Fundamentals of telecommunications

Training materials for wireless trainers

The Abdus Salam
International Centre
for Theoretical Physics

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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.

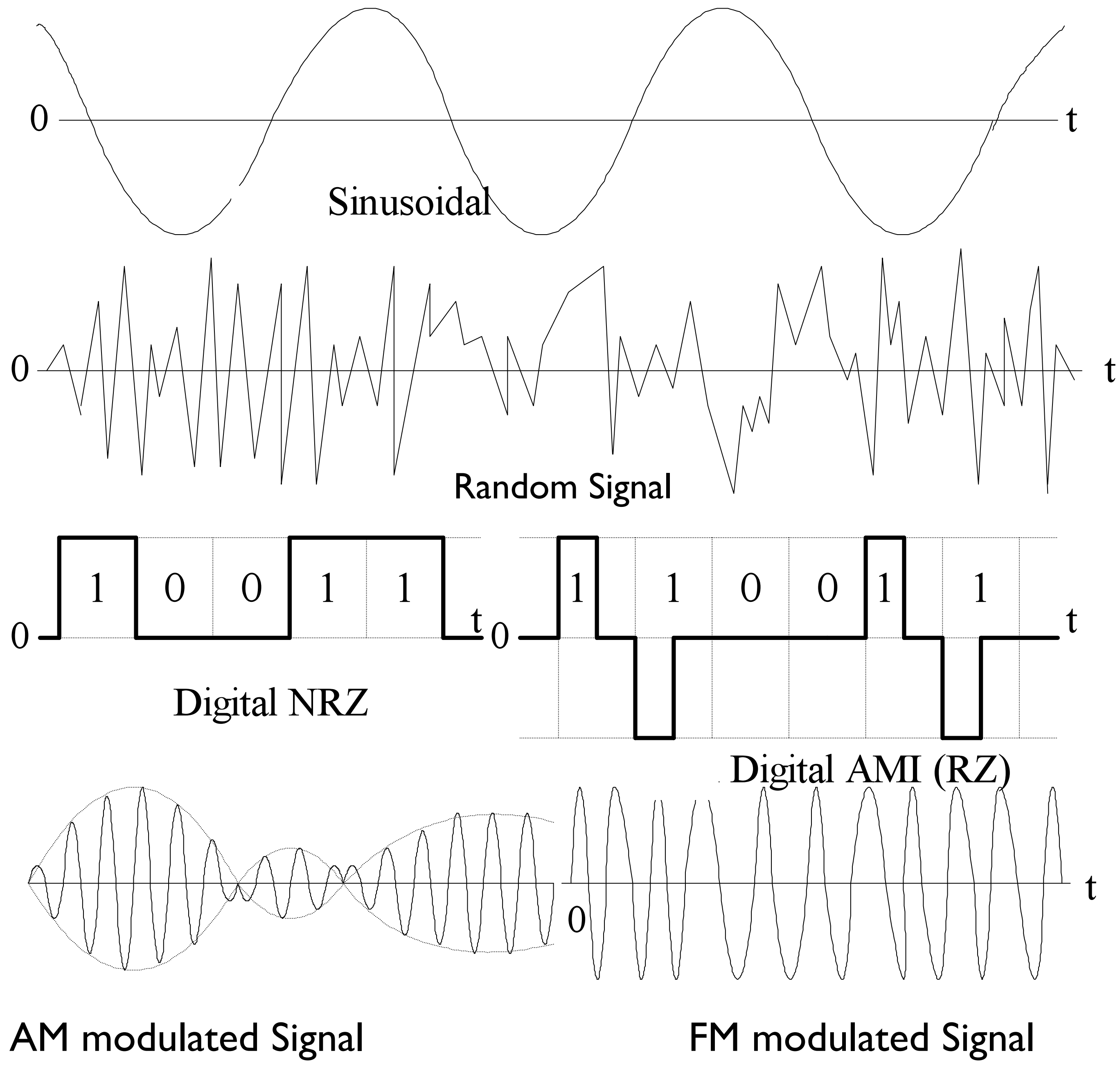
The sampling frequency must be at least twice the maximum frequency present in the signal in order to carry ***all*** the information contained in it.

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 analog signals are voice and video, examples of digital signals are written text and the morse code used in telegraphy. Any analog signal can be converted to a digital one containing the same information. Digital signals are more robust and easier to store and transport, that is why nowadays digital signals prevail

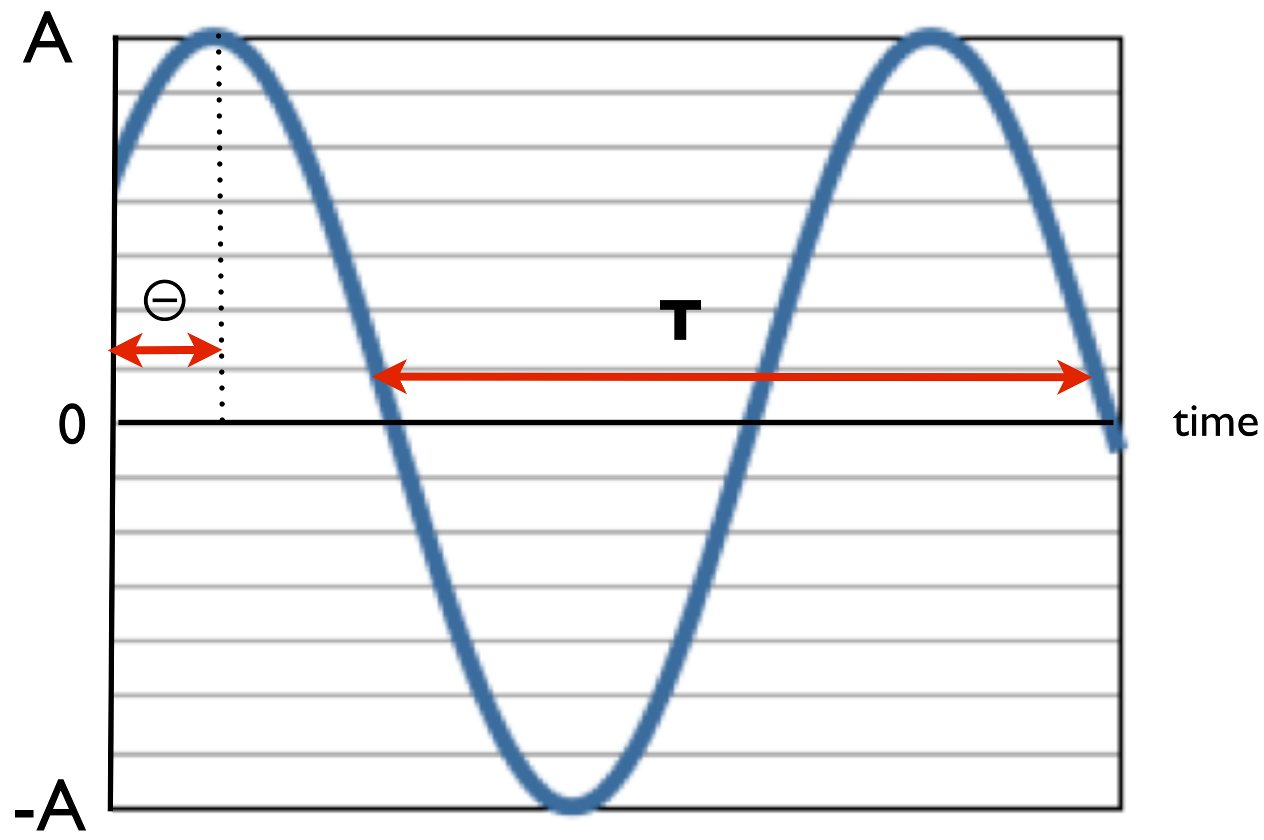
Examples of Signals



These are some examples of typical signals encountered in telecommunication systems.

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, $T = 1/f_o$

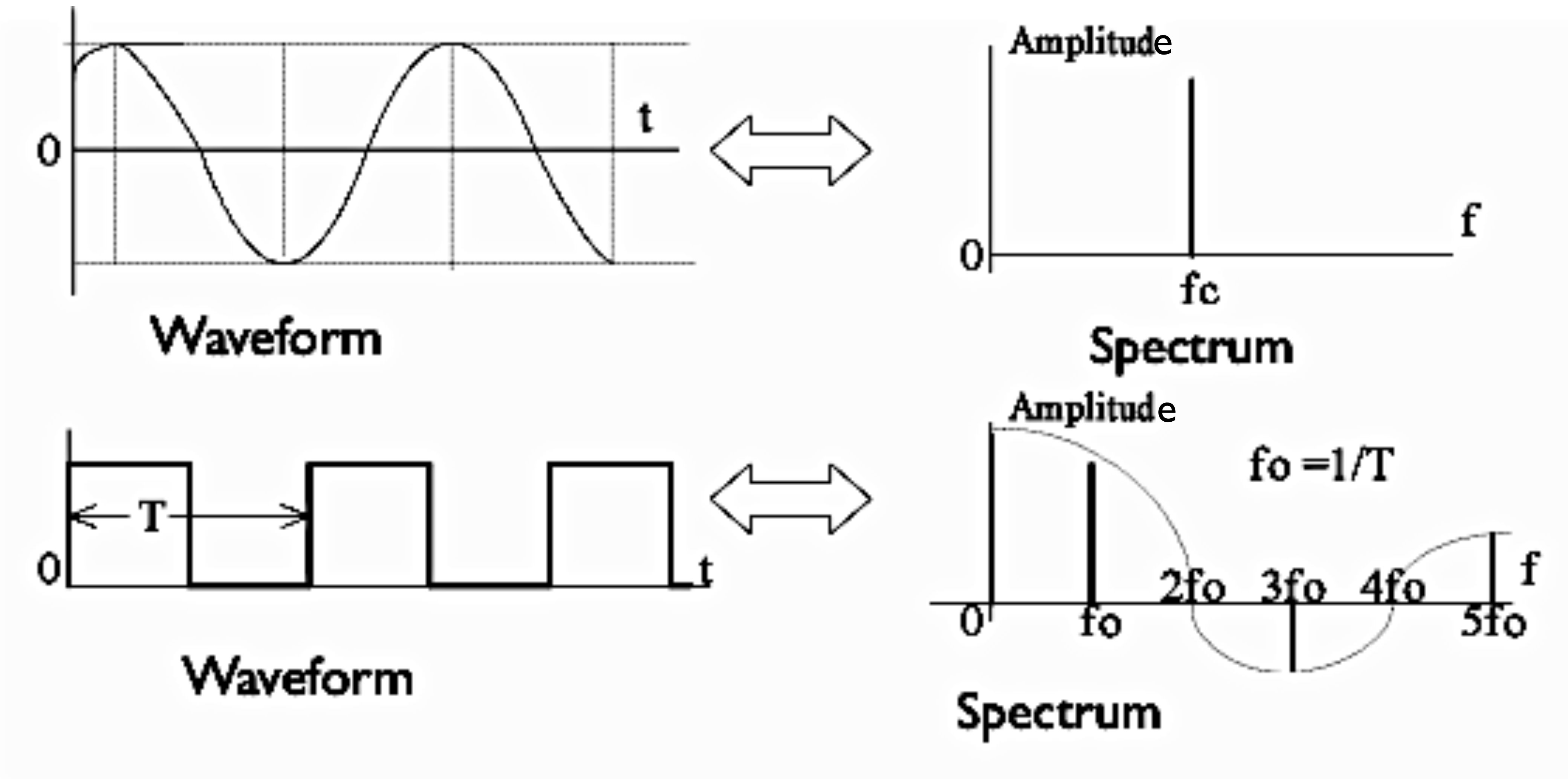
Θ = Phase

The sinusoidal signal is very important and can be expressed by a simple mathematical formula.

It contains a single frequency.

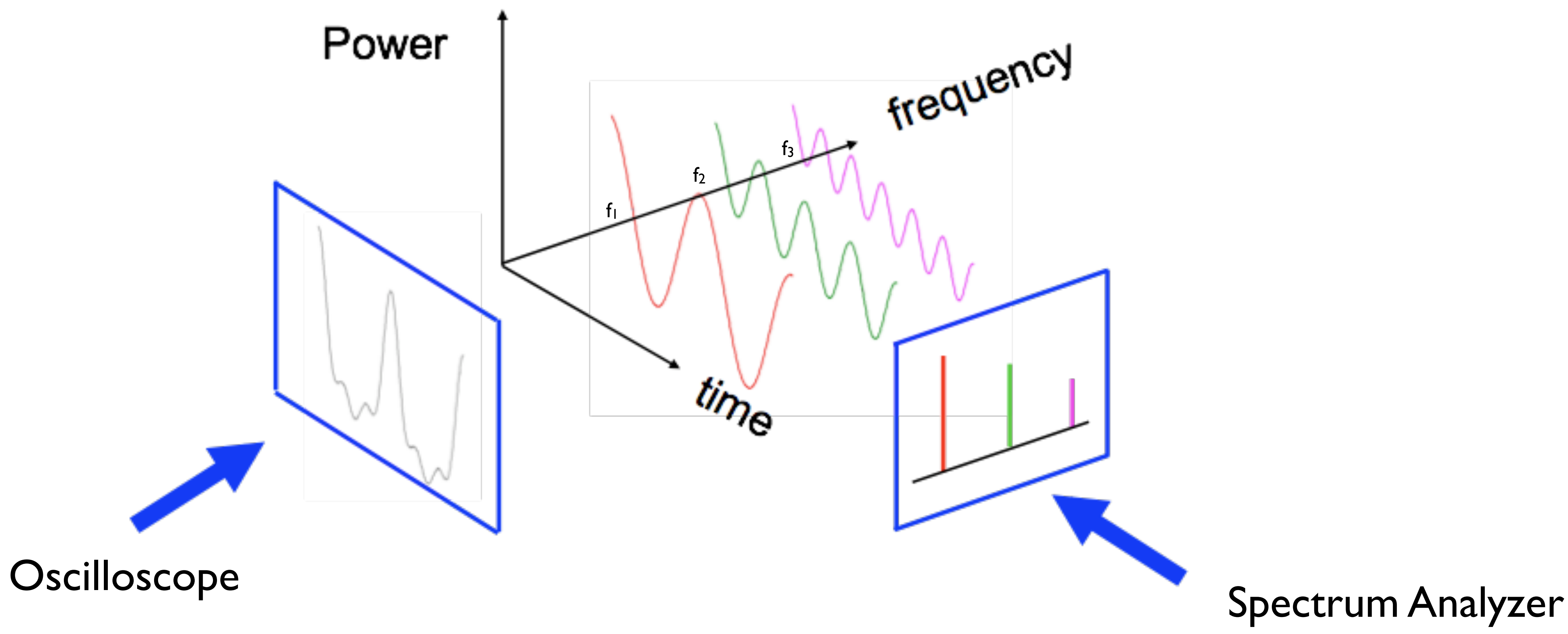
The phase is the offset from zero of the signal, when the offset is 90° we can also express the signal as $v(t)=A \sin(\omega t)$

Signals and Spectra



A signal can be characterized by its behavior over time or by its frequency components, which constitute its spectrum.
Any periodic signal is composed of many sinusoidal components, all of them multiples of the fundamental frequency, which is the inverse of the period of the signal.

Spectral analysis and filters



The graph shows that we can look at a signal from the perspective of its evolution over time, or we can look at it from the perspective of its frequency component. When we look at it from this perspective, we are dealing with the spectrum of the signal. The waveform can be displayed by an instrument called an oscilloscope, while the spectrum can be displayed by what is called a Spectrum Analyzer.

The spectrum distribution relays very important information about a the signal and allows for the intuitive understanding of the concept of filtering electrical signals.

In the example shown, the signal is formed by the superposition of three sinusoidal components of frequency f_1,f_2 and f_3.

If pass this signal through a device that will remove f_2 and f_3, the output is a pure sinusoidal with the f_1 frequency. We call this operation "**Low Pass filterin**g" because it removes the higher frequencies.

Conversely, we can apply the signal to a "**High Pass Filte**r", a device that will remove f_1 and f_2 leaving only a sinusoidal signal at the f_3 frequency.

Other combinations are possible, giving rise to a variety of filters.

No physical device can transmit all the infinite frequencies of the electromagnetic spectrum, so any device will always perform some kind of filtering in the signal that goes through it.

The bandwidth of a signal is the difference between the highest and the lowest frequency that it contains and is expressed in Hz.

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*.

Although signal are always real, spectra are complex functions that, in general, will have a real and an imaginary part.
By the same token, it is customary to deal with negative frequencies, as well as with negative times.
This simplifies the mathematical treatment of the 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

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

$$\omega = 2\pi f$$

You will also often encounter the Fourier Transform expressed as a function of ω, remember that ω = 2πf, and also the letter "j "used for indication of the imaginary symbol. The Fourier transform operator is indicated by a $\mathscr{F}$ (script F). The inverse Fourier transform is indicated by $\mathscr{F}^{-1}$

Properties of the Fourier transform

Linearity: $\mathscr{F}$[x1(t) + x2(t)] =$\mathscr{F}$[x1(t)]+ $\mathscr{F}$[x2(t)]

Time shifting: $\mathscr{F}$[x(t−to)] = $e^{-i2\pi ft_o}$ X(f)

Change of scale: $\mathscr{F}$[Ax(t)] =1/|A| X(f/A)

Frequency shifting: **$\mathscr{F}$-1[X(f-fo)] = x(t)** ⟦*e^(-i2πtfo)*⟧

Symmetry: direct and inverse Fourier transform are similar operations

Fast Fourier Transform

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

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

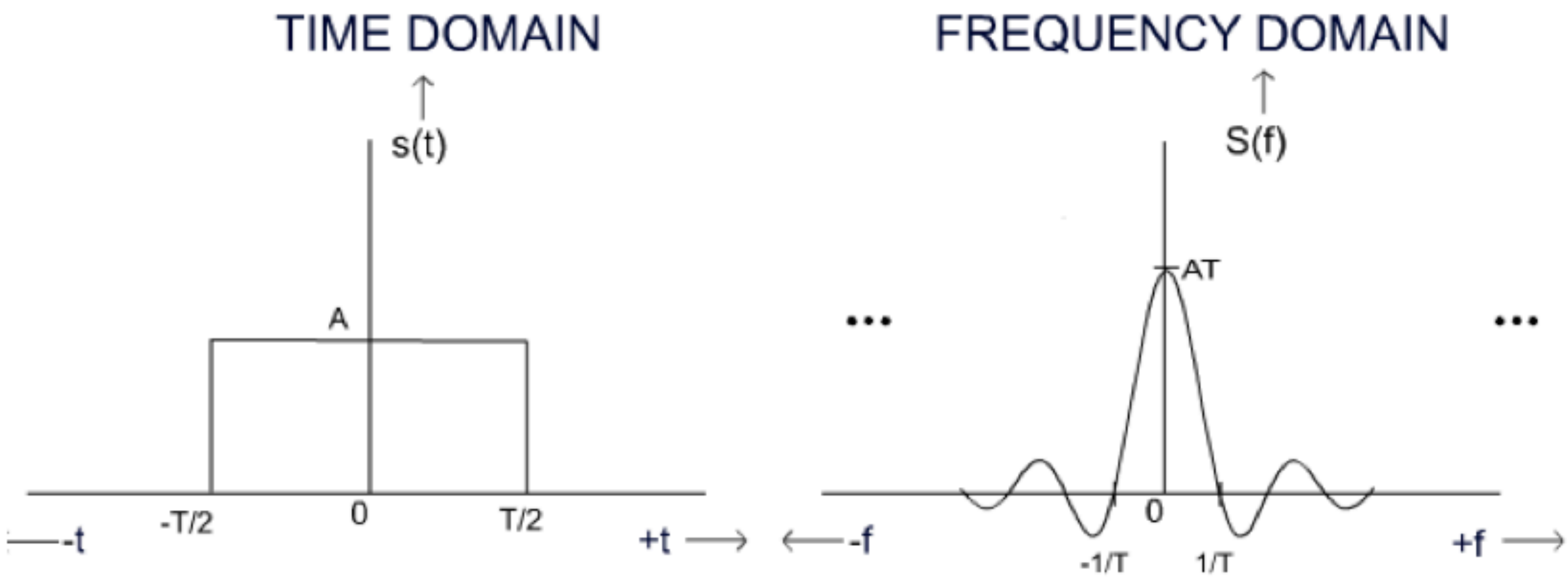
http://www.westga.edu/~jhasbun/osp/Fourier.htm
Applet for Fourier transform, sampling and quantifying

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

So, if we have a digital signal, we can perform the Fourier transform very easily in the digital domain.

