Principle of Electronics Communication (DI03000121)

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Course Agenda I

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
3. Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)
4. Lecture 1.4: Block diagram of Analog and Digital communication system
5. Lecture 1.5: Modulation: Definition & its classification

Course Agenda II

6. Lecture 1.6: Classification based on analog & pulse signal as carrier
7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
8. Lecture 2.1: Need of modulation & Types of modulation techniques
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

Course Agenda III

11. Lecture 2.4: Types of AM band spectrum (DSB, SSB) and their applications
12. Lecture 2.5: Power relations in AM wave, bandwidth, Power saving in SSB
13. Lecture 2.6: Generation of AM: DSBSC signal using balanced modulator circuit
14. Lecture 14: Demodulation of AM signal: Envelope detector using diode
15. Lecture 2.8: Block diagram of super heterodyne receiver
16. Lecture 2.9: Working of each block of super heterodyne receiver with waveforms

Course Agenda IV

17. Lecture 2.10: Characteristics: Selectivity, Sensitivity, Image frequency rejection and Fidelity
18. Lecture 3.1: Frequency modulation: Mathematical representation of FM signal, time & frequency domain
19. Lecture 3.2: Bessel's function, frequency deviation ratio, modulation index (numerical)
20. Lecture 3.3: Types of frequency modulation (Narrow Band and Wide Band FM)
21. Lecture 3.4: Phase Modulation: definition, waveforms and applications

Course Agenda V

22. Lecture 3.5: Pre-emphasis and De-emphasis
23. Lecture 3.6: Generation techniques for FM wave: Phase locked loop FM modulator
24. Lecture 3.7: Basics and types of FM demodulators: Ratio detector
25. Lecture 3.8: PLL as FM demodulator
26. Lecture 3.9: FM Receiver: Block diagram and working with waveforms
27. Lecture 3.10: Compare AM, FM and PM

Course Agenda VI

Course Agenda VII

35. Lecture 4.8: PCM Transmitter
36. Lecture 4.9: PCM Receiver & System Review
37. Lecture 5.1: Properties of EM (electromagnetic) wave
38. Lecture 5.2: Antenna fundamentals: Resonant/Nonresonant, ideal antenna, principle of Tx/Rx antenna
39. Lecture 5.3: Antenna parameters: Radiation pattern, polarization, bandwidth, beam width
40. Lecture 5.4: Antenna parameters: antenna resistance, directivity, power gain, antenna gain

Course Agenda VIII

41. Lecture 5.5: Dipole antenna and radiation pattern
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44. Lecture 5.8: Antenna: Yagi-Uda antenna and smart antenna
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Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain

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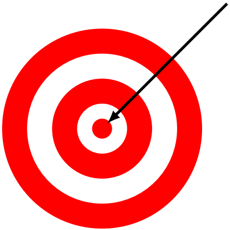
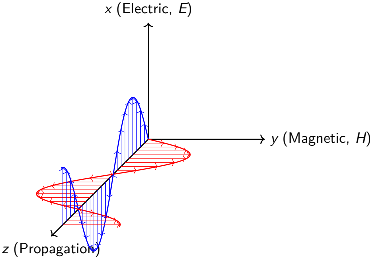
Learning Objectives

- Define Electromagnetic (EM) waves and their generation.
- List various frequency bands in the EM spectrum.
- Map specific frequency bands to their practical application domains.
- Solve basic numericals relating frequency and wavelength.



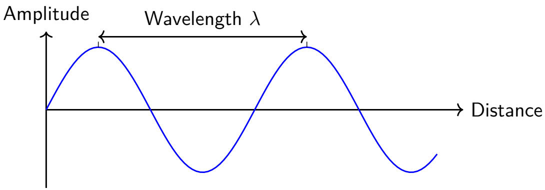
What is an Electromagnetic Wave?

- EM waves are synchronized oscillations of electric and magnetic fields.
- They propagate at the speed of light in a vacuum ($c \approx 3 \times 10^8$ m/s).
- Require no physical medium for propagation (can travel through a vacuum).
- Generated by accelerating electrical charges.



Relationship between Frequency and Wavelength

- Frequency (f): Number of cycles per second, measured in Hertz (Hz).
- Wavelength (λ): Distance over which the wave's shape repeats, measured in meters (m).
- Universal Wave Equation for EM waves: $c = f \times \lambda$
- Inversely proportional: Higher frequency means shorter wavelength.



Numerical Example: Calculating Wavelength

Problem: An FM radio station broadcasts at a frequency of 100 MHz. What is the wavelength of the transmitted EM wave?

Step-by-step Calculation

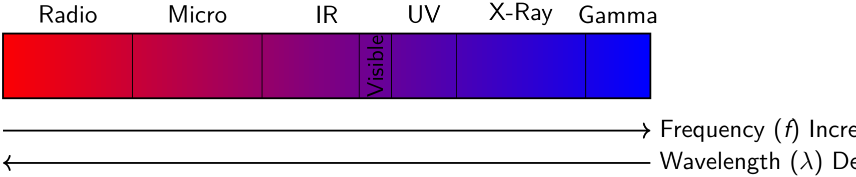
Given:
$f = 100$ MHz $= 100 \times 10^6$ Hz
$c = 3 \times 10^8$ m/s
Formula: $\lambda = \frac{c}{f}$

Solution:

$\lambda = \frac{3\times10^8}{100\times10^6} = \frac{300}{100} = 3$ meters

The Electromagnetic Spectrum

- The complete range of all types of EM radiation.
- Spans from extremely low frequencies (radio waves) to very high frequencies (gamma rays).
- For communication systems, we primarily focus on the Radio Frequency (RF) and Microwave spectrum.
- RF spectrum: 3 kHz to 300 GHz.



ITU Frequency Bands: VLF to HF

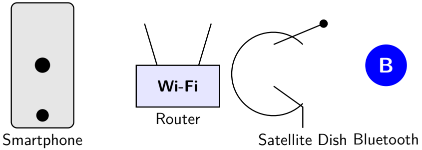
Band Name	Frequency Range	Primary Application
VLF (Very Low Freq)	3 – 30 kHz	Maritime navigation
LF (Low Frequency)	30 – 300 kHz	AM longwave radio
MF (Medium Freq)	300 – 3000 kHz	AM broadcasting
HF (High Frequency)	3 – 30 MHz	Shortwave radio, amateur radio

ITU Frequency Bands: VHF to EHF

Band Name	Frequency Range	Primary Application
VHF (Very High Freq)	30 – 300 MHz	FM radio, TV broadcasting
UHF (Ultra High Freq)	300 – 3000 MHz	Mobile phones, Wi-Fi, TV
SHF (Super High Freq)	3 – 30 GHz	Satellite comm., Radar
EHF (Extremely High Freq)	30 – 300 GHz	5G networks, Radio astronomy

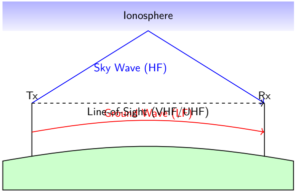
Application Focus: UHF and SHF

- UHF and SHF are heavily crowded because they offer good bandwidth and reasonable antenna sizes.
- **UHF Applications:** 4G LTE, Bluetooth, GPS, Wi-Fi (2.4 GHz).
- **SHF Applications:** Wi-Fi (5 GHz), Direct-to-Home (DTH) TV, Point-to-point microwave links.
- Higher frequencies offer more data capacity but suffer from higher attenuation (signal loss).



Why do we use different frequency bands?

- **Propagation characteristics:** Lower frequencies (HF) can bounce off the ionosphere for global reach.
- **Bandwidth requirements:** High-definition video needs high bandwidth, only available at higher frequencies (SHF).
- **Antenna size:** Antenna size is proportional to wavelength. Higher frequency = smaller antenna.
- **Interference:** Regulated by government bodies to prevent overlapping transmissions.



