

Chapter 3

EEG recording

Data Acquisition

- Most acquisition systems can be subdivided into **analog** and **digital** components.
- Observing a biological process normally starts with the connection of a **transducer or electrode pair** to pick up a signal.
- The next stage is amplification. In most cases the amplification takes place in two steps using a separate preamplifier and amplifier.
- Then signal is usually filtered to attenuate undesired frequency components (band pass filtering and/or by notch filtering).
- A critical step is to attenuate frequencies that are too high to be digitized by the ADC. This operation is performed by the **antialiasing filter**.
- Finally, the sample-and-hold (S/H) circuit samples the analog signal and holds it to a constant value during the ADC process.

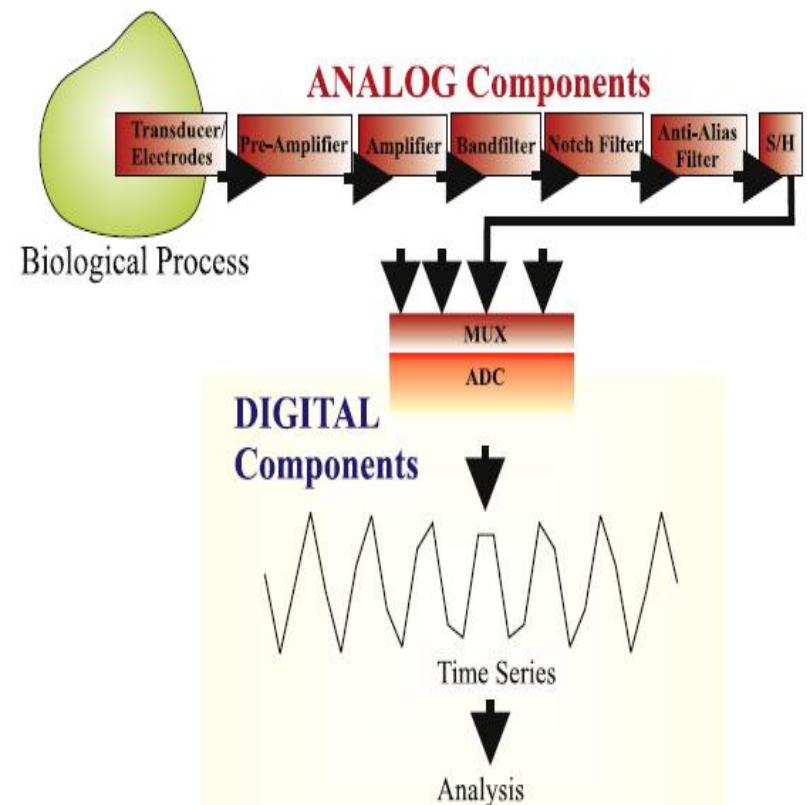
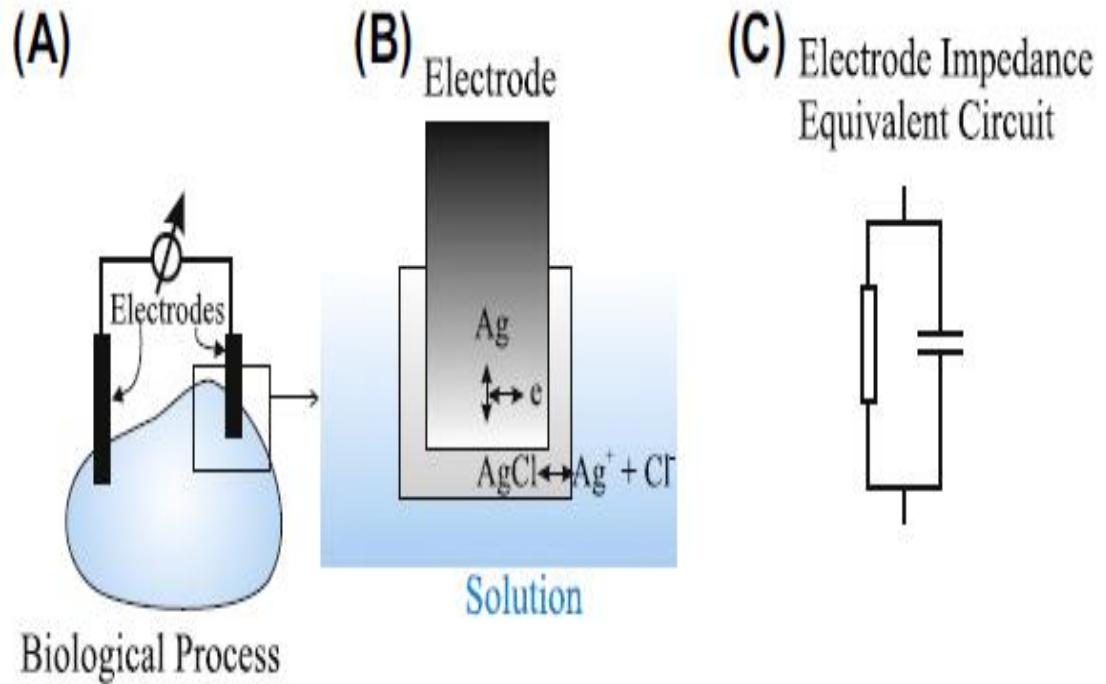


