Introduction to Communications and IOT

Praanesh Balakrishnan Nair

February 4, 2025

Contents

1 Introduction

1.1 What a Signal is

- It's a Quantitative Representation of Information
- The most basic representation of a signal is in the form of a graph (t on X-axis and $f(t)$ on Y-axis)

1.2 Types of Communication

1.2.1 Wired / Wireless

1. **Wired**:
    - Via Coaxial cables or Fibre-Optic Cables
2. **Wireless**:
    - Via Electromagnetic waves or rays

1.2.2 Unidirectional / Bidirectional

1. **Simplex**:
    - One-way
    - Eg. Broadcast, FM
2. **Half-Duplex**:
    - Two-way, but only one direction at a time
    - Eg. walkie-talkie
3. **Duplex**:
    - Two-way, and both directions are simultaneously possible

1.2.3 Analogue / Digital

1. **Analog**:
    - Both t and $f(t)$ are continuous
2. **Digital**:
    - Both t and $f(t)$ are discrete

3. **Continuous-Time**:
   - t is continuous, but $f(t)$ is discrete
4. **Discrete-Time**:
   - t is discrete and $f(t)$ is continuous

1.2.4 Transmission Technique

Before knowing this, you must know what bandwidth is:

Bandwidth:

- Range of frequencies a signal operates.
- In other words:

  Bandwidth = (Highest Frequency of the Wave/Signal)−(Lowest Frequency of the Wave/Signal)

- Fast, irregular variations in frequency $\propto$ Bandwidth

1. **Baseband**:
   - Digital Signals which are sent via TDM (Time Division Multiplexing)
   - One signal uses the entire bandwidth
2. **Broadband**:
   - (I'll add this later)

2 Characteristics of a Signal

2.1 Standard Notation of a Standard Sinusoidal Signal

- For a graph where X-axis = θ and Y-axis = $sin(\theta)$, the measure of input is θ.
- To actually measure a signal against time, X-axis = t (time) and Y-axis = $sin(\theta)$
- Here's what we do for that: $sin(\theta + \phi) = sin(\omega t + \phi)$

2.2 Angular Frequency

- ω = Angular Frequency/Velocity
- $= \frac{Angle}{Time}$
- $= \frac{2\pi}{T}$

2.3 Frequency

- $f = \frac{1}{T}$
- So, $\omega = \frac{2\pi}{T}$ can also be written as $\omega = 2\pi f$

2.4 Phase

- θ or ωt is the X-coordinate.
- Phase ϕ is added to the X-coordinate, so the wave shifts to the left by ϕ
- In a way, it's an offset to a wave. (Check `https://www.geogebra.org/m/rzzqtx6q` for some Visualization)
- For example, if a sine wave is offset by $\frac{1}{6}th$ of a cycle, then the phase would be $\frac{1}{6} * 360^0 \Rightarrow$ Phase $= 60^0$