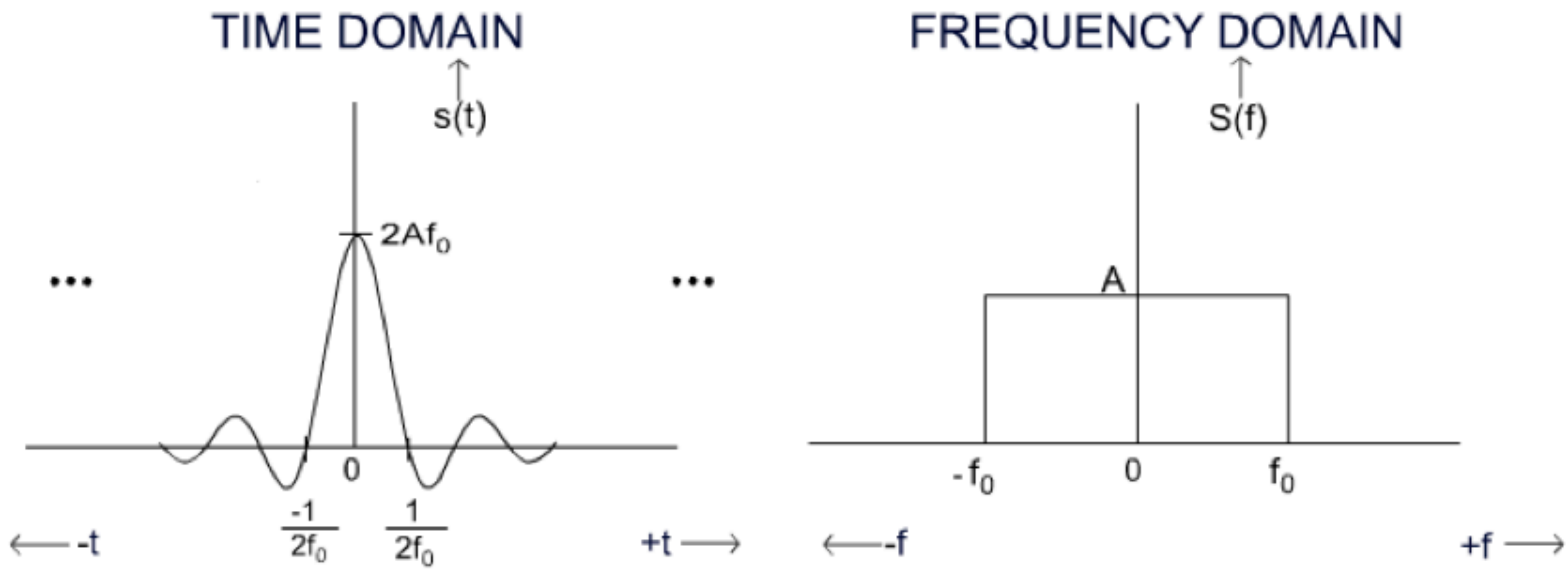
Common signals and spectra



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

The rectangle function (a single pulse) transforms into the sinc function in the frequency domain. Notice that the spectrum goes all the way to infinity, this is due to the abrupt change of value of the signal, which are not physically possible. In reality there will always be a certain rise and fall time of the pulse and this will limit the bandwidth.
Notice that the narrower the pulse duration T, the broader the frequency occupancy, given by the null of the sinc function that occurs at 1/T.

Common signals and spectra



$$s(t) = 2Af_0 \frac{\sin(2\pi f_0 t)}{(2\pi f_0 t)} \quad \longleftrightarrow \quad \begin{aligned} S(f) &= A \quad & |f| < f_0 \\ &= A/2 \quad & |f| = f_0 \\ &= 0 \quad & |f| > f_0 \end{aligned}$$

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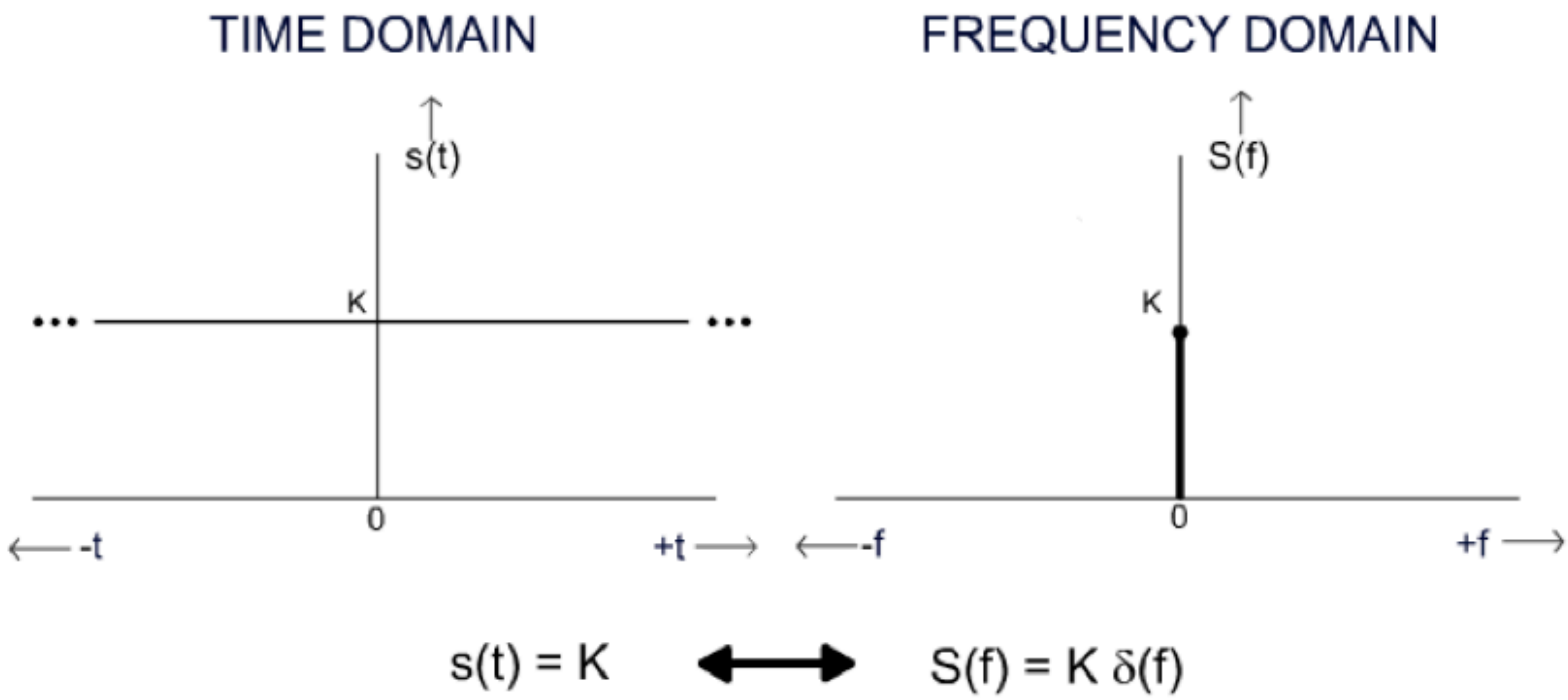
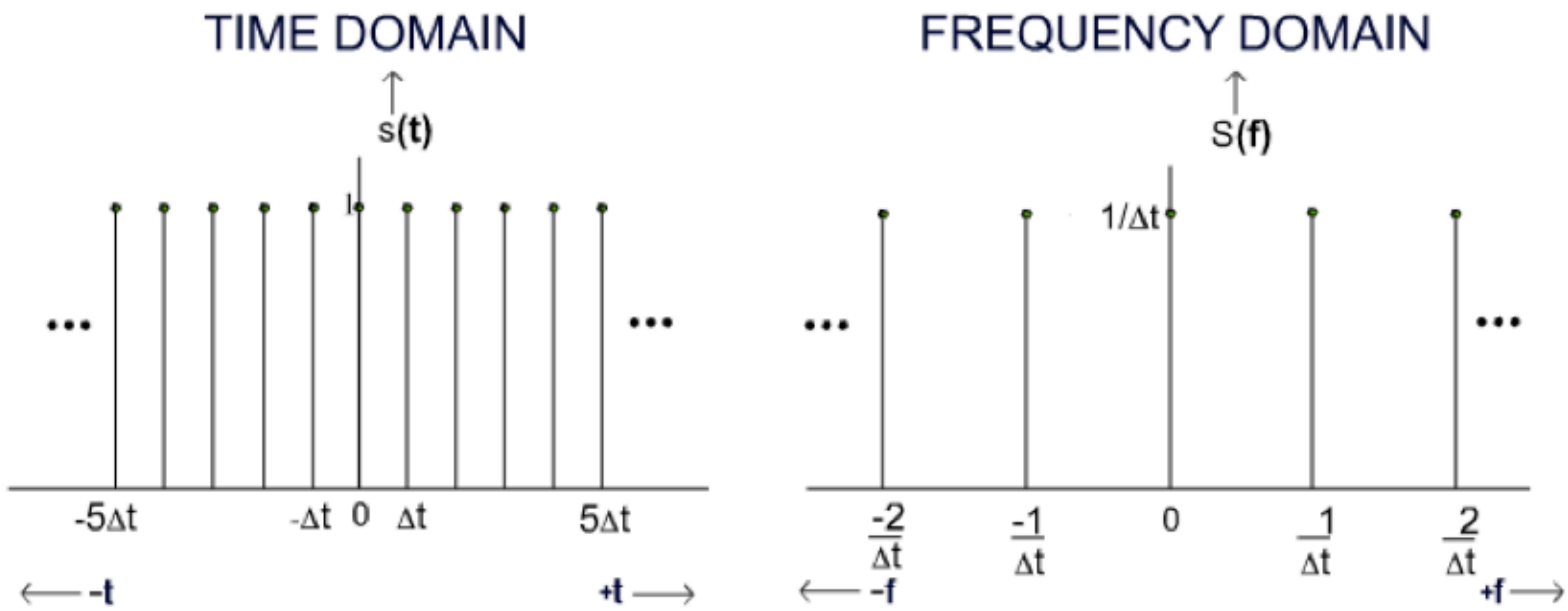
This illustrates the symmetry property of Fourier transforms

Common signals and spectra



The spectrum of a constant signal has only one component in frequency, at the origin (DC). In mathematical terms, a Dirac impulse. $\delta(f)=0$, for all $f \neq 0$, but the area under $\delta(f)$ is 1.

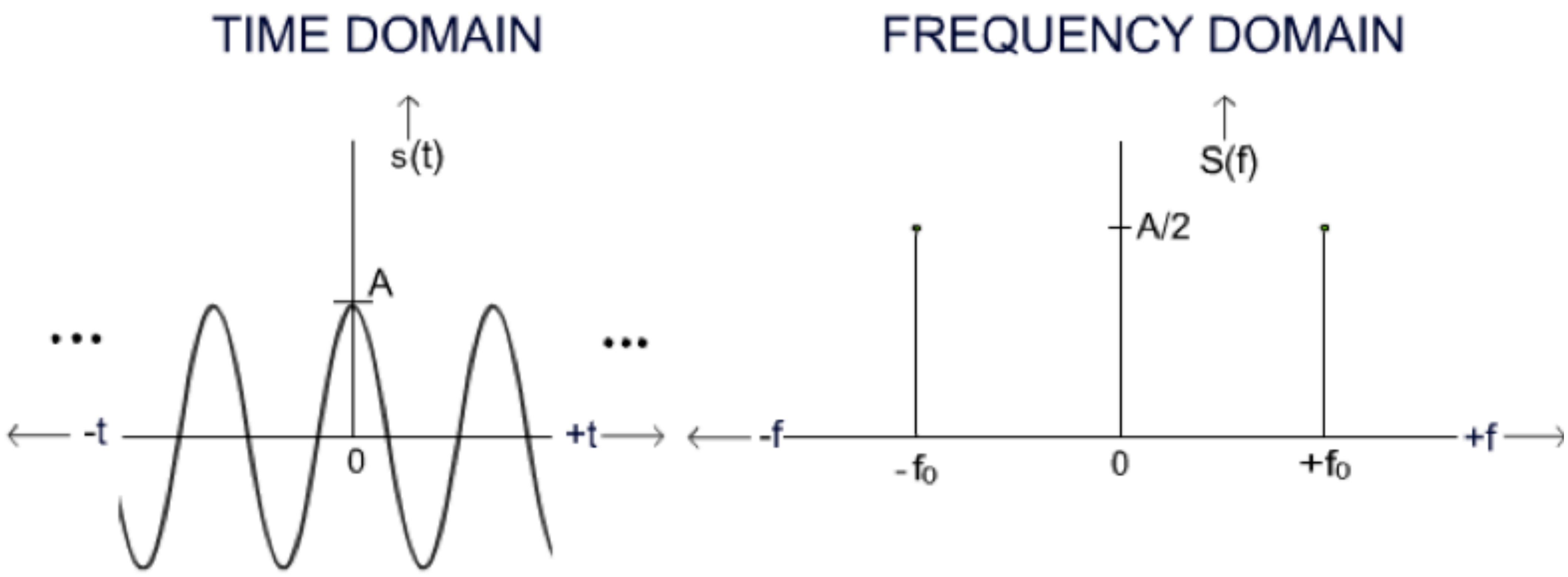
Common signals and spectra



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

A sequence of equally spaced Dirac impulses in time transforms in another sequence of Dirac impulses equally spaced in frequency.