Figure2.1 , Signal processing for Neuroscientist, second edition by Wim van Drongelen

Electrodes

Metal electrodes are often used to measure potentials which must be bathed in an **ionic solution**. A fundamental problem with such direct measurements of electricity in solutions is the interface between the metal and solution.

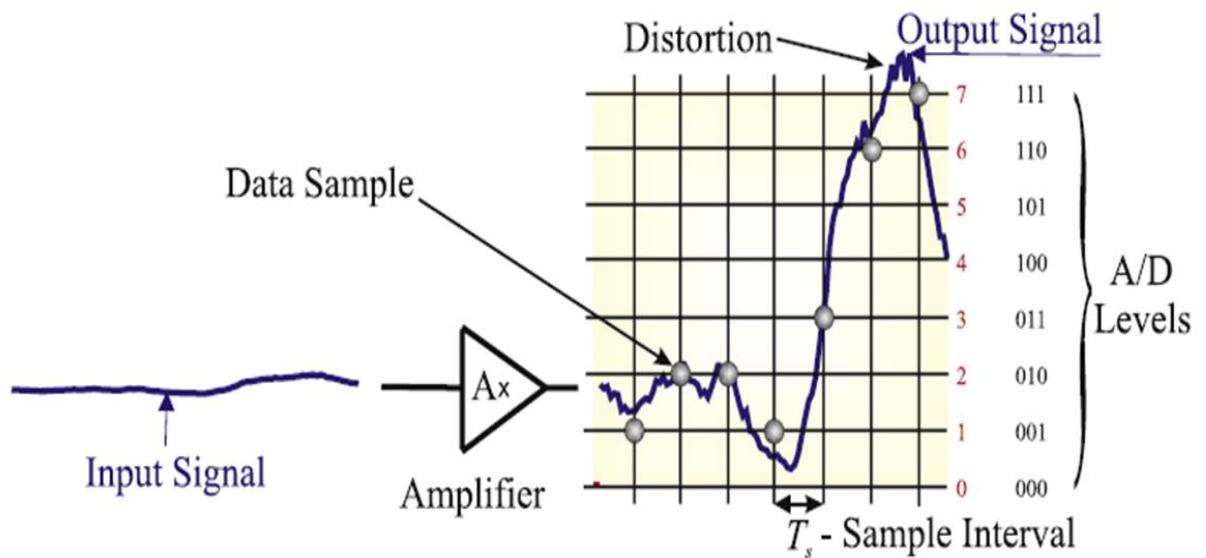
This boundary generates an electrode potential that is material and solution Specific. The electrode potential is usually not a problem when bio potentials are read from electrode pairs made of the same material. One widely used electrodes are silver electrode with a silver chloride coating. This facilitates the transition from ionic (**Ag or Cl**) to electronic conduction, reducing the electrode capacitance at the solution interface, and consequently facilitating the recording of signals with low-frequency components.