3 Time Domain vs Frequency Domain

In both cases, Y-Axis = *Amplitude*. Only X-Axis changes

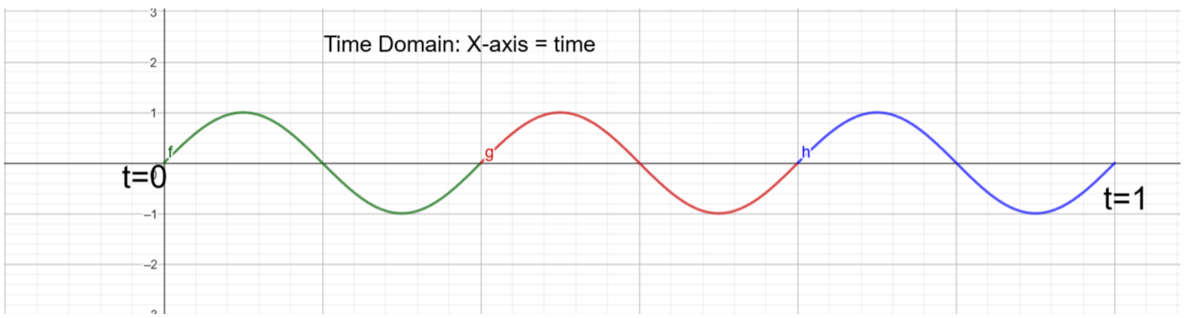

Figure 1: Time Domain

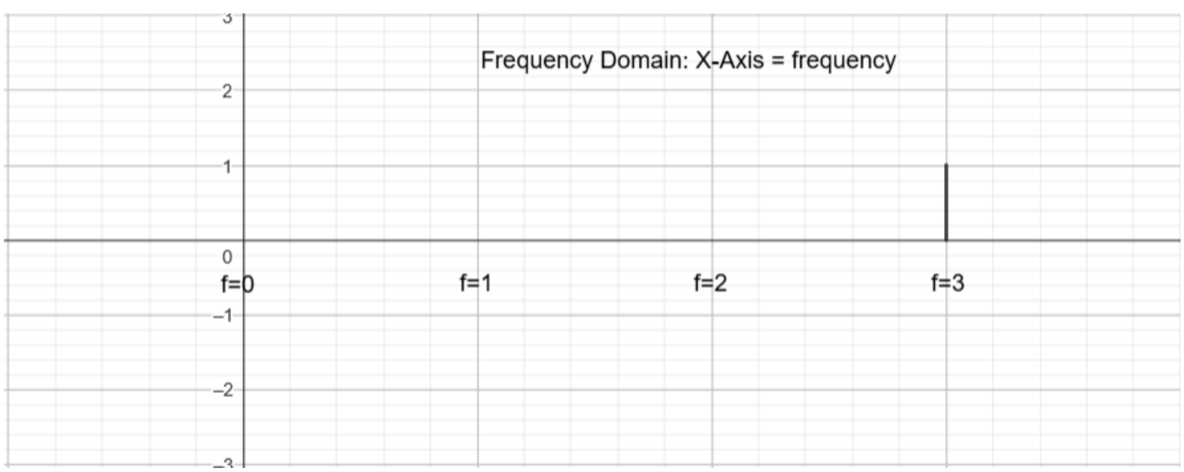

Figure 2: Frequency Domain

4 Odd Signals vs Even Signals

- **Odd Signals/Functions**: $y(-x) = y(x)$
- **Even Signals/Functions**: $y(-x) = -y(x)$

5 Energy and Power of a Signal

5.1 Prerequisite knowledge

- Let's assume we have a sinusoidal voltage and current
- $P = \frac{V^2}{R} = I^2 R$
- This means that the power of a signal is some **constant** times **voltage squared** or **current squared**
- Let us have a general signal $x(t)$ which can either be sinusoidal voltage or sinusoidal current

$$x(t) = V \textbf{ or } x(t) = I$$

- So Instantaneous Power $= P = (x(t))^2$

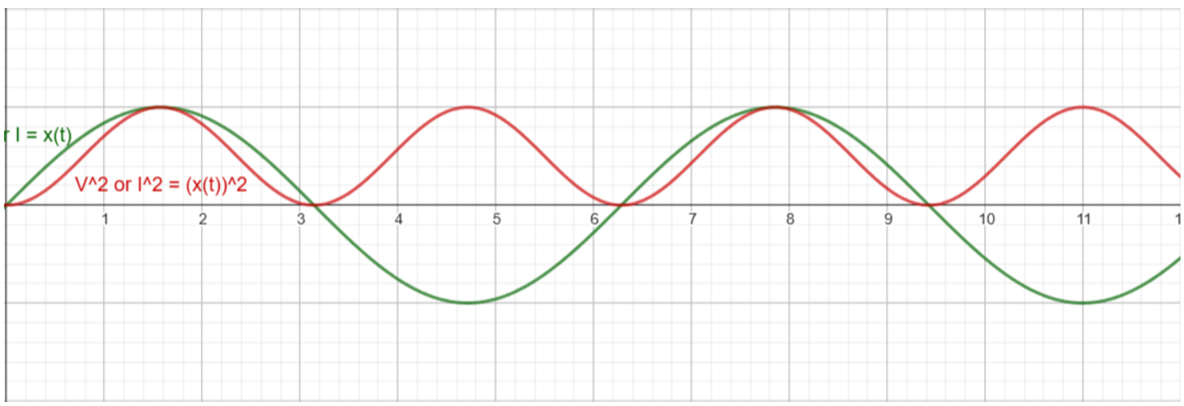

Figure 3: Green Curve showing V or I and the Red Curve showing P

5.2 Energy

- Energy $=$ Power $*$ time
- But the above formula is only applicable for discrete values.
- So the energy of a signal would be the area of the Power-Time Graph

