2. TELECOMMUNICATIONS BASICS

The purpose of any telecommunications system is to transfer *information* from the sender to the receiver by a means of a communication *channel*.
The information is carried by a *signal*, which is certain physical quantity that changes with time.
The signal can be a voltage proportional to the amplitude of the voice, like in a simple telephone, a sequence of pulses of light in an optical fibre, or a radio-electric wave irradiated by an antenna.
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***. Any analog signal can be converted into a digital signal by appropriately ***sampling*** and then coding it. The sampling frequency must be at least twice the maximum frequency present in the signal in order to carry ***all*** the information contained therein. Random signals 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. A signal can be characterised by its behaviour over time or by its frequency components, which constitute its spectrum. Some examples of signals are shown in Figure TB 1.

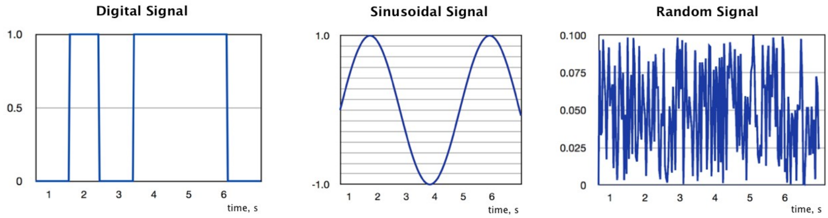

Figure TB 1: Examples of signals

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. So a signal can be characterised either by a graph of its amplitude over time, called a waveform, or a graph of of the amplitudes of its frequency components, called a spectrum.

Figure TB 2: Waveforms, Spectrum and filters

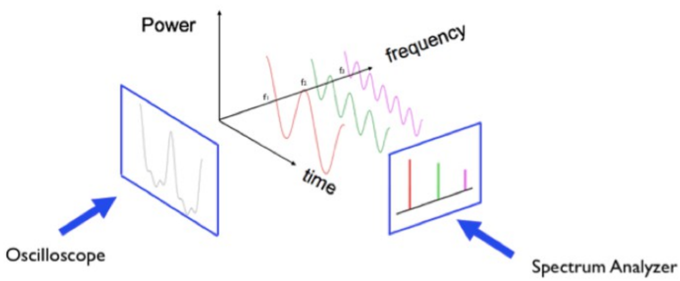

Figure TB 2 shows how the same signal can be seen from two different perspectives. The waveform can be displayed by an instrument called an oscilloscope, while the spectrum can be displayed by what is called a Spectrum Analyzer. The spectrum distribution relays very important information about the signal and allows for the intuitive understanding of the concept of filtering of 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 we pass this signal through a device that will remove f_2 and f_3, the output is a pure sinusoidal at frequency f_1.

We call this operation "**Low Pass filtering**" because it removes the higher frequencies. Conversely, we can apply the signal to a "**High Pass Filter**", 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 radio-electric spectrum, so every device will always perform some extent of filtering to 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 (number of cycles per second).

While travelling through the communication channel, the signal is subject to *interference* caused by other signals and is also affected by the electrical *noise* always present in any electrical or optical component. ***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.

Consequently, the received signal will always be a distorted replica of the transmitted signal, from which the original information must be retrieved by appropriate means to combat the effect of interference and noise. Furthermore, the received signal will be subject to *attenuation* and *delay* that increase with the distance between the transmitter and the receiver.

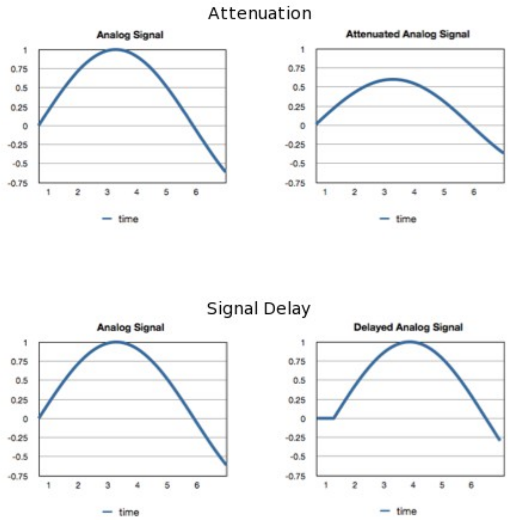

Figure TB 3: Attenuation and delay

Although it is relatively simple to restore the amplitude of signal by means of an electrical *amplifier*, the components of the amplifier will add additional noise to the signal, so at very long distances where the received signal is feeble, the amplifier will produce a signal so garbled with noise that the information originally transmitted will no longer be retrievable.