- (A) A setup with silver- silver chloride electrodes with
- (B) a detail of the chloride layer and
- (C) a simplified electronic equivalent circuit.

Analogue to digital conversion

- The nature of biomedical signals is analog: i.e., continuous both in amplitude and time. Modern data acquisition and analysis frequently depends on digital signal processing, and therefore the signal must be converted into a discrete representation. The time scale is made discrete by sampling the continuous wave at a given interval; the amplitude scale is made discrete by an analog-to-digital converter (A/D converter or ADC), which can be thought of as a truncation or rounding of a real-valued measurement to an integer representation.
- An important characteristic of an ADC is its amplitude resolution, which is measured in **bits**.



Analog-to-digital conversion (ADC). An example of an analog signal that is amplified A and digitized showing seven samples (marked by the dots) taken at a regular sample interval T_s , and a 3-bit A/D conversion. There are $2^3 = 8$ levels (0-7) of conversion. The decimal (0-7) representation of the digitizer levels is in red, the 3-bit binary code (000-111) in black. Note that, in this example, the converter represents the amplified signal values as integer values based on the signal value rounded to the nearest discrete level

Analogue to digital conversion...

Graphical representation of the Dirac function in continuous and discrete time. (A) The unit impulse (d, top row) and unit step (U, bottom row) function. The unit impulse can be considered as the derivative of the unit step. The unit impulse can be considered a square wave with duration T and amplitude 1/T

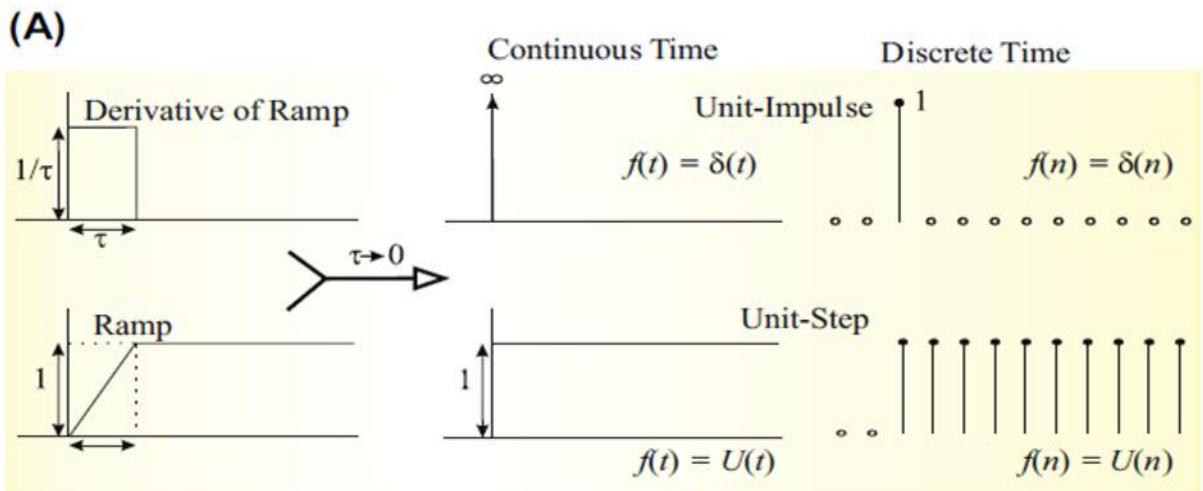


TABLE 2.1 Dirac Delta Function

Continuous Time	Discrete Time
$\delta(t) = 0 \text{ for } t \neq 0$	$\delta(n) = 0 \text{ for } n \neq 0$
$\int_{-\infty}^{\infty} \delta(t) dt = 1$	$\sum_{n=-\infty}^{\infty} \delta(n) = 1$