Quick Check

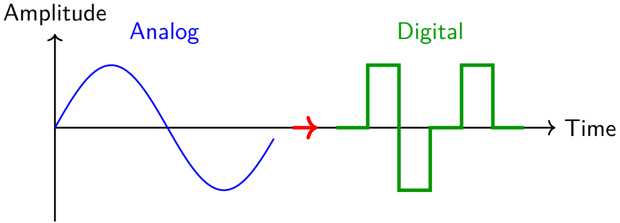
- **Q1:** What is the relationship between frequency and wavelength?
- **Q2:** In which band does FM Radio operate?
- **Q3:** Calculate the wavelength for a 3 GHz Wi-Fi signal. ($c = 3 \times 10^8$ m/s)

Key Takeaways

- ✓ EM waves consist of electric and magnetic fields and travel at the speed of light.
- ✓ Frequency and wavelength are inversely proportional: $c = f\lambda$.
- ✓ The radio spectrum is divided into bands (VLF to EHF) by the ITU.
- ✓ Choice of frequency band depends on bandwidth needed, propagation type, and antenna size constraints.

Next Lecture Preview

- Lecture 2: Signals and its representation
- Difference between Analog and Digital Signals
- Understanding Pulses and Impulses
- How information is converted into electrical signals



Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse

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Recap of Previous Lecture

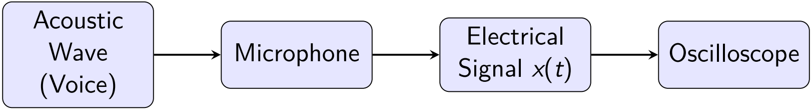
- In Lecture 1, we learned about EM waves, their spectrum, frequency bands (VLF to EHF), and their applications.

Course Overview & Today's Agenda

- ❖ Definition of a Signal
- ❖ Analog Signals (Continuous)
- ❖ Digital Signals (Discrete)
- ❖ Comparison: Analog vs Digital
- ❖ Pulse Signals
- ❖ Impulse Signals (Dirac Delta)

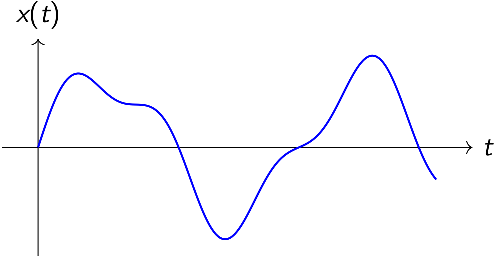
What is a Signal?

- A signal is a physical quantity that varies with time, space, or any other independent variable.
- In communication, a signal carries information from a transmitter to a receiver.
- Examples: Voice (acoustic), Voltage (electrical), Light intensity (optical).
- Mathematically represented as a function of time: $x(t)$.



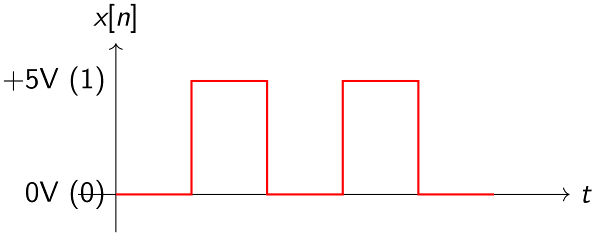
Analog (Continuous-Time) Signals

- An analog signal varies continuously with respect to time.
- It can take an infinite number of values within a given range.
- Most naturally occurring signals are analog (e.g., human voice, temperature variations).
- Represented smoothly on a graph with no breaks or jumps.



Digital (Discrete) Signals

- A digital signal can only take on a finite set of discrete values.
- Usually represented in binary format (0s and 1s).
- Highly resilient to noise compared to analog signals.
- Time is also discrete (sampled at specific intervals).

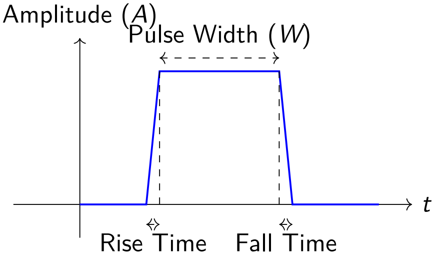


Analog vs Digital Signals

Analog	**Digital**
Continuous time & amplitude	Discrete time & amplitude
Prone to noise degradation	High noise immunity
Difficult to store & process	Easily stored (memory) and processed (CPUs)
Examples: Cassette tapes, Vinyl	Examples: CDs, MP3, SSDs

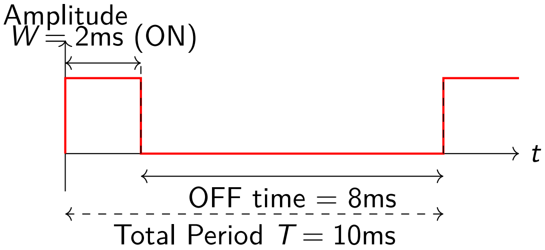
The Pulse Signal

- A pulse is a rapid, transient change in the amplitude of a signal from a baseline value to a higher or lower value, followed by a rapid return.
- Used extensively in radar, digital electronics, and pulse modulation techniques.
- Key parameters: Pulse Width (W), Amplitude (A), Rise time, Fall time.



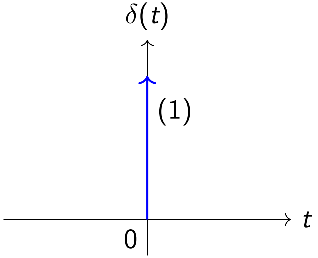
Numerical Example: Pulse Duty Cycle

- **Duty Cycle** is the percentage of time a pulse is 'ON' during one complete cycle.
- **Problem:** A periodic pulse wave has a total time period $T = 10$ ms and a pulse width $W = 2$ ms. Calculate the Duty Cycle.
- **Formula:** Duty Cycle (%) $= \left(\frac{W}{T}\right) \times 100$
- **Solution:** Duty Cycle $= \left(\frac{2}{10}\right) \times 100 = 20\%$



The Impulse Signal (Dirac Delta Function)

- An ideal impulse is a signal that is infinitely high, infinitely narrow, with a total area of 1.
- Mathematically denoted as $\delta(t)$.
- $\delta(t) = 0$ for $t \neq 0$, and $\int_{-\infty}^{\infty} \delta(t)dt = 1$.
- Theoretical concept used to test systems (Impulse Response).



Properties of the Impulse Signal

- **Sifting Property:** When multiplied by a continuous function and integrated, it 'sifts' out the value at the impulse location.

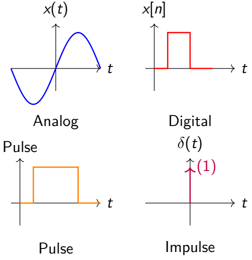
Sifting Property

$$\int_{-\infty}^{\infty} x(t)\delta(t - t_0)dt = x(t_0)$$

- Used heavily in sampling theory (converting analog to digital).
- Represents a sudden shock to a system (e.g., a lightning strike on an antenna).

Summary of Signal Representations

- **Analog:** $x(t)$, smooth, continuous.
- **Digital:** $x[n]$, discrete steps, binary logic.
- **Pulse:** Rectangular burst, defined by width and amplitude.
- **Impulse:** $\delta(t)$, infinitely narrow spike at $t = 0$.



Quick Check

Questions

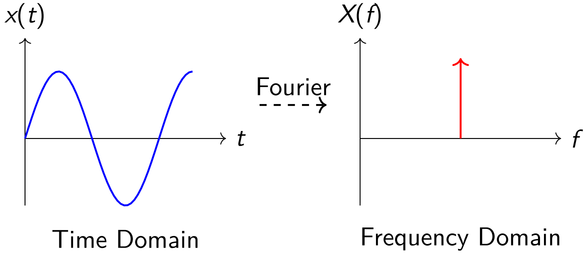
1. Which type of signal is more immune to noise: Analog or Digital?
2. What is the mathematical notation for an impulse signal?
3. If a pulse is ON for 4 ms and OFF for 6 ms, what is the duty cycle?

Key Takeaways

- ● Signals carry information and can be represented mathematically as functions of time.
- ● Analog signals are continuous; digital signals are discrete and robust against noise.
- ● Pulses have finite width and amplitude, used in timing and digital comms.
- ● Impulse signals are theoretical infinite spikes used for system analysis.

