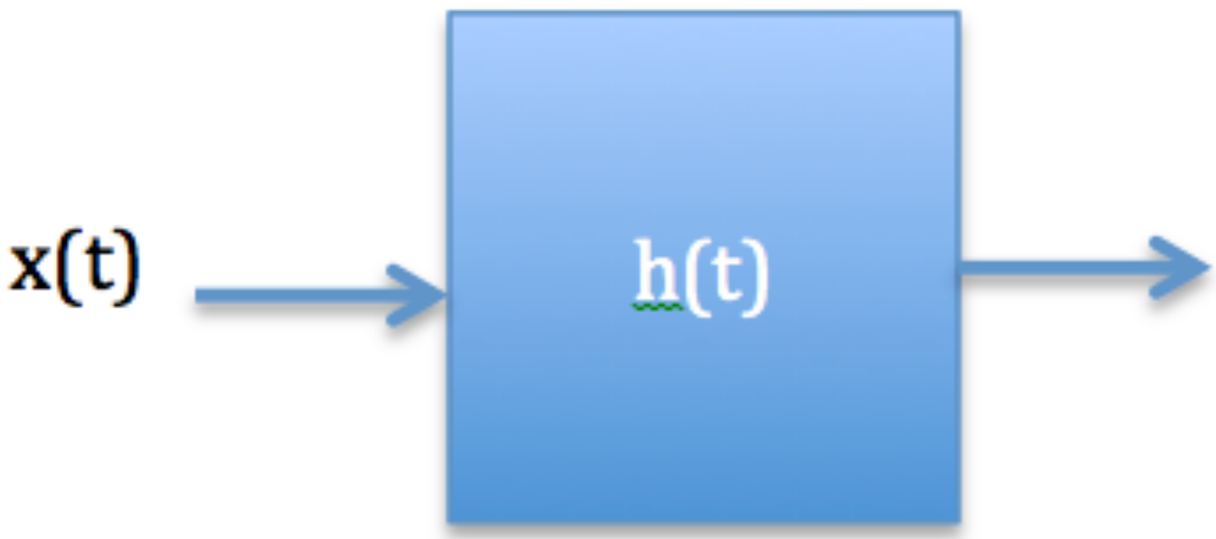


$$s(t) = A\sin(2\pi f_0 t) \longleftrightarrow S(f) = -i\,A/2\;\delta(f - f_0) + i\,A/2\;\delta(f + f_0)$$

Linear time invariant systems



Linear time invariant systems
time domain



$$y(t) = \int_{-\infty}^{\infty} x(\tau)h(t-\tau)\,d\tau$$

the convolution is also written as

y(t)=x(t)✪h(t)

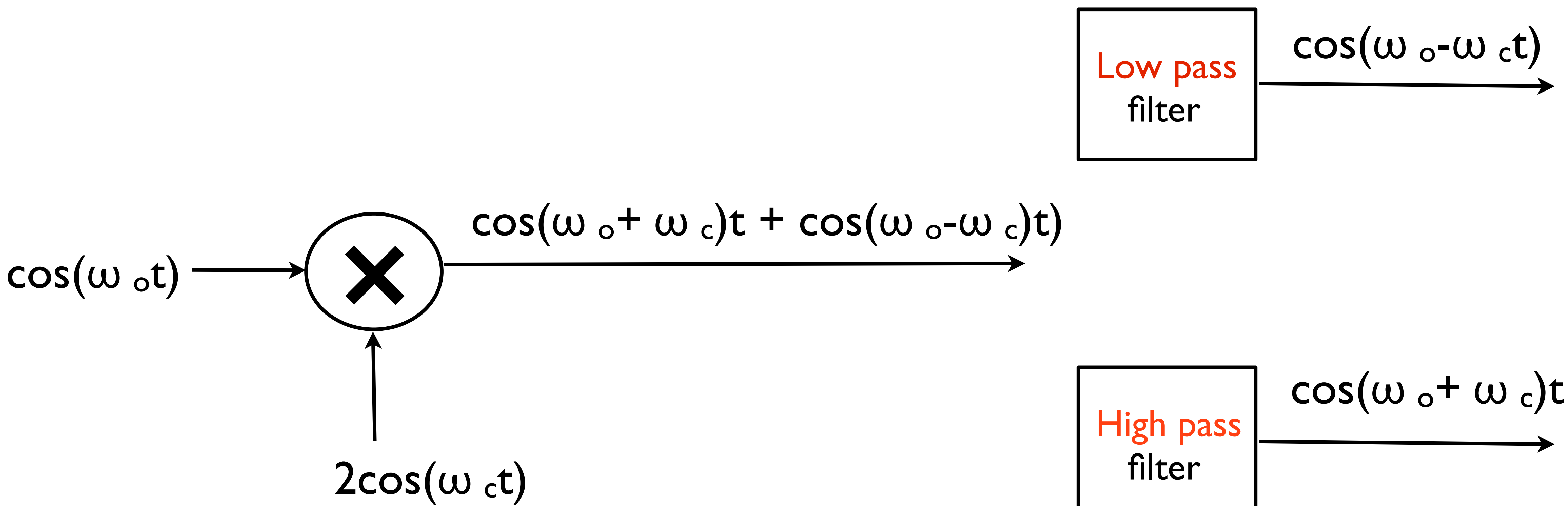
A very important feature of the convolution operation is that the convolution of a signal with a Dirac delta function reproduces de signal centered at the place of occurrence of the delta function

Linear time invariant systems in frequency domain

H(f) is the transfer function of the system

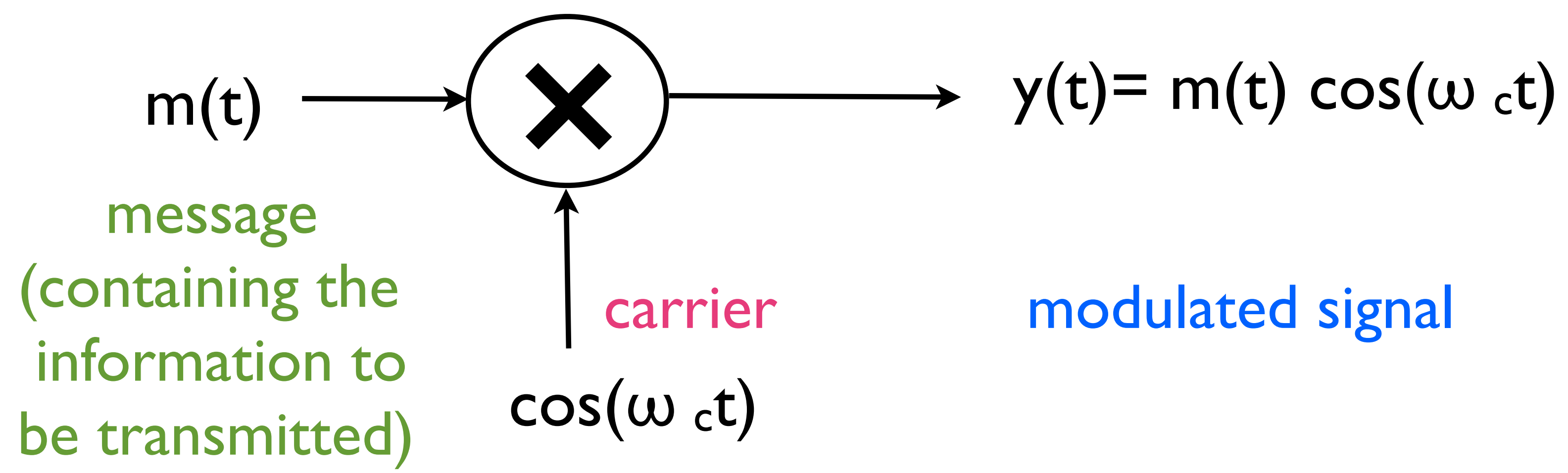
$$y(t) = \int_{-\infty}^{\infty} X(f)\, H(f) e^{-i\omega t}\, df$$

Basic frequency converter (frequency mixer)

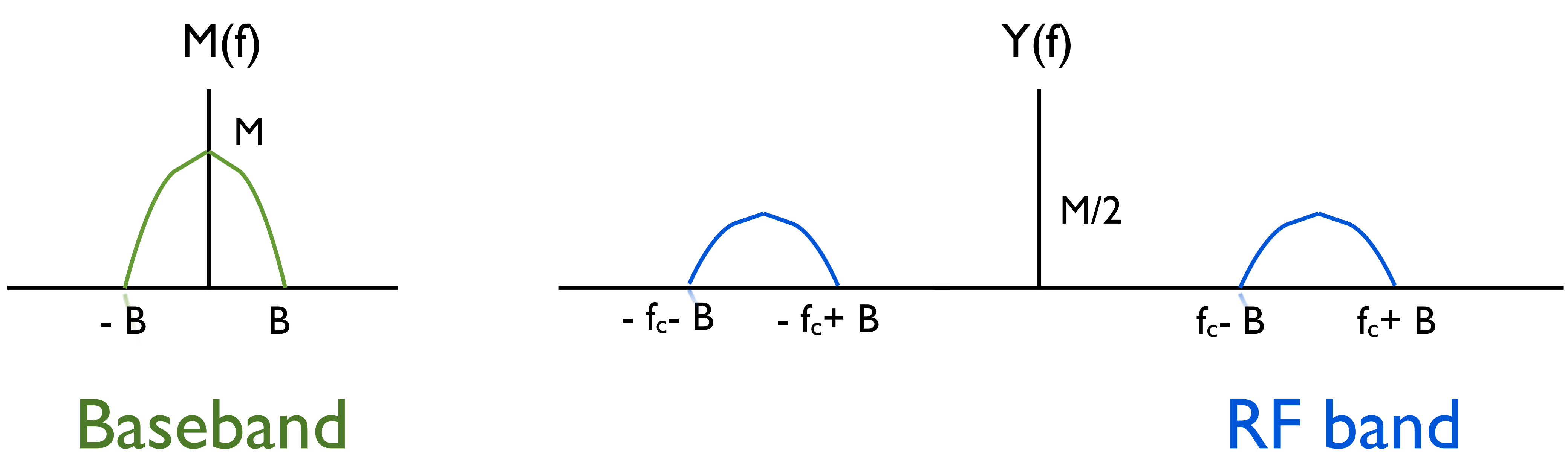


$$2\cos(\omega_o t)\cos(\omega\, t)=\cos(\omega_o+\omega_c)t+\cos(\omega_o-\omega_c)t$$

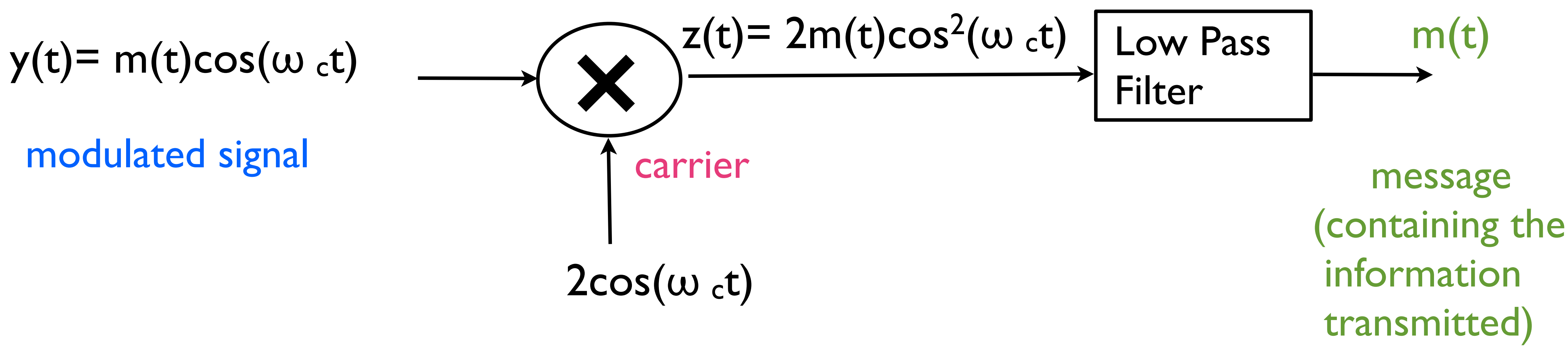
Amplitude modulation



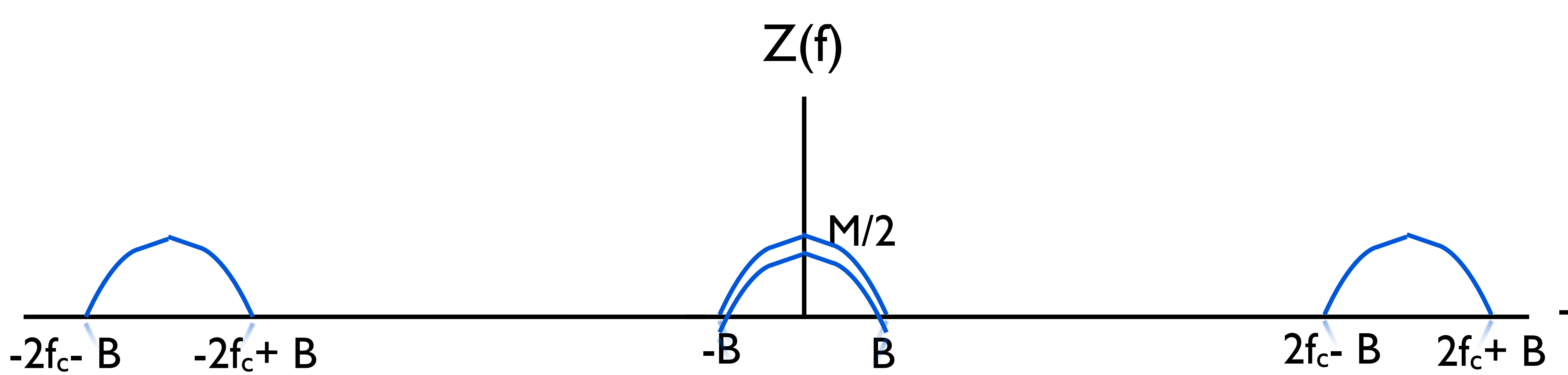
in the frequency domain:

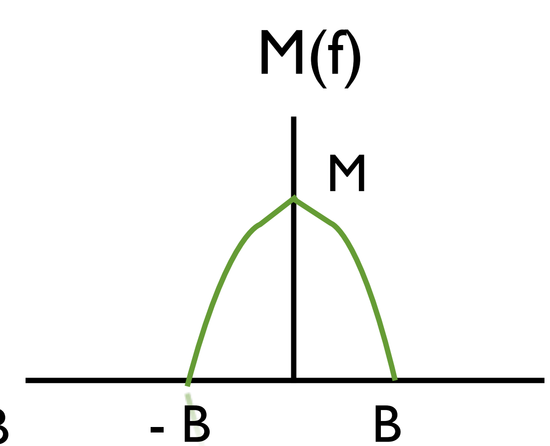


Amplitude demodulation

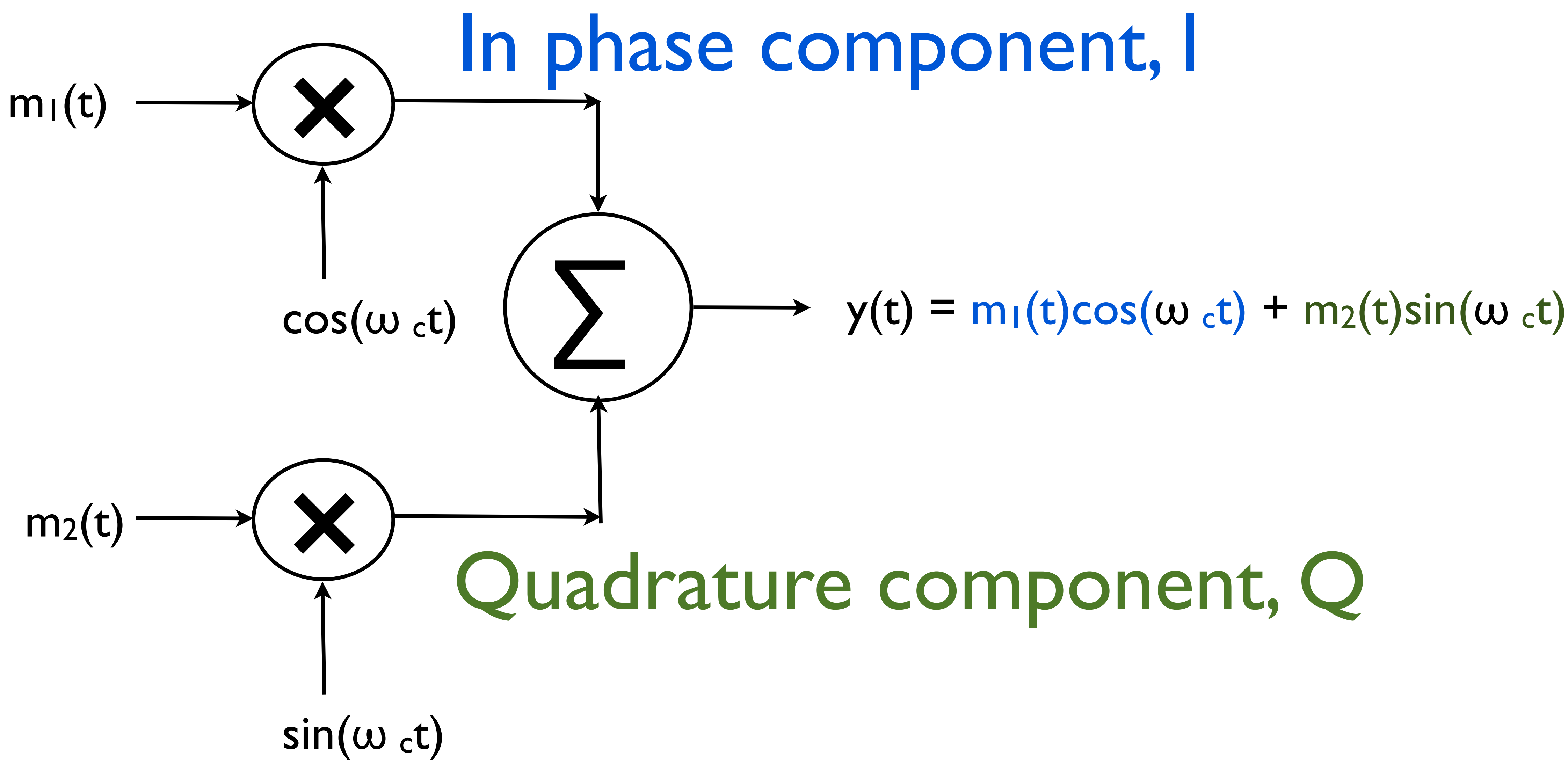


in the frequency domain:

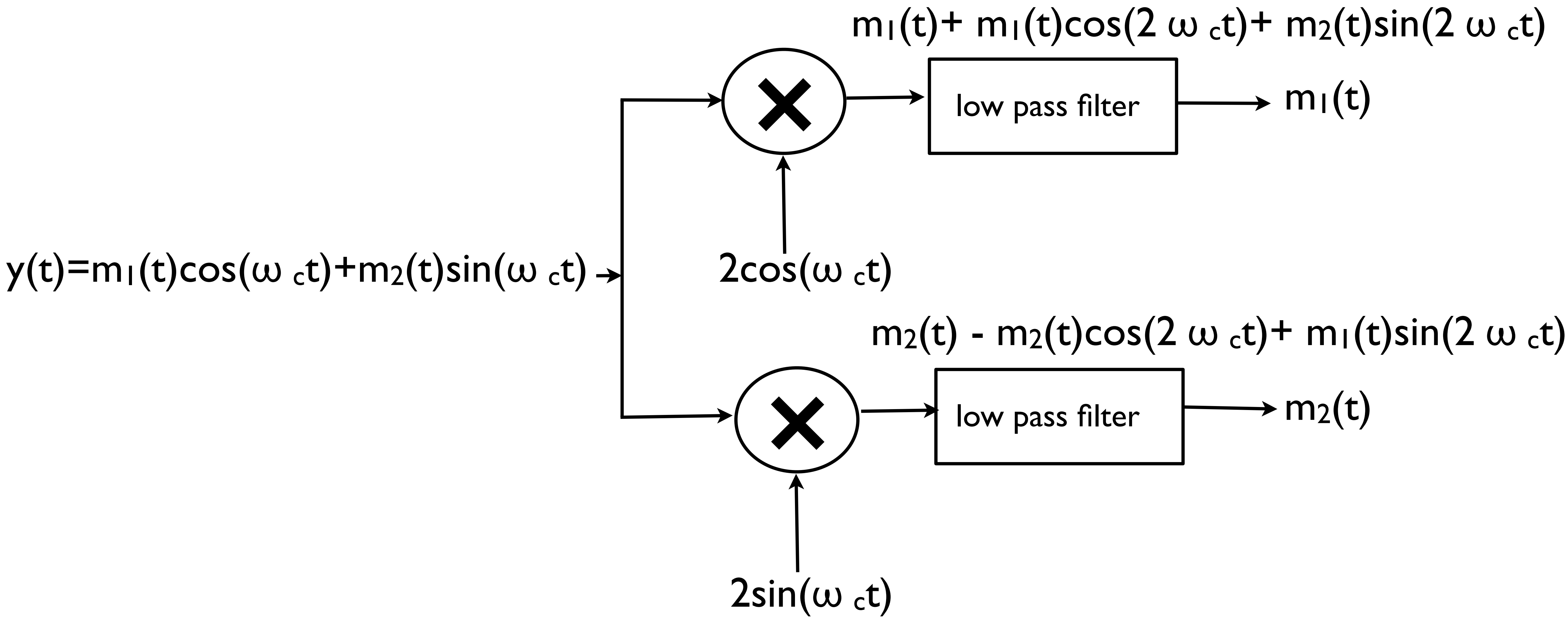




Orthogonality

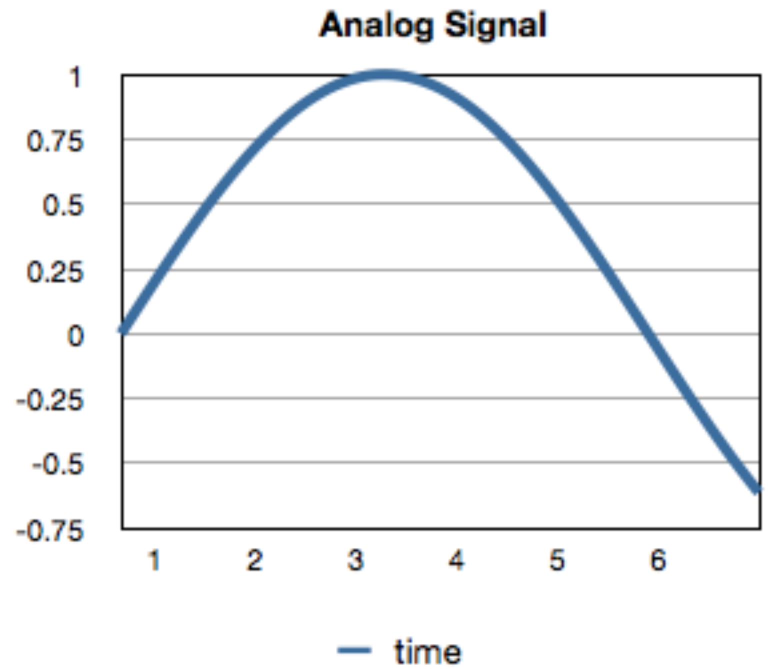


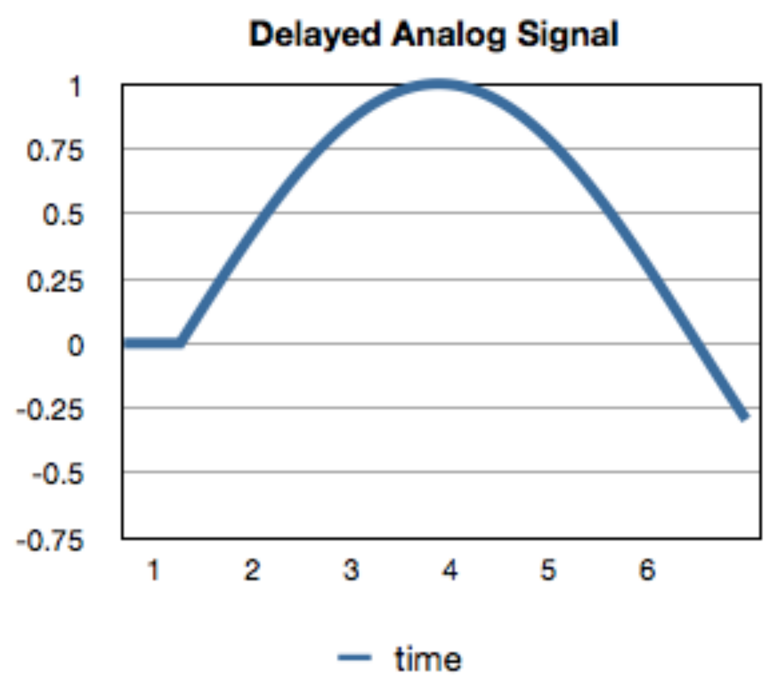
Orthogonality



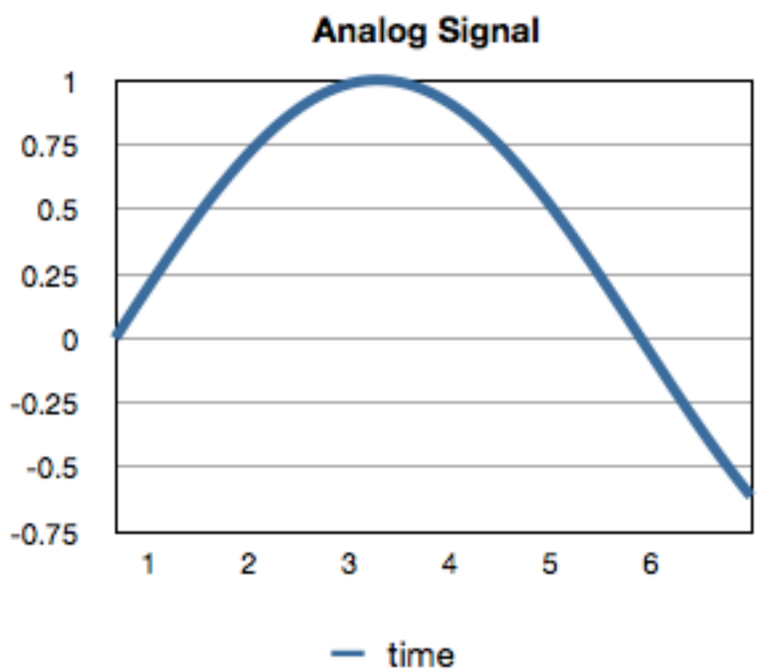
Communication System

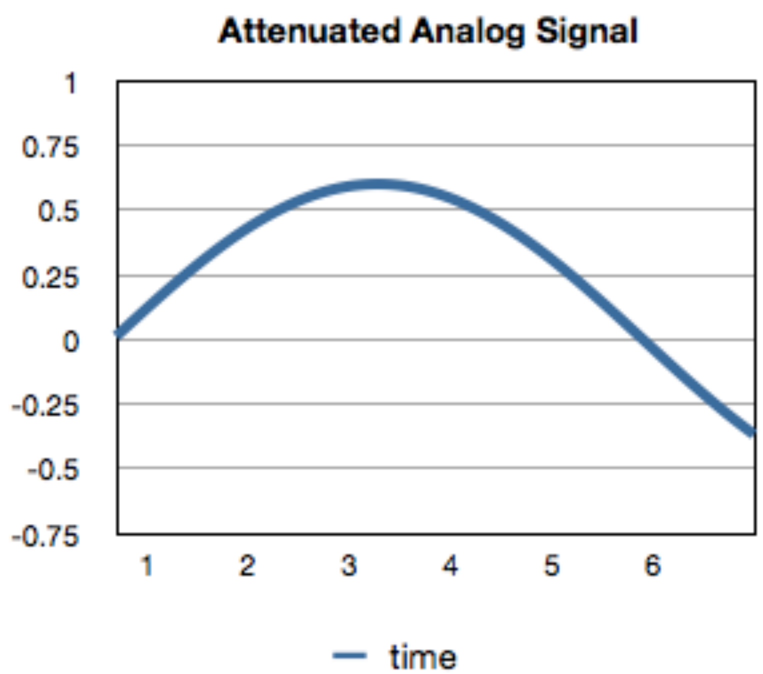
Signal Delay





Attenuation



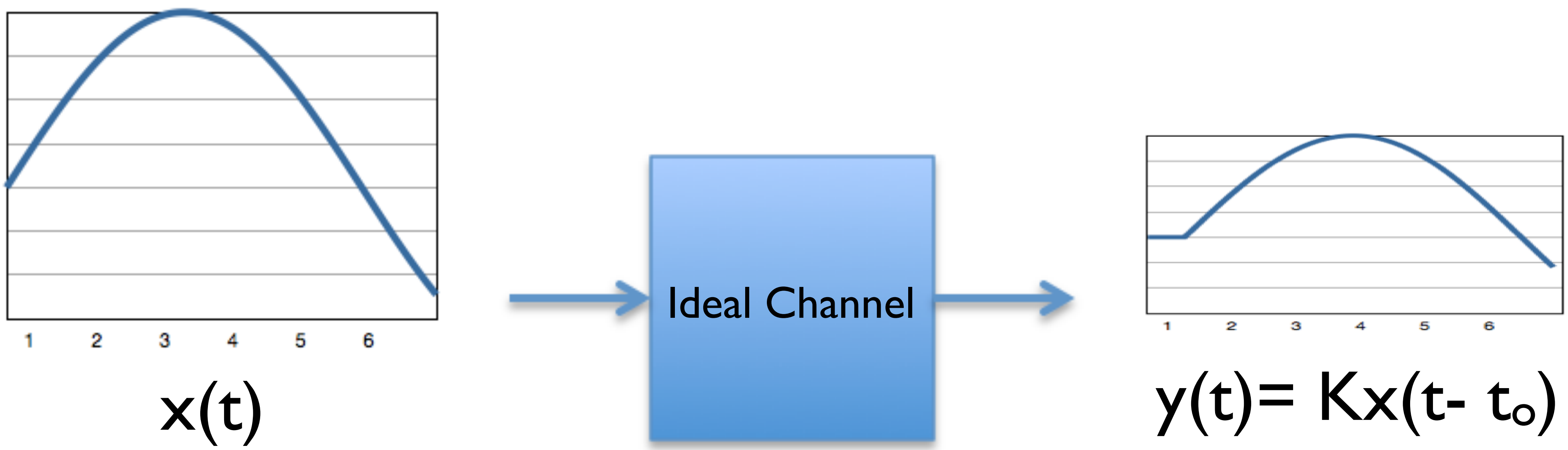


Transmitted Signal

Received Signal

Ideal Channel

An ideal communication channel will have an output y(t) that is an attenuated and delayed replica of its input x(t)



So its transfer must be constant and have phase linearly dependent on the frequency:

$$H(f) = K\, e^{-i2\pi t_o}$$

Amplifiers

Amplifiers are an example of an ideal channel over its frequency of operation:

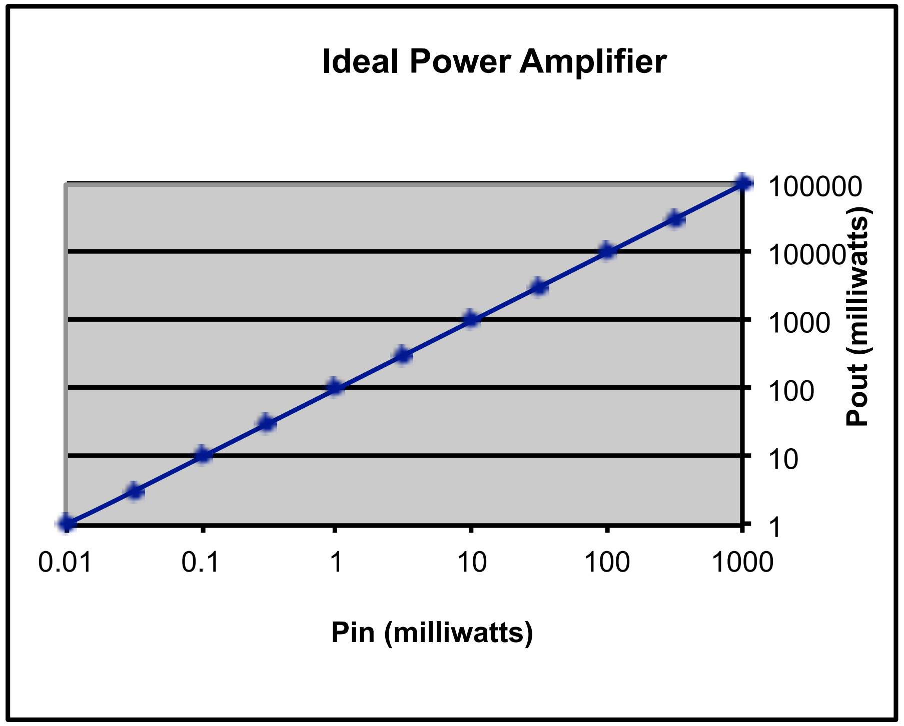
$$y(t)= A\, x(t)$$

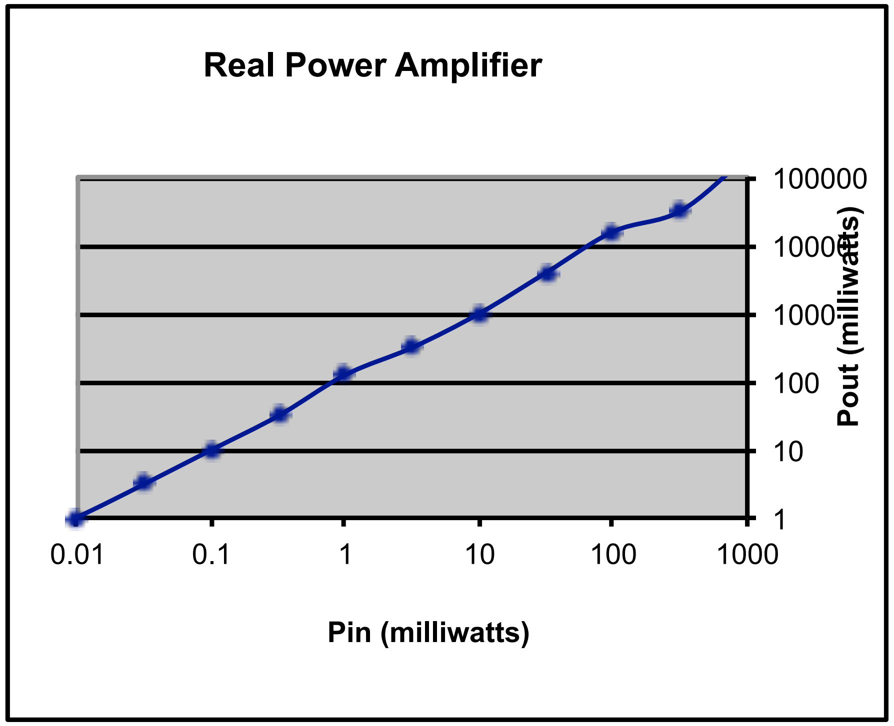
Real amplifiers will always have some amount of non linearities, so their output will also contain higher order terms:

$$y(t)= Ax(t) + Bx^2(t)+Cx3(t)+.....(A>>B>>C)$$

So besides amplification, a real amplifier will also act as a frequency converter and the output will include terms corresponding to the sum, the difference and the multiples of the input frequencies.

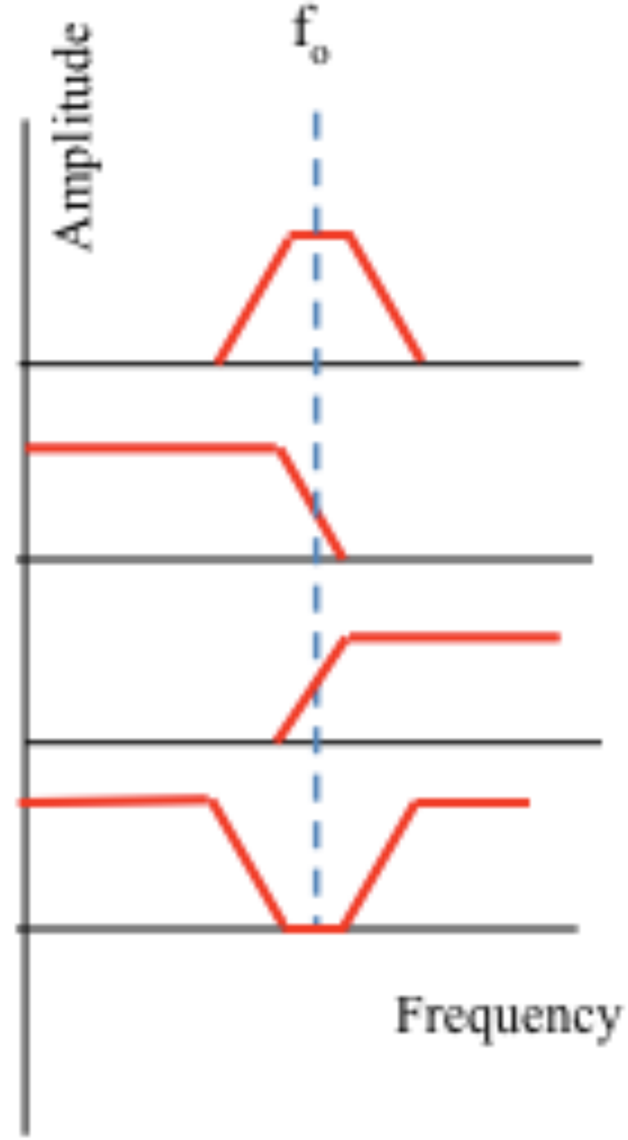
Amplifiers


Ideal Power Amplifier
100000
10000
1000
100
10
1
Pout (milliwatts)
0.01
0.1
1
10
100
1000
Pin (milliwatts)

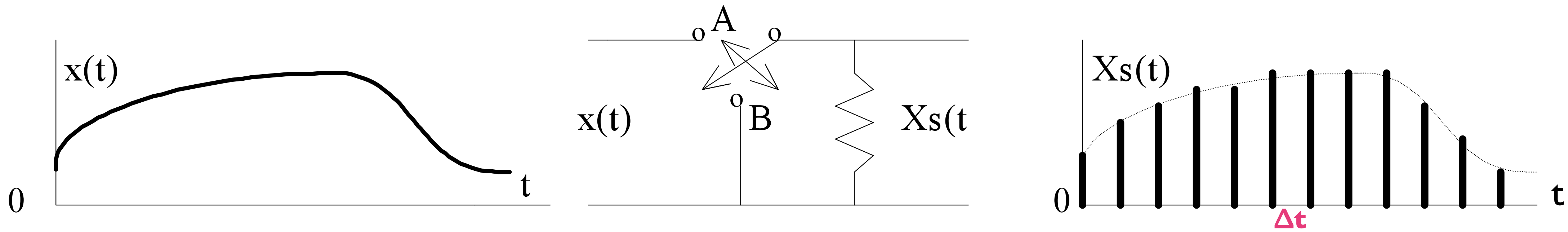

Real Power Amplifier
100000
10000
1000
100
10
1
Pout (milliwatts)
0.01
0.1
1
10
100
1000
Pin (milliwatts)