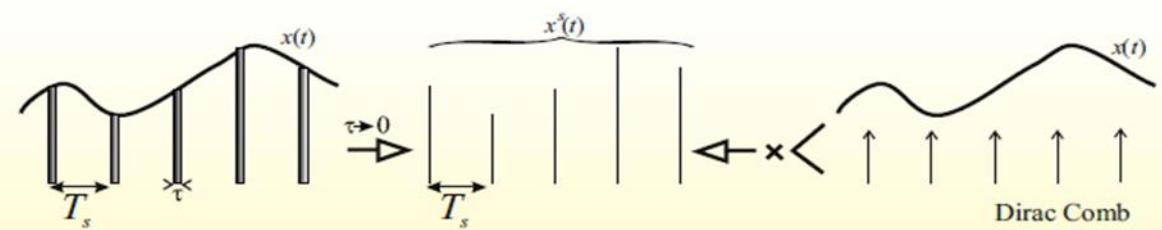
Analogue to digital conversion...

The continuous signal is also discretized (sampled) in time. To obtain a reliable sampled representation of a continuous signal, the sample interval (T_s) or sample frequency ($F_s = 1/T_s$) must relate to the type of signal that is being recorded.

In order to develop a mathematical description of sampling, unit impulse (Dirac impulse) function is introduced .

Sampling a continuous function $x(t)$ by multiplication with the Dirac comb generates discrete time function $x^s(t)$.

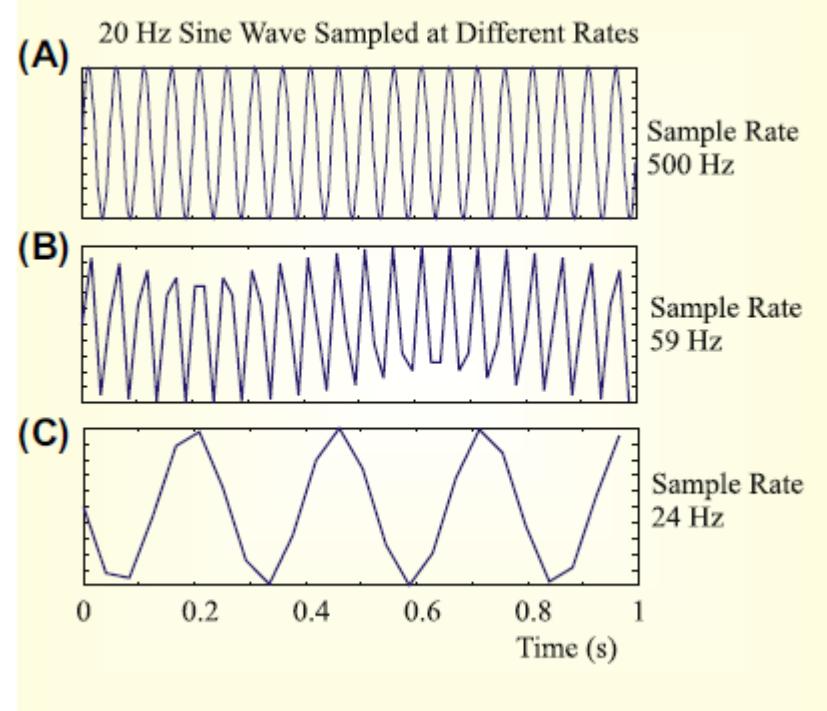
(B)



$$x^s(nT_s) = \sum_{n=-\infty}^{\infty} x(nT_s)\delta(t - nT_s) = x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

Analogue to digital conversion...

From time domain observation, it may be obvious that the sample rate at which one obtains $x_s(t)$ must be sufficient to represent the change in the continuous signal $x(t)$. Several examples are shown in this figure. As illustrated schematically in the figure, it seems that sampling a 20 Hz sine wave at a rate of $2 \times 20 = 40$ Hz at least conserves the frequency content of the signal. If these samples were taken exactly at the peaks and valleys of the sine wave, the sampled wave would look like a 20 Hz triangular wave. If not sampled at the peaks and valleys, the waveform will have an even more severely distorted appearance.