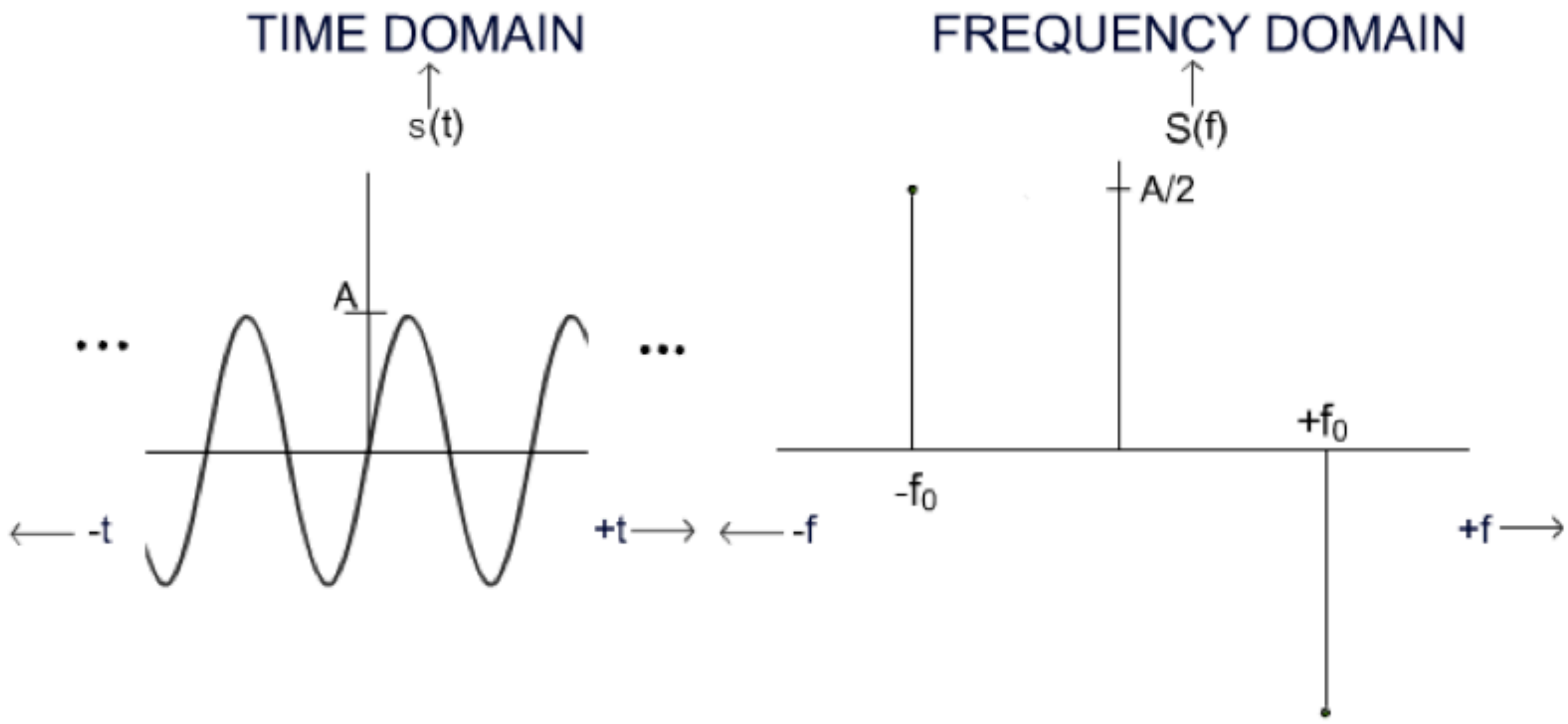
Common signals and spectra



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

The cosine function has an even and real spectrum

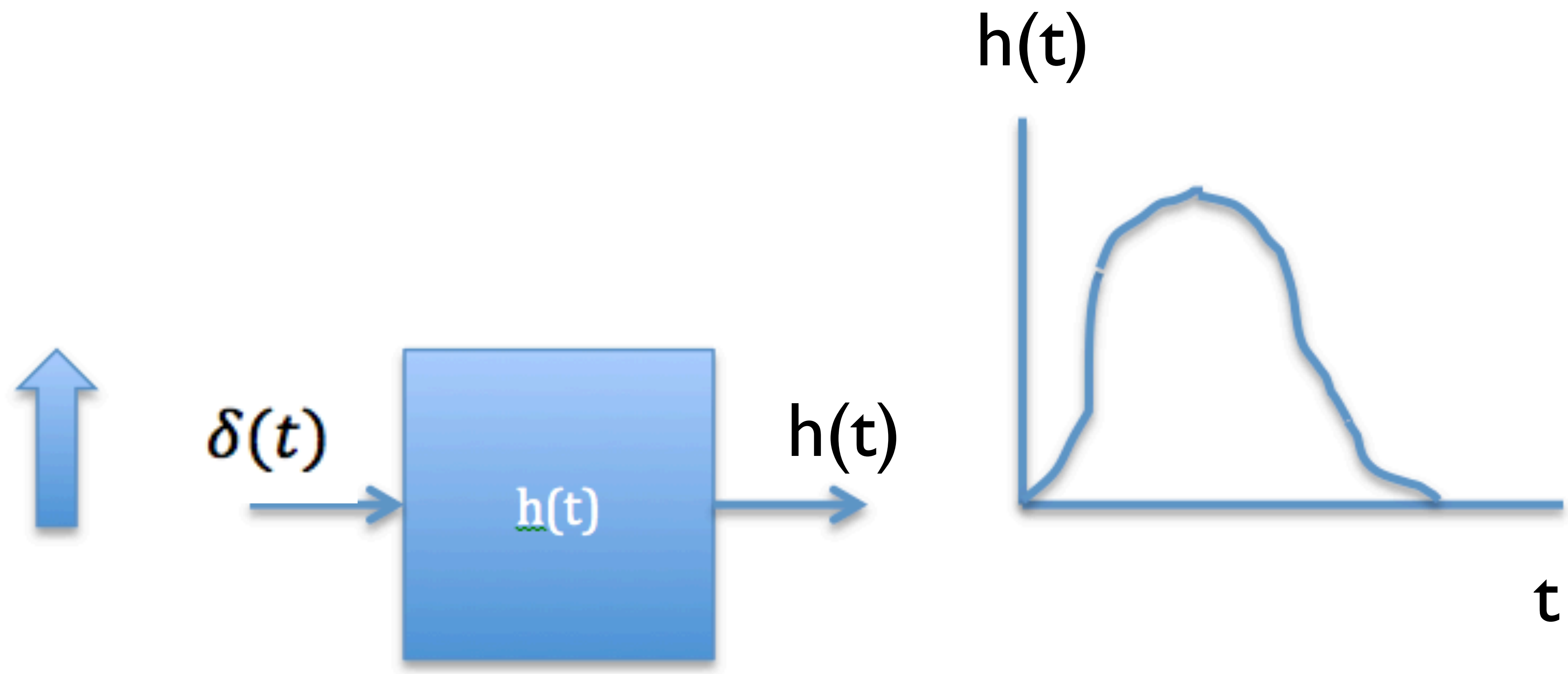
Common signals and spectra



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

The sine function has an odd and imaginary spectrum

Linear time invariant systems



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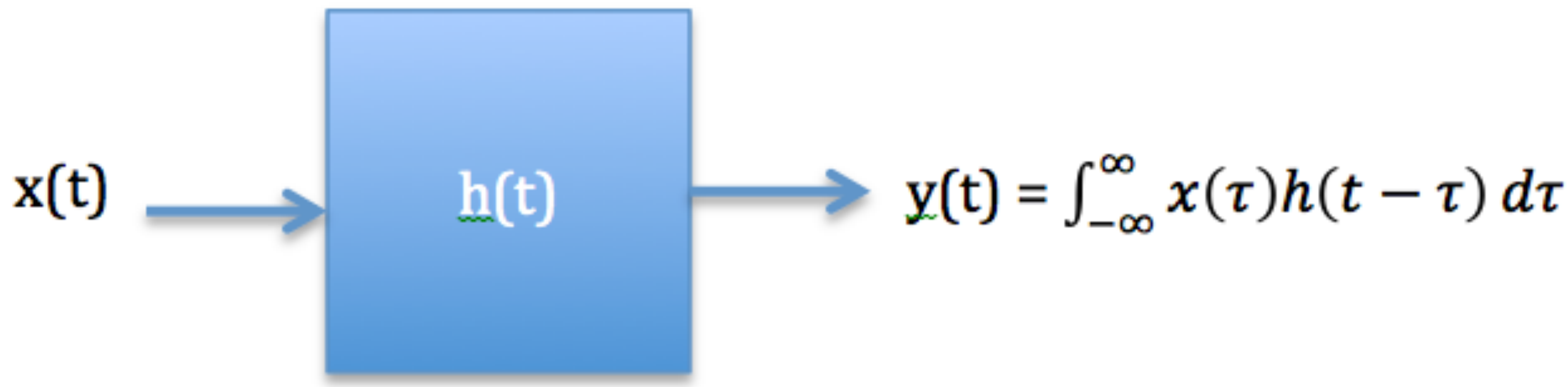
A linear time invariant system is characterized in the time domain by its impulse response, that is the output h(t) corresponding to a Dirac delta function input.

Linear time invariant systems time domain



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

h(t) is the impulse response of the system, that is the output when an impulse signal is applied to the input.
The output to an arbitrary x(t) input is its convolution with the impulse response.
The convolution operation is:
a) linear, it obeys the superposition principle, $[x_1(t)+x_2(t)]$✪h(t) = $x_1(t)$✪h(t) + $x_2(t)]$✪h(t)
b) commutative, x(t)✪h(t) = h(t)✪x(t)
c) domain independent, it can be performed in the time or in the frequency domain
d) not physically doable but we can approximate it
The convolution of two time signals means the product of their spectra

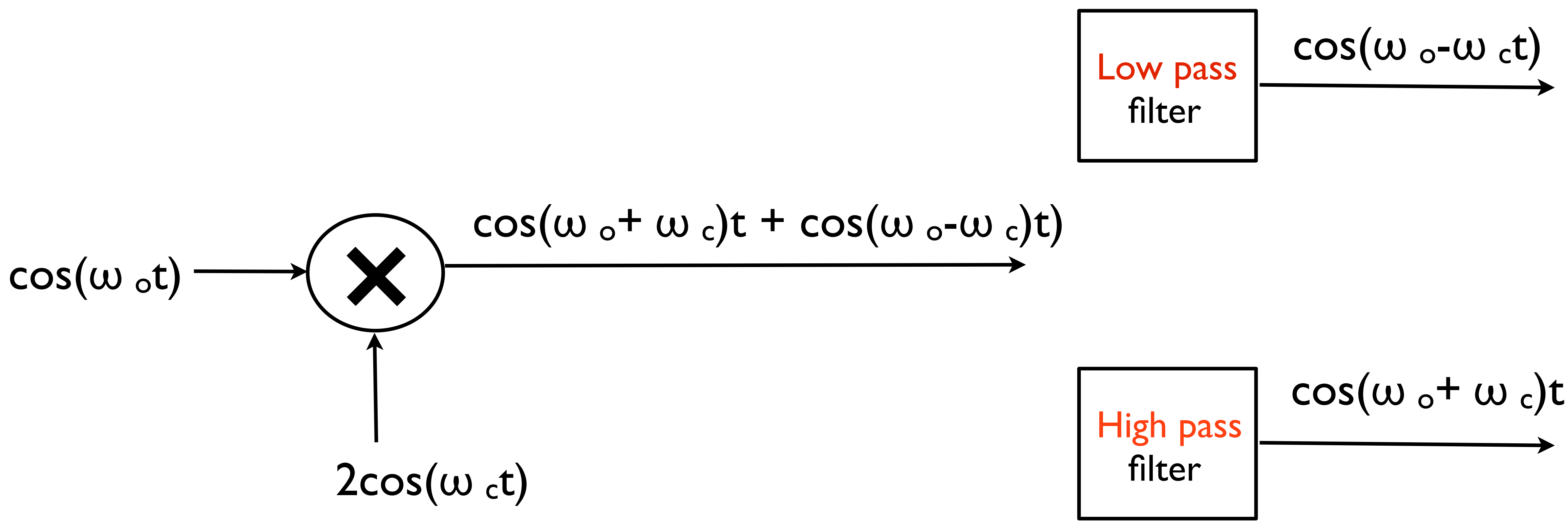
Linear time invariant systems in frequency domain

H(f) is the transfer function of the system

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

So there are two methods to find the response of a linear time invariant system, by performing the convolution in the time domain or by calculating the inverse Fourier transform of the product of the spectrum of the input with the transfer function of the system.

Basic frequency converter (frequency mixer)



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

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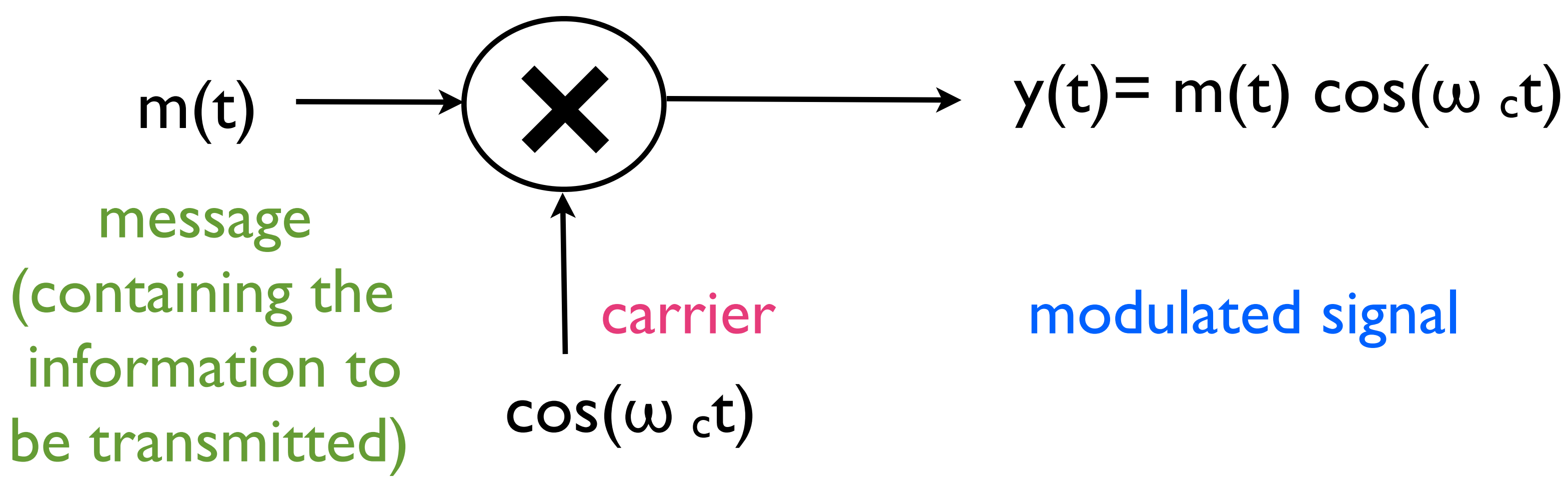
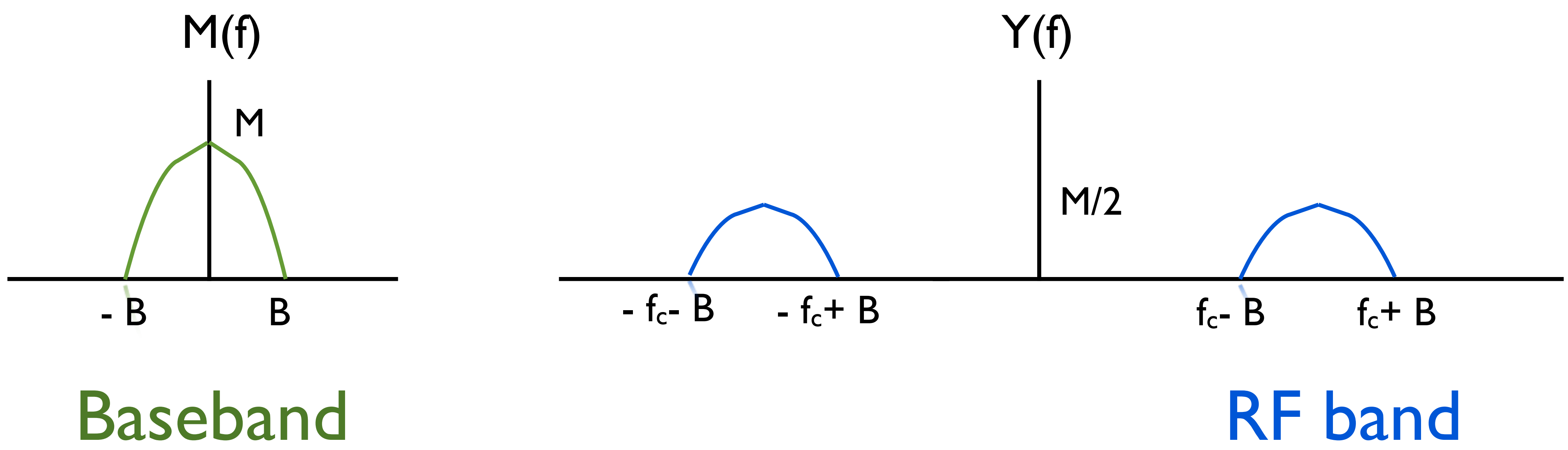
The basic frequency translator can be used to increase the frequency of the input signal, as is often the case at the transmitter side, or to decrease the frequency, as is needed at the receptor.
When the output frequency is higher, the circuit is called an **upconverter**, the opposite is called a **downconverter**.
We can also use a bandpass filter if we desire to translate the signal to a given intermediate frequency (IF).
This arrangement is used in heterodyne receivers.

Amplitude modulation



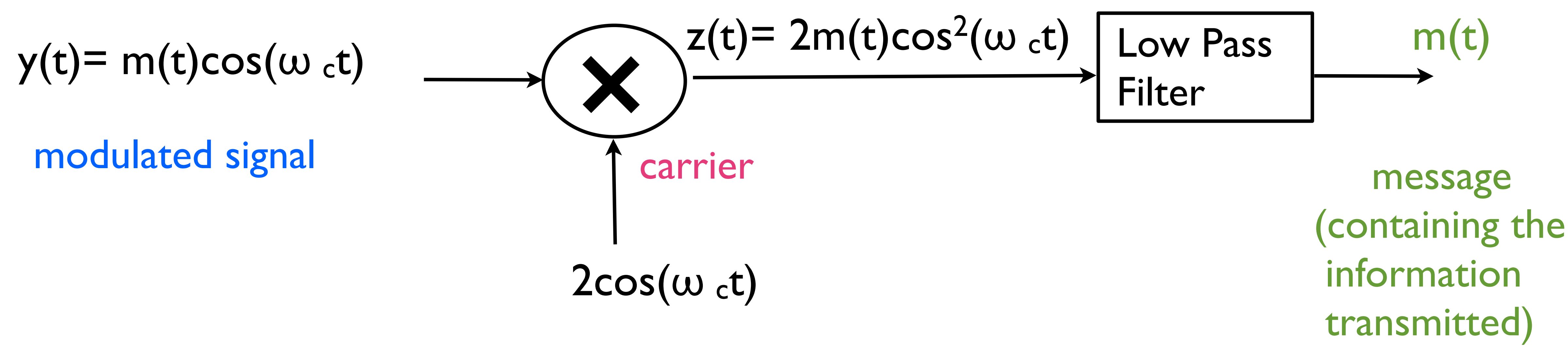
in the frequency domain:



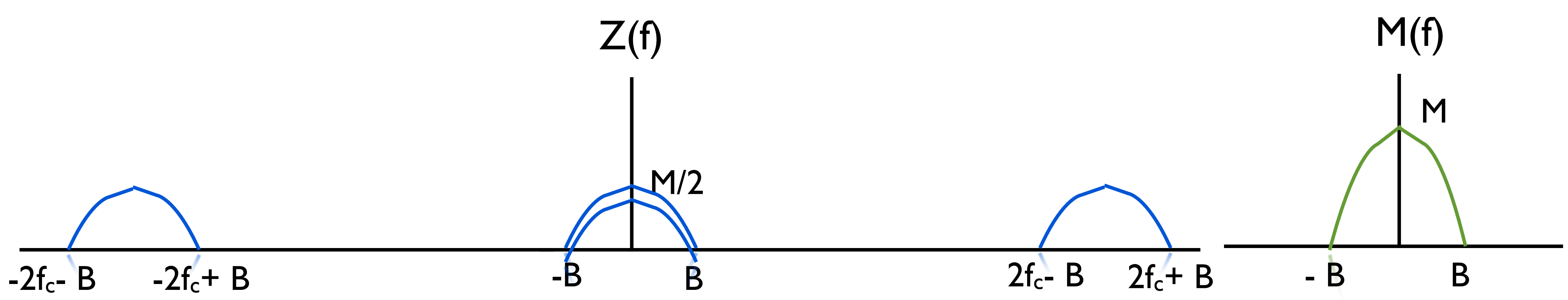
The same circuit used for a mixer can also be used for amplitude modulation, by inputting the message to be modulated.

$\cos(\omega_c t)$ is called the carrier signal, $m(t)$ is the message to be transmitted, with frequencies around the origin and so they are called **baseband**, while the modulated frequencies constitute the **RF band**.

Amplitude demodulation



in the frequency domain:



The same circuit used for a mixer can also be used for amplitude modulation, by inputting the message to be modulated.