Filter Types

- Bandpass
- Lowpass
- High Pass
- Bandstop



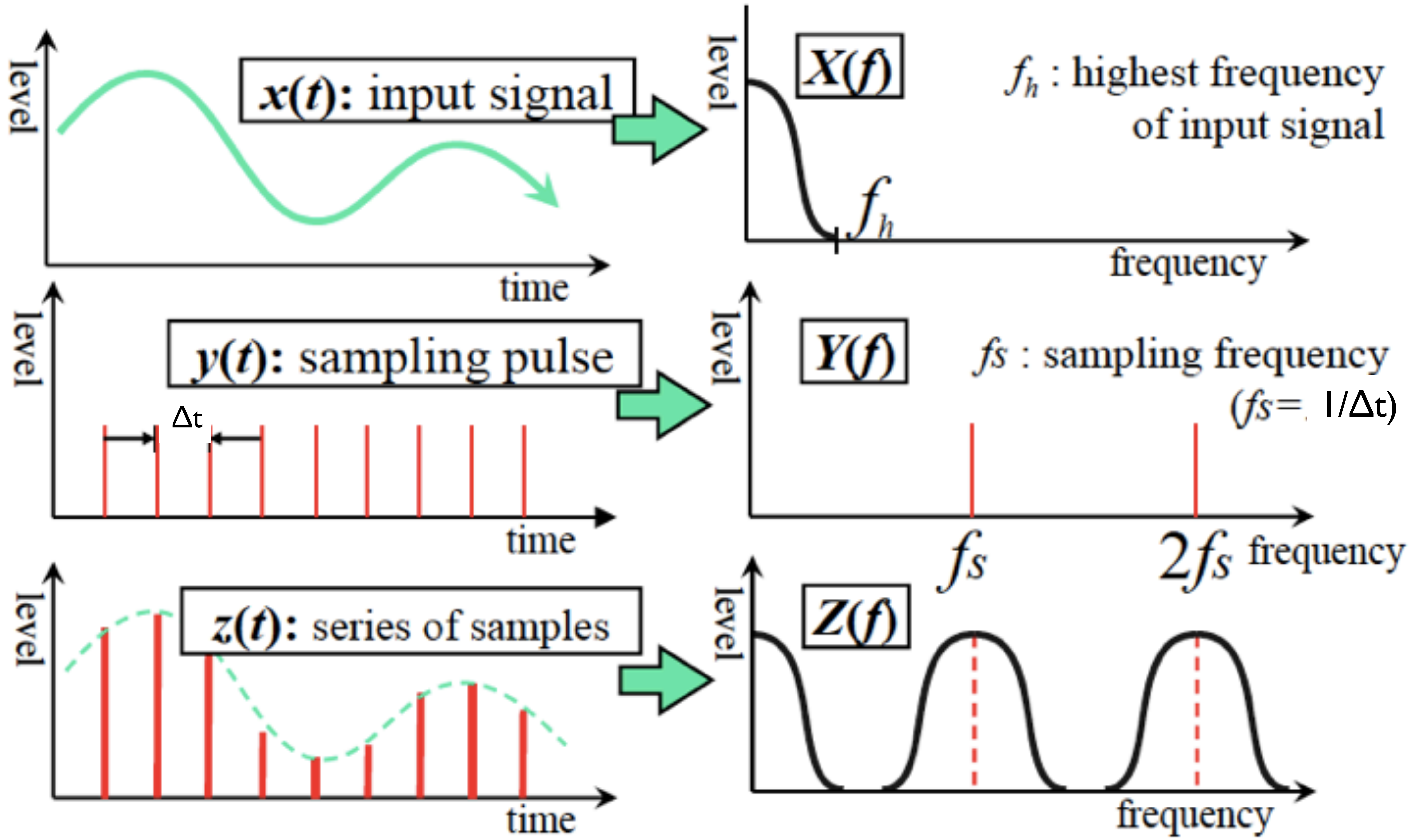
Sampling



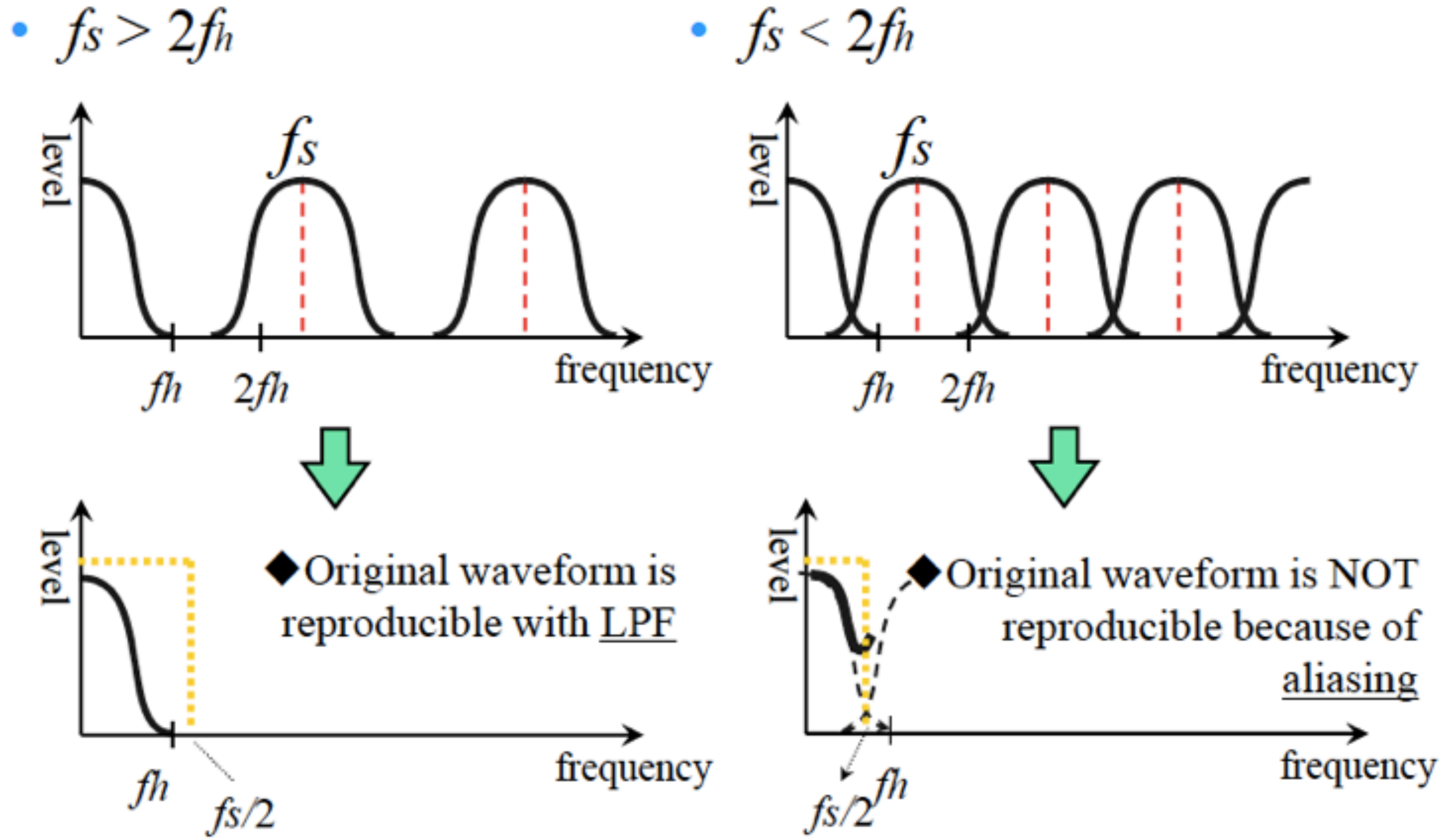
Analog Signal Sampling Circuit Sampled Signal

The sampling frequency f_s must be at least twice the highest frequency f_h present in the analog signal.
The original signal can be recovered from its samples by means of a low pass filter with cutoff frequency f_h. This is called an interpolation filter.
Sampling implies multiplication of the signal by a train of impulses equally spaced every $\Delta t = 1/f_s$

Sampling


level
x(t): input signal
time
level
X(f)
f_h : highest frequency
of input signal
f_h
frequency
level
y(t): sampling pulse
Δt
time
level
Y(f)
fs : sampling frequency
(fs= 1/Δt)
fs
2fs
frequency
level
z(t): series of samples
time
level
Z(f)
frequency

Aliasing and interpolation filter



Digital conversion

The sampled signal is still analog because the value of each sample spans a range of continuos values. To obtain a digital signal, we have to limit the infinite number of possible values of the signal to a specific set of predefined values in what is known as the quantization process. This process entails a quantization error similar to the error caused when we round off a number. The quantization error cannot be recovered, but we can make it as small as required by increasing the number of values that a sample is allowed to adopt.

Once the sampled signal is quantized it can be coded to convert to a truly digital signal.

This is normally done with an ADC (Analog to Digital Converter).

The recovery of the original signal is done by a DAC.

Why Digital?

Noise does not accumulate when you have a chain of devices like it happens in an analog system: CD Versus Vinyl, VHS Vs DVD.

The same goes for the storing of the information.

Detection of a digital signal is easier than an analog signal, so digital signal can have greater range.

Digital signals can use less bandwidth, as exemplified by the "***digital dividend***" currently being harnessed in many countries.

Digital circuits are easier to design and can achieve greater integration levels than analog circuits.

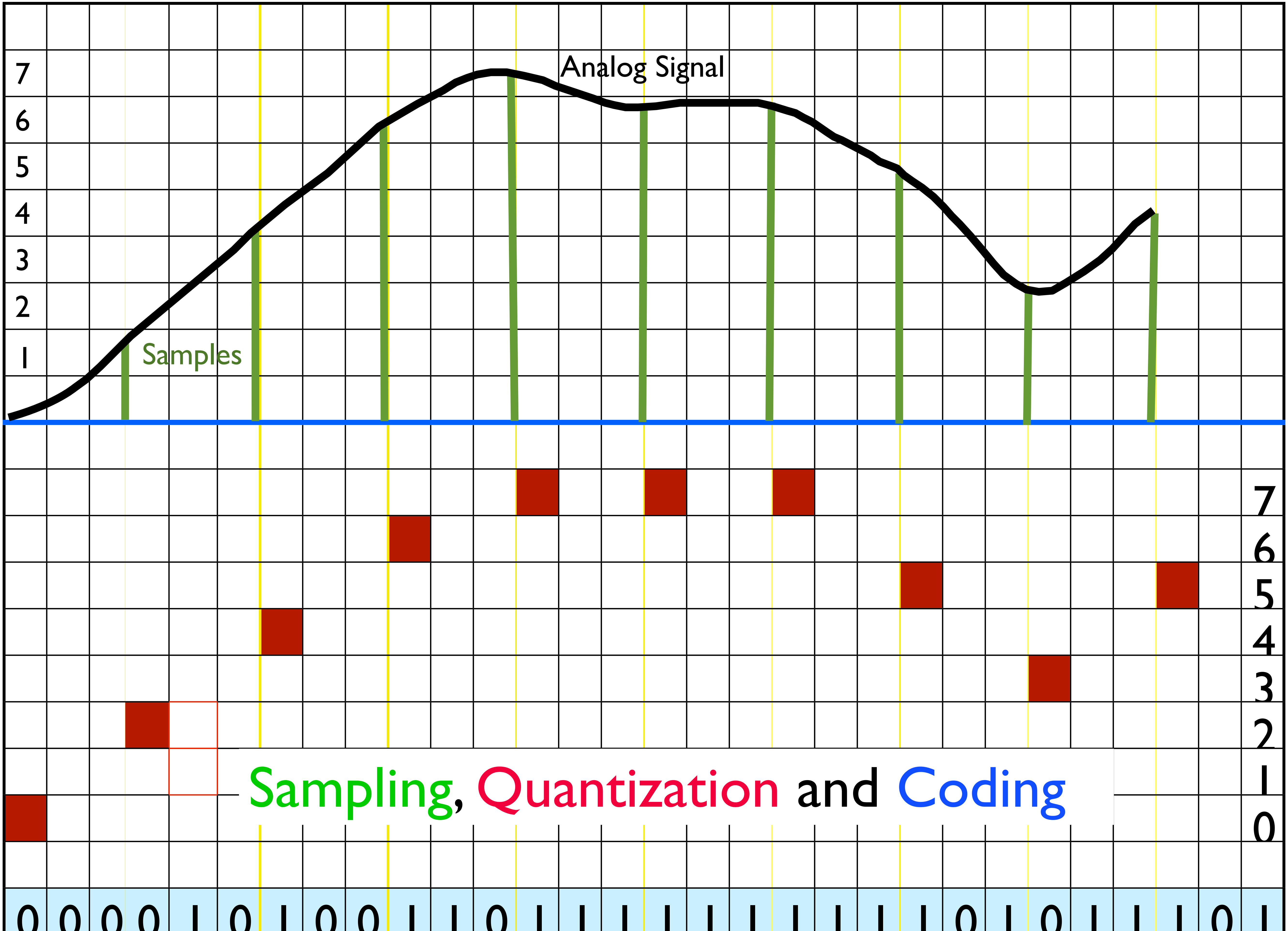
Digital signals can be encoded in ways that allow the recover from transmission errors, albeit at the expense of throughput.