Next Lecture Preview

- **Lecture 3: Standard Signals in Time and Frequency Domain**
- Saw-tooth, Sinusoidal, and Rectangular signals
- Understanding how signals look in the frequency domain (Spectrum)
- Basic concepts of Fourier representation



Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)

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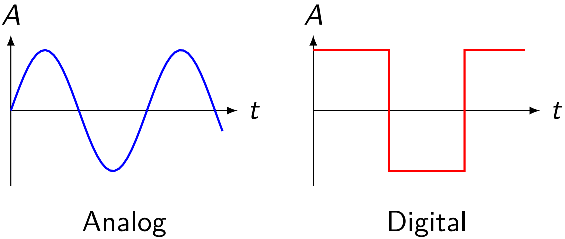
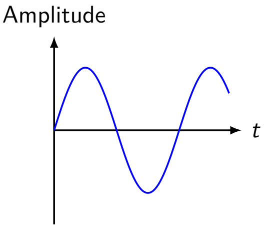
Recap of Previous Lecture

- In Lecture 2, we covered the definitions of analog, digital, pulse, and impulse signals.

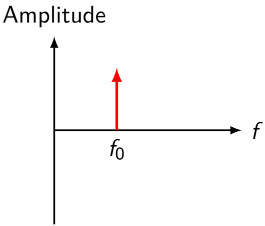


Time Domain vs. Frequency Domain

- **Time Domain:** How a signal's amplitude changes over time. (Observed on an Oscilloscope)
- Graph: X-axis = Time (t), Y-axis = Amplitude (V or I).
- **Frequency Domain:** How much of the signal's energy lies within each given frequency band over a range of frequencies. (Observed on a Spectrum Analyzer)
- Graph: X-axis = Frequency (f), Y-axis = Amplitude or Power.



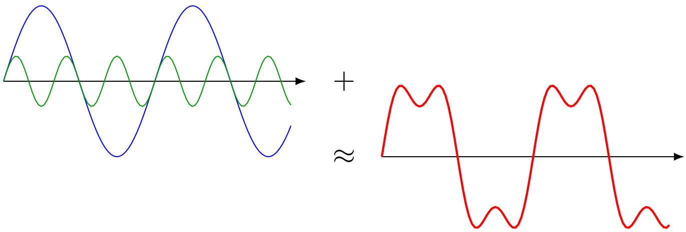
Oscilloscope (Time)



Spectrum Analyzer (Frequency)

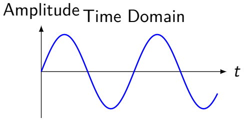
The Core Idea: Fourier Analysis

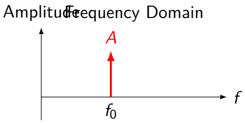
- Jean-Baptiste Joseph Fourier proved that **any periodic signal** can be constructed by adding together multiple sinusoidal waves (sine and cosine) of different frequencies and amplitudes.
- A sine wave is pure; it has only one frequency.
- Complex shapes (like square or saw-tooth) contain a fundamental frequency plus multiple "harmonics" (multiples of the fundamental frequency).



Sinusoidal Signals

- The most fundamental signal in nature and communication.
- Mathematical form: $x(t) = A \sin(2\pi f t + \phi)$
- Where A is amplitude, f is frequency, ϕ is phase.
- **Frequency Domain:** A pure sine wave contains energy at exactly ONE frequency.





Numerical Example: Sine Wave Properties

Problem

A sinusoidal voltage is given by $v(t) = 10\sin(2000\pi t)$. Find its Amplitude, Frequency in Hz, and Time Period.

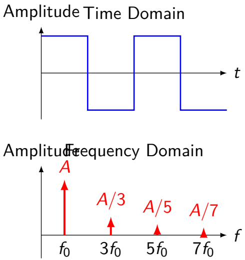
- **Compare with standard form:** $v(t) = A\sin(2\pi ft)$
- **Amplitude (A):** 10 Volts
- **Angular frequency ($2\pi f$):** $2000\pi \implies 2f = 2000 \implies f = 1000\text{ Hz} = 1\text{ kHz}$
- **Time Period (T):** $T = 1/f = 1/1000 = 1\text{ ms}$

$$v(t) = \color{red}{10}\sin(\color{blue}{2000\pi} t)$$

$$v(t) = \color{red}{A}\sin(\color{blue}{2\pi f}t)$$

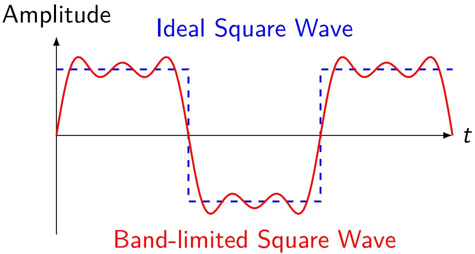
Rectangular (Square) Signals

- A periodic signal that alternates instantaneously between two levels.
- Used heavily in digital circuits (clocks, binary data).
- **Frequency Domain:** Contains the fundamental frequency f_0 AND an infinite number of **ODD harmonics** $(3f_0, 5f_0, 7f_0, \dots)$.
- Amplitude of harmonics decreases as frequency increases $(1/n)$.



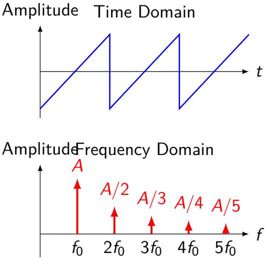
Bandwidth of a Square Wave

- Because a perfect square wave has sharp, vertical edges, it requires infinite frequencies (infinite harmonics) to reproduce perfectly.
- In reality, systems have limited bandwidth.
- Passing a square wave through a limited-bandwidth channel rounds off the sharp edges.
- This is why high-speed digital cables (like HDMI) need very high bandwidth.



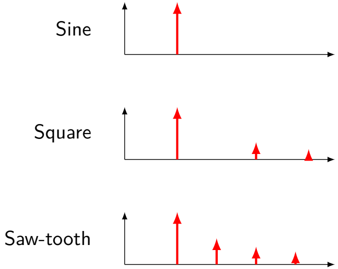
Saw-tooth Signals

- A waveform that ramps upward linearly and then drops sharply to the starting value.
- Used in CRT TV displays, synthesizers, and sweeping circuits.
- **Frequency Domain:** Contains the fundamental frequency f_0 AND **ALL harmonics** (both even and odd: $2f_0, 3f_0, 4f_0, \ldots$).
- Amplitude of harmonics decreases as $1/n$.



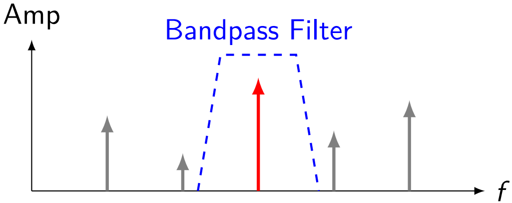
Comparing the Spectra

- **Sine Wave:** Single frequency component. Narrowest possible bandwidth.
- **Square Wave:** Fundamental + Odd Harmonics. Wide bandwidth.
- **Saw-tooth Wave:** Fundamental + All Harmonics. Very wide bandwidth.
- Rule of thumb: The sharper the edges in the time domain, the wider the spread in the frequency domain.



Why Frequency Domain Matters

- Communication channels (like air, fiber optics, cables) are characterized by their frequency bandwidth.
- By looking at a signal's frequency spectrum, we know if it will "fit" into the channel.
- Filters are designed in the frequency domain to block unwanted noise or select a specific signal (like tuning a radio).



Quick Check

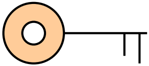
Questions

1. What instrument is used to view the frequency domain of a signal?
2. Does a square wave contain even harmonics, odd harmonics, or both?
3. If a sine wave has an angular frequency of 100π rad/s, what is its frequency in Hz?

?

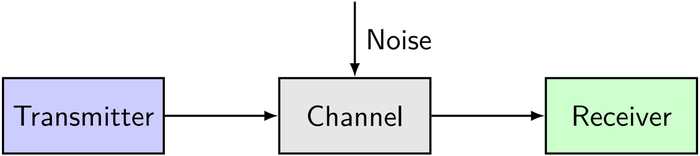
Key Takeaways

- Signals can be analyzed in both time (amplitude vs time) and frequency (amplitude vs frequency) domains.
- A pure sine wave consists of a single frequency component.
- Complex periodic signals (square, saw-tooth) are made up of a fundamental frequency plus multiple harmonics (Fourier Series).
- Sharp transitions in time require high-frequency components.