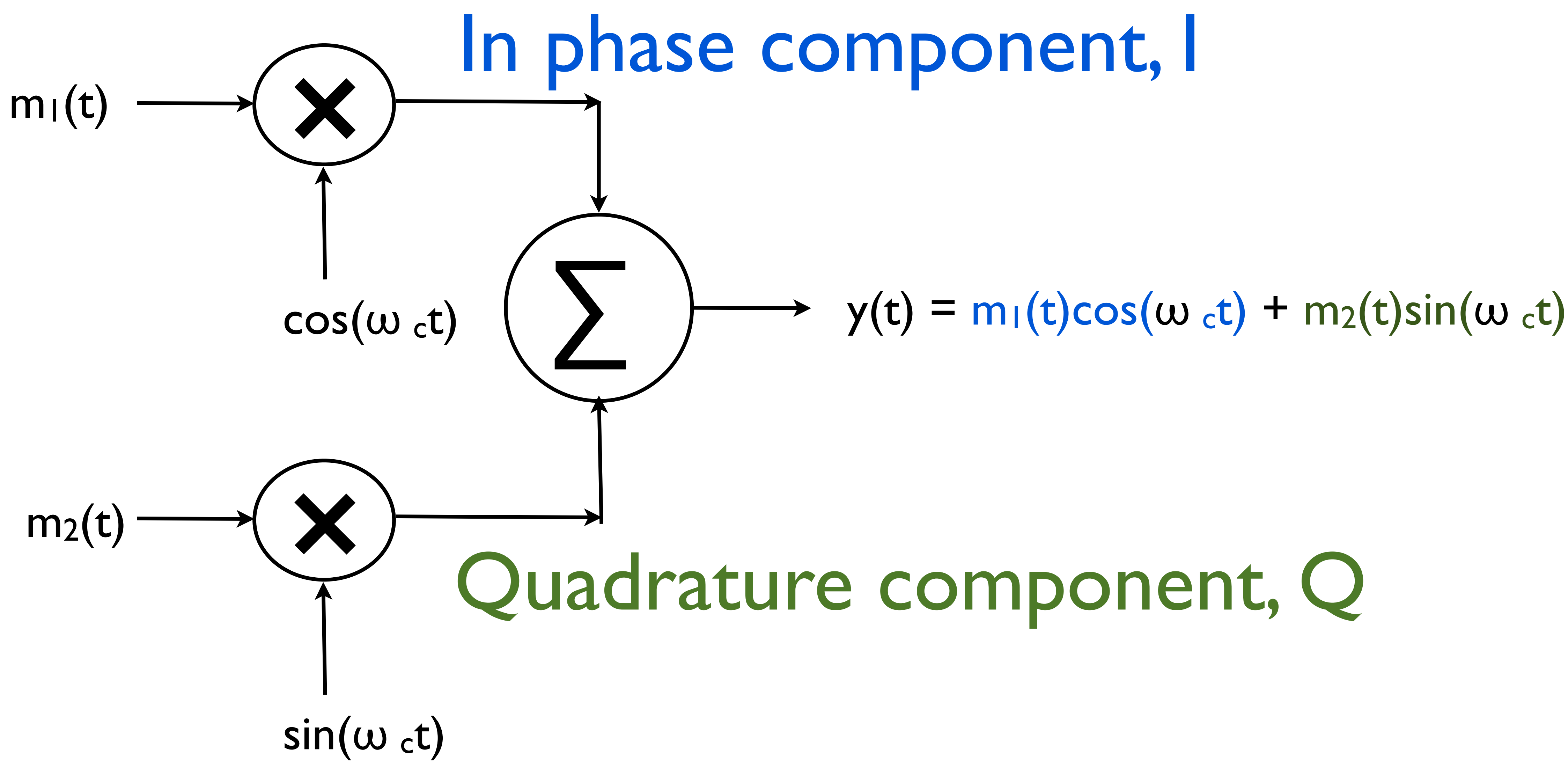
$\cos(\omega_c t)$ is called the carrier signal,

$m(t)\cos(\omega_c t)[2\cos(\omega_c t)] = m(t) + m(t)\cos(2\omega_c t)$

the low pass filter eliminates the component centered at 2fc

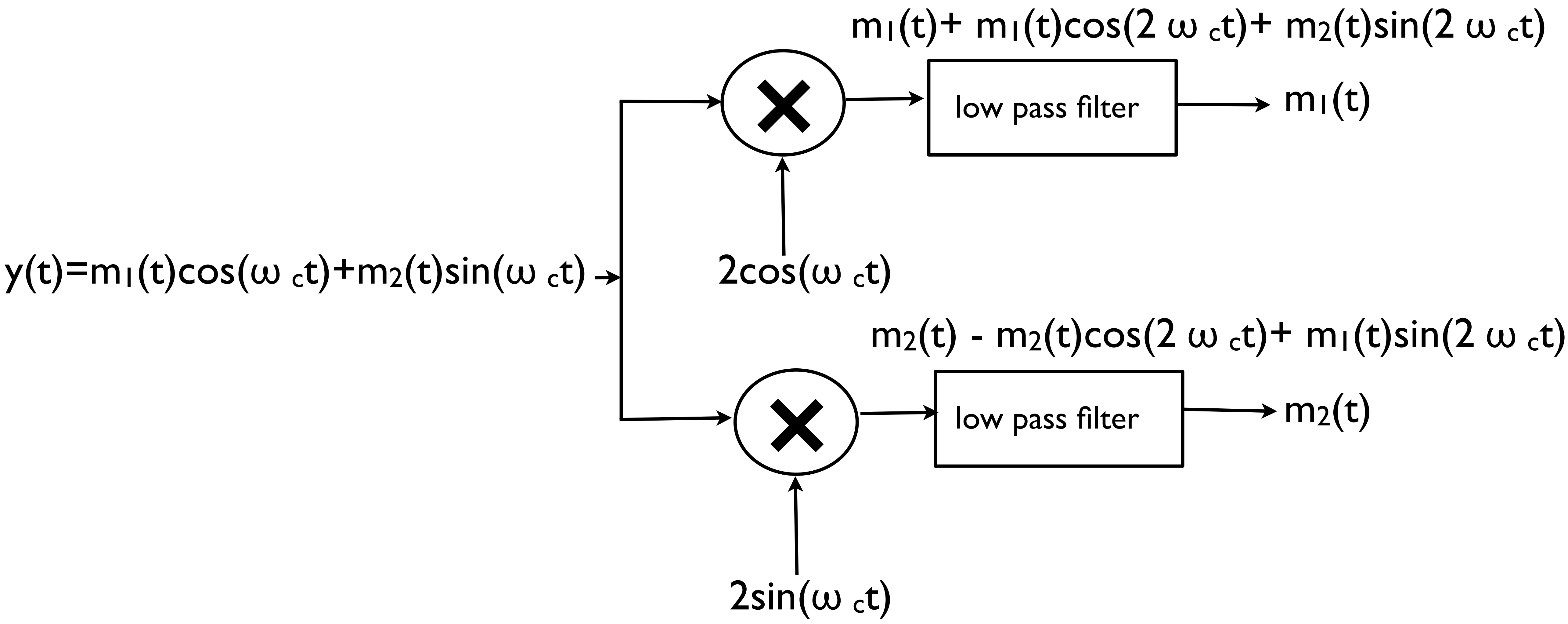
Notice the superposition of the message spectra at the origin.

Orthogonality



Although the spectra M1(f) and M2(f) occupy the same frequency interval, they can be separated because of the orthogonality of the sine and cosine functions.

Orthogonality



In the upper branch: $[m_1(t)\cos(w_c t)+m_2(t)\sin(w_c t)]\ 2\cos(w_c t)= m_1(t)+ m_1(t)\cos(2w_c t) + m_2(t)\sin(2w_c t)$

The low pass filter eliminates the components centered around 2fc so the output is simply $m_1(t)$.

In the lower branch: $[m_1(t)\cos(w_c t)+m_2(t)\sin(w_c t)]\ 2\sin(w_c t)= m_2(t) - m_2(t)\cos(2w_c t) + m_1(t)\sin(2w_c t)$

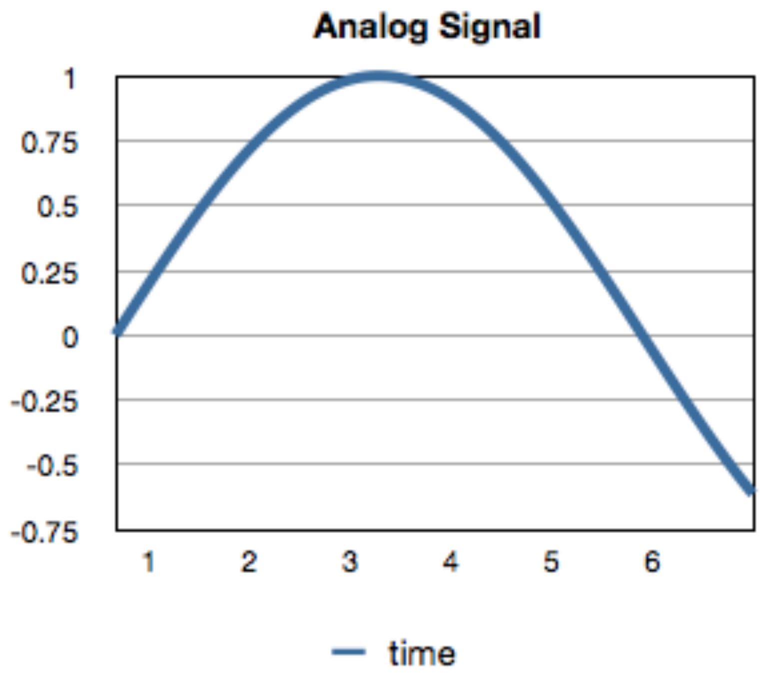
The low pass filter eliminates the components centered around 2fc so the output is simply $m_2(t)$.

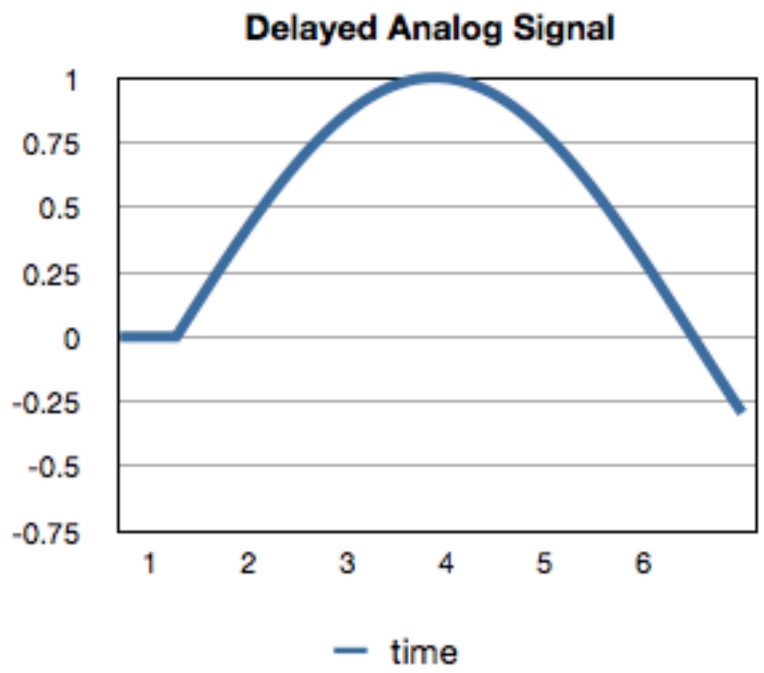
The orthogonality allows the sharing of the same frequency interval by two independent signals without interference, they are separated by phase differences.

Communication System

The basic communication system is formed by a transmitter TX, a communication channel and a receiver RX
The transmitter injects a signal into the channel that delivers it to the receiver.
The receiver must recover the information contained in the received signal despite the limitations introduced by the channel.
The channel can be a physical one, like a copper cable or an optical fiber, or simply air or even vacuum that will transmits electromagnetic waves.
Any channel is subject to some kind of electromagnetic "noise" and interference, will attenuate the signal and will change its shape (distortion).
Since it takes some time for the signal to traverse the channel, the received signal will have some latency with respect to the transmitted signal. This "latency" might change over time and contribute to "jitter" in the received signal.
The signal might also reach the receiver by means of different trajectories, and in this case the different received versions will interact as a consequence of the "multipath".
Multipath can completely obliterate a signal but it can also used advantageously in some modern communications techniques which employ MIMO (Multiple Input Multiple Output) in order to extract the information contained in signals with different trajectories to increase the transmission rate or the range.

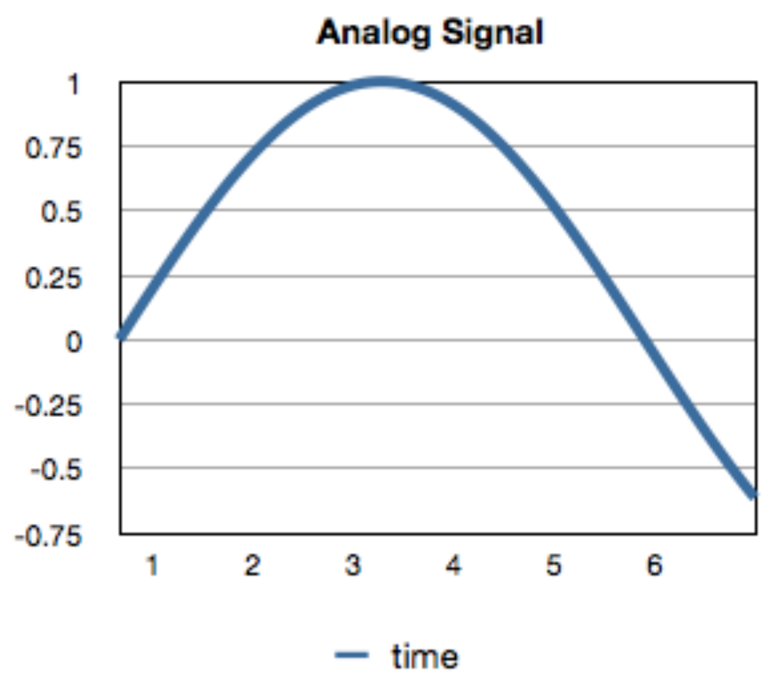
Signal Delay



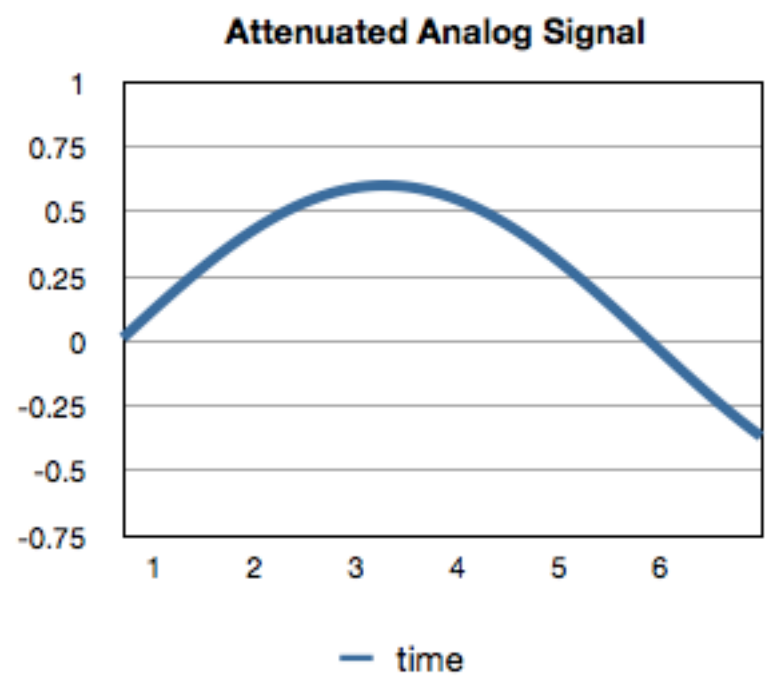


The bottom graph shows a delayed replica of the top signal. Delay is a very important aspect of the quality of a communication system. For a good voice quality, the total delay (also called latency) should be less than 150 ms, according to ITU-T G.114 recommendation.
But when communicating through a geostationary satellite, the minimum delay is imposed by the propagation time up to the satellite orbiting at 36000 km above earth, and back down, for a total of 72000 km, which, at 300000 km/s, represents 0.24 s.
The delay of the signal signifies a phase change in the spectrum, $x(t-t_o)= X(f)e^{-iwft_o}$.

Attenuation



Transmitted Signal

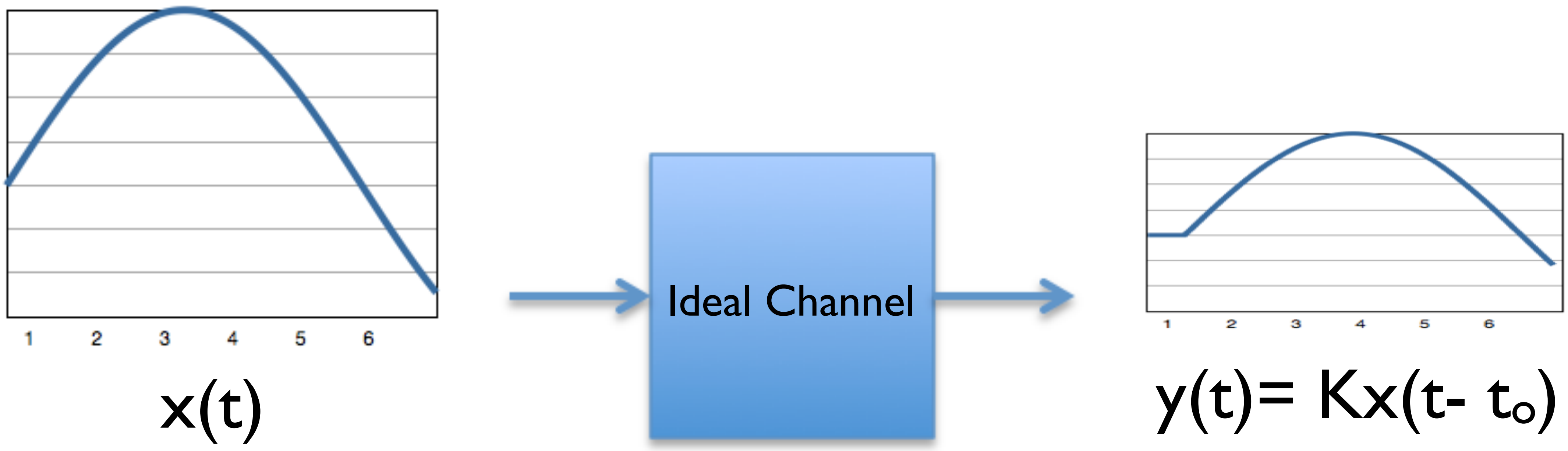


Received Signal

Although the effect of attenuation can easily be overcome with an amplifier, the amplifier will also enhance any noise introduced by the channel and inevitably introduces some extra noise of its own.

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:

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

This, of course, can only be possible over a limited ranges of frequencies, giving rise to the ideal filters.

Amplifiers

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

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

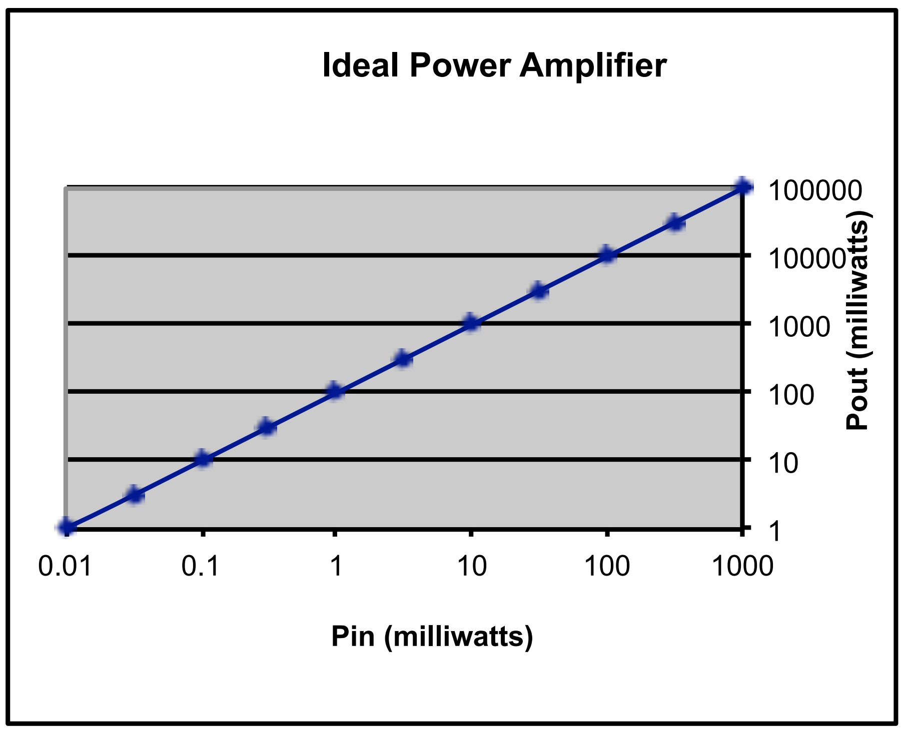
Real amplifiers will always have some amount of non linearities, so their output will also contain higher order terms:

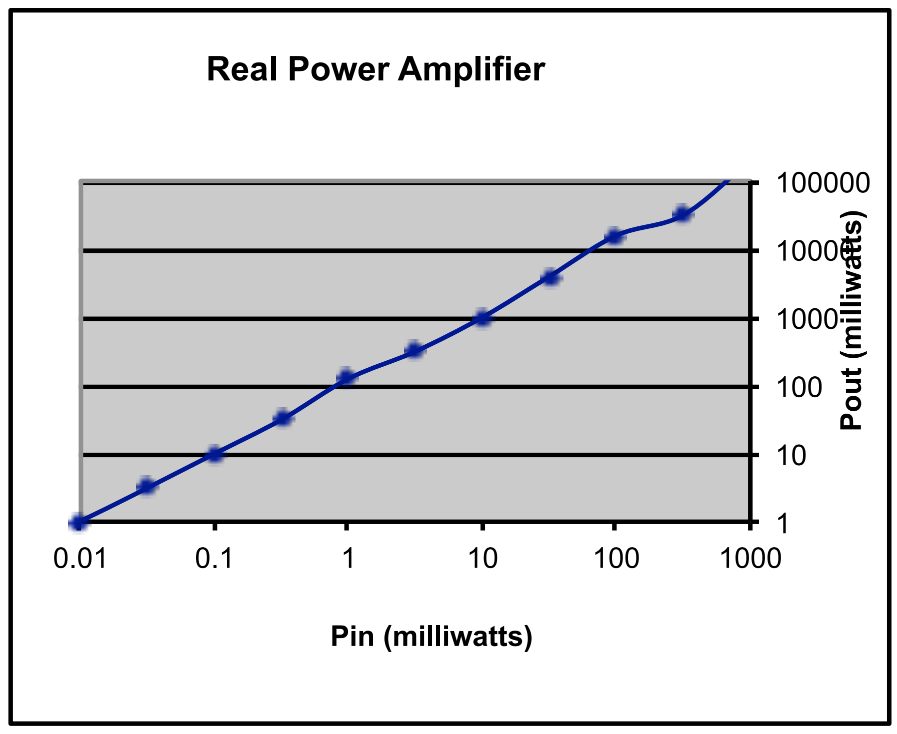
$$y(t)= Ax(t) + Bx^2(t)+Cx3(t)+\ldots\ldots(A>>B>>C)$$

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.