One way to address this problem consists in converting the continuous quantity carrying the information into a sequence of very simple *symbols* which can be easier to recognise even at great distance. For instance, the flag of a ship is a convenient way to distinguish the nationality of the ship even at distances at which the letters on the hull cannot be read.
This technique has been extended to carry generalised messages by assigning different position of flags to every letter of the alphabet, in an early form of long distance telecommunications by means of *digital* or *numeric* signals.
The limitation of this method is obvious; to be able to distinguish among, say, 26 symbols corresponding to each letters of the alphabet, one must be quite close to the communicating ship.

On the other hand, if we code each letter of the alphabet in a sequence of only two symbols, these symbols can be distinguished at much longer distance, for example the dot and dashes of the telegraph system.
The process of transforming a continuous analog signal into a discontinuous digital one is called Analog to Digital Conversion (**ADC**), and conversely we must have a Digital to Analog Converter (**DAC**) at the receiving end to retrieve the original information.
This is the reason why most modern telecommunication systems use digital binary signals to convey all sorts of information in a more robust way. The receiver must only distinguish between two possible symbols, or in other words between two possible values of the received *bit* (**_bi_nary dig_it_**). For instance, the CD has replaced the vinyl record, and analogue television is being replaced by digital television. Digital signals can use less bandwidth, as exemplified by the "***digital dividend***" currently being harnessed in many countries which consists in bandwidth that has become available thanks to the transition from analog to digital transmission in TV broadcasting.
Although in the process of converting from an analog to a digital information system there is always some loss of information, we can engineer the system so as to make this loss negligible.

Figure TB 4: Undersampled Image

For example, in a digital camera we can choose the number of bits used to record the image.
The greater the number of bits (proportional to the amount of *megapixels*), the better the rendering, but more memory will be used and longer time to transmit the image will be needed.
So most modern communication systems deal with digital signals, although the original variable that we want to transmit might be analog, like the voice.

It can be shown that any analog signal can be reconstructed from discrete samples if the sampling rate is at least twice as high as the highest frequency content of the signal.

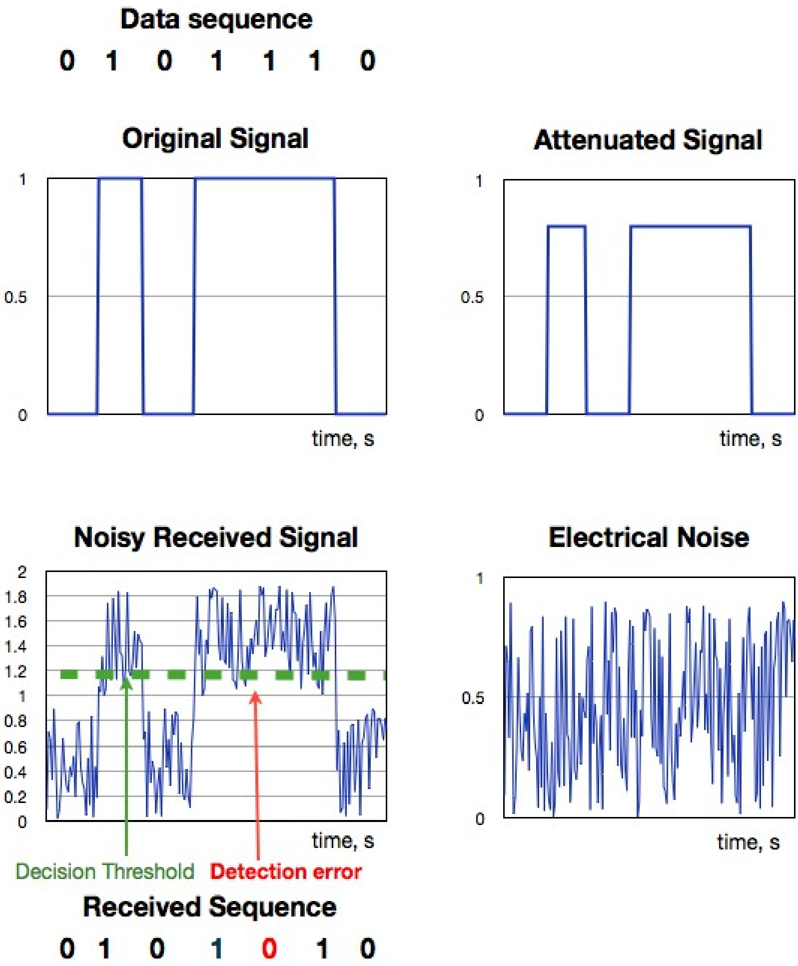

Figure TB 5: detection of a noisy signal

Then each sample is coded in as many bits as necessary to achieve the desired amount of precision.