Image Sampling

Normal, 72pixels/inch

Sampled Image, 10 pixels/inch

Sampling, Quantization and Coding



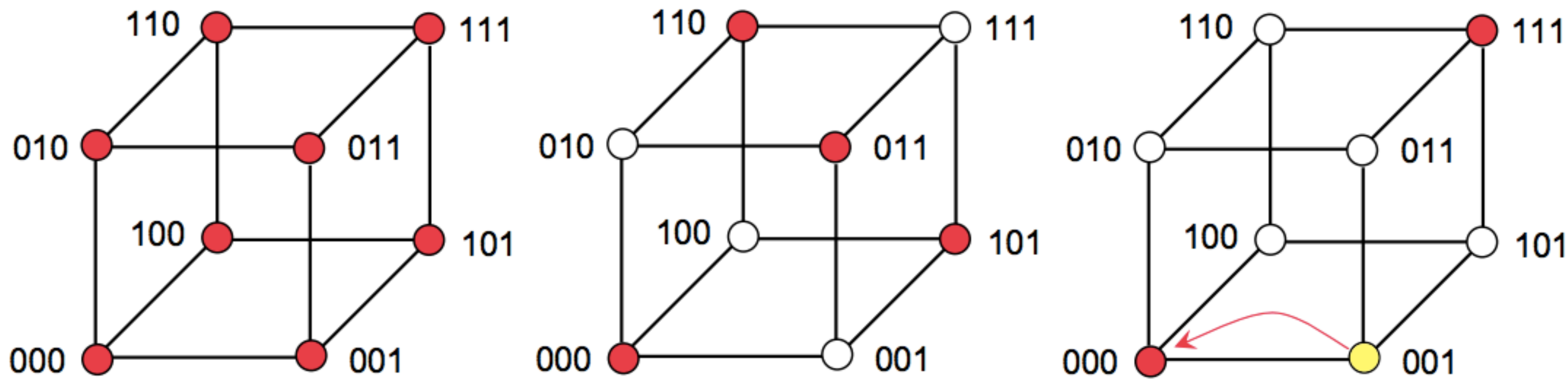
Recovering from errors

Two strategies to recover from transmission errors:

- Automatic Repeat-Request (ARQ)
- Line coding
- Both make use of REDUNDANCY to overcome transmission errors
- Forward Error Correction (FEC) adds a certain amount of redundant bits to the information bits prior to transmission
- Parity check is the simplest form of error detection

Example of Coding

q=2, n=3 → $\mathbf{x} = [x_0 \; x_1 \; x_2]$ ● code word, i.e. $\mathbf{x} \in \Gamma$ ○ no code word $\mathbf{x} \notin \Gamma$



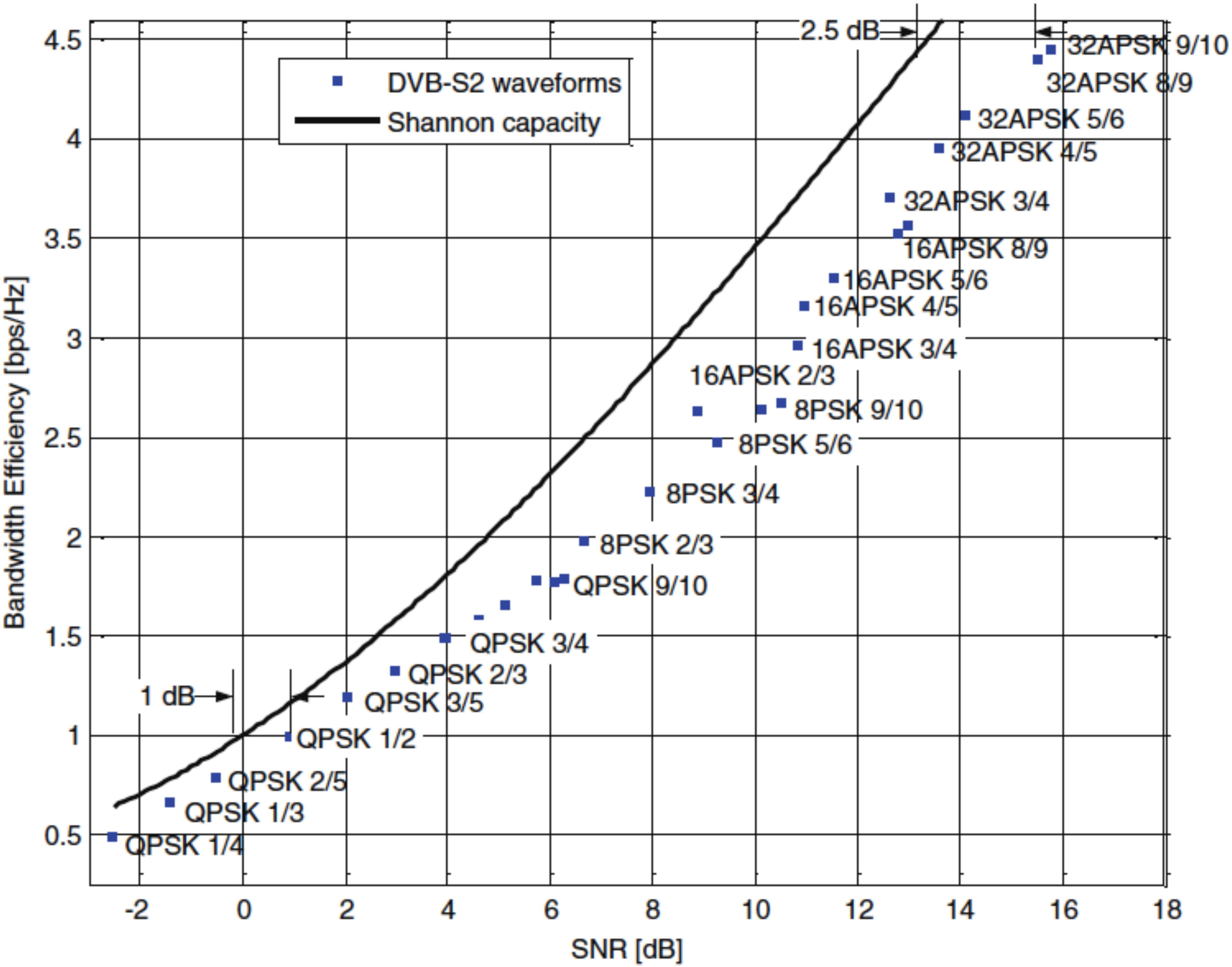
$d_{\min} = 1$ $d_{\min} = 2$ $d_{\min} = 3$

- Code rate $R_c = 1$
- No error correction
- No error detection

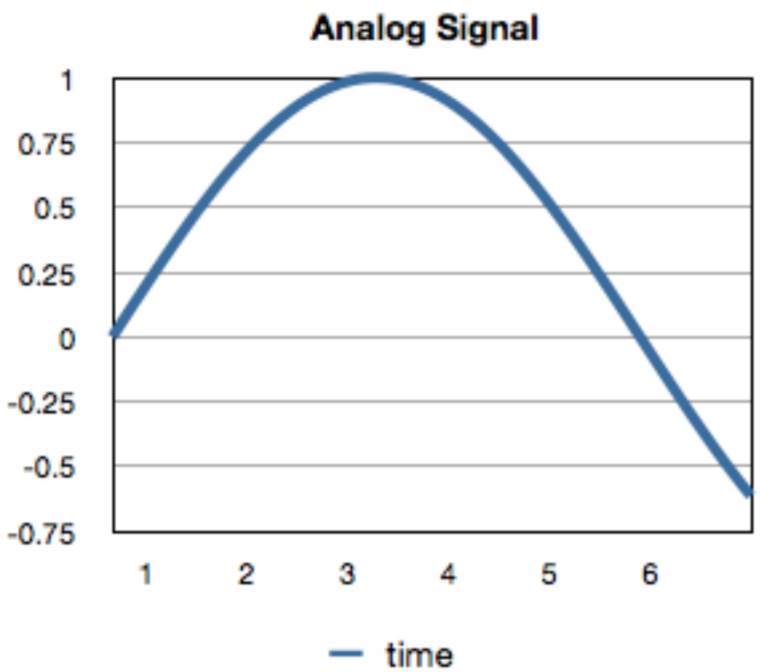
- Code rate $R_c = 2/3$
- No error correction
- Detection of single error

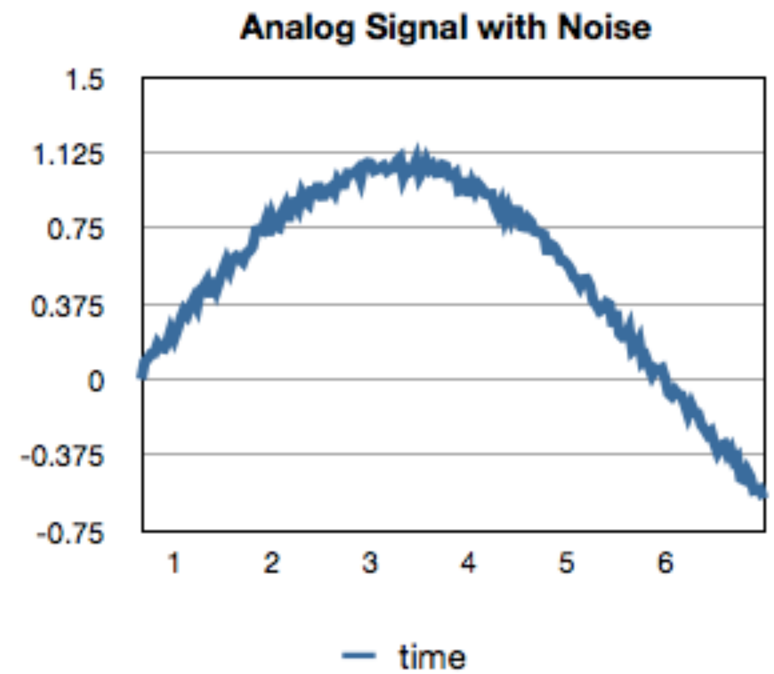
- Code rate $R_c = 1/3$
- Correction of single error
- Detection of 2 errors