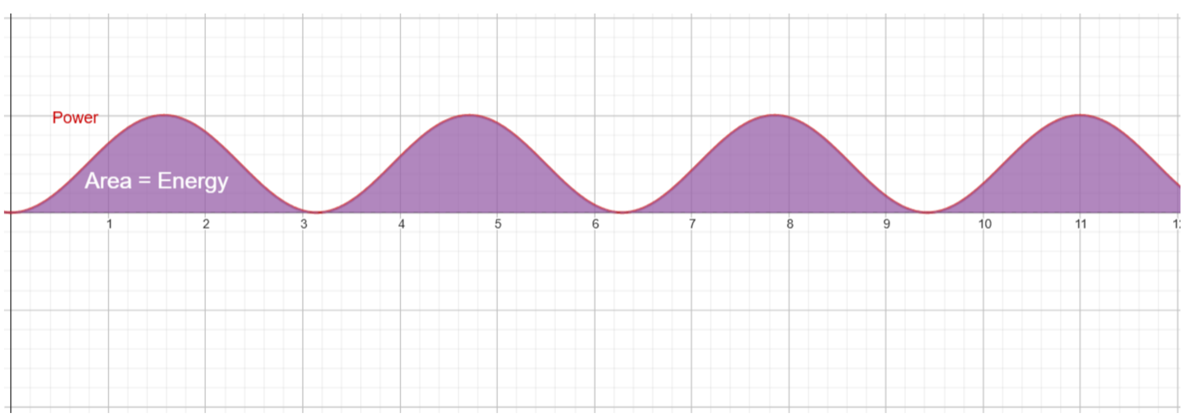

Figure 4: Area under the Red Curve

$$\text{Energy} = \int \mathrm{P}dt = \int_{\frac{-T}{2}}^{\frac{T}{2}} (x(t))^2 dt$$

- The limits are actually from 0 to T, but having them from $\frac{-T}{2}$ to $\frac{T}{2}$ simplifies calculations.

5.3 Power

- Power is just $\frac{\text{Energy}}{\text{Time}}$.
- Power $= \frac{\int_{\frac{-T}{2}}^{\frac{T}{2}} (x(t))^2 dt}{T}$

6 Complex Sinusoids

- **In phase**: Two signals are said to be in phase if they have phase difference 0
- **Quadrature**: Two signals are said to be in quadrature if they have a phase difference 0
- A complex sinusoid is given as $cos(\theta) + jsin(\theta)$
- $cos(\theta)$ is the real component plotted on
- Now $cos(\theta)$ is taken to be on the $Inphase - Time$ plane, and $sin(\theta)$ is taken to be on the $Quadrature - Time$ plane
- This results in a helical structure.
- Number of rotations about the *time* axis, per unit time, is the frequency of the complex sinusoid.
- Anti-Clockwise rotation means Positive frequency, so clockwise rotation means negative frequency

7 Sampling

7.1 What it is

- Converting a continuous time signal into a discrete time signal by taking samples of the signals at discrete time intervals
- Say we have a continuous sinusoidal signal:

$$s(t) = Acos(2\pi Ft + \phi)$$

- In its discrete form, instead of a parameter t, you'd have parameters n and T_s:

$$s[n] = Acos(2\pi FnT_s + \phi)$$

or

$$s[n] = Acos(2\pi F\frac{n}{F_s} + \phi)$$

Here, T_s = Sampling Time Period and F_s = Sampling Frequency

7.2 Sampling Theorem or Nyquist Theorem

- F_s is the number of samples taken per second i.e. the **sampling rate**. Likewise, T_s is the time taken to record one sample
- If F_s is too less, you won't be able to capture the wave correctly. You'll end up over-simplifying the wave.
- This is called **aliasing**, and it's where high-frequency components appear as low-frequency components because of insufficient sampling rate.
- Nyquist Theorem states that:

$$F_s \geq 2B$$

where B is the highest bandwidth present in the signal

Another way of saying this would be:

$$B \leq \frac{F_s}{2}$$

8 Filters

8.1 Analog Filters

1. **Low Pass Filters**: Keeps frequencies below a cutoff, and cuts off everything after it
2. **High Pass Filters**: Keeps frequencies after a cutoff, and cuts off everything below it
3. **Band Pass Filters**: Keeps frequencies inside a range (above a lower cutoff, and below a higher cutoff), and cuts off everything outside
4. **Band Reject/Stop Fitlers**: Keeps frequencies outside a range, and cuts off everything inside

8.2 Digital Filters

- They're mathematical algorithms used on discrete time signals
- Before knowing this, you must know about impulse signals:
    - **Impulse Signal** $\partial[n]$:
        - $\partial[n] = 1$ if $n = 0$, and $\partial[n] = 0$ for any other value of n
        - This function is used for representing frequencies in digital signals
    - **Impulse Response** $h[n]$: Output of a system, if the input is an impulse signal.