Next Lecture Preview

- Lecture 4: Block diagram of Communication Systems
- Components of an Analog Communication System
- Components of a Digital Communication System
- Transmitter, Channel, Receiver, and Noise



Lecture 1.4: Block diagram of Analog and Digital communication system

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Course Overview & Today's Agenda

- Basic Elements of Communication
- Block Diagram of Analog Communication System
- Functions of Analog Blocks
- Block Diagram of Digital Communication System
- Functions of Digital Blocks
- Analog vs Digital Systems

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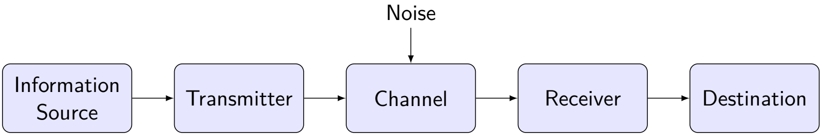
Recap

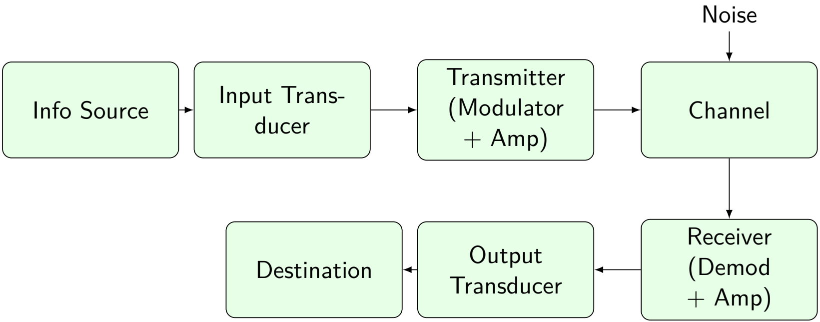
In Lecture 3, we studied standard signals (sine, square, saw-tooth) and their representations in both time and frequency domains.

Basic Elements of Any Communication System

- **Information Source:** Generates the message (Voice, video, data).
- **Transmitter (Tx):** Modifies the message into a suitable form for transmission.
- **Communication Channel:** The physical medium that bridges the Tx and Rx (Cable, free space, optical fiber).
- **Receiver (Rx):** Extracts the original message from the transmitted signal.
- **Destination:** The final recipient of the message.



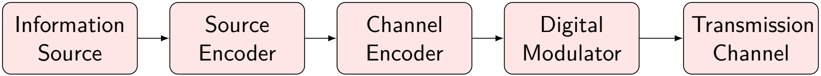
Analog Communication System: Block Diagram



Functions of Analog Blocks

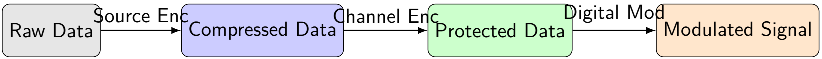
- **Input Transducer:** Converts physical quantity to electrical signal (e.g., Microphone converts sound to voltage).
- **Modulator:** Superimposes low-frequency message onto a high-frequency carrier wave (Crucial for wireless).
- **Amplifier (Tx):** Boosts signal power for transmission.
- **Demodulator:** Recovers the original low-frequency message from the carrier at the receiver.
- **Output Transducer:** Converts electrical signal back to physical form (e.g., Speaker).

Digital Communication System: Block Diagram (Tx)

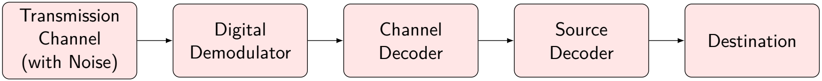


Functions of Digital Blocks (Tx)

- **Source Encoder:** Compresses data, removes redundancy (e.g., MP3 encoding). Converts analog to digital if needed.
- **Channel Encoder:** Adds redundant bits for error detection and correction (e.g., Parity bits). Protects against noise.
- **Digital Modulator:** Maps the digital bits (0s and 1s) into analog waveforms suitable for the channel (e.g., ASK, PSK).

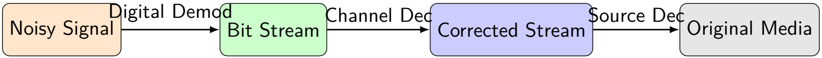


Digital Communication System: Block Diagram (Rx)



Functions of Digital Blocks (Rx)

- **Digital Demodulator:** Estimates the transmitted bits from the received noisy waveform.
- **Channel Decoder:** Uses the redundant bits to detect and correct errors caused by the channel.
- **Source Decoder:** Decompresses the data back to its original form (e.g., digital audio).
- **D/A Converter (if needed):** Converts the final digital data back to analog for human consumption.



Numerical Example: Data Rate

Problem

A source encoder compresses a video file of 100 Megabytes to 20 Megabytes. It needs to be transmitted in 10 seconds. Find the required data rate in Mbps (Megabits per second).

- **Data size:** 20 MB $= 20 \times 8 = 160$ Megabits (Mb)
- **Time:** 10 seconds
- **Data Rate:** Size/Time
- **Solution:** 160 Mb/10 s $= 16$ Mbps

Comparison: Analog vs Digital Systems

Analog System	**Digital System**
Simple hardware	Complex hardware (encoders/decoders)
No error correction	Features error correction
Degrades with noise	Robust against noise
Inefficient use of bandwidth	Highly efficient due to source compression
Not secure	Highly secure through encryption

Quick Check

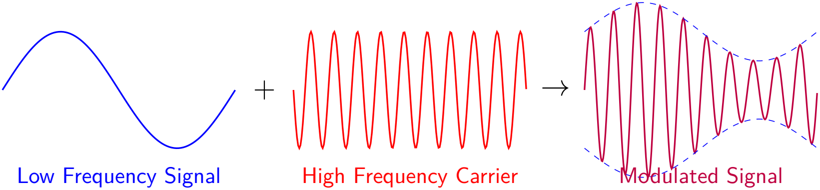
- **Q1:** What is the function of an input transducer?
- **Q2:** Which digital block is responsible for error correction?
- **Q3:** Which digital block compresses the data?

Key Takeaways

- All systems share Tx, Channel, and Rx.
- Analog systems rely on modulators and transducers.
- Digital systems utilize Source Encoding (compression) and Channel Encoding (error correction).
- Digital systems are more complex but offer superior noise immunity and security.

Next Lecture Preview

- Lecture 5: Modulation: Definition & its classification
- Why do we need modulation?
- How do we map low-frequency signals to high-frequency carriers?
- Antenna height calculations.



Any Questions?

Lecture 1.5: Modulation: Definition & its classification

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Course Overview & Today's Agenda

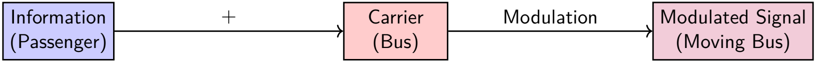
Previous Lecture Recap

Recap

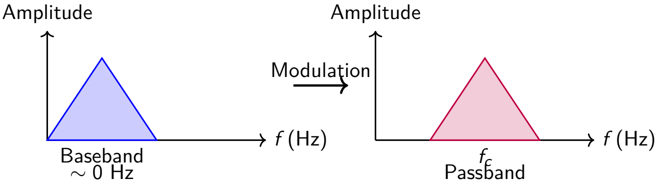
In Lecture 4, we analyzed the block diagrams of analog and digital communication systems, highlighting the role of modulators and encoders.

What is Modulation?

- **Modulation** is the process of varying one or more properties of a periodic waveform, called the *carrier signal*, with a modulating signal that contains information to be transmitted.
- Think of it as the message (passenger) boarding a high-frequency carrier (the bus) to travel a long distance.
- Properties that can be varied: Amplitude, Frequency, Phase.