We are neglecting the delay always introduced by the amplifier since it is not relevant to the present discussion and will only add complexity to the math.
Suppose $x(t)= x_1(t)+x_2(t)$
A real amplifier might have an output given by $y(t)= 100x(t) + x^2(t)$, so the output would be $y(t)=100x_1(t)+[x_1(t)2 +x_2(t)]^2=100x_1(t)+x_1^2(t)+2x_1(t)x_2(t) +x_2^2(t)$, and will behave also as a mixer for the input signals.

Amplifiers

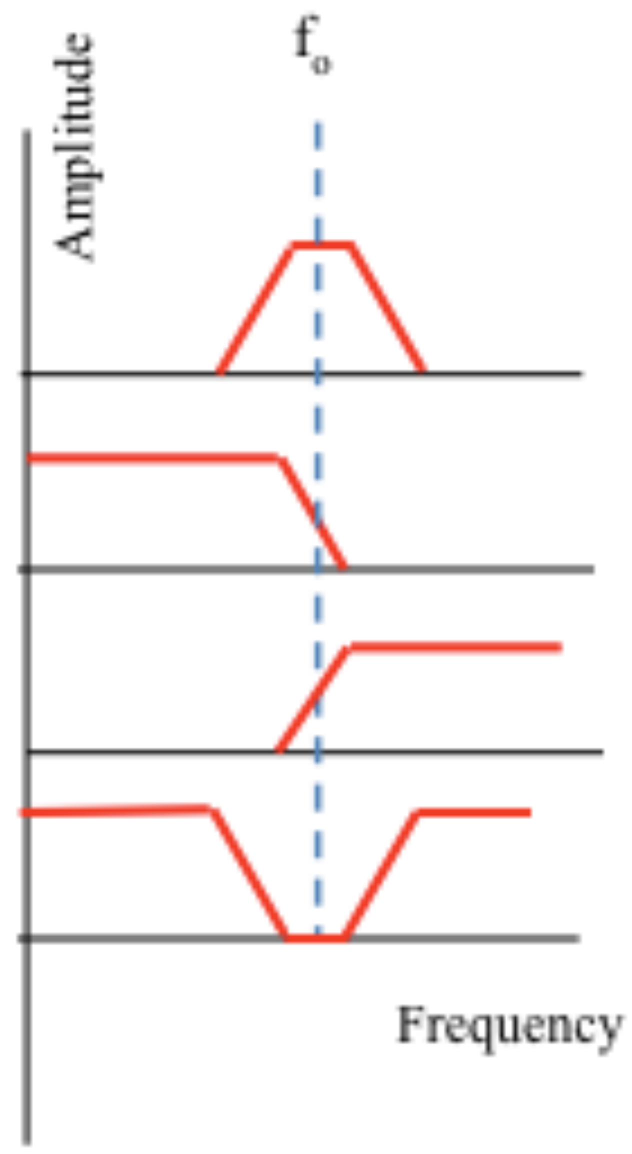




Graphs of an ideal power amplifier in which the output is a linear function of the input and a real amplifier that shows some distortion

Filter Types

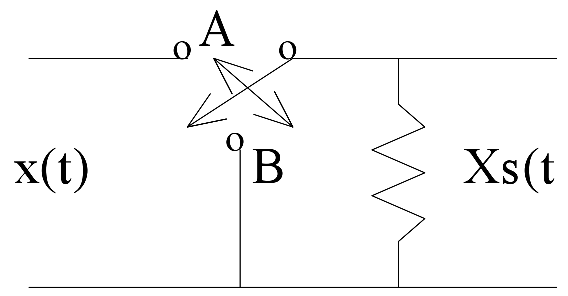
- Bandpass
- Lowpass
- High Pass
- Bandstop



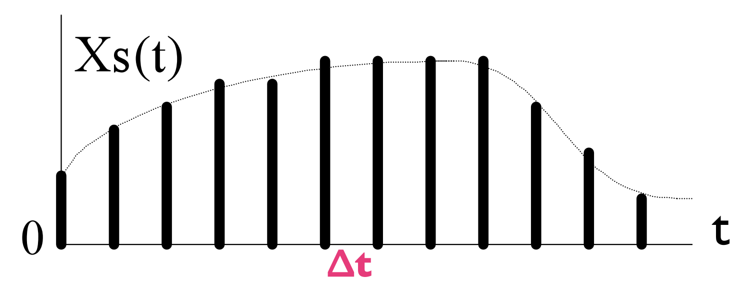
An electrical filter is a device that selects a particular range of frequencies, heavily attenuating others.
A bandpass filter will only let through the frequencies between the lower cutoff and the upper cutoff. Can also be specified by a center frequency f_o and a bandwidth B.
A lowpass filter will block all the frequencies above the cutoff f_o.
A highpass filter will only allow the frequencies above f_o.
A bandstop filter, also called a notch filter, will remove the frequencies around f_o.
Keep in mind that the skirt (transition region) of a filter is never vertical, so filters will not perfectly block frequencies near f_o and will allow a certain amount of the frequencies that were supposed to blocked.
Furthermore, some type of filters will allow a limited amount of gain variation in the pass band, this is called “ripple”.
Regulatory bodies specify this limitations by means of the “spectrum mask” which states clearly the amount of attenuation required in each frequency interval of interest.

Sampling

Analog Signal



Sampling Circuit

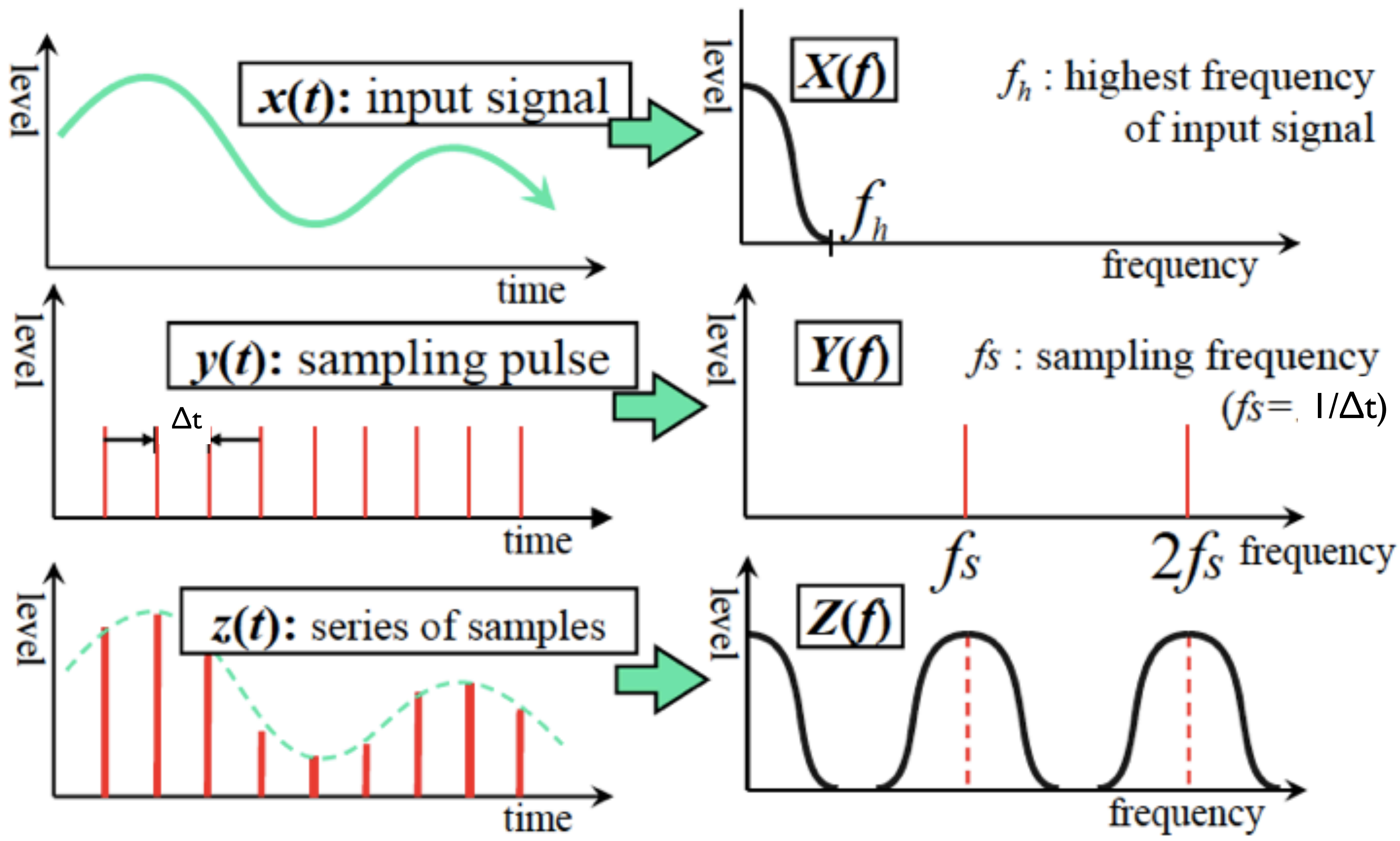


Sampled Signal

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 $\Delta t = 1/f_s$

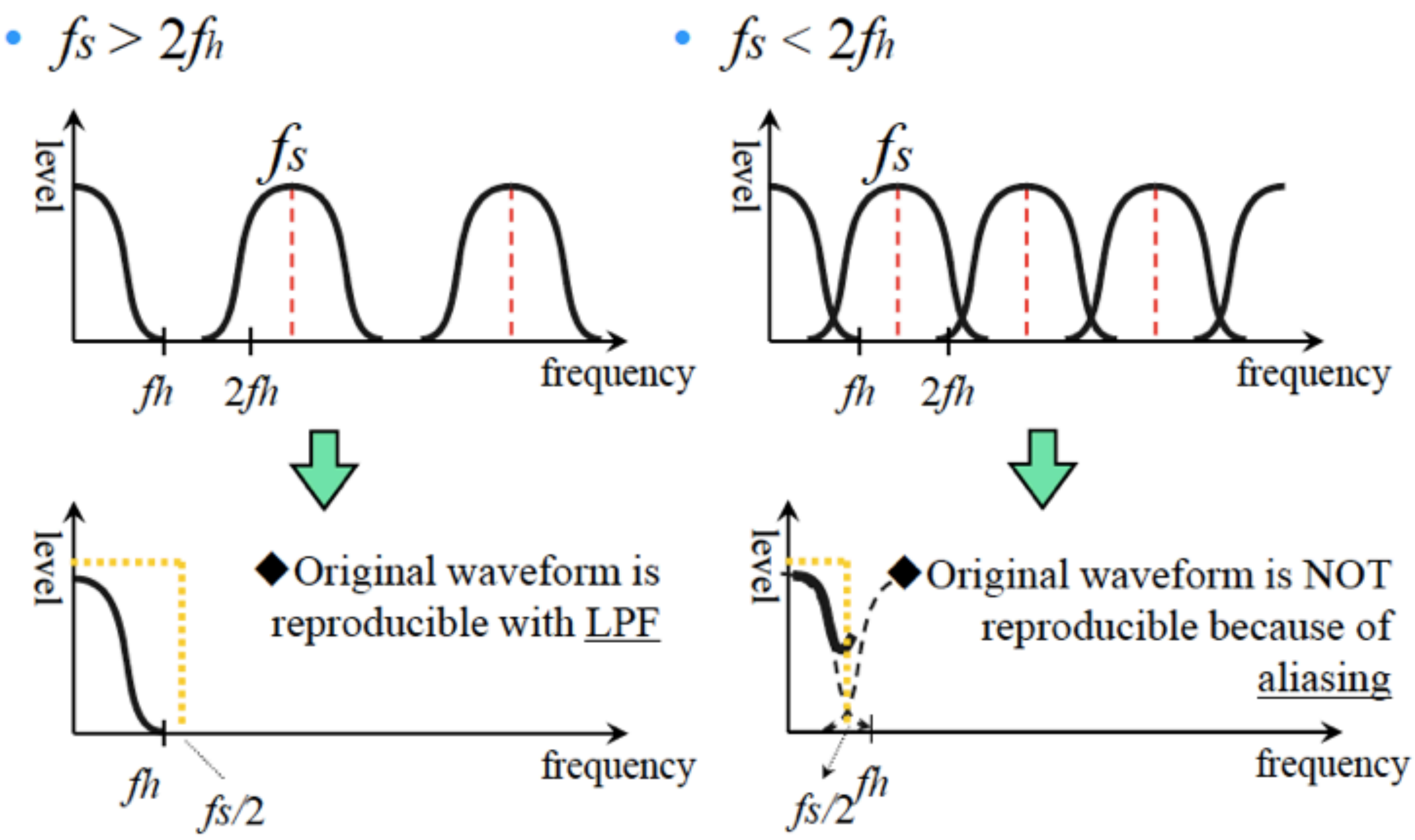
The minimum sampling frequency required is called the Nyquist frequency.
By symmetry, multiplying two signals in the time domain corresponds to performing the convolution of their spectra in the frequency domain.
The spectrum of a train of Dirac impulses separated every Δt seconds is another train of Dirac impulses separated every f_s hertz in the frequency domain. Therefore the sampling of the signal causes infinite replications of its spectrum every f_s Hz. To avoid overlapping, f_s must be at least twice f_h.
By low pass filtering the sampled signal, all the replicas of the spectrum above f_h are removed and we are left with the original signal.

Sampling



The sampled signal z(t) is the product of x(t) with y(t). In the frequency domain, this corresponds to the convolution of X(f) with Y(f) which produces a replication of the spectrum of x(t).

Aliasing and interpolation filter



If the sampling frequency is lower than twice the highest frequency present in the signal, overlap of the spectrum will occur and the recovered signal after filtering will be distorted.
To avoid this, the signal to be sampled is low pass filtered **prior** to the sampling process, to make sure that no frequency higher than half the sampling frequency is presented to the sampling circuit.
This low pass filter is therefore called an **antialising** filter.
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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.

In practice, the sampling and coding of the signal is done by a device known as ADC, normally an integrated circuit or a portion of a larger system on a chip. In the receiver, the opposite operation is performed by a Digital to Analog Converter (DAC) that will restore the original analog signal

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.

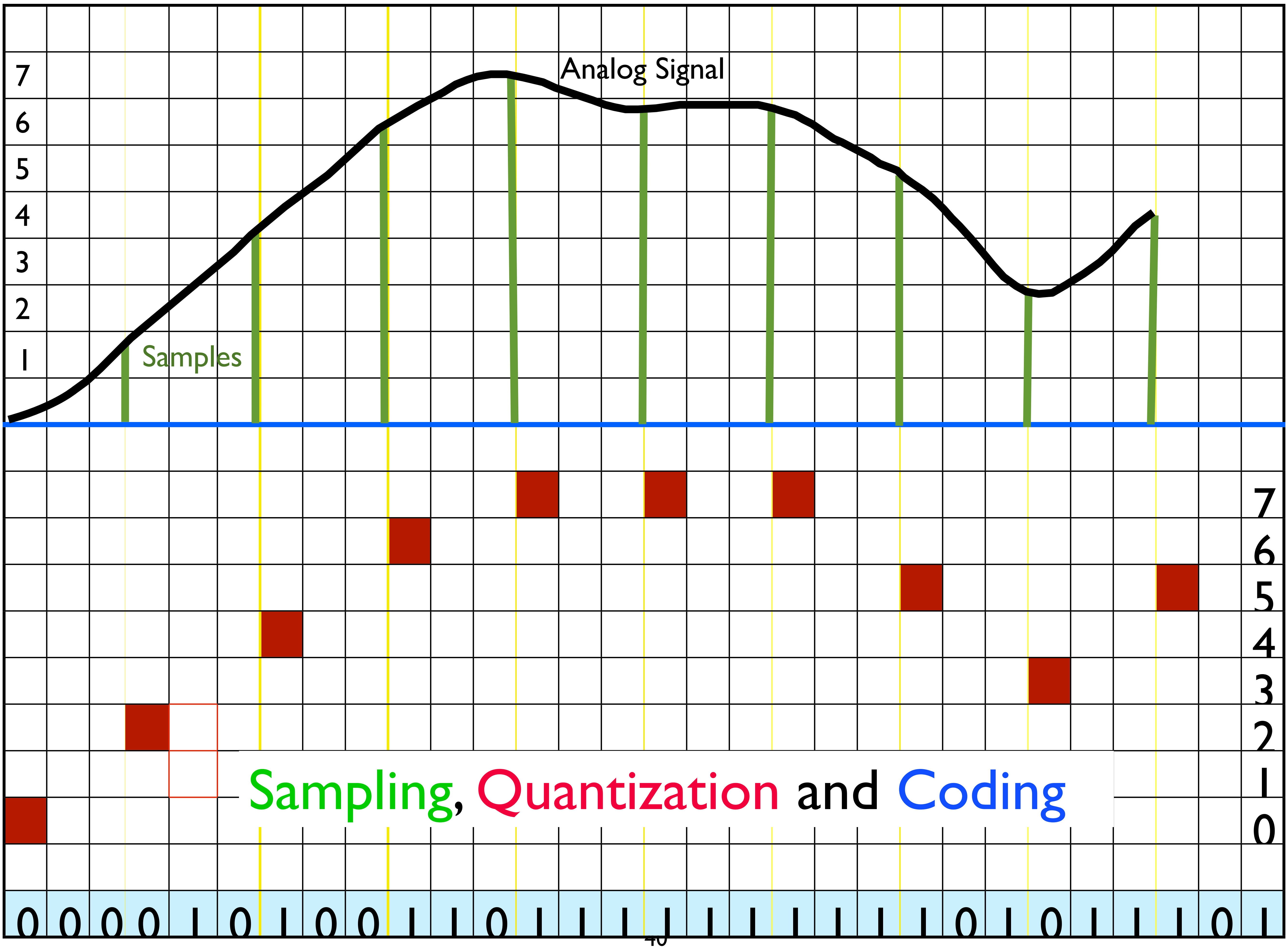
Digital signals can be encoded in ways that allow the recover from transmission errors, albeit at the expense of throughput.