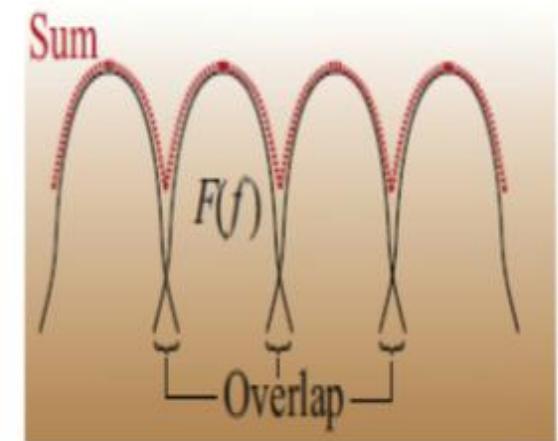
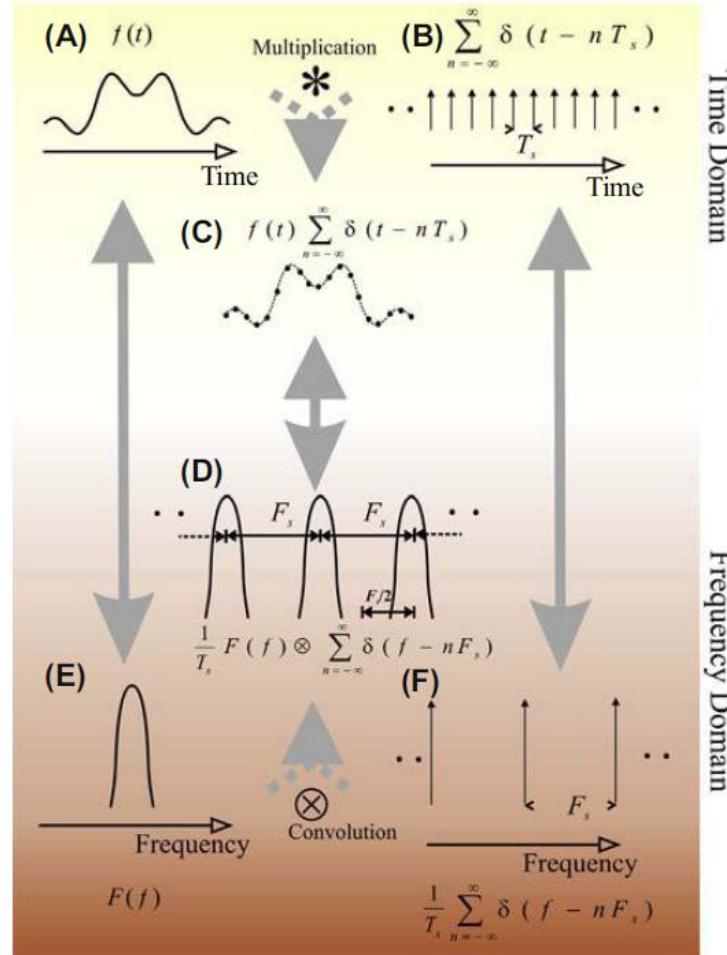


The minimum sampling rate (in this example 40 Hz) is called the **Nyquist sampling** frequency or the **Nyquist limit**. Thus, the sampling rate determines the highest frequency that can be represented by the sampled signal. This value (half the sample rate) is often indicated as the Nyquist frequency of the sampled signal.

Analogue to digital conversion...

Fourier transform of a sampled function. Sampling a function $f(t)$ (A) in the time domain can be represented by a multiplication of $f(t)$ with a train of d functions with an interval T_s , as depicted in (B), resulting in a series of samples (C). The Fourier transform of the sampled version is a periodic function, as shown in (D).