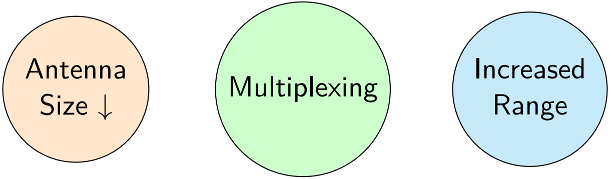


- **Baseband Signal:** The original information signal (e.g., voice from 300 Hz to 3400 Hz). Also called the modulating signal. Low frequency.
- **Carrier Signal:** A high-frequency sinusoidal wave used to carry the baseband signal. (e.g., 100 MHz).
- **Passband (Modulated) Signal:** The resulting high-frequency signal after modulation. Transmitted through the antenna.



Why do we need Modulation?

- **1. Practicability of Antenna Height:** Low frequencies require impossibly large antennas.
- **2. Multiplexing:** Allows multiple signals to share the same channel without mixing up (e.g., multiple FM radio stations).
- **3. Reducing Interference:** Shifts signals away from low-frequency environmental noise.
- **4. Increased Range:** High frequencies travel further as EM waves with less power.



The Antenna Height Problem

- For efficient electromagnetic radiation, the transmitting antenna length (L) must be at least one-quarter of the wavelength.
- Formula: $L = \frac{\lambda}{4}$
- Since $\lambda = \frac{c}{f}$, then $L = \frac{c}{4f}$.
- Low frequency $\implies$ High wavelength $\implies$ Giant antenna.

Antenna Length Equation

$$\lambda = \frac{c}{f}$$

$$L = \frac{\lambda}{4} = \frac{c}{4f}$$

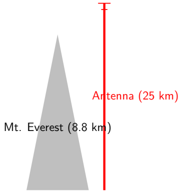
Numerical Example: Unmodulated Voice Antenna

Problem: Calculate the minimum antenna length required to transmit a raw audio signal of 3 kHz (3000 Hz) directly. ($c = 3 \times 10^8$ m/s)

Step 1: Calculate wavelength: $\lambda = \frac{c}{f} = \frac{3\times10^8}{3000} = 100,000$ meters (100 km).

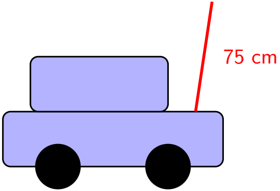
Step 2: Calculate antenna length: $L = \frac{\lambda}{4} = \frac{100{,}000}{4} = 25,000$ meters.

Conclusion: You would need a 25 km tall antenna to transmit voice directly! Impossible.



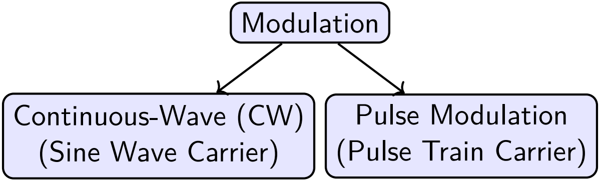
Numerical Example: Modulated Signal Antenna

Problem: Now, assume we modulate that voice signal onto an FM carrier of 100 MHz. Calculate the new minimum antenna length.
Step 1: Calculate wavelength: $\lambda = \frac{c}{f} = \frac{3\times10^8}{100\times10^6} = 3$ meters.
Step 2: Calculate antenna length: $L = \frac{\lambda}{4} = \frac{3}{4} = 0.75$ meters (75 cm).
Conclusion: 75 cm is the size of a standard car antenna. Highly practical.



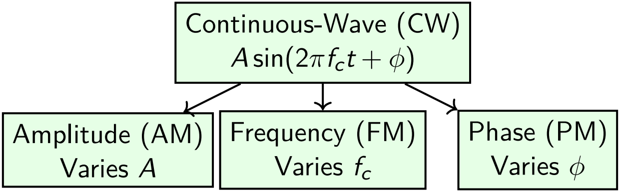
Classification of Modulation

- Modulation techniques are broadly classified based on:
  1. **The type of Baseband Signal:** Is the message Analog or Digital?
  2. **The type of Carrier Signal:** Is the carrier a continuous Sinusoidal wave or a sequence of Pulses?
- These combinations give us Continuous-Wave (CW) Modulation and Pulse Modulation.



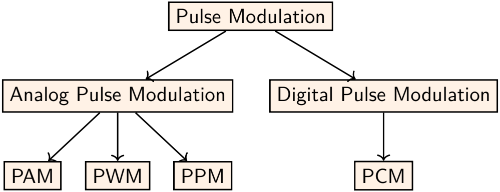
Continuous-Wave (CW) Modulation

- Carrier is a continuous sine wave: $A\sin(2\pi f_c t + \phi)$.
- **Amplitude Modulation (AM):** Message varies the Amplitude (A).
- **Frequency Modulation (FM):** Message varies the Frequency (f_c).
- **Phase Modulation (PM):** Message varies the Phase (ϕ).



Pulse Modulation

- Carrier is a train of rectangular pulses.
- **Pulse Amplitude Modulation (PAM):** Pulse height varies.
- **Pulse Width Modulation (PWM):** Pulse duration varies.
- **Pulse Position Modulation (PPM):** Pulse timing varies.
- **Pulse Code Modulation (PCM):** Digital representation (Unit 4).



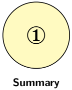
Quick Check

Questions

- **Q1:** What is the relation between required antenna length and wavelength?
- **Q2:** Does modulation increase or decrease the required antenna size?
- **Q3:** Name the three parameters of a sine wave that can be modulated.

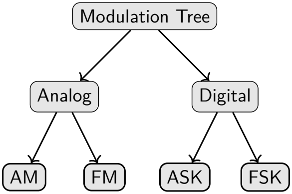
Key Takeaways

- Modulation embeds a low-frequency baseband signal onto a high-frequency carrier.
- It is strictly required to make antenna sizes physically practical ($L = \lambda/4$).
- It allows multiplexing (multiple users in the air at once).
- Modulation is classified into Continuous-Wave (AM, FM, PM) and Pulse Modulation.



Next Lecture Preview

- Lecture 6: Classification based on analog & pulse signal as carrier
- Detailed look at the modulation family tree.
- Analog-to-Analog, Digital-to-Analog mappings.
- Understanding ASK, FSK, and PSK concepts.



Questions?

Lecture 1.6: Classification based on analog & pulse signal as carrier

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Agenda

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
3. Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)
4. Lecture 1.4: Block diagram of Analog and Digital communication system
5. Lecture 1.5: Modulation: Definition & its classification
6. Lecture 1.6: Classification based on analog & pulse signal as carrier
7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
8. Lecture 2.1: Need of modulation & Types of modulation techniques
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

Recap

Previously on Communication Systems

In Lecture 5, we defined modulation and proved mathematically why high-frequency carriers are needed for practical antenna sizes.

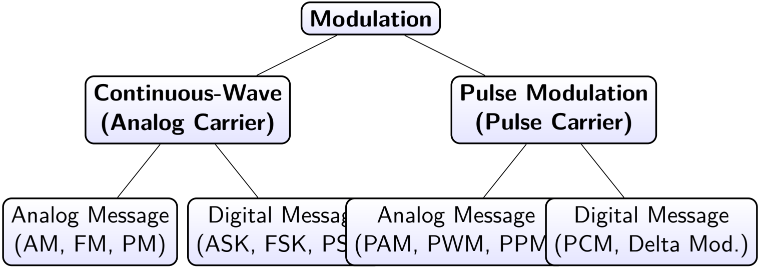
The Two Key Variables

To classify any modulation scheme, ask two questions:

- **1. What is the Message (Baseband)?**
  - Is it Analog (continuous voice) or Digital (1s and 0s)?
- **2. What is the Carrier?**
  - Is it a Continuous Sine Wave (Analog Carrier) or a stream of Pulses (Pulse Carrier)?

