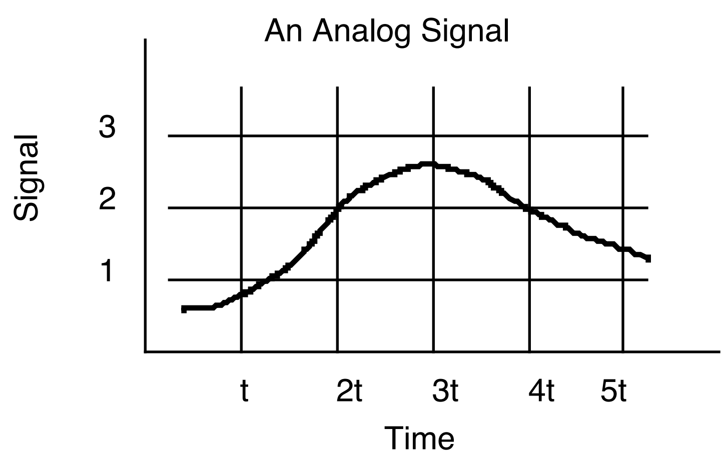
About Electronic Data Gathering:
Analog to Digital Conversion, Filtering and Amplification[i]

Biology 390 -- Physiology

I. <u>Data Gathering with a Computer</u>:

A. <u>The Basics of Analog to Digital Conversion</u>: Basically, any data gathering enterprise needs the following -- (i) some sort of **collector/transducer to collect and convert a biological signal into an electrical signal** (if it isn't already one); (ii) a **signal conditioner** -- a device that filters out extraneous signals (sometimes called noise and distortion) and, if needed amplifies the signal of interest. Another way of putting things is that the conditioner improves the signal to noise ratio, that is, it increases the conspicuousness of the signal; (iii) a **display and storage device**. It is not necessary that all of these elements always be present. Examples of transducer/collectors are microphones; signal conditions are filters and preamplifiers and display/storage devices include chart recorders, oscilloscopes, and computers.

Often the signal from a collector/transducer is what we refer to as an **ANALOG** signal. That is, it is a potentially continuous variable instead of one that can only have a series of discrete values. Action potentials, muscle force, and blood pressure are good examples of an analog variables



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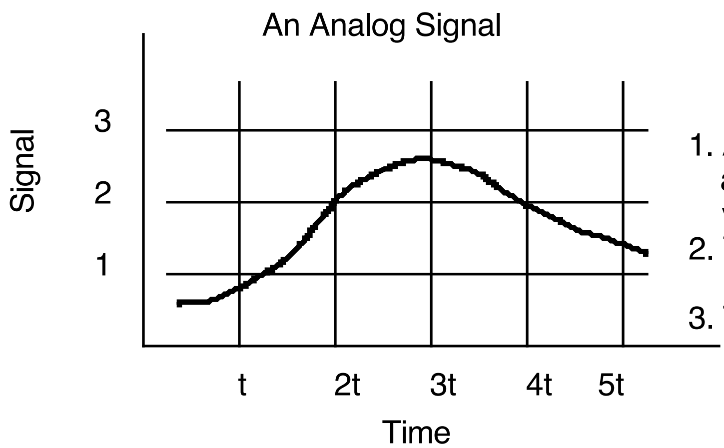
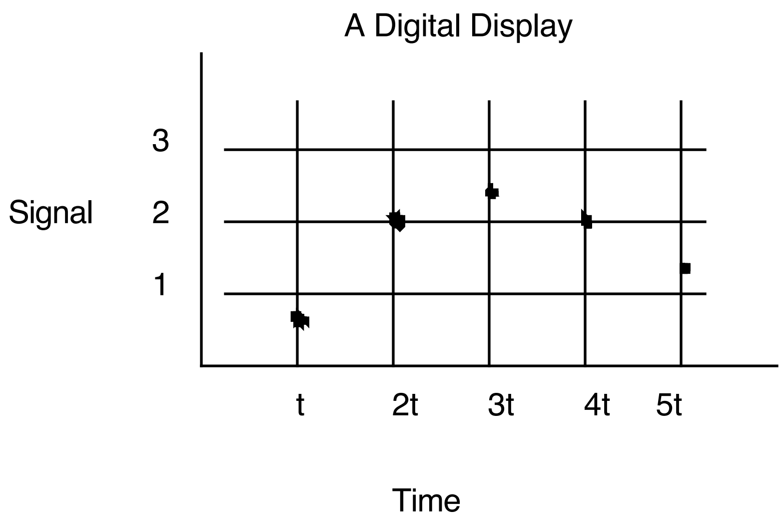
Modern digital computers cannot work with analog signals; they can only deal with values that can be represented as exact numbers. Thus, if we are dealing with an analog phenomenon such as the electrical activity of the heart, we must "digitize" it -- that is, accurately represent it as a series of discrete voltages at regular time interval. A device called an **<u>ANALOG to DIGITAL CONVERTER</u>** (a.k.a. **A/D converter** (read as "<u>A to D converte</u>r")) does this. Essentially, it produces a list of time/voltage points that can be stored in the computer's memory and then manipulated to make a display that resembles to some degree the original signal:



The Conversion to Digital Values

1. At each time Interval (shown as a vertical line, the instantaneous voltage is read.
2. The values between these interval are, therefore, ignored.
3. Thus, we would have these values:

time (x)	voltage (y)
t	0.8
2t	2.0
3t	2.6
4t	2.0
5t	1.4



Notice that the individual points can either be graphed as such or the computer can connect them using any of a number of mathematical techniques -- from as simple as connect the dots to rather complicated averaging curves. Note also that the accuracy of the digital representation of the actual event is largely determined by the rate of sampling -- obviously high rates of sampling fill in more of the graph and leave less to guessed by the computer program!

Filling in the lines is, of course, nothing more than a hypothesis about what has happened between the measurements.

B. <u>Bits, Conversion Rates, and Signal Depth</u>: Digital computers work using binary numbers, any one place in such a number can one of two values. The number of places used in making a binary representation of a number is the number of **<u>bits</u>** used. Thus, to represent two different numbers or things only (e.g., 0 and 1) 1 place is needed in the binary system. This is a one-bit representation. Likewise, to represent 4 different things (e.g., 0-3) two places are needed; this is a 2-bit representation. More generally:

1. number of things than can represented = $\mathbf{2^{\#\ of\ bits}}$

so, if my representations are all 4 places long (e.g., 0000 to 1111), then I can represent 2^4 or 16 different things!

When computers try to measure something analog by assigning some number to its value, they set aside a certain standard size memory unit (in terms of bits) to make this measurement. The **size of this number (# of bits)** will **help to determine the precision of the representation of the number** by the computer. **the other factor that matters** is the **range being represented**.

Thus suppose that voltages between 0 and 1 volt are being represented by an 8- and 16-bit system. The 8-bit system will divide that range into 256 (2^8) different values -- thus voltages will be assigned according to 4 mV steps; by contrast the 16 bit computer can read 65,536 different values and therefore will be able to read voltages to a 15 μV precision.

Now, note that if the voltage scale is expanded to, for example 0 to 100 V, it will be divided up into the same number of steps in each case but the difference between each step will be 100X larger -- the 8 bit system will only be able to tell 0.4V differences apart! The bit size that data is represented with in a computer is therefore related to precision and is often called the **bit depth** of the signal -- all else being equal, the more depth, the more precisely and accurately the computer represents an analog signal.

Acceptable Digitization Rates: Besides bit depth, the other factor that is important in understanding the limitations and usefulness of computer data is the digitization rate. We covered this idea above. But there is one very important additional consideration. When digitizing an analog signal the digitization must occur at a rate at least as great as the **Nyquist Frequency** (N_f) that is 2X the highest frequency one wishes to digitize. Thus, to digitize physiological events where the variable is changing rapidly (the equivalent of a high frequency) we need a high sampling rate. In the case of other variables (for instance, breathing) much lower rates can be used. While it is generally good to use a high rate, in some cases (for example, records of breathing, experience shows that nothing is really added. The advantage of using less memory and faster calculation (of a smaller data set) thus outweighs the normal advantage of greater precision.