The Fourier transform of the sampled function can be obtained from the **convolution** of the Fourier transform $F(f)$ of $f(t)$ (shown in (E)) and the Fourier transform of the train of unit impulses with an interval $F_s=1/T_s$, as shown in (F). From this diagram it can be appreciated that the width of $F(f)$ should fall within period F_s (i.e., the maximum value of the spectrum of the sampled signal must be less than $F_s/2$) in order to avoid overlap in the spectra .



To remove the effect of aliasing in digitized signals, the analog measurement chain must remove/attenuate all frequencies above the Nyquist frequency by using a filter (antialiasing filter). To avoid distortion in the time domain (as seen in the example where the wave is digitized at 59 Hz) sampling at 5 times the maximum frequency is common.

EEG ACQUISITION

- To acquire EEG signals, the first step was given in 1958 by Herbert Jasper, who suggested a system for naming and placement of electrodes on the scalp, which is now worldwide used, called “International 10–20 System” .
- electrodes on the edges of the scalp are 10% distant of the horizontal line connecting the **Nasion** to the **Inion** through the preauricular point, where this percentage is related to the length of the line connecting the nasion to the inion through the vertex. All electrodes are positioned at a distance of 20% between each other. Then, numbers “10” and “20” of the “International 10–20 System” refer to these percentage values.

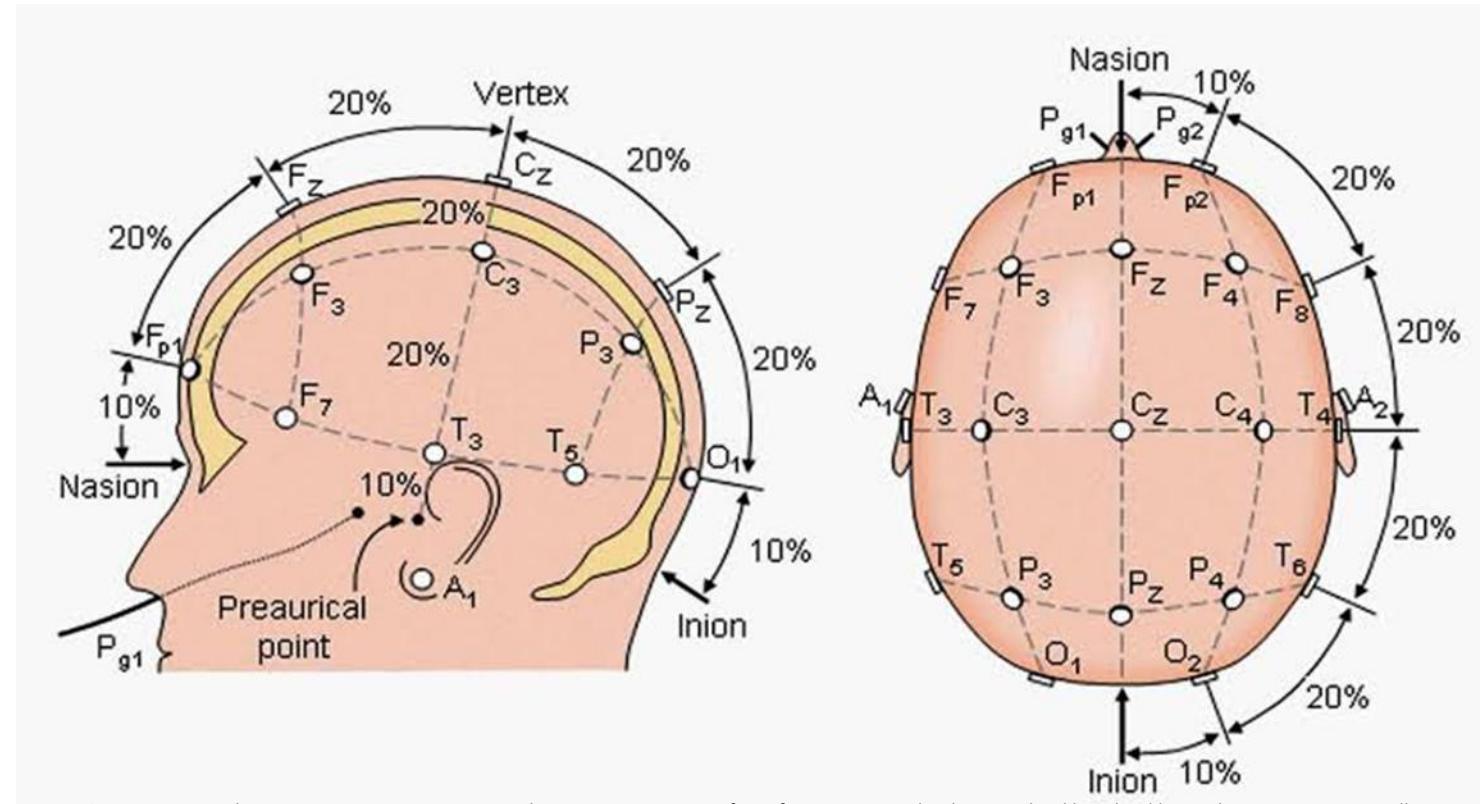
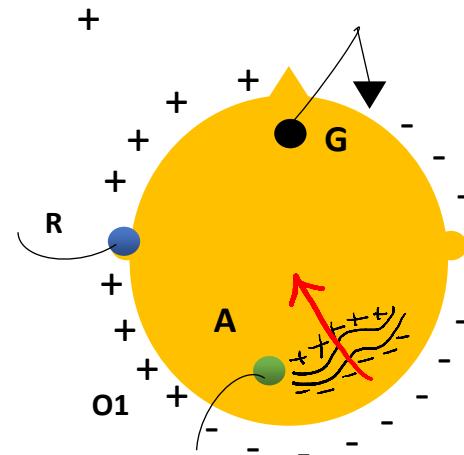