8.2.1 Finite Impulse Response (FIR)

1. What it is
    - Output depends only on current and past input
    - Output does NOT depend on past output

$$y[n] = \sum_{i=0}^{M} b_i x[n-i]$$

where $y[n]$ = output of filter, $x[n]$ = input signal, b_i = filter coefficients, M = filter order = number of taps

- Eg. $y[n] = 0.25x[n] + 0.5x[n-1] + 0.25x[n-2]$
  - This is called a 3-tap FIR Filter
  - The impulse response for this filter would be
    * $h[n] = [y[0], y[1], y[2], y[3], ..., y[n]]$
    * $h[n] = [0.25, 0.5, 0.25, 0, 0, 0, ...]$

2. Characteristics

- Stable (Phase response is linear, and can be good for image/audio processing)
- Phase Accurate
- Computationally expensive
- Impulse response decays to 0

8.2.2 Infinite Impulse Response (IIR)

1. What is is

- Output depends on past input AND past output i.e. it uses **feedback**.

$$y[n] = \sum_{i=0}^{N} a_i y[n-i] + \sum_{i=0}^{M} b_i x[n-i]$$

Here, it's just whatever FIR was, but you're also doing the same thing for the previous **outputs** too.

2. Characteristics

- If order is low, frequency cutoffs will be sharp (Phase response is non-linear) (Can be good for real-time communication)
- Computationally efficient (you'll need less parameters)
- Can cause phase distortion
- Impulse response never decays to 0 (hence, it's infinite)

9 Modulation

$$\text{Base Signal/ Message} + \text{Carrier Signal} = \text{Modulated Signal}$$

9.1 Amplitude Modulation (AM)

- Amplitude of carrier signal is changed
- Say the message signal is:

$$m(t) = A_m sin(2\pi f_m t)$$

And the original carrier signal is:

$$c(t) = A_c sin(2\pi f_c t)$$

- The modulated signal i.e. the new carrier signal is

$$s(t) = (A_c + A_m sin(2\pi f_m t))sin(2\pi f_c t)$$

9.2 Modulation Index (m)

$$m = \frac{\text{Amplitude of Message Signal}}{\text{Amplitude of Carrier Signal}} = \frac{A_m}{A_c}$$

- If $m < 1 \Rightarrow$ UnderModulated (Not Using full bandwidth)
- If $m = 1 \Rightarrow$ Uses full bandwidth and is 100% Modulated
- If $m > 1 \Rightarrow$ OverModulated, causes distortion

9.3 Side Band

- We already know that $s(t) = (A_c + A_m sin(2\pi f_m t))sin(2\pi f_c t)$
- $s(t) = A_c sin(2\pi f_c t) + A_m sin(2\pi f_m t) sin(2\pi f_c t)$ ($\because$ Multiply the $sin(2\pi f_c t)$ term inside)
- $s(t) = A_c \sin 2\pi f_c t + \frac{A_m}{2} \cos 2\pi t(f_c - f_m) - \frac{A_m}{2} \cos 2\pi t(f_c + f_m)$ ($\because \sin A \sin B = \frac{\cos(A-B)}{2} - \frac{\cos(A+B)}{2}$)
- $f_c + f_m$ = upper band
- $f_c - f_m$ = lower band
- Band width = (upper band) - (lower band) = $2f_m$

9.4 Power of AM

$$Power_{AM} = Power_{Carrier}(1 + \frac{m^2}{2}) = Power_{Carrier} + 2 * Power_{1Sideband}$$

9.5 AM Demodulation

9.5.1 Synchronous/ Coherent demodulation

- Multiply recieved signal with a sine wave of same frequency and phase

9.5.2 Asynchronous/ Non-Coherent demodulation

- Doesn't need frequency or phase matching
- Uses diode, resistor, capacitor

9.6 Frequency Modulation (FM)

- We change frequency of carrier signal

$$s(t) = A_c cos(2\pi f_c t + \beta sin(2\pi f_m t))$$

where $\beta = \frac{\Delta f}{f_m}$, Δf = Frequency Deviation

- Bandwidth of FM is given by:

$$\text{Bandwidth} = 2(\Delta f + f_m)$$