Image Sampling

Normal, 72pixels/inch

Sampled Image, 10 pixels/inch

To visualize the effect of sampling, the image of the left has a normal resolution, while the one on the right is shown at 10 pixels per inch, emphasizing the effect of sampling and quantization.



Codification of a signal with 8 possible levels performed with 3 bit words. The voltage ranges from 0 to 7 volts divided in 8 quantization levels. The original analog signal is black. The sampling instant are represented by the yellow vertical lines.
The sampled signal is green. The sampled signal can have any value, so it is discrete in time but analog in voltage. The quantization process (Quantized signal in red), transforms the sampled signal into a digital multilevel signal (can only have discrete -numeric- values) which can be coded into a digital binary signal (will only have two possible values) shown with light blue background.
For example, the first sample has a value 1.8, it is quantized as the number 2, and coded in the 3 bit long word 0 1 0, the second sample with a value of 4.0 is quantized as the number 4 and coded as 1 0 0, the third sample has a value of 6.2, quantized as the number 6 and binary coded as 1 1 0, the fourth, fifth and sixth sample have slightly different values, but they are all quantized in the same number 7 (this shows the quantizing error incurred), coded as 1 1 1.
Although the quantization error cannot be corrected, its magnitude can be made as small as required by increasing the number of bits per sample. For instance, voice is normally coded with 8 bits per sample, but high fidelity music requires 14 bits per sample.

Recovering from errors

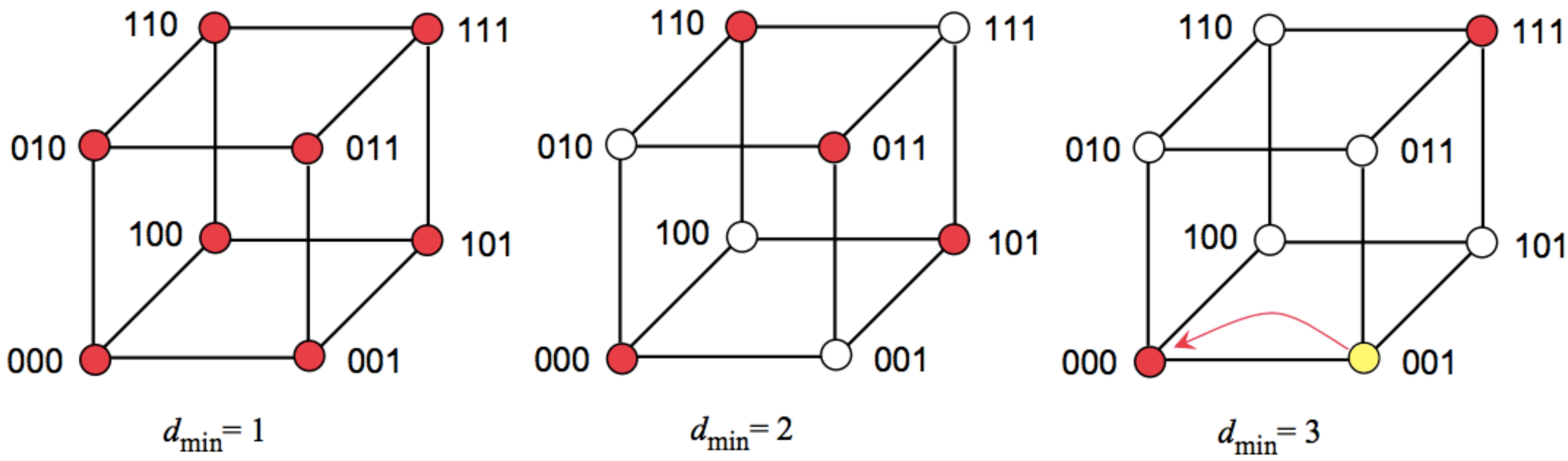
Two strategies to recover from transmission errors:

- 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

Channel coding is referred to the processes done in both transmitter and receiver consisting on allocating extra bits, parity bits, at the price of consuming extra bandwidth.
The redundancy allows the receiver to detect a limited number of errors that may occur anywhere in the message, and often to correct these errors without retransmission. FEC gives the receiver the ability to correct errors without needing a reverse channel to request retransmission of data, but at the cost of a fixed, higher forward channel bandwidth. FEC is therefore applied in situations where retransmissions are costly or impossible, such as one-way communication links, multicast and in satellite transmission.

Example of Coding

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

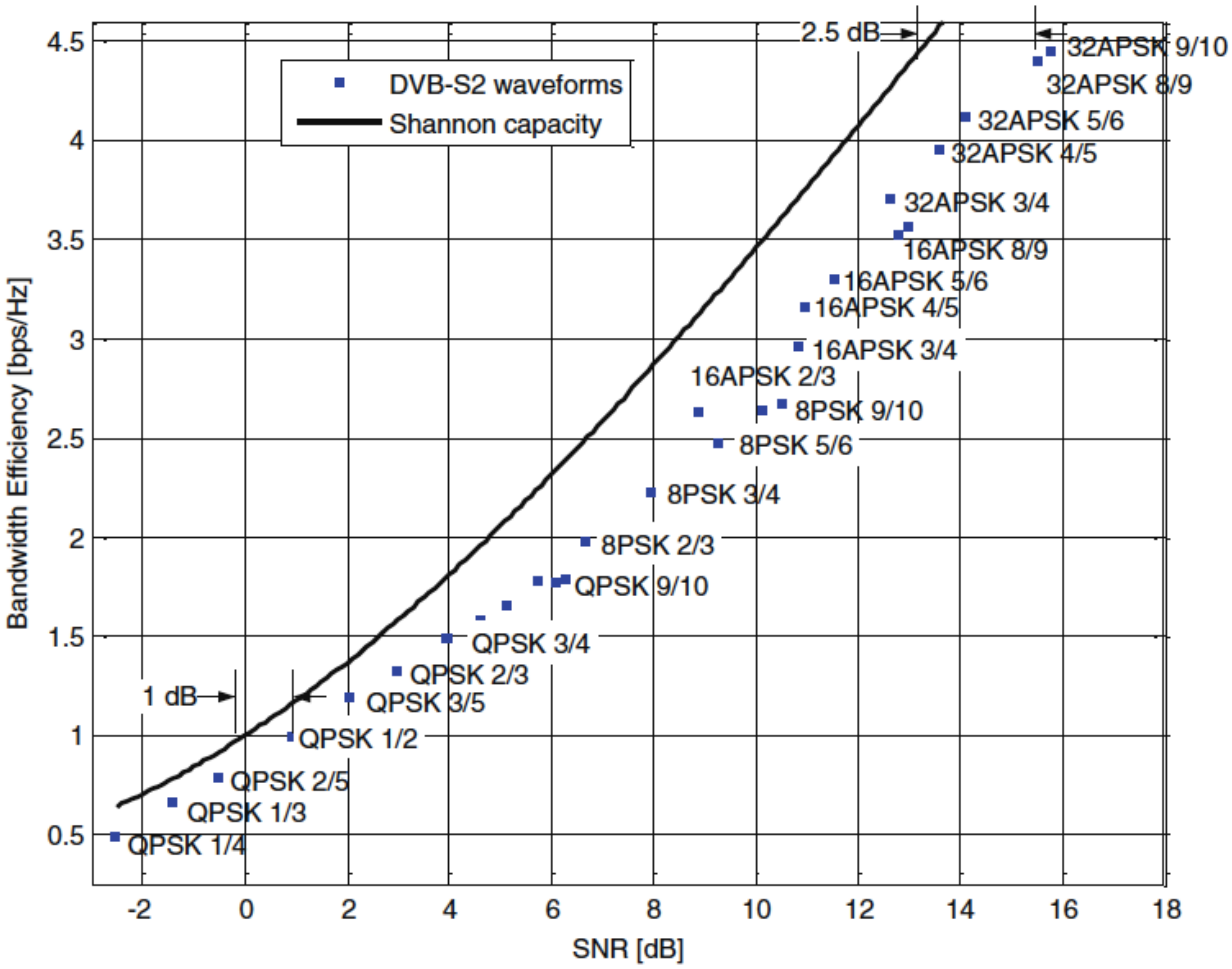


- Code rate $R_c = 1$
- No error correction
- No error detection

- 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



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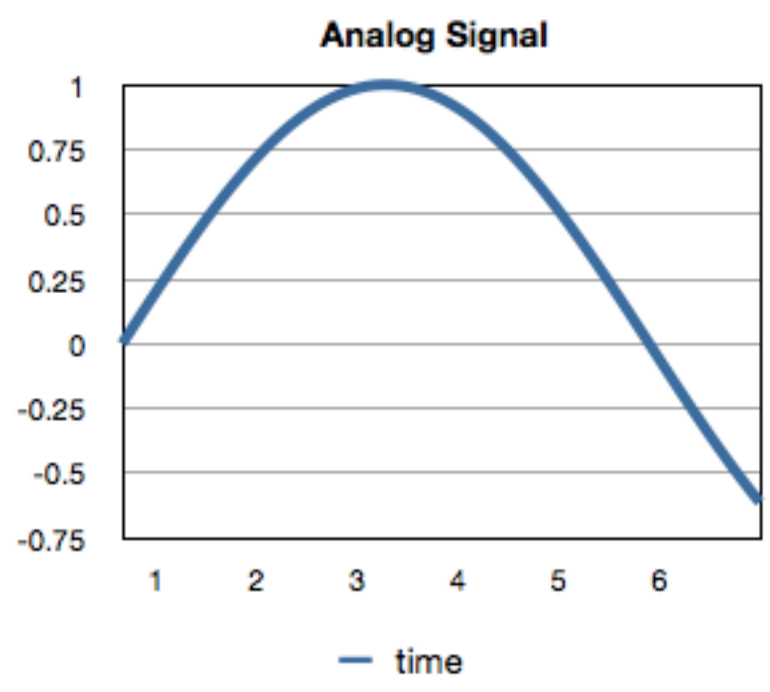
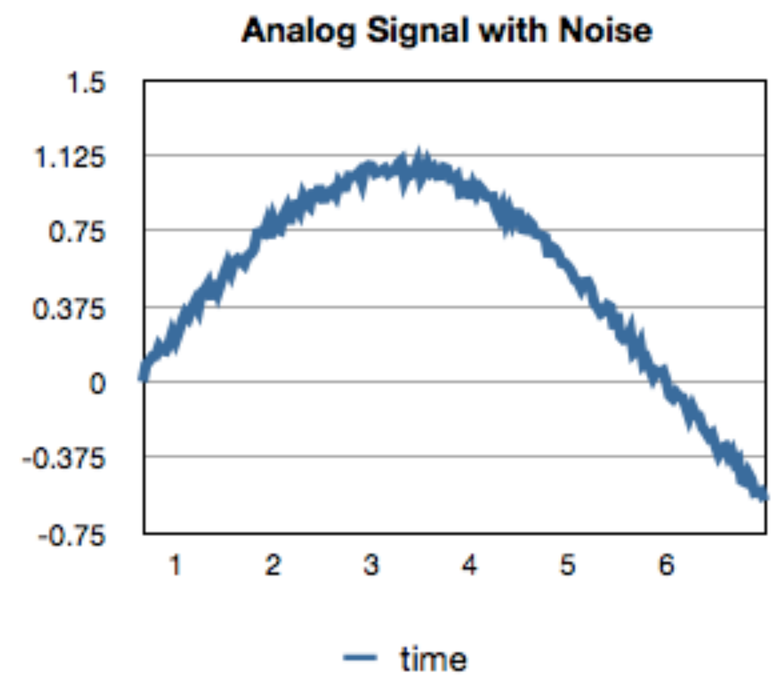
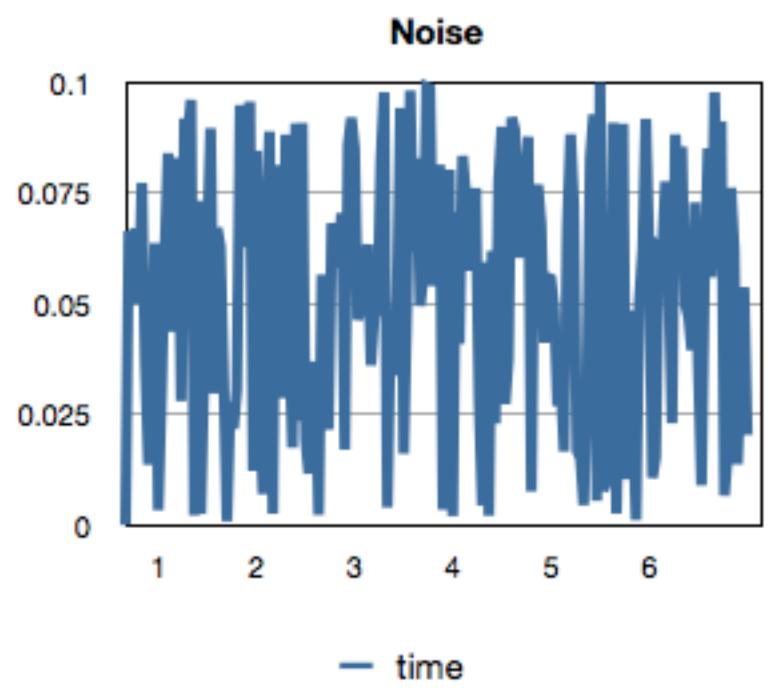
Noise in an analog Signal







Noise can completely masquerade the transmitted signal. Telecommunications engineers have strived for a century to find better ways to recover the information contained in the signal contaminated by noise.
Noise is due to the random movement of electrons in any electronic device. As the temperature increases, so the does the amount of random movements.
The most prevalent type of Noise is Thermal Noise, called White Noise, because it has a power spectrum flat over all frequencies of interest. The White Noise power is proportional to the temperature in kelvins (K) and the bandwidth, and the proportionality factor is Boltzmann constant, k= 1.38x10^-23 J/K
So, the greater the bandwidth of the receiver, the greater the amount of noise that will enter the system.

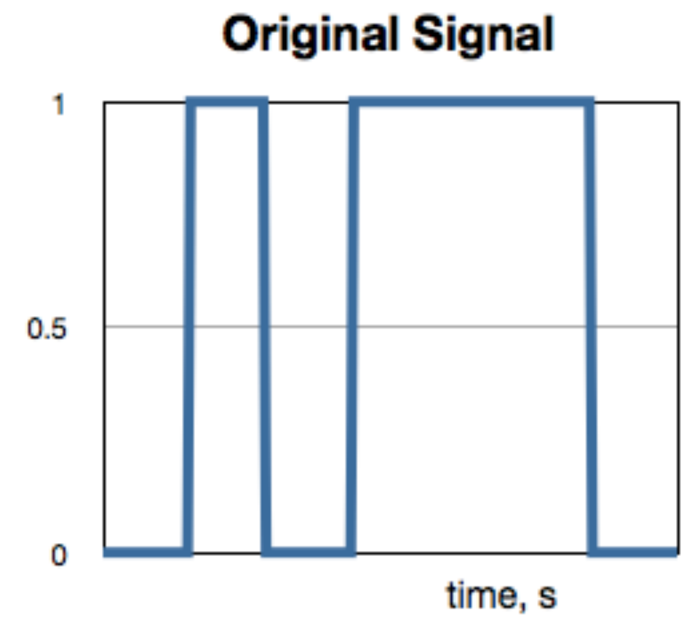
Electronic Noise

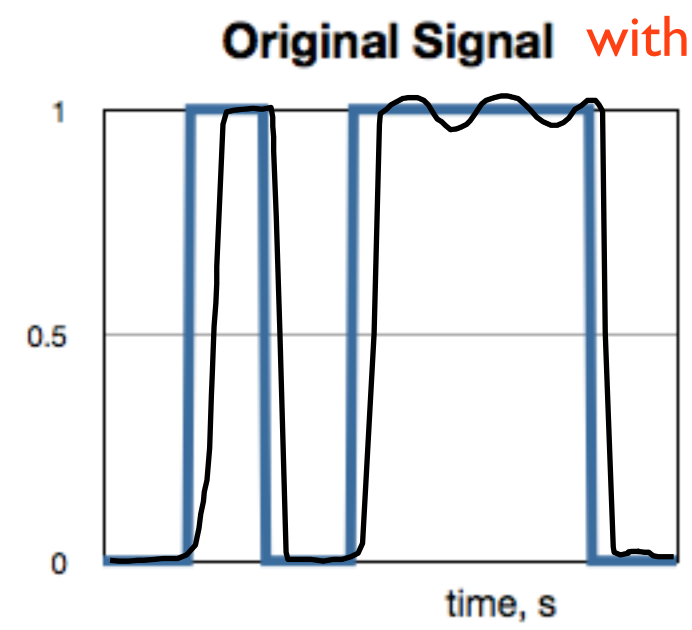
- Noise poses the ultimate limit to the range of a communications system
- Every component of the system introduces noise
- There are also external sources of noise, like atmospheric noise and man made noise
- Thermal noise power (always present) is frequency independent and is given (in watts) by kTB,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

The symbol for absolute temperature is K, not degrees kelvin. Boltzmann constant is lower case k.
$dBm = 10 \log_{10}[(kTB)/0.001]$
Noise power density in W/Hz is $N_o=kT$
For an 8 MHz wide channel the thermal noise is −135 dBm
The amount of noise introduced by a component is often expressed as the Noise Figure in dB, which must be added to the thermal noise to calculate the total amount of noise affecting the receiver.

Bandwidth Limitation





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Every real channel will be limited in bandwidth, the effect is that of low pass filtering the signal, which means that any sharp transition in the input signal will be smoothed out by the channel and also some "ringing" will appear when the original signal was stable. Normally this effect will be taken care by the proper sampling of the received signal, but if it is too pronounced it can lead to transmission errors.

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.

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Transmitting the word "**quiz**" is an example of information content. When transmitting the first letter the receiver has to guess among all the 28 possible letters of the alphabet ($Pe=1/28$, $\log_2(28) = 4.8$ bits), but after receiving the letter **q**, the only possibility is that the following letter will be a **u**, so the transmission of the **u** carries no information, The transmission of the **i** carries a little more information because we know that we must receive a vowel, so we only have to guess among 5 possibilities, $I= \log_2(5) = 2.3$ bits

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.

Coming back o the transmission of the word “quiz”, if a transmission error prevents the reception of the first letter, the reception of the second letter now becomes very important because it let’s us reconstruct the word and recover the lost letter.