	Continuous Carrier	**Pulse Carrier**
Analog Message	AM, FM, PM	PAM, PWM, PPM
Digital Message	ASK, FSK, PSK	PCM, DM

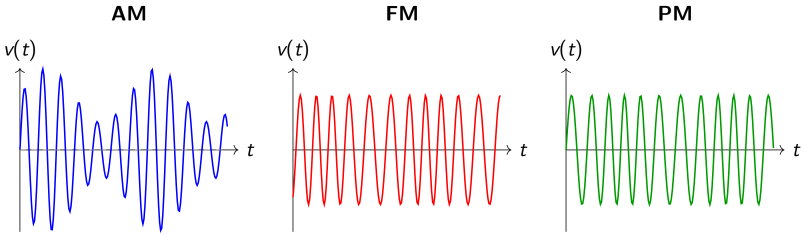
The Complete Modulation Hierarchy



1. Analog Message + Continuous Carrier

The traditional form of radio broadcasting.

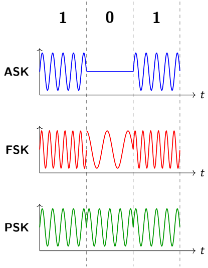
- **Amplitude Modulation (AM):** Amplitude of carrier varies with message. (Used in AM Radio).
- **Frequency Modulation (FM):** Frequency of carrier varies with message. (Used in FM Radio, analog TV audio).
- **Phase Modulation (PM):** Phase of carrier varies. (Used in some specialized analog systems).



2. Digital Message + Continuous Carrier

Used when sending digital data over an analog medium (like air or old telephone lines). Also known as **Digital Shift Keying**.

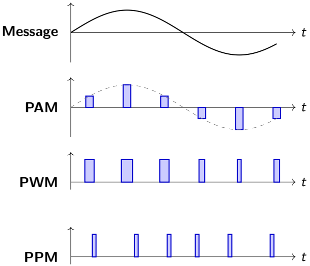
- **ASK (Amplitude Shift Keying):** Binary 1 is full amplitude, 0 is zero amplitude.
- **FSK (Frequency Shift Keying):** Binary 1 is high frequency, 0 is low frequency.
- **PSK (Phase Shift Keying):** Binary 1 is 0-degree phase, 0 is 180-degree phase.



3. Analog Message + Pulse Carrier

The carrier is a fast sequence of square pulses. The analog message modifies these pulses.

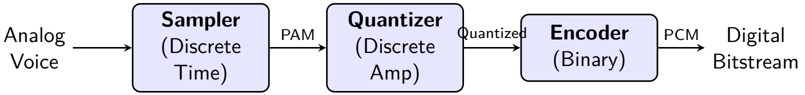
- **PAM (Pulse Amplitude Modulation):** The height of each pulse matches the analog message at that instant.
- **PWM (Pulse Width Modulation):** The width of the pulse changes. Wider pulse = higher analog value.
- **PPM (Pulse Position Modulation):** The timing (position) of the pulse shifts based on the analog value.



4. Digital Message + Pulse Carrier

The ultimate digital system. The message is digital, and it's sent as digital pulses.

- **Pulse Code Modulation (PCM):** The analog signal is sampled, quantized into discrete levels, and then encoded into binary pulses.
- This is how all modern digital audio (CDs, MP3s, PC audio) is created.
- Other variants: Delta Modulation (DM), Adaptive DM.



Numerical Example: Digital Bit Rate

Problem: In a PCM system, an analog voice signal is sampled 8000 times per second. Each sample is converted into an 8-bit digital code. Calculate the bit rate.

Step-by-Step Solution

Given:

- Sampling Rate (f_s) $= 8000$ samples/sec
- Bits per sample (n) $= 8$ bits/sample

Formula:

$$\text{Bit Rate } (R_b) = f_s \times n$$

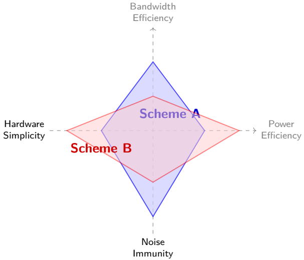
Calculation:

$$\begin{aligned} R_b &= 8000 \times 8 \\ &= 64,000 \text{ bits per second} \\ &= \mathbf{64} \text{ kbps} \end{aligned}$$

Why so many types?

Each technique has a trade-off between:

1. **Bandwidth efficiency:** How much frequency spectrum it consumes.
2. **Power efficiency:** How much transmitter battery power it requires.
3. **Noise immunity:** How well it survives a noisy channel.
4. **Hardware complexity:** Cost of building the Tx and Rx.



Summary Matrix

Message	Carrier	Category	Examples
Analog	Continuous Sine	Analog Continuous	AM, FM, PM
Digital	Continuous Sine	Digital Continuous	ASK, FSK, PSK
Analog	Pulse Train	Analog Pulse	PAM, PWM, PPM
Digital	Pulse Train	Digital Pulse	PCM, DM

Quick Check!

1. **Q1:** What does FSK stand for?
2. **Q2:** If I vary the width of a pulse based on a voice signal, what modulation is this?
3. **Q3:** Which modulation technique is used for standard digital telephone audio?

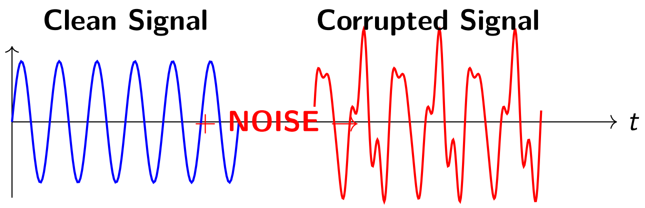
Key Takeaways

- Modulation is classified by the nature of the message (Analog/Digital) and the carrier (Continuous/Pulse).
- Continuous carriers lead to AM/FM (analog data) or ASK/FSK (digital data).
- Pulse carriers lead to PAM/PWM (analog data) or PCM (digital data).
- System choice depends on trade-offs in bandwidth, power, and complexity.

Next Lecture Preview

Lecture 7: Noise in communication systems

- What is noise and where does it come from?
- Signal-to-Noise Ratio (SNR).
- Noise Figure calculations.



Lecture 1.7: Noise in communication system, classification, S/N, noise figure

Complete Lecture Deck – All 45 Lectures

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July 22, 2026

Recap of Lecture 6

In Lecture 6, we mapped out the complete classification tree of modulation techniques (AM, FM, ASK, PCM, etc.).

Course Overview & Today's Agenda

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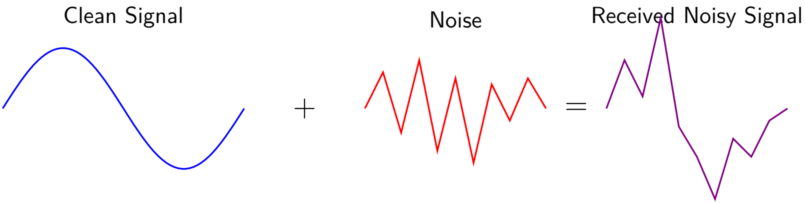
8. Lecture 2.1: Need of modulation & Types of modulation techniques

What is Noise?

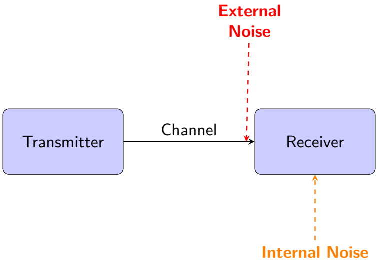
- **Noise** is an unwanted, random electrical signal that interferes with the transmission and reception of the message signal.
- It limits the operating range of systems.
- It limits the maximum data rate (Shannon's Channel Capacity).
- Unlike interference from another specific transmitter, true noise is random and unpredictable.



Broad Classification of Noise

Noise is classified by where it originates:

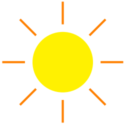
- **1. External Noise:** Generated outside the communication system (in the channel).
- **2. Internal Noise:** Generated by the electronic components within the transmitter and receiver themselves.



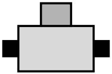
External Noise

- **Atmospheric Noise (Static):** Caused by lightning discharges and thunderstorms. Primarily affects lower frequencies (AM radio).
- **Extraterrestrial Noise:**
  - *Solar Noise:* Radiation from the sun.
  - *Cosmic Noise:* Radiation from distant stars and galaxies.
- **Industrial (Man-made) Noise:** Automobile ignitions, electric motors, power lines, fluorescent lights.