Figure 1.17. Introduction to Non-Invasive EEG-Based Brain–Computer Interfaces for Assistive Technologies ,Edited by Teodiano Freire Bastos-Filho

The electrodes are named by a capital letter corresponding to the initial of the brain lobe where they are placed (F = Frontal, C = Central, P = Parietal, O = Occipital, and T = Temporal), followed by an even number for the right hemisphere and an odd number for the left hemisphere. Electrodes on the frontal pole are named Fp, and the letter “A” is used for electrodes placed in the ear (from “auricular”). For the electrodes in the line connecting the nasion to the inion, the letter “z” is added, which indicates “zero”, rather than a number, indicating the central division of the brain hemispheres.

EEG ACQUISITION...

- If the electrical potential of the scalp was measured with only one EEG electrode relative to the **ground of the acquisition** circuit, the circuit would measure only the **static electricity** difference between the scalp and the circuit, which is much larger than the neural activity.
- However, even using the voltage difference between two electrodes in the scalp to create an EEG channel, any noise that affects the ground or the power of the acquisition circuit would also mask the neural activity.
- To solve this problem, the EEG amplifier system uses **differential amplifiers** with three electrodes to create a channel. Thus, the differential amplifier amplifies the voltage difference between V_{AG} and V_{RG} ($C = V_{AG} - V_{RG}$) and the common noise that affects the ground of both measures is eliminated .
- The ground electrode is usually positioned on the frontal bone to minimize the noise with **muscular origin**, and its potential is canceled during differential amplification. Then, its location is not as important as the location of the reference electrode.
- There is no ideally “neutral” place to position the reference electrode, so it should be taken in mind that the EEG signal from one channel always reflects the contribution of both active and reference electrodes. The reference electrode is usually placed on an ear lobe or both.