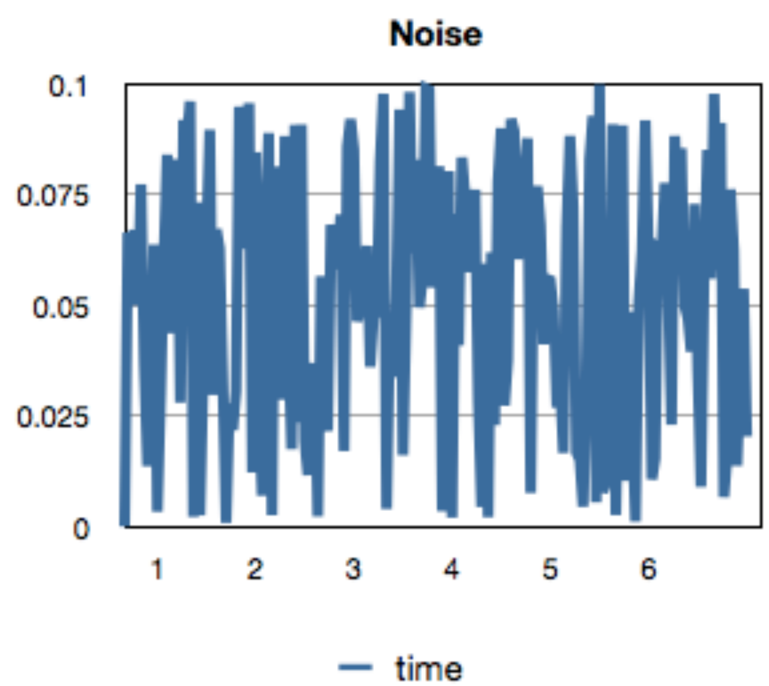
Modulation and coding examples



Noise in an analog Signal





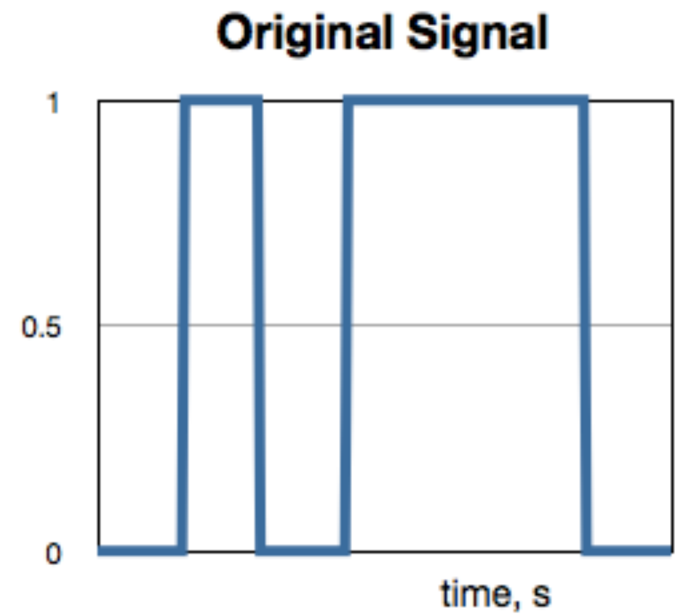


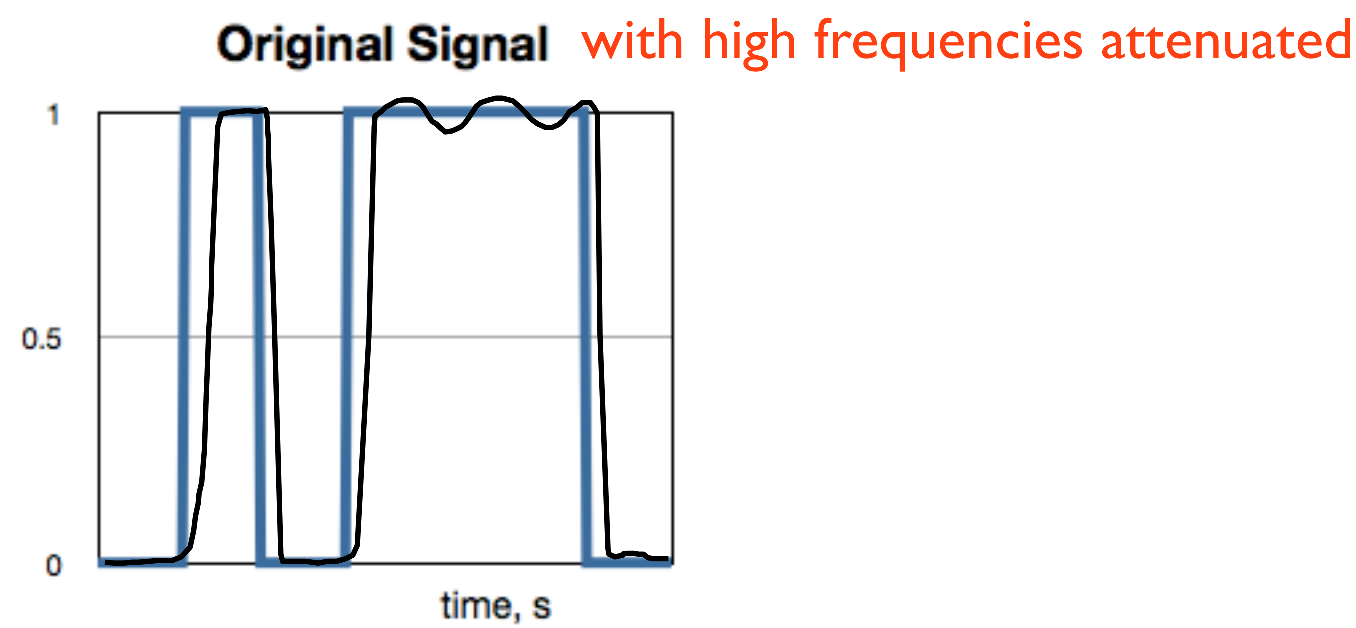
Electronic Noise

- Noise poses the ultimate limit to the range of a communications system
- Every component of the system introduces noise
- There are also external sources of noise, like atmospheric noise and man made noise
- Thermal noise power (always present) is frequency independent and is given (in watts) by kTB, where:

 k is Boltzmann constant, 1.38×10^{-23} J/K
 T is absolute temperature in kelvins (K)
 B is bandwidth in Hz
 At 26 °C (T= 273.4+26) the noise power in dBm in 1 MHz is:
 $-174 + 10 \log_{10}(B) = - 144$ dBm

Bandwidth Limitation





Interference

Any signal different from the one that our system is designed to receive that is captured by the receiver impairs the communication and is called interference.

Intra-channel interference originates in the same channel as our signal.

Co-channel interference is due to the imperfection of the filters that will let in signals from adjacent channels.

Information Measurement

$$I = \log_2 (1/Pe)$$

The information carried by a signal is expressed in bits and is proportional to the logarithm of the inverse of the probability of the occurrence of the corresponding event.
The more unlikely an event to happen, the more information its happening will carry.
Transmitting a message of an event that the receiver already knows carries no information.
The amount of information transmitted in one second is the ***capacity*** of the channel, expressed in bit/s.

Redundancy

Sending twice the same information is a waste of the system capacity that reduces the ***throughput***.

Nevertheless, if an error occurs, the redundancy can be used to overcome the error.

Every ***error correcting code*** must use some sort of redundancy.

Digital Channel Capacity

$$C = B \log_2 \{1+S/(N_o B)\}$$

Capacity, bit/s

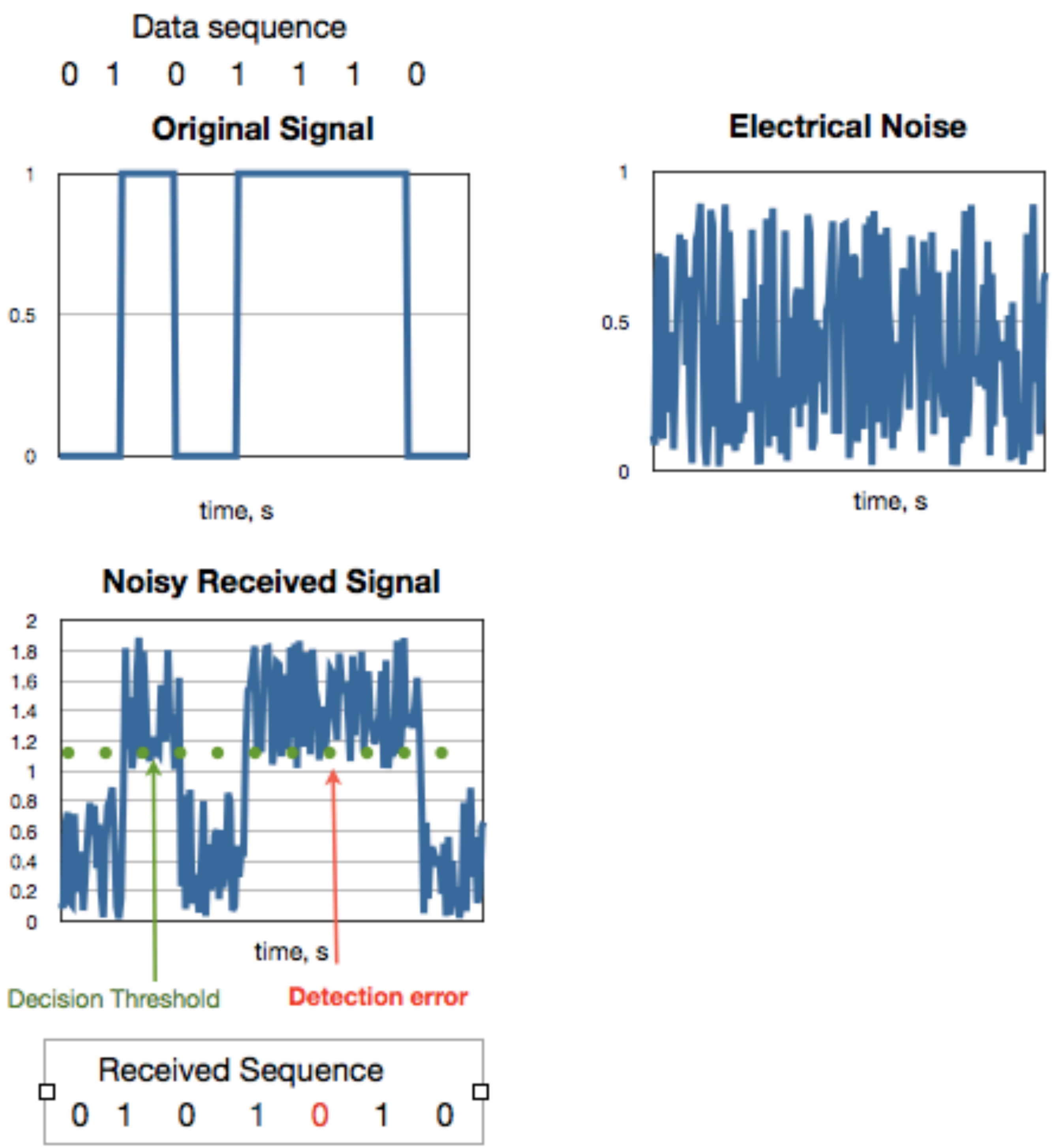
B, bandwidth, Hz

Signal power, W

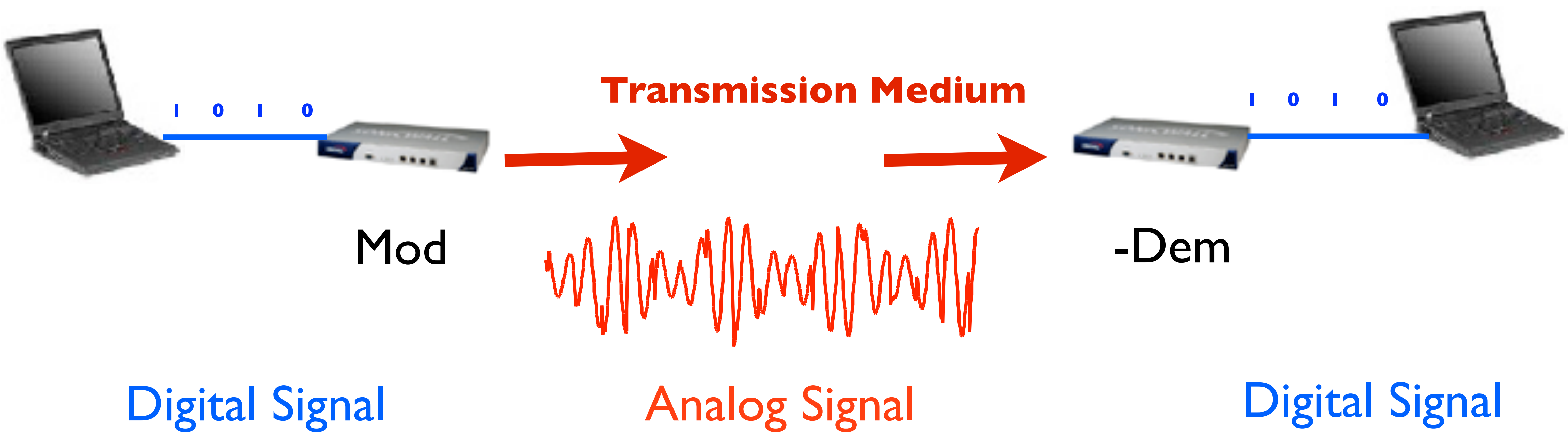
Noise Power density, W/Hz

The capacity, also called throughput is the number of bits transmitted in one second.
The spectral efficiency C/B is the number of bits per second transmitted in one Hz

Detection of a noisy signal

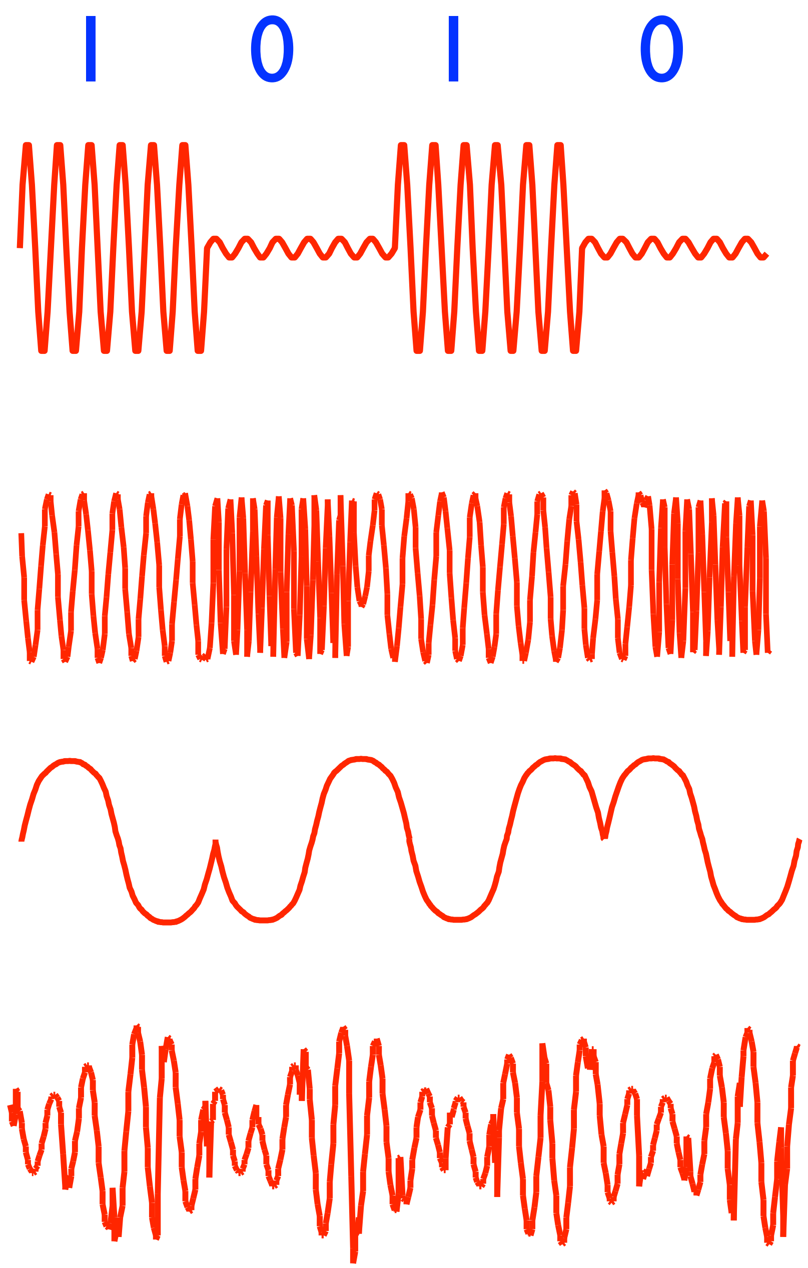


MoDem



1 0 1 0
Transmission Medium
1 0 1 0
Mod
-Dem
Digital Signal
Analog Signal
Digital Signal