These bits can now be efficiently stored or transmitted, since for the recovery of the information one needs to distinguish among only two states, and not among the infinite nuances of an analog signal.

This is shown in Figure TB 5, where the original data consists of the **0 1 0 1 1 1 0** sequence. The **0's** are represented as zero volts and the **1's** as 1 V. As the signal moves towards the receiver, its amplitude will diminish. This effect is called "attenuation" and is shown in the figure. Likewise, there will also be a delay as the signal moves from the transmitter to the receiver, the variability in the delay of the received signal is called ***jitter***. Attenuation, noise or jitter (or their combination) 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 in a digital system we can still recover the information contained by sampling the received signal at the correct time 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 1.1 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 faithfully (the bandwidth of the system), higher frequencies will be blocked, 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.

Modulation

The robustness of the digital signal is also exemplified by the fact that it was chosen for the first trials of radio transmission. Marconi showed the feasibility of long distance transmission, but pretty soon realised that there was a need to share the medium among different users.

This was achieved by assigning different *carrier* frequencies which were *modulated* by each user's *message*. *Modulation* is a scheme to modify the *amplitude*, *frequency* or *phase* of the carrier according with the information one wants to transmit. The original information is retrieved at destination by the corresponding demodulation of the received signal.

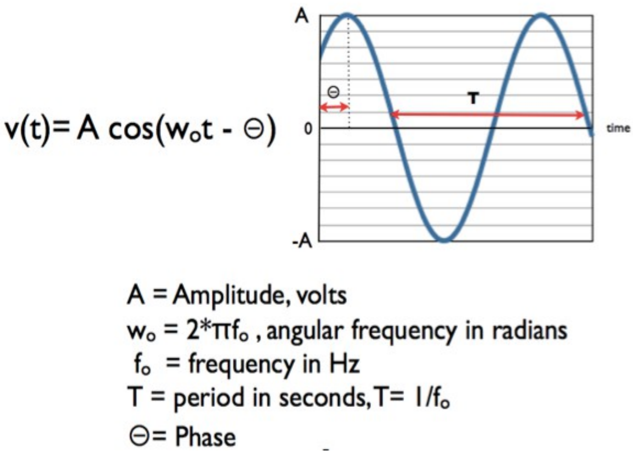

Figure TB 6: Sinusoidal Carrier Signal

Figure TB 6 shows a carrier signal with Amplitude **A**, phase **θ**, and frequency $\mathbf{f_o}$ which is the reciprocal of the period **T**.
The combination of different modulation schemes has resulted in a plethora of modulation techniques depending on which aspect one wants to optimise: robustness against noise, amount of information transmitted per second (*capacity* of the link in bits/second) or *spectral efficiency* (number of bits/s per Hertz).
For instance, **BPSK** -Binary Phase Shift Keying- is a very robust modulation technique but transmits only one bit per symbol, while **256 QAM** -Quaternary Amplitude Modulation- will carry 8 bits per symbol, thus multiplying by a factor of eight the amount of information transmitted per second, but to correctly distinguish amongst the 256 symbols transmitted, the received signal must be very strong as compared with the noise (a very high **S/N** -Signal/Noise ratio- is required).
The ultimate measure of quality in digital transmission is the **BER** -Bit Error Rate- which corresponds to the fraction of erroneously decoded bits.

Typical values of BER range between 10^{-3} and 10^{-9}.
The modulation also allows us to choose which range of frequency we want to use for a given transmission. All frequencies are not created equal and the choice of the carrier frequency is determined by legal, commercial and technical constraints.

Multiplexing and duplexing

In general, the sharing of a channel among different users is called *multiplexing*.
This is shown in Figure TB 7.