Atmospheric



Extraterrestrial

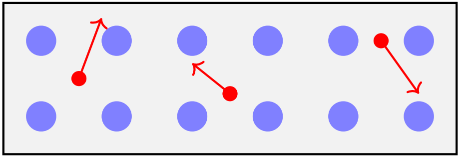


Industrial

Internal Noise

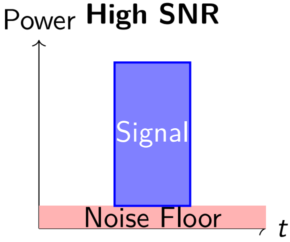
- **Thermal (Johnson-Nyquist) Noise:** Caused by random, temperature-dependent motion of electrons in conductors (resistors). Present in ALL electronic circuits.
- **Shot Noise:** Caused by the discrete, particle nature of electrons crossing a junction (like in diodes and transistors).
- **Flicker (1/f) Noise:** Occurs at low frequencies in active devices.

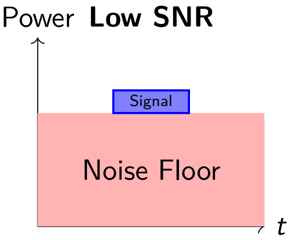
Resistor Lattice



Signal-to-Noise Ratio (S/N or SNR)

- SNR is the ratio of signal power to noise power at a given point in the system.
- Formula: $\text{SNR} = \frac{P_{\text{signal}}}{P_{\text{noise}}}$
- A higher SNR means the signal is much stronger than the noise (good quality).
- A low SNR (approaching 1) means the signal is barely distinguishable from the noise.





SNR in Decibels (dB)

- Because signal and noise powers can vary by factors of millions, we express SNR on a logarithmic scale using Decibels (dB).
- Formula: SNR (dB) $= 10\log_{10}\left(\frac{P_{\text{signal}}}{P_{\text{noise}}}\right)$
- If Signal Power = Noise Power, SNR = 1, which is 0 dB.
- Good communication usually requires an SNR of 10 dB to 30 dB or more.

$$\boxed{\text{SNR}_{\text{dB}} = 10\log_{10}(\text{SNR}_{\text{linear}})}$$

Numerical Example: SNR in dB

Problem: A receiver captures a signal power of 10 mW (milliwatts). The measured noise power is 10 μW (microwatts). Calculate the SNR in dB.

Solution:

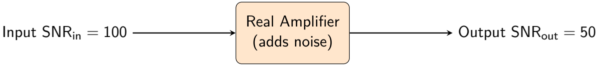
1. **Step 1:** Convert to same units. $P_s = 10 \times 10^{-3}$ W, $P_n = 10 \times 10^{-6}$ W
2. **Step 2:** Calculate linear ratio: $\frac{P_s}{P_n} = \frac{10 \times 10^{-3}}{10 \times 10^{-6}} = \frac{10^{-2}}{10^{-5}} = 1000$
3. **Step 3:** Convert to dB:
   SNR (dB) $= 10 \log_{10}(1000) = 10 \times 3 = 30$ dB

Key Concept: Every factor of 10 in linear ratio adds 10 dB.

- Ratio $= 10 \rightarrow 10$ dB
- Ratio $= 100 \rightarrow 20$ dB
- Ratio $= 1000 \rightarrow 30$ dB

Noise Factor (F) and Noise Figure (NF)

- Every amplifier adds its own internal noise to the signal.
- Therefore, the SNR at the output is always *worse* than the SNR at the input.
- **Noise Factor (F):** Ratio of Input SNR to Output SNR (linear).
- $F = \frac{\text{SNR}_{\text{in}}}{\text{SNR}_{\text{out}}}$
- Ideal amplifier: $F = 1$. Real amplifier: $F > 1$.



$$F = \frac{100}{50} = 2$$

Noise Figure (NF)

- **Noise Figure (NF)** is simply the Noise Factor expressed in decibels (dB).
- Formula: NF (dB) $= 10\log_{10}(F)$
- OR: NF $= \text{SNR}_{\text{in (dB)}} - \text{SNR}_{\text{out (dB)}}$
- An ideal amplifier has a Noise Figure of 0 dB.
- A lower Noise Figure means a higher quality, 'low-noise' amplifier (LNA).

$$\text{NF} = 10\log_{10}(F)$$

$$\text{If } F = 2, \text{ NF} \approx 3 \text{ dB}$$

Quick Check

1. **Q1:** Is lightning considered external or internal noise?
2. **Q2:** If signal power is 100 times the noise power, what is the SNR in dB?
3. **Q3:** What is the Noise Figure of an ideal, noiseless amplifier?

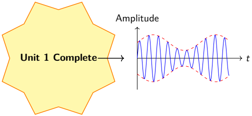
$\boxed{?}$

Key Takeaways

- Noise is random electrical interference that degrades signal quality.
- It is classified into External (atmospheric, cosmic, man-made) and Internal (thermal, shot).
- SNR measures signal quality; higher is better. Usually expressed in dB.
- Noise Figure measures how much a component degrades the SNR; lower is better.

Unit 1 Conclusion (Preview)

- We have completed Unit 1: Basics of Communication System.
- We covered EM waves, Signals (Time/Freq), Block Diagrams, Modulation Needs, and Noise.
- Next time, we begin **Unit 2: Amplitude Modulation (AM)**.
- We will apply all these basic concepts to mathematically analyze AM systems.



Any Questions?

Lecture 2.1: Need of modulation & Types of modulation techniques

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

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July 22, 2026

Recap of Unit 1

In Unit 1, we learned about different types of signals (analog and digital), basic blocks of communication systems, and noise.

Course Overview & Today's Agenda

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
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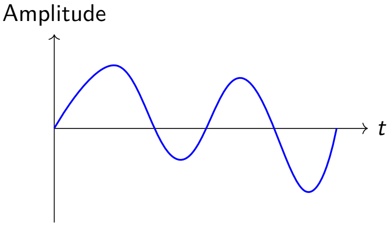
5. Lecture 1.5: Modulation: Definition & its classification
6. Lecture 1.6: Classification based on analog & pulse signal as carrier
7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
8. Lecture 2.1: Need of modulation & Types of modulation techniques

Learning Objectives

- Define the term 'Modulation'
- Calculate antenna length based on frequency
- List at least 3 reasons why modulation is necessary
- Classify various modulation schemes

What is a Baseband Signal?

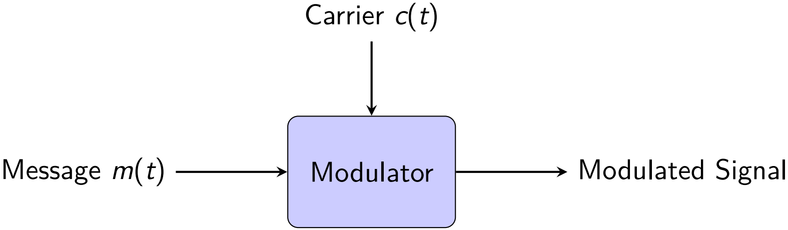
- Baseband signals are original information signals
- E.g., human voice (20 Hz - 20 kHz), video signal (0 - 5 MHz)
- Typically low frequency in nature
- Cannot travel long distances over free space easily



Low-frequency baseband signal

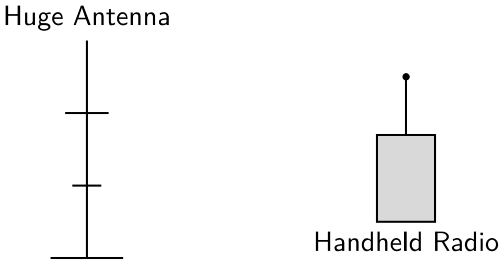
What is Modulation?

- The process of varying one or more properties of a periodic waveform (carrier signal) with a modulating signal that contains information to be transmitted.
- Carrier signal is a high-frequency continuous wave.
- Properties varied: Amplitude, Frequency, or Phase.