Comparison of modulation techniques



Digital Sequence

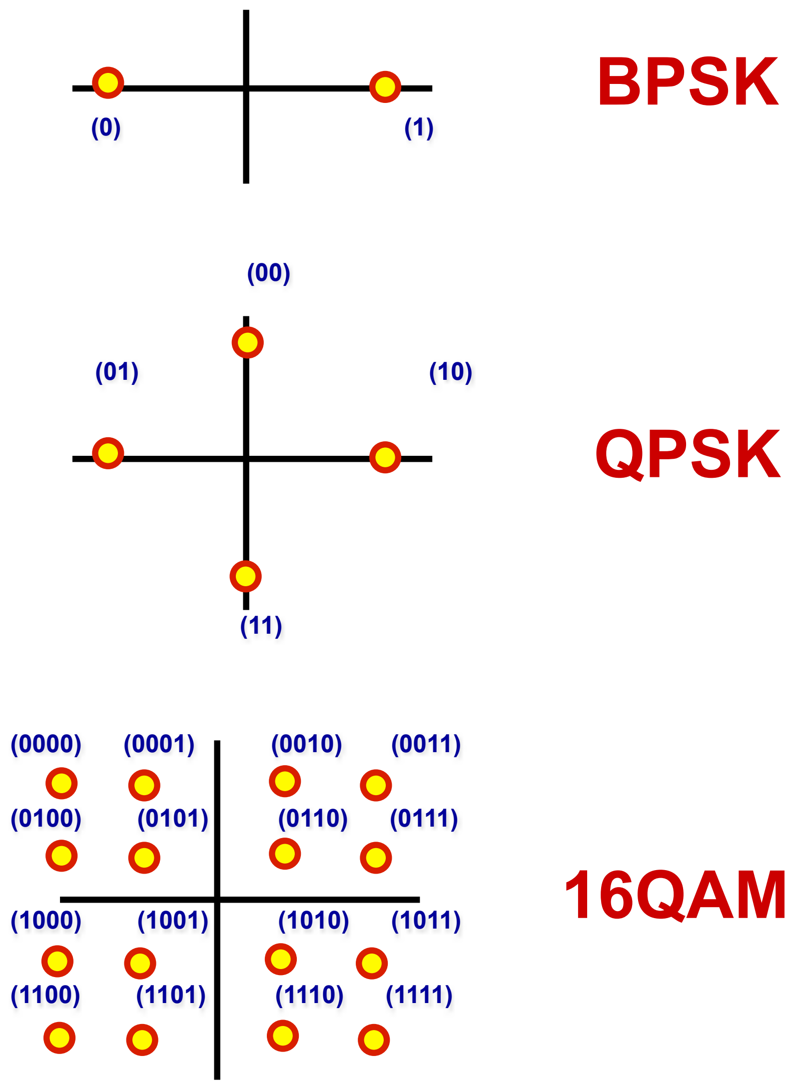
ASK modulation

FSK modulation

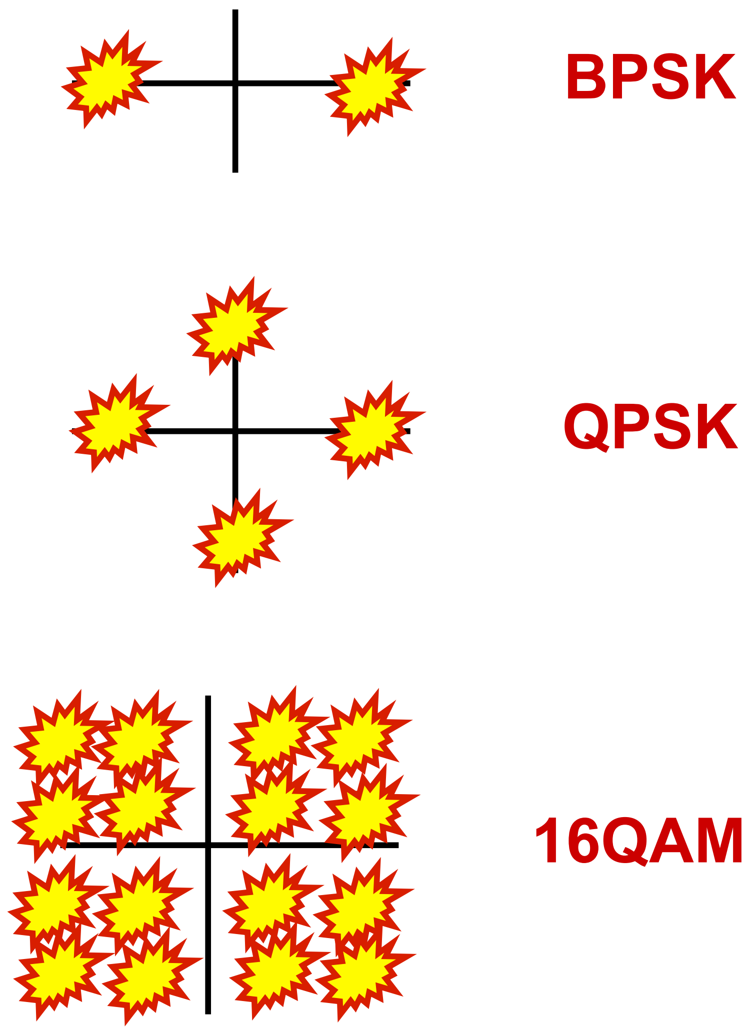
PSK modulation

QAM modulation, changes
both amplitude and phase

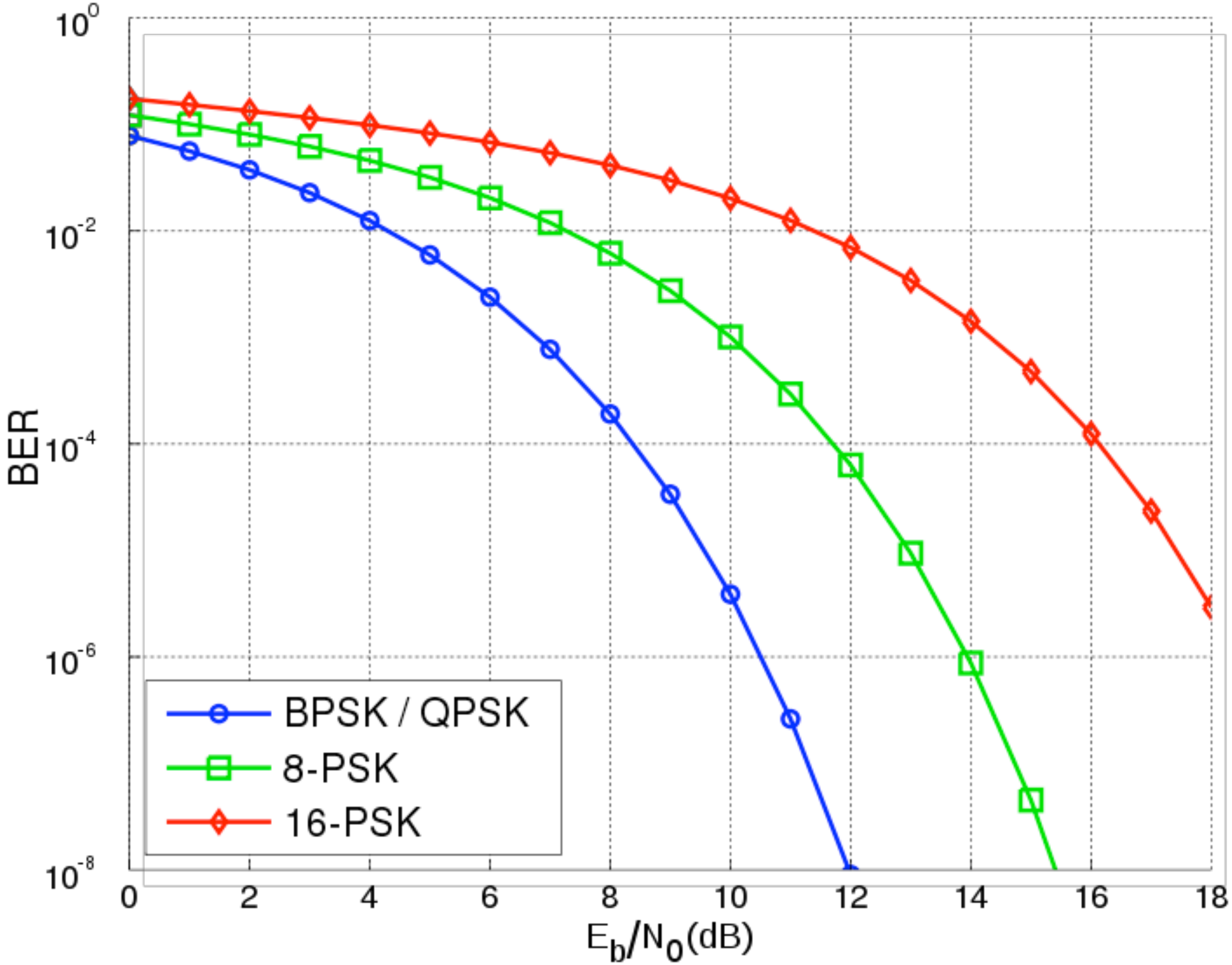
Binary Modulation Constellation



Effect of noise in the detection



BER Versus E_b/N_o



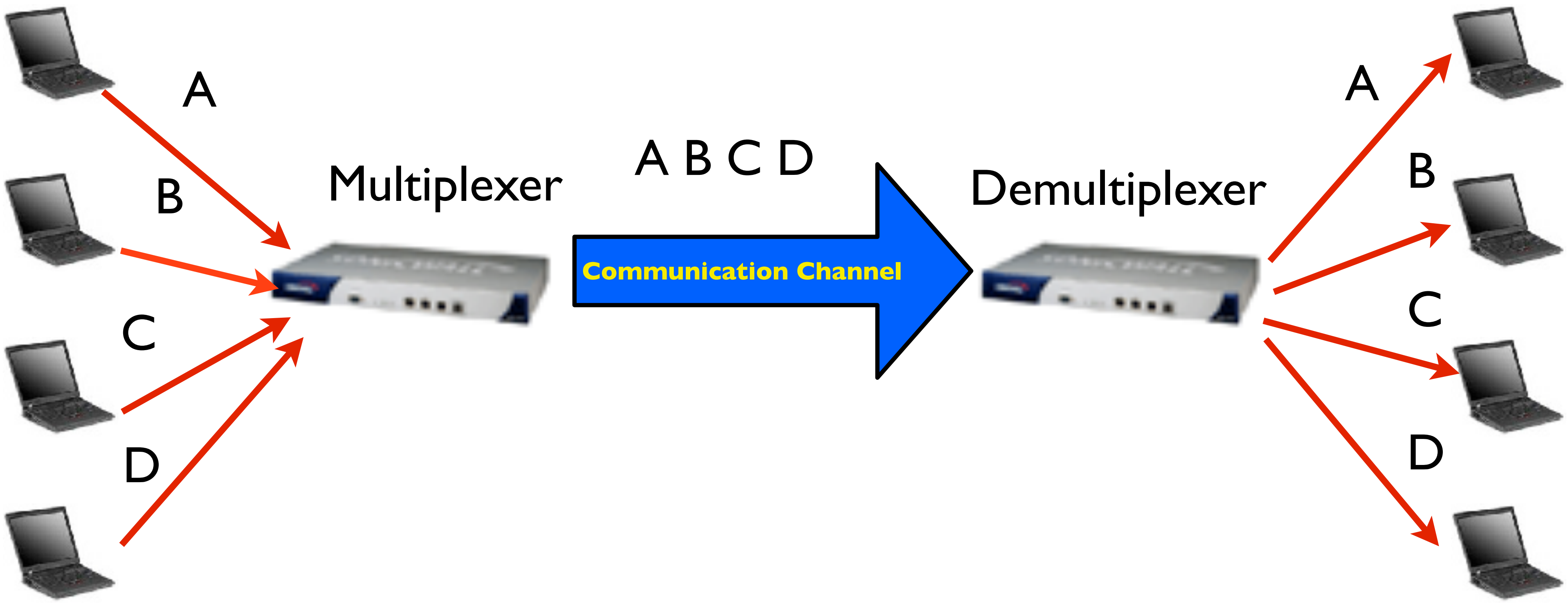
From Wikipedia: http://en.wikipedia.org/wiki/Bit_error_rate