! An example of digitization. Modern audio recordings for CDs are generally made at 44.2 kHz -- the highest frequencies most of us can hear are around 15 kHz so these are digitized at about 3X their value. However, most of what we hear from the CD is digitized at a far higher rate relative to its frequency!

II. Conditioning the Signal Prior to Input to the Measuring Device:

A. **introduction**: For a number of reasons, it is desirable to make some modifications of signals prior to measurement. Most of these have to do with the concept of noise and distortion and so here is a brief introduction to those topics.

B. **Noise and Distortion:** These are to be avoided (usually) because they introduce error referred to as **artifact** into recordings.

1. **Noise**: any sort of signal that is undesired

2. **Distortion**: This is the process where the signal is changed, usually in the connecting, amplifying, and recording equipment. You must always be on guard against this and noise because both types of artifacts make interpretation difficult or misleading. Some of the types of distortion that we need to worry about are:

(a) clipping: where the peaks of signals are cut-off when they exceed a certain amplitude; clearly information about amplitude has been lost.

(b) threshold: where signals below a certain level are ignored and those above are passed on; the somewhat but not exactly like the opposite of clipping.

3. **Signal to Noise Ratio:** Obviously, given some of the problems with noise just mentioned and the extremely weak nature of signals of the signals under investigation, it will be important to make the signal stand out as much as possible from the background noise.

a. The ratio of the two is called the **Signal to Noise Ratio (S/N).** When different signals are compared in electronics (and acoustics) one common means of making comparisons is by using a ratio.

b. In large part because of the way our sense of hearing works, a log scale of ratios has come into common use and so it is important that you understand this. Since we hear sounds that differ by many orders of magnitude, and in part due to the way we perceive differences in sounds, it was convenient to take the log of the ratio of the two signals being compared. The log of the ratio of two signals that differ by a factor of 10 is 1.0; this unit of difference is defined as a **bel**. In order to expand the scale somewhat from this, the scale is multiplied by a 10 and the unit differences are referred to as **decibels** (**dB**). Thus there are 10 dB in one bel. This is the last you will hear about bels -- they are just not used. But you do need to know about decibels. Mathematically, the exact formula used to calculate the dB difference between two signals varies according to whether it is a ratio of power of one signal to another (in which case the formula is:

2. $$dB = 10 * \log \frac{P_1}{P_2}$$

where P_1 and P_2 are the two different power levels being compared (and P_2 is usually that of the noise).

On the other hand, if voltages (or pressures or forces) are being compared a different formula is used since power $\propto$ voltage^2. This is the relationship for voltage:

3. $$dB = 20 * \log \frac{V_1}{V_2}$$

where V_1 is usually the voltage of the signal and V_2 is usually the voltage level of the noise. They can also refer simply to any two different signal voltage levels. This is the formula that we will be most concerned with and therefore it is important to know a couple things about dB comparisons of voltages.

> Notice that many things besides sounds can be represented in dB. However, probably the only time we will use dB is to consider signal to noise ratios. Large values mean that we can have high confidence in our data since it stands out from and will not be confused with the background noise.
>
> **?** If the voltage of a signal is doubled, what is the is the increase in dB?

(ANS: 6 dB)
What if it is doubled again? (compare both to the original value and the first doubled value).
(ANS: 12 dB from original, 6 dB from the "doubled" signal)

C. **Amplification**:

1. There are three general reasons for amplifying a signal:

(a) to boast the power, voltage, or current to a level that will allow it to be measured or displayed with available instruments. This is the most obvious reason for using an amplifier. There is only one warning here -- **too much amplification is bad** because it

(i) can make signals that are really nothing more than artifacts appear to be significant (by making them appear big!)

(ii) over-drive the recording device and run the record off the viewing space.

(b) **Maintain the S/N**.

(i) If I amplify a signal, I will amplify both the noise and the signal, the S/N should be unaffected. So what?

(ii) That is true, but ignores a benefit that comes from ampflying the signal shortly after it is transduced from the source.