Digital Channel Capacity

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

Capacity, bit/s

B, bandwidth, Hz

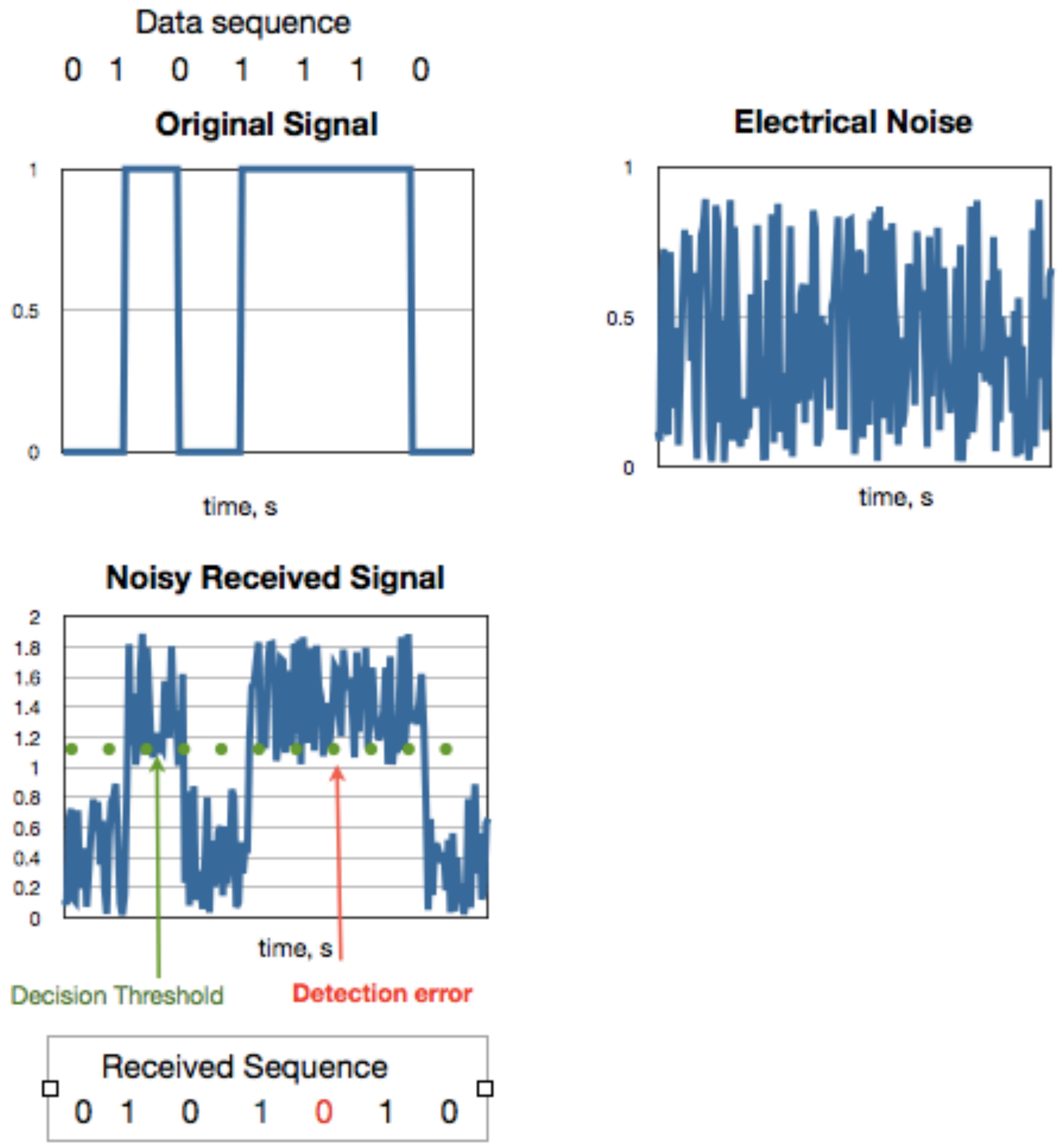
Signal power, W

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

Total noise power is N_oB in W. The energy per bit E_b in J is also a significant parameter to compare different digital modulation schemes, E_b/No is equal to S/N divided by the number of bits per symbol. One symbol may transmit several data bits.
The bandwidth efficiency, or spectral efficiency, is an important figure of merit for communication systems because bandwidth is a scarce and valuable resource, so designers strive to pack as many bit/s in a certain amount of bandwidth as possible. This increases the complexity of the system and also the required S/N in order to correctly decode the received signal. That is why most systems provide greater capacity at shorter distances where the signal power is greater.
Since capacity in bit/s is proportional to bandwidth in Hz, it is common to speak of “bandwidth” in bit/s. Bear in mind that the number of bits carried by each hertz can be as low as 1/2 or as high as 8.

Detection of a noisy signal

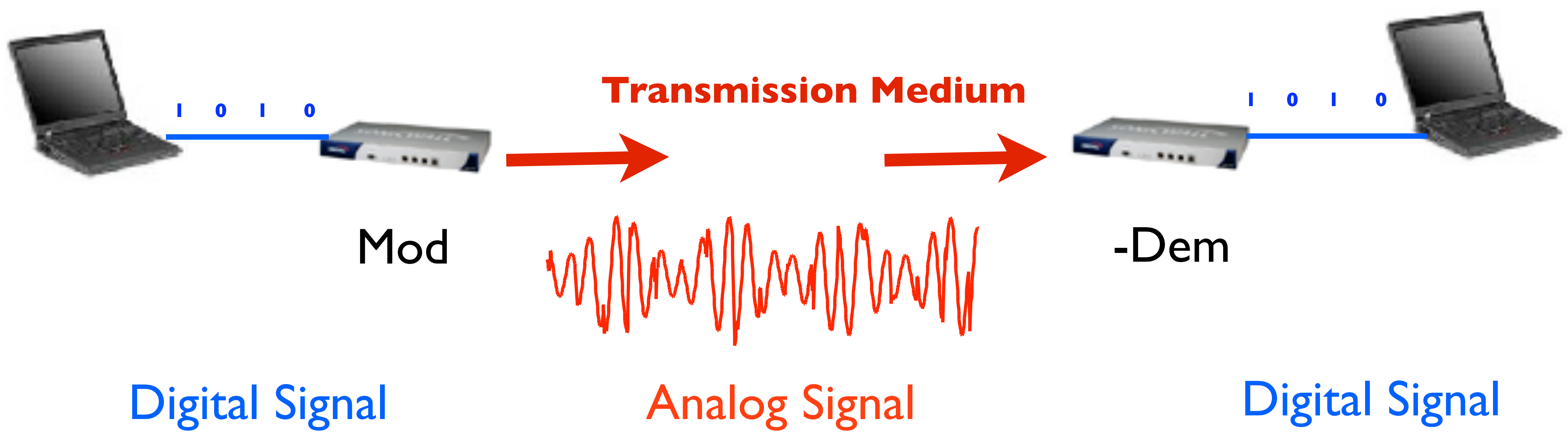


In the figure, the original data consists of the **0 1 0 1 1 1 0** sequence. The **0s (zeros)** are represented as zero volts and the **1s** as 1 V. As the signal moves towards the receiver, its amplitude will diminish. This effect is called "attenuation" and is shown in the slide 18. Likewise, there will also be a delay as the signal moves from the transmitter to the receiver. Each of these impairments, if severe enough, can cause a detection error.

An amplifier can be used to overcome the attenuation, but the electrical noise always present in the system will add to the received signal. The noisy received signal is therefore quite different from the original signal, but since it is a digital system we can still recover the information contained by sampling the received signal and comparing the value at the sampling time with a suitable threshold voltage. In this example the noise received signal has a peak of 1.8 V, so we might choose e threshold voltage of 0.9 V. IF the received signal is above the threshold, the detector will output a digital **1**, otherwise, it will output a **0**. In this case we can see that because of the effect of the noise the fifth bit was erroneously detected as a zero.

Transmission errors can also occur if the sampling signal period is different from that of the original data (difference in the clock rates), or if the receiver clock is not stable enough (jitter). Any physical system will have an upper limit in the frequencies that will transmit without attenuation (the bandwidth of the system), so the abrupt rise and fall of the voltage will be smoothed out as the signal goes through the channel. Therefore, we must make sure that each of the elements of the system has enough bandwidth to handle the signal. On the other hand, the greater the bandwidth of the receiver system, the greater the amount of the noise that will affect the received signal.

MoDem



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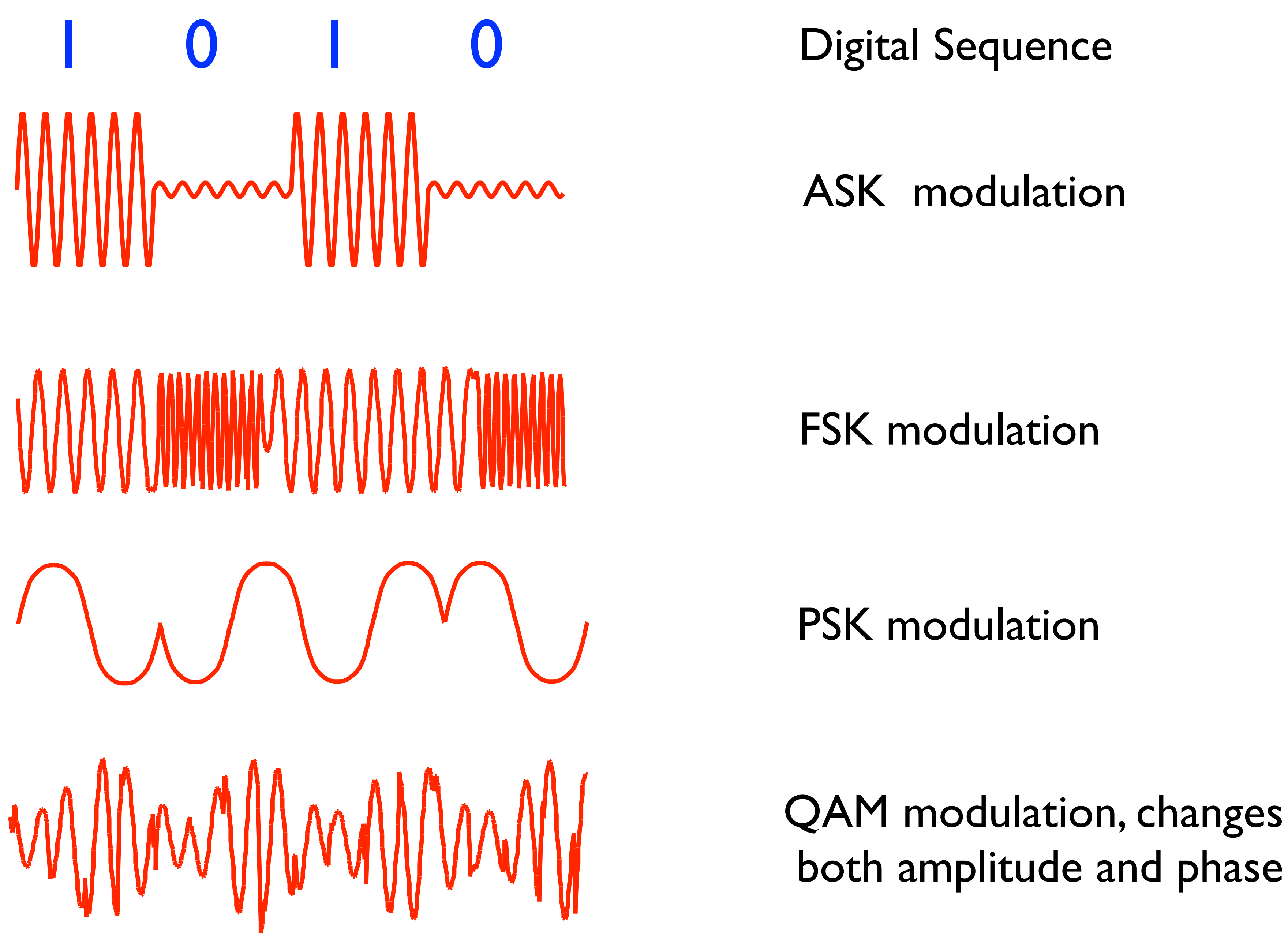
In order to transmit a digital signal at a reasonable distance it has to be processed by a modulator.
The modulator can:
Select the frequency at which the signal will be transmitted over the channel.
Allow for different signals to share the same modulation channel, in a process known as multiplexing.
Adapt the signals parameters to suit the requirements of a given channel (bandwidth, spectral properties, noise robustness, etc.).
Provide the flexibility to exchange spectral efficiency for robustness, as needed.
Of course, at the receiving end, the inverse operation, called demodulation, needs to be performed. So in bidirectional systems a single device will perform both operations and therefore will be called a modem.

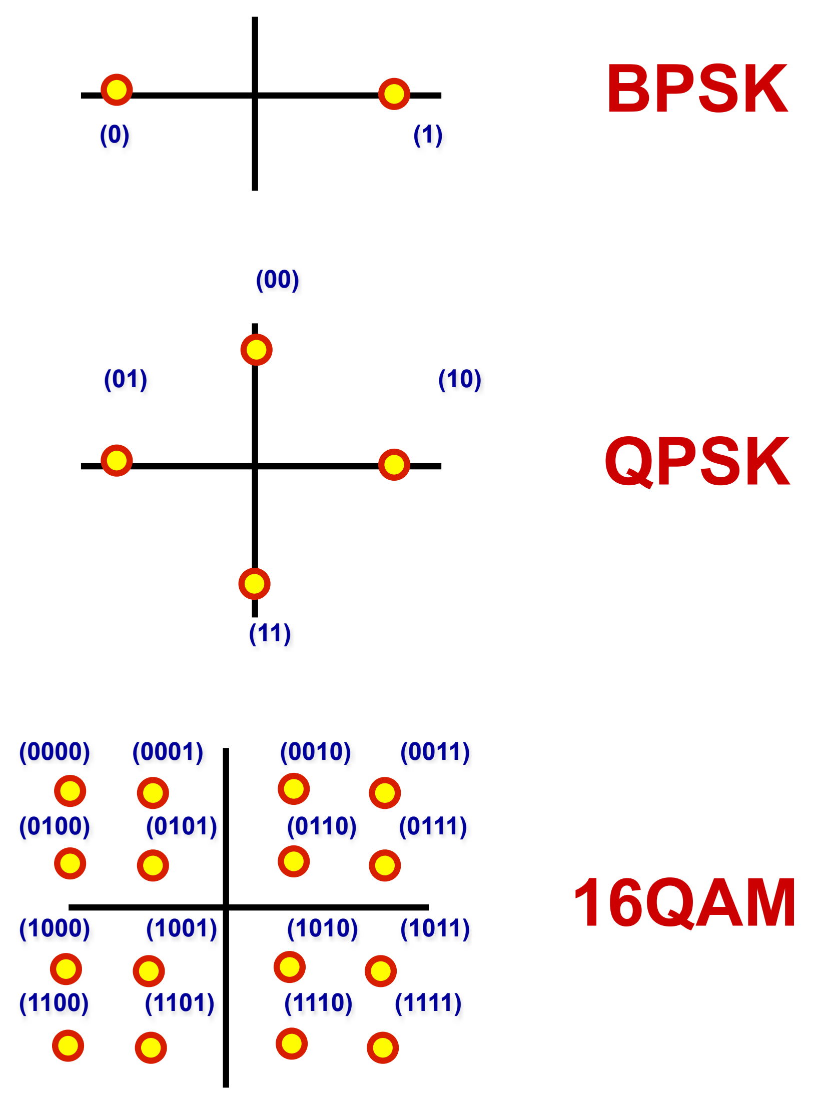
The word modem is a combination of the words modulation and demodulation which is precisely what a modem does. A modem can also be viewed as a device that takes digital information, transfers it on to a medium to allow transportation of the information, and at the other end, removes the information from the medium and restores it to it's original digital form. This brings up two distinguishing characteristics of a modem, the type of information it accepts and the media that it operates upon. In the case of WiFi modems, the information is data 10BT or 100BT Ethernet format and the media radio.
The type of medium employed by the modem dictates the type of modulation it will employ, The medium can be a copper cable, an optical fiber or an electromagnetic wave in free space.
Although the modem is a separate building block, it is often embedded in a laptop or in a wireless router.

Comparison of modulation techniques

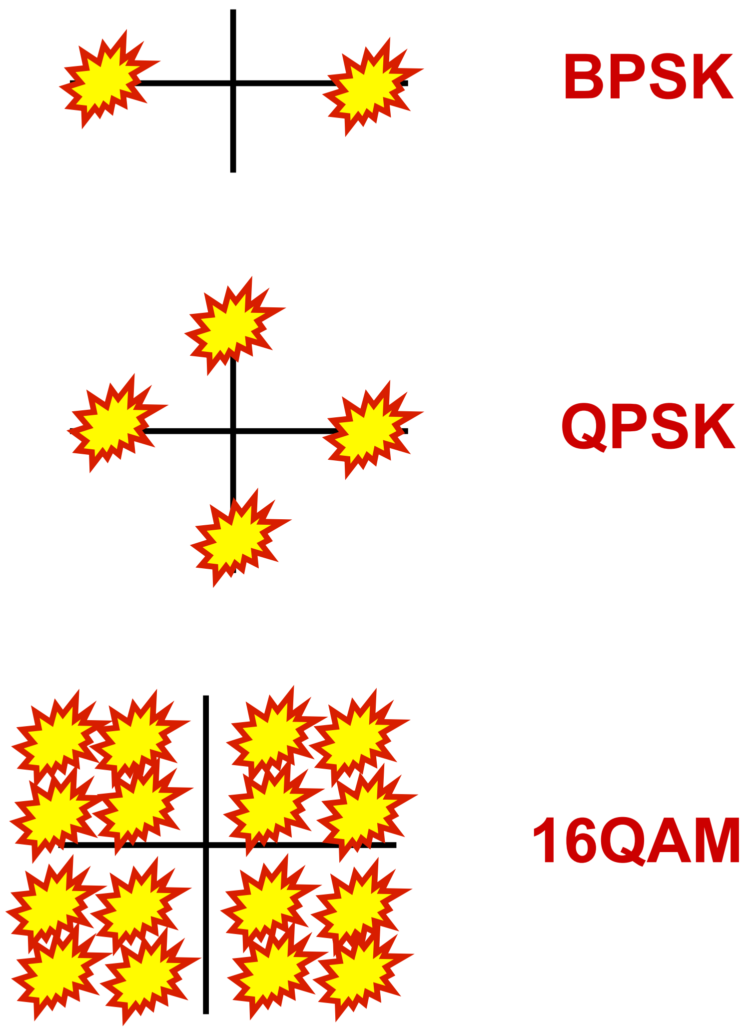


The digital sequence 1 0 1 0 is shown modulating a sinusoidal carrier in ASK (Amplitude Shifting Keying), FSK (Frequency Shifting Keying), PSK (Phase Shifting Keying) and QAM (Quadrature Amplitude Modulation). Quadrature modulation is another term used for binary phase modulation. There is a great number of modulation techniques derived from these basic schemes.