(G) grounding electrode

(R) reference electrode

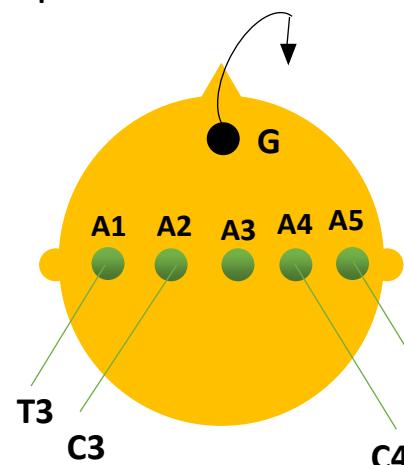
(A) active electrode

The ground electrode is positioned on the frontal region, the reference electrode is positioned on the left ear, and the electrode (A) is positioned on the occipital lobe of the left hemisphere to create the channel O1. This figure also shows the resulting dipole of an active area of the cortex and its potential distribution on the scalp. In this example, VAG is lower than VRG, and therefore, the calculated potential for O1 will be negative.

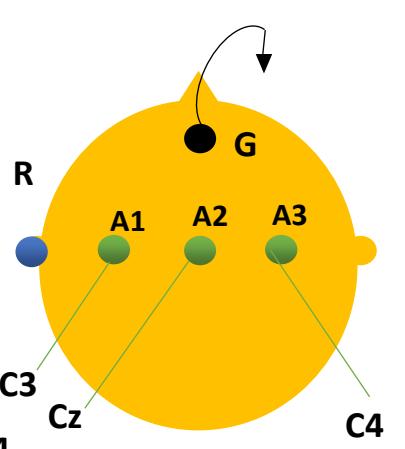
EEG ACQUISITION...

For the EEG acquisition with multiple locations, there are three distinct ways of electrode derivations to create the channels: bipolar method, unipolar method (or common electrode/reference), and free reference methods, which apply spatial filter, such as Laplacian, local average reference (LAR), and common average reference (CAR).

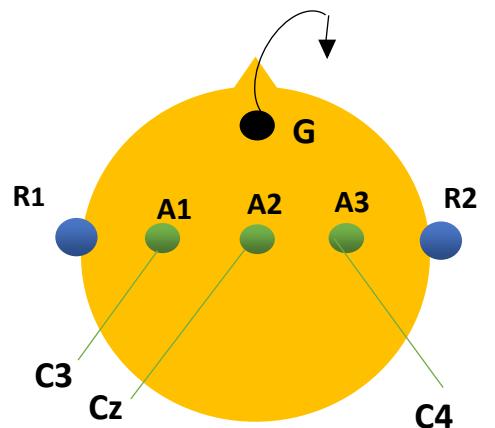
bipolar reference method



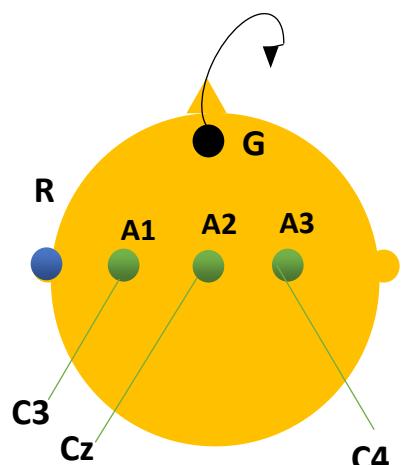
unipolar reference method



biauricular reference method



Common average reference method



$$T_3 = V_{A_1G} - V_{A_2G}$$

$$C_3 = V_{A_2G} - V_{A_3G}$$

$$C_4 = V_{A_4G} - V_{A_3G}$$

$$T_4 = V_{A_5G} - V_{A_4G}$$

$$C_3 = V_{A_1G} - V_{RG}$$

$$C_z = V_{A_2G} - V_{RG}$$

$$T_4 = V_{A_3G} - V_{RG}$$

$$V'RG = \frac{VR1G + VR2G}{2}$$

$$C_3 = V_{A_1G} - V_{R'G}$$

$$C_z = V_{A_2G} - V_{R'G}$$

$$T_4 = V_{A_3G} - V_{R'G}$$

$$V'RG = M = \frac{C_3 + C_z + C_4}{3}$$

$$C_3' = C_3 - M$$

$$C_z' = C_z - M$$

$$C_4' = C_4 - M$$