(a) one problem in bioelectrical recordings is what happens between the recording electrode and the measuring device.

(b) when a signal is first picked up, there is some noise in it and a certain S/N. However, **both signal and noise are likely to be very weak**.

(c) As the signal moves through wires towards the measuring device, additional stray signals (as mentioned above) will be picked up and the S/N ratio will degrade.

(d) If the signal (and initial noise, although we'll see about this in a moment) are both boosted right after being picked up by using what is referred to as a **preamplifier** then, relative to the new signal, any additional noise will be small and the S/N ratio will not degrade significantly as the information moves towards the measuring device!

(c) match impedance: this is a more difficult subject that is related to both amplification and S/N ratio improvement. I mention it for completeness.

D. **Filtering**: as was just mentioned, S/N ratios can be prevented from degrading by using a preamplifier near the recording electrodes. I also mentioned that usually a preamplifier also contains means to improve the S/N. These devices are called filters.

1. One thing that filters often modify is the **bandwidth**; This can be taken as the range of frequencies that are allowed to pass through a circuit. Filters can restrict the bandwidth to a certain range by truncating both higher and lower frequencies or by removing just high or low frequencies.

2. Most filters do not abruptly cut off all frequencies outside of some range (and in fact there are a number of reasons that such behavior might not even be desirable).

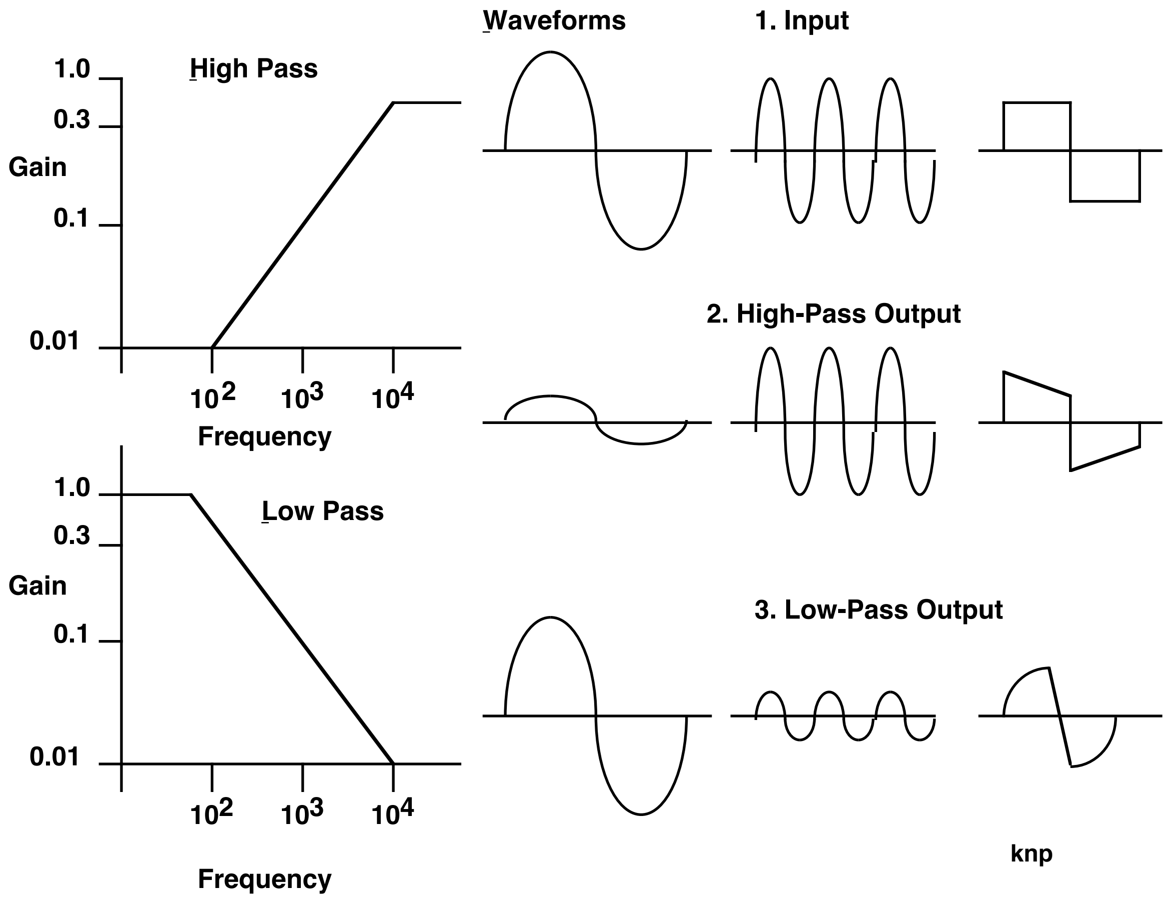
(a) The point where attenuation has reduced an inputted signal by - 3 dB is called the **cutoff point**. Note that the attenuation begins therefore begins at a different frequency. .