Binary Modulation Constellation



Effect of noise in the detection



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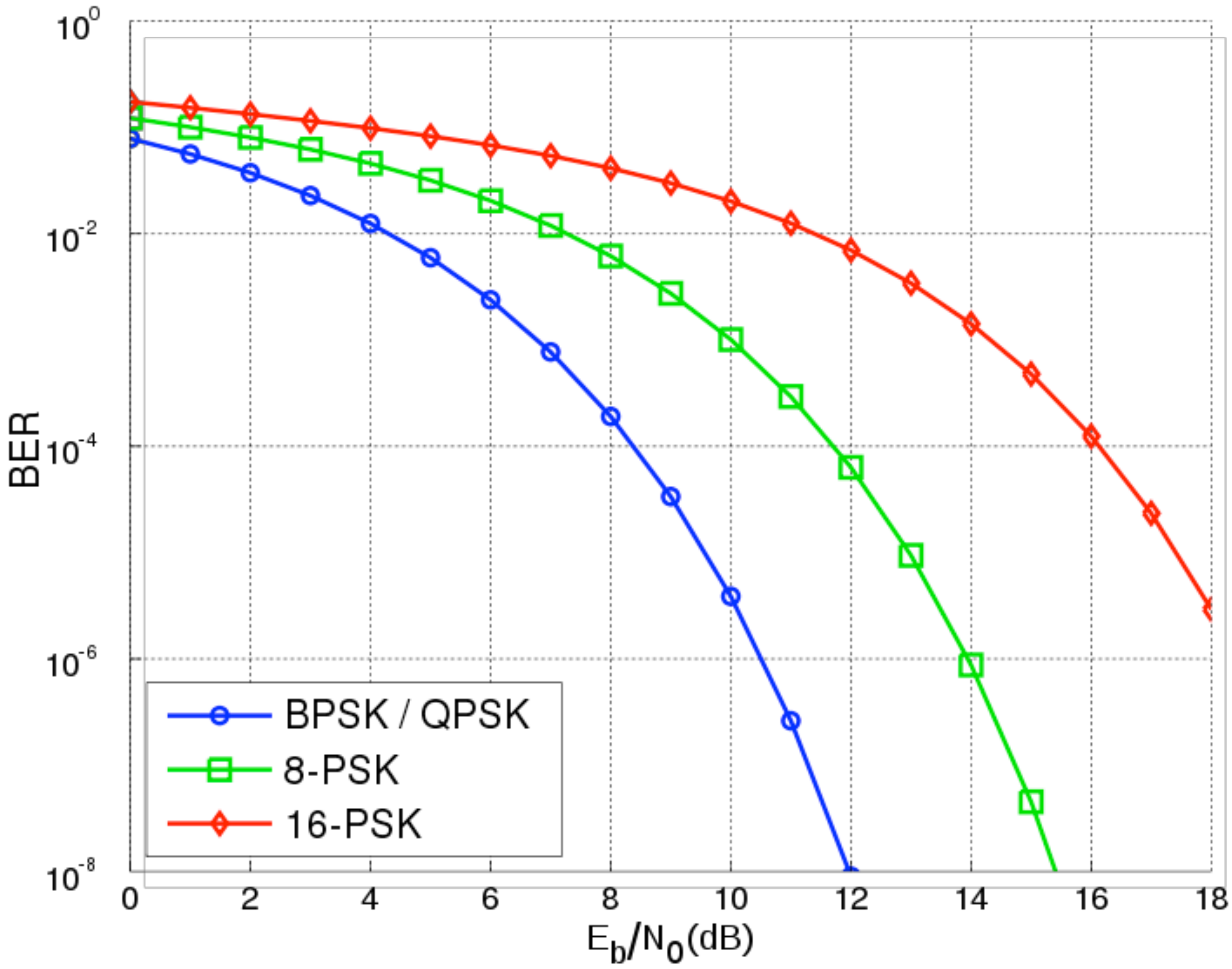
BER Versus E_b/N_o



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

The **bit error rate** (**BER**) is the number of erroneous received bits divided by the total number of transferred bits, often expressed as a percentage.

E_b is energy per bit. N_o is noise spectral density. An Additive White Gaussian noise (AWGN) channel is assumed, as a typical case.

The graph shows the very non-linear relationship between the BER and the E_b/N_o for different modulation schemes. For a BER of 10^{-4}, the signal energy to noise ratio has to be 8.3 dB in very robust modulation schemes like BPSK (Bipolar Phase Shifting Keying). If we use 8-PSK (eight states Phase Shifting Keying), we need an $\mathbf{E_b/N_o}$ of 10.7 dB. This is the price to be paid in order to be able to distinguish among 8 possible states of the received signal instead of 2. But now we can pack 3 bits per each transmitted symbol. If we use 16-PSK, we can code 4 bits per symbol, but we will need 16 dB of $\mathbf{E_b/N_o}$.

There is always a trade-off between the $\mathbf{E_b/N_o}$ and the data rate we can achieve with the same error probability. We can pack more bits per symbol, thus achieving better throughput, but we need to have stronger signal. In practice this means that the closer you are to the transmitter, the faster the data transfer that you can get.

Often a BER of 10^{-5} is the target for radio channel. What will the required S/N be for 8-PSK modulation?

$\mathbf{E_b/N_o} = (\mathrm{bits}/_{\mathrm{Symbol}})\mathbf{S/N}$

On fiber optics communication channels there is very little noise, so a BER of 10^{-9} is the norm.

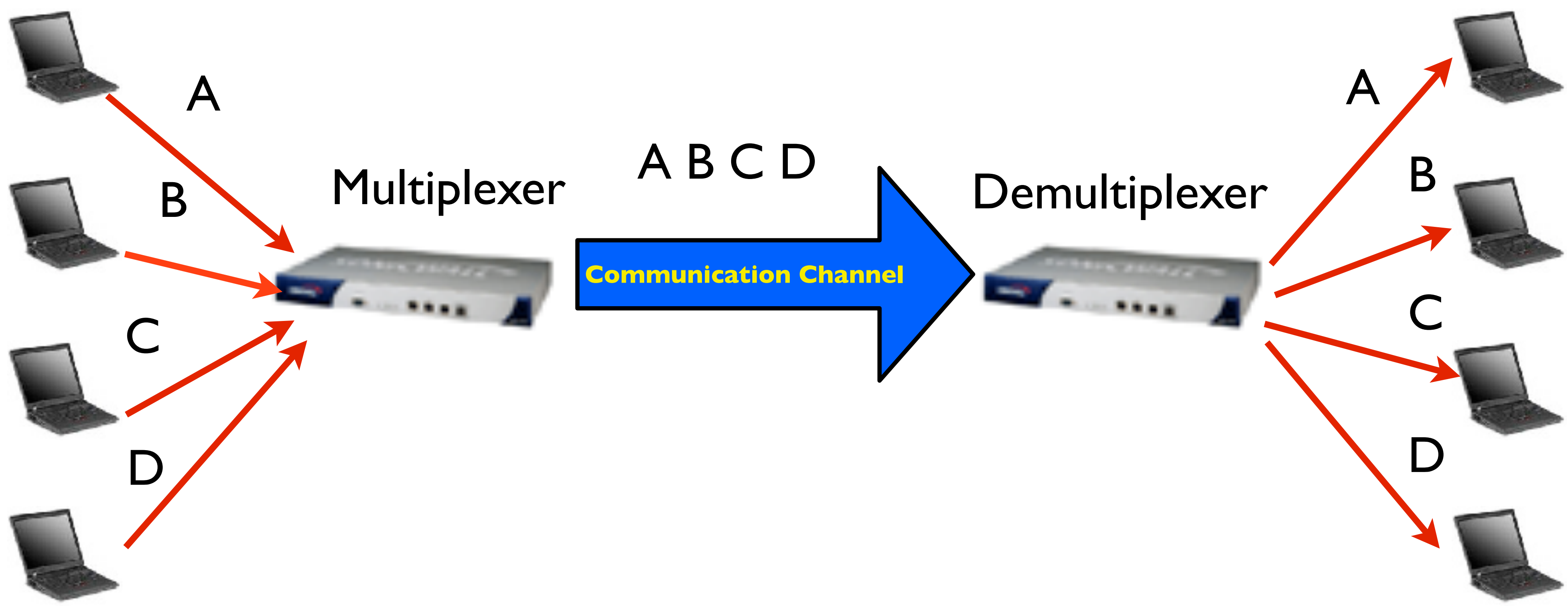
Comparison of modulation types

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

E_b: Energy per bit
N_o: Noise spectral density, W/Hz
In this example we assume a different model for the channel and aim to a BER of 10^{-6}, we get an interesting comparison of modulation schemes.
Notice that both 16-PSK and 16-QAM (Quadrature Amplitude Modulation) offer the same number of bits per transmitted symbol, yet there is a 3 bit difference in the required E_b/N_o. This is due to the fact that noise in the channel will affect more the amplitude than the phase of the signal, at higher Eb/No ratios. This is not the case at a Eb/No of 10.1 dB where the performance of 4-PSK and 4-QAM is the same.

Multiplexing



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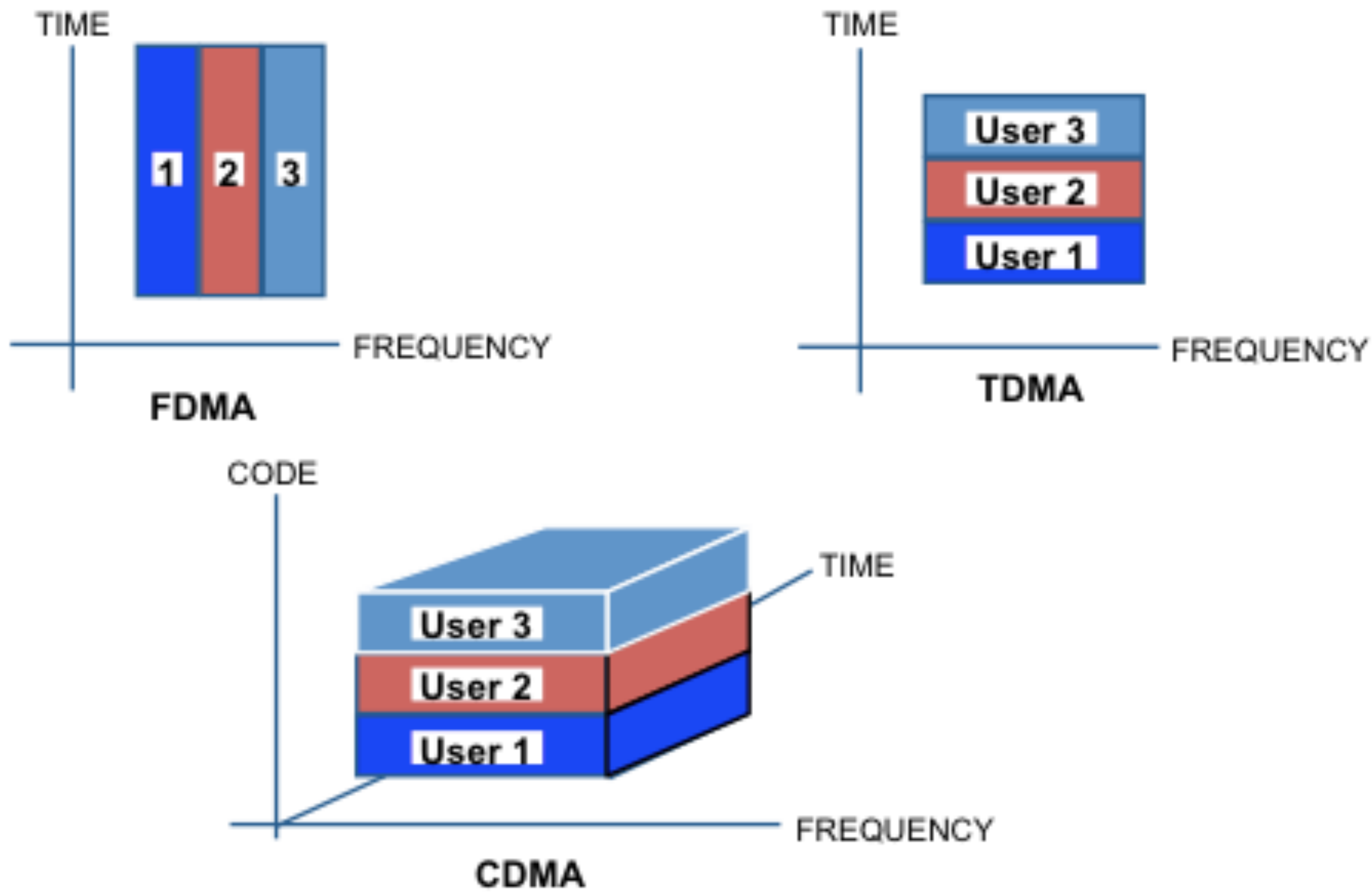
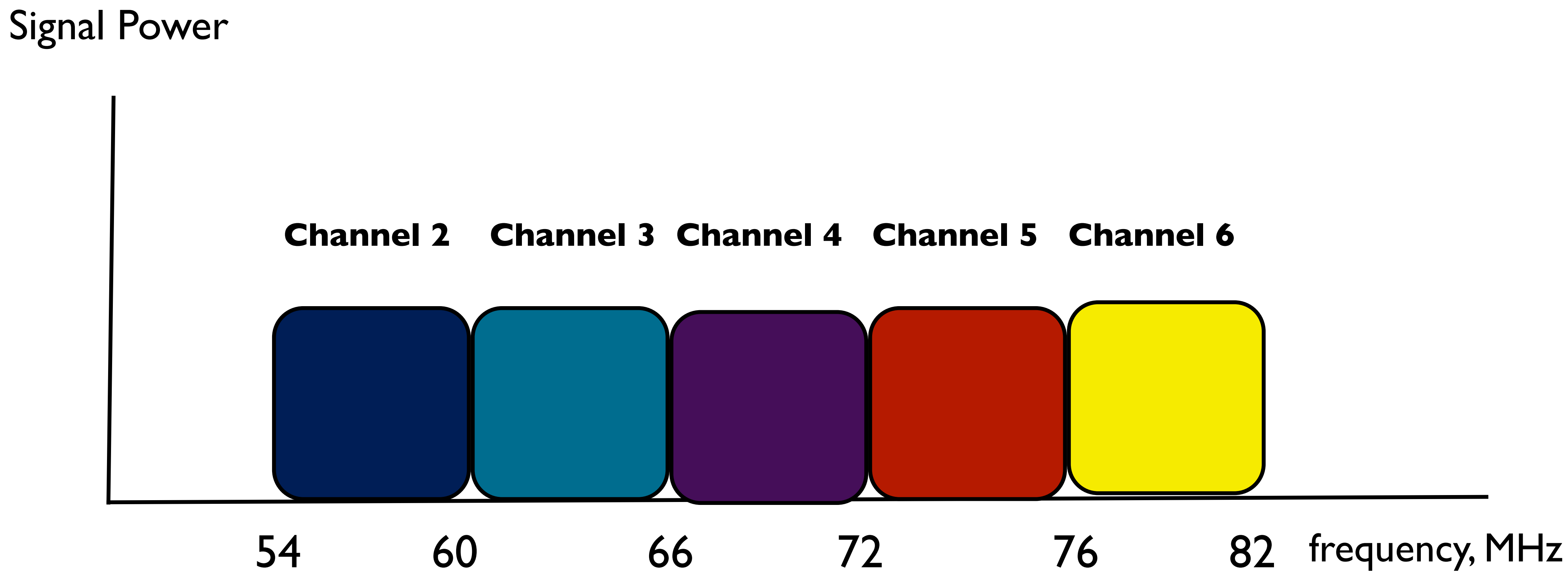
Multiplexing is the sharing of a single communication channel among different users. The communication channel can be a copper wire, an optical fiber, or the space between a transmitting and a receiving antenna.
Different users can be distinguished by means of different frequencies, time slots, codes or regions of space.
At the receiving end the opposite operation must be performed to retrieve the individual streams and deliver them to the corresponding destination

Medium sharing techniques



In FDMA (Frequency Division Multiple Access), each user has a different frequency band allocated.
In TDMA (Time Division Multiple Access), each user has a different time slot allocated, while the same frequency is shared among all the users of the service.
IN CDMA (Code Division Multiple Access), the users are distinguished by means of a special mathematical code, while sharing the same frequency and time slots.

Example:
U.S. Television Channels Allocation



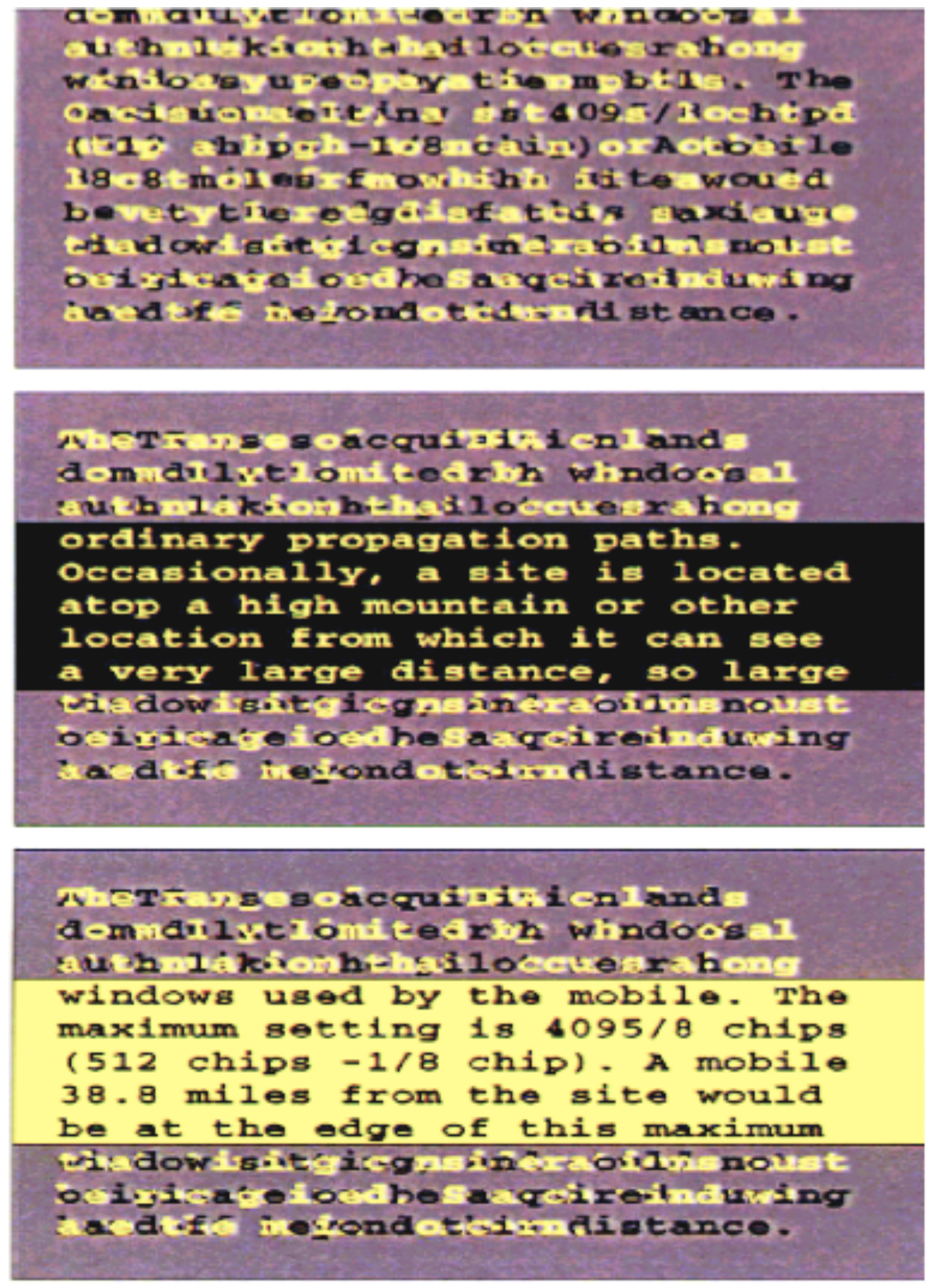
Nominal bandwidth per channel is 6 MHz, but since the filters used at the transmitter are not perfect, there is always some “bleeding” of the signal in adjacent channels.
So, in any given area if channel 2 is being used, channel 3 will be left vacant and the next usable channel will be channel 4, and then 6 and so on. These “guard bands” that most be kept empty to avoid interference are a waste of valuable spectrum, and, in the case of TV, are called “White Spaces”.
Digital TV has a narrower spectrum and these White Spaces can be claimed to be used for other applications in what is called the “Digital Dividend”.
Alternatively, sophisticated modulation techniques can be employed to remove the adjacent channel’s interference, by means of OFDM (Orthogonal Frequency Diversity Modulation) which is employed in many newer systems to make better use of the available spectrum.

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



This visual analogy is aimed at explaining how two different messages can be superimposed in the same medium and later separated by the proper decoding technique.

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/