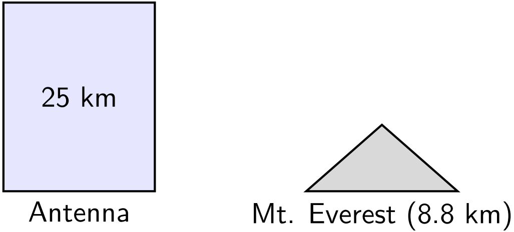
Why Can't We Transmit Baseband Directly?

- Baseband signals have low frequencies.
- Low frequency means long wavelength ($\lambda = c/f$).
- Antenna size is proportional to wavelength (usually $\lambda/4$).
- Result: Impractically large antennas are required.



Need for Modulation #1: Antenna Height

- Let's calculate antenna length for a voice signal.
- Given frequency (f) $= 3$ kHz, speed of light (c) $= 3 \times 10^8$ m/s.
- Wavelength (λ) $= c/f = (3 \times 10^8)/(3000) = 100,000$ meters $= 100$ km.
- Required antenna length ($\lambda/4$) $= 100$ km$/4 = 25$ km!



Need for Modulation #1: Modulated Case

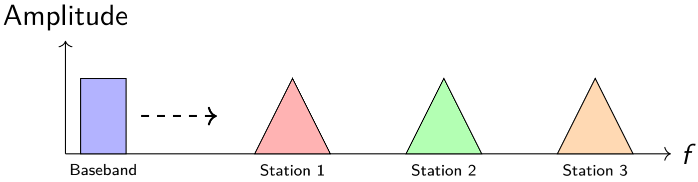
- Now let's use a 1 MHz carrier frequency (AM radio band).
- Given $f = 1$ MHz $= 10^6$ Hz.
- Wavelength $(\lambda) = c/f = (3 \times 10^8)/(10^6) = 300$ meters.
- Required antenna length $(\lambda/4) = 300/4 = 75$ meters.
- 75 meters is practical and easy to construct.

Baseband (3 kHz): 25,000 meters
Modulated (1 MHz): 75 meters

Massive Size Reduction!

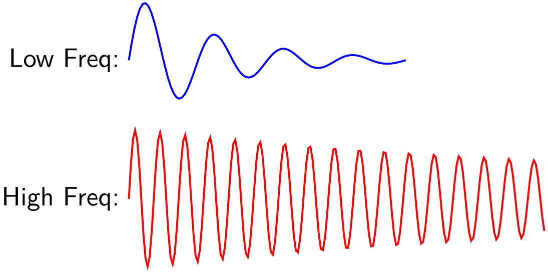
Need for Modulation #2: Multiplexing

- Without modulation, all radio stations transmitting voice would use the exact same 20Hz - 20kHz band.
- Signals would mix and interfere in the air.
- Modulation translates each station's baseband signal to a unique high-frequency carrier (e.g., 93.5 MHz, 98.3 MHz).
- Allows multiple signals to be transmitted simultaneously without interference.



Need for Modulation #3: Operating Range

- Low-frequency baseband signals suffer heavy attenuation (loss of power) over distances.
- High-frequency modulated signals radiate effectively from antennas.
- Travel longer distances with less attenuation in free space.
- Increases the overall communication range.



Need for Modulation #4: Narrowbanding

- Consider audio band 50 Hz to 10 kHz. Ratio of highest to lowest freq $= 10000/50 = 200$.
- A single antenna cannot cover this wide relative range efficiently.
- Shifted to 1 MHz carrier: range is 1,000,050 to 1,010,000 Hz.
- Ratio ≈ 1. Signal becomes narrowband (ratio close to 1).
- Allows use of a single tuned antenna.

Wideband: $\frac{f_{\text{max}}}{f_{\text{min}}} \gg 1$ (Hard to design antenna)

Narrowband: $\frac{f_{\text{max}}}{f_{\text{min}}} \approx 1$ (Easy to design antenna)

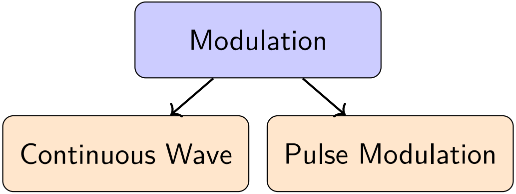
Summary of Need for Modulation

- 1. Reduces antenna height to practical sizes.
- 2. Avoids mixing of signals (enables multiplexing).
- 3. Increases the range of communication.
- 4. Allows adjustment of bandwidth (narrowbanding).
- 5. Improves quality of reception.

Types of Modulation Techniques

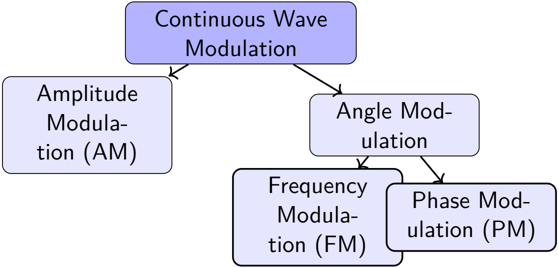
Modulation is broadly classified into two main categories based on the carrier:

- **1. Continuous Wave (CW) Modulation:** The carrier is a continuous sine wave.
- **2. Pulse Modulation:** The carrier is a periodic sequence of rectangular pulses.



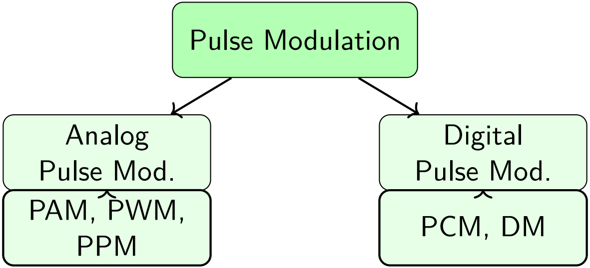
Classification of Continuous Wave Modulation

- **Amplitude Modulation (AM):** Amplitude of carrier varies in accordance with message signal.
- **Angle Modulation:**
  - **Frequency Modulation (FM):** Frequency of carrier varies.
  - **Phase Modulation (PM):** Phase of carrier varies.



Classification of Pulse Modulation

- **Analog Pulse Modulation** (Pulse amplitude, width, or position varies):
  - PAM (Pulse Amplitude Modulation)
  - PWM (Pulse Width Modulation)
  - PPM (Pulse Position Modulation)
- **Digital Pulse Modulation:**
  - PCM (Pulse Code Modulation), Delta Modulation

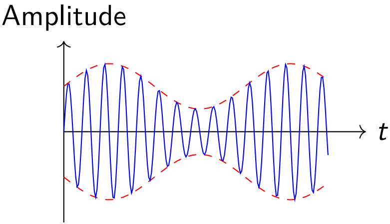


Key Takeaways

- Modulation involves changing a carrier wave parameter (amplitude, frequency, phase) using a message signal.
- Crucial for reducing antenna height and enabling multiplexing.
- Main types: Continuous Wave (AM, FM, PM) and Pulse Modulation.
- Our focus moving forward in this unit: Amplitude Modulation.

Next Lecture Preview

- Lecture 9: Amplitude Modulation (AM)
- Mathematical representation of AM
- Understanding modulation index
- Analyzing waveforms



Questions?

Any Questions?

Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Recap of Lecture 8

In the last lecture, we learned why modulation is essential (antenna height, multiplexing) and saw the classification of modulation techniques.

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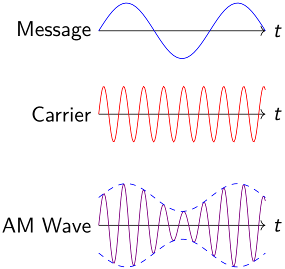
Learning Objectives

- Define AM precisely.
- Write down equations for message, carrier, and AM waves.
- Calculate modulation index (m).
- Identify consequences of over-modulation.

Definition of Amplitude Modulation

- Amplitude Modulation is the process in which the maximum amplitude of the high-frequency carrier wave is varied in accordance with the instantaneous amplitude of the low-frequency message signal.
- The frequency and phase of the carrier remain completely unchanged.



The Signals Involved

- **1. Modulating Signal (Message):**
  - Contains the information.
  - Low frequency.
  - $e_m = E_m \sin(\omega_m t)$
- **2. Carrier Signal:**
  - Transports the information.
  - High frequency.
  - $e_c = E_c \sin