(b) As frequencies get further and further from the cutoff point, attenuation increases. The rate at which attenuation occurs at frequencies beyond the cutoff is called the **rolloff** and it is generally about 6 dB per octave; this means that the amplitude is halved every doubling of (or halving) of frequency depending on the type of filter (see below). **<u>The important message here is that attenuation is not abrupt; failure to understand this can result in unwanted noise in a processed signal</u>.**

3. There are three general types of filters we will need to be familiar with:

(a) **<u>High-Pass (or Low Cutoff)</u>**: these filters progressively remove the frequencies starting somewhat above a cutoff frequency.

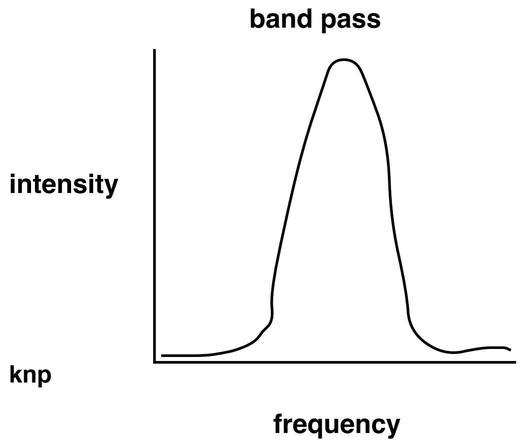
(b) **<u>Low-Pass (or High Cutoff)</u>**: just the opposite of high-pass; these remove frequencies starting just below the cutoff frequency.

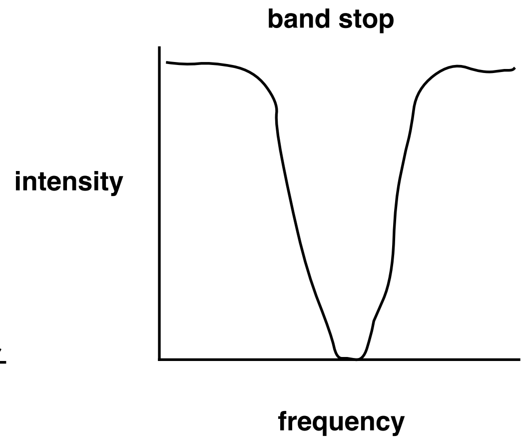


The effects of low and high pass filtering on sine waves of different frequencies and on a square wave. You should realize that a square wave is actually a complex waveform (review the information on Fourier transformations). See if the alteration of the square wave is what you would expect by each filter.
(source: Loeb, G.E. and C. Gans, Electromyography for Experimentalists . University of Chicago Press, 1986. after fig. 12.2.)

(c) **<u>Band-Pass Filters</u>**: These allow some range of frequencies to pass but they have both an upper and lower frequency cutoff point. **<u>The filters that are used in the construction of spectrographs and spectrograms are all of the band pass type; the differ mainly in how sharply they cut off energy at wavelengths on either side of their window</u>**.

(d) **Band-Stop Filter**: the opposite of a band pass; it allows frequencies above and below some range to pass.





Important: We will see that the Biopac data gathering and analysis package has many types of filters; please know the basics as outlined above and we will learn to use them in lab.