Comparison of modulation types

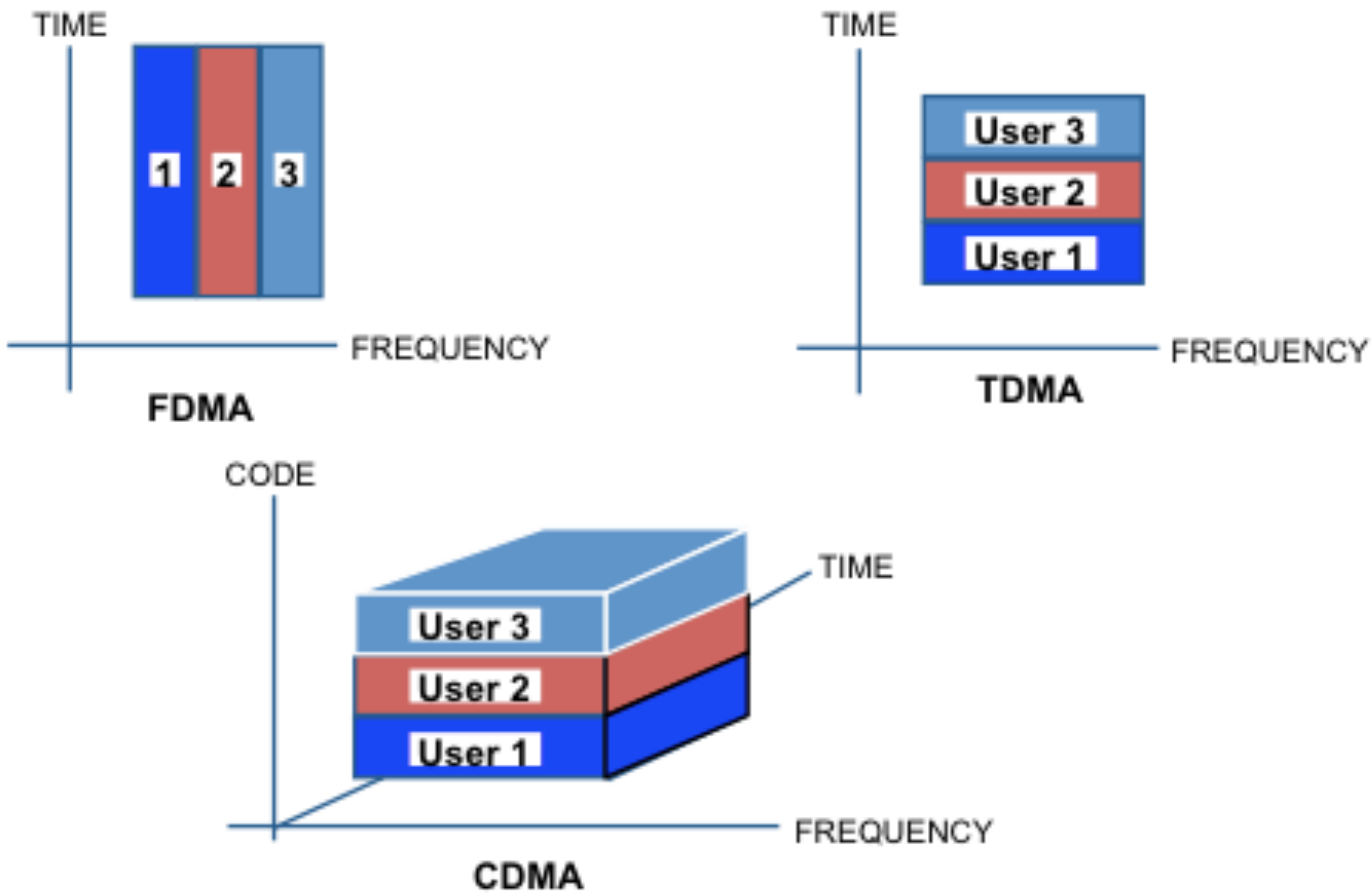
BER of 10^{-6}

Mod. Type	Bits/Symbol	Required E_b/N_o
16 PSK	4	18 dB
16 QAM	4	15 dB
8 PSK	3	14.5 dB
4 PSK	2	10.1 dB
4 QAM	2	10.1 dB
BFSK	1	13.5 dB
BPSK	1	10.5 dB

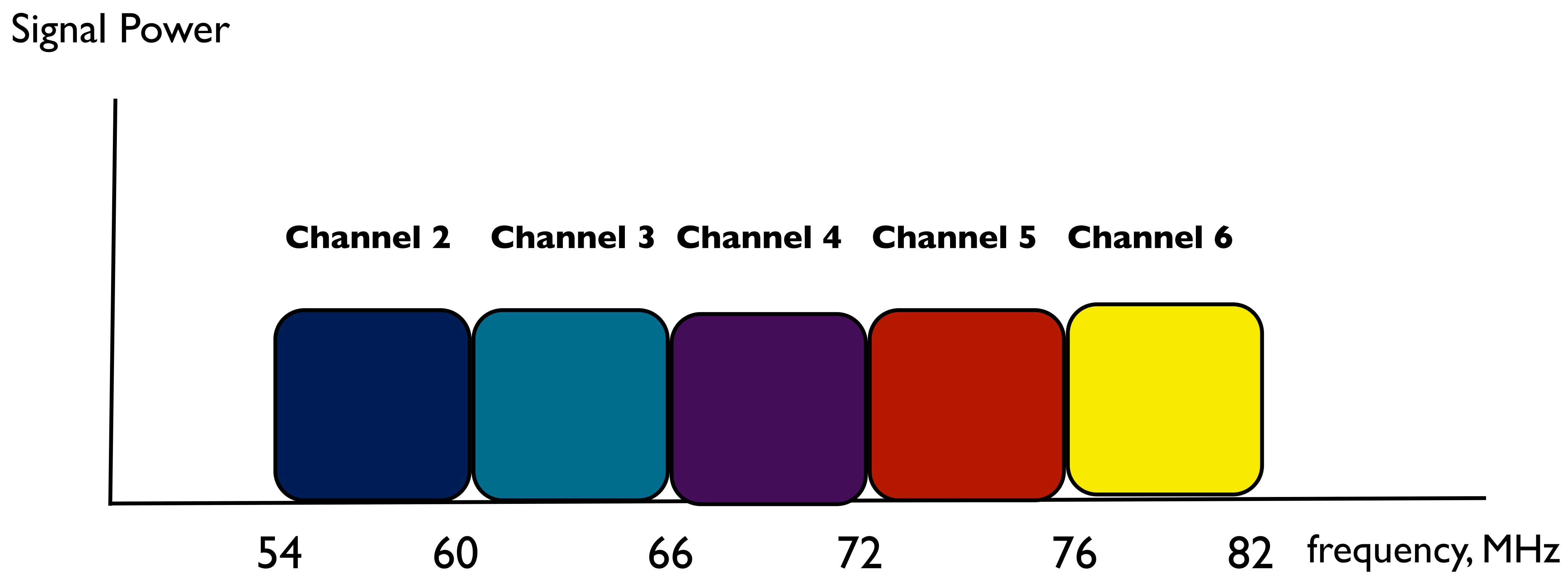
Multiplexing



Medium sharing techniques

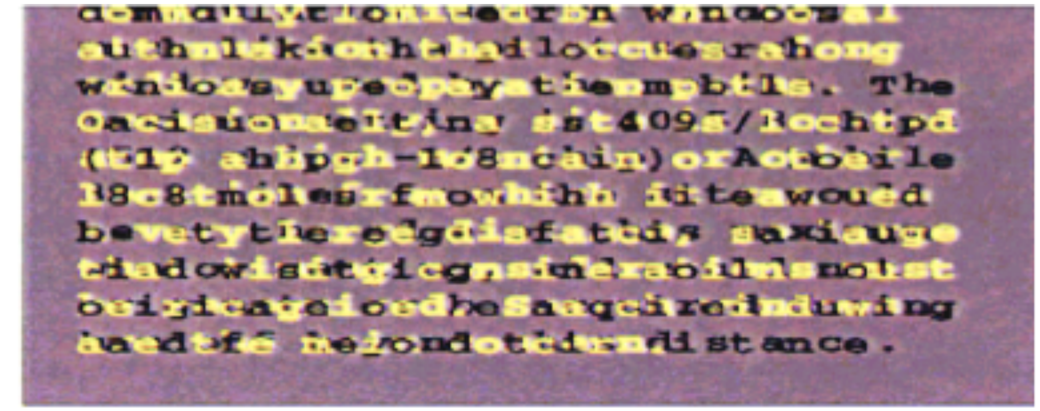


Example:
U.S. Television Channels Allocation

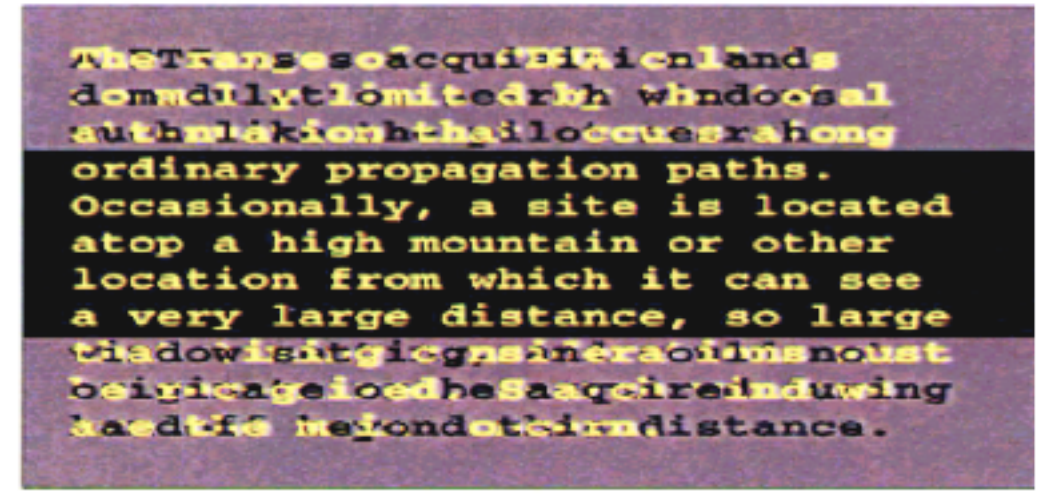


CDMA analogy

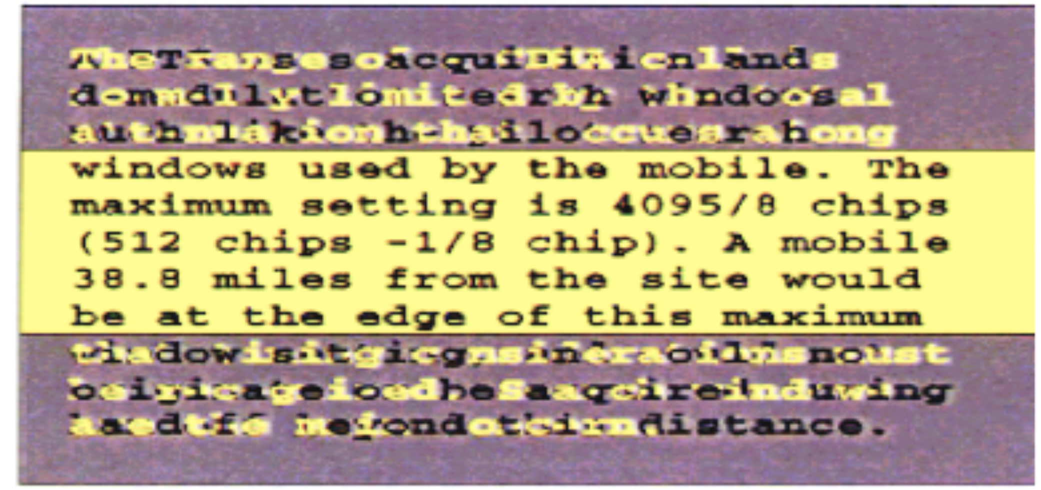
Two messages superposed, one in yellow and one in blue



A blue filter reveals what is written in yellow



A yellow filter reveals what is written in blue



Types of transmissions

Simplex:
one way only, example: TV Broadcasting

Half-duplex:
the corresponding stations have to take turns to access the medium, example: walkie-talkie. Requires hand-shaking to coordinate access. This technique is called **TDD** (Time Division Duplexing)

Full-duplex:
the two corresponding stations can transmit simultaneously, employing different frequencies. This technique is called **FDD** (Frequency Division Duplexing). A guard band must be allowed between the two frequencies in use.

Conclusions

The communication system must overcome the noise and interference to deliver a suitable replica of the signal to the receiver.
The capacity of the communication channel is proportional to the bandwidth and to the logarithm of the S/N ratio.
Modulation is used to adapt the signal to the channel and to allow several signals to share the same channel.
Higher order modulation schemes permit higher transmission rates, but require higher S/N ratio.
The channel can be shared by several uses that use different frequencies, different time slots or different codes.

Thank you for your attention

For more details about the topics presented in this lecture, please see the book ***Wireless Networking in the Developing World***, available as a free download in many languages:

http://wndw.net/