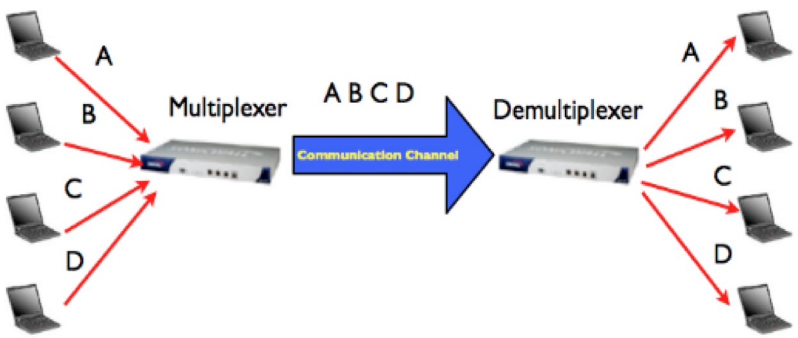

Figure TB 7: Multiplexing

Assigning different carrier frequencies to different users is called **FDMA** -Frequency Division Multiple Access-.
An alternative technique consists in assigning different *time slots* to different users, in what is known as **TDMA** -Time Division Multiple Access-, or even different codes in **CDMA** -Code Division Multiple Access- where the different users are recognised at the receiver by the particular mathematical code assigned to them. See Figure TB 8.
By using two or more antennas simultaneously, one can take advantage of the different amount of fading introduced in the different paths to the receiver establishing a difference among users in what is known as **SDMA** - Space Division Multiple Access-, a technique employed in the **MIMO** -Multiple Input,Multiple Output- systems that have gained popularity recently.

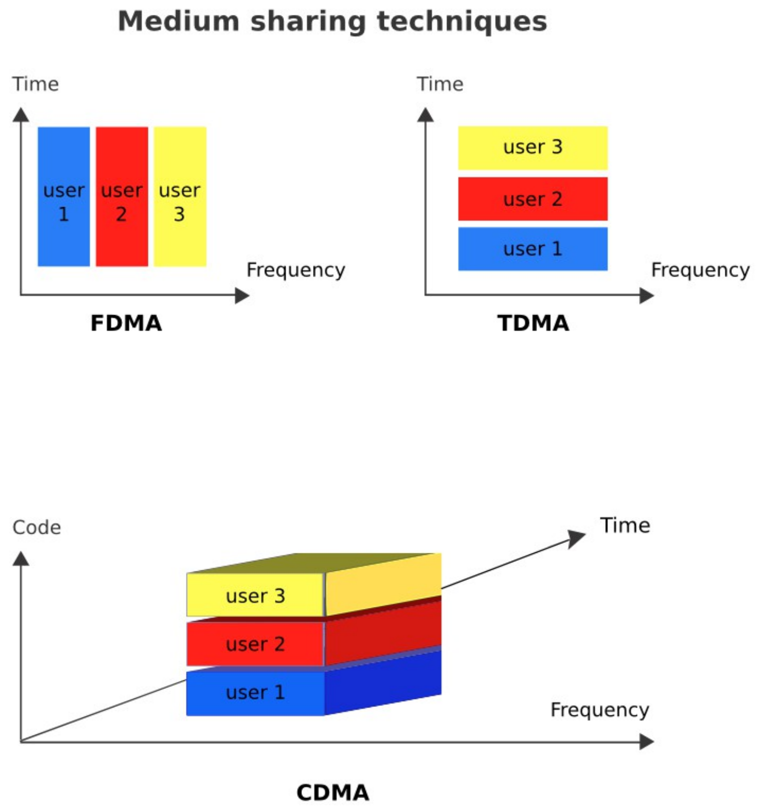

Figure TB 8: Medium Sharing techniques

Most communication systems transmit information in both directions, for instance from the Base Station to the subscriber in what is called the ***downlink***, and from the subscriber to the base station in the ***uplink***.

To accomplish this, the channel must be shared between the two directions giving rise respectively to **FDD** -Frequency Division Duplexing- and **TDD** -Time Division Duplexing-.

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 in bits/second is proportional to the bandwidth in Hz 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 allow for a higher transmission rate, but require higher S/N ratio.

The channel can be shared by several users that occupy different frequencies, different time slots, different codes or by taking advantage of different propagation characteristics in what is called spatial multiplexing.

For more information and slides covering this topic please visit http://wtkit.org/groups/wtkit/wiki/820cb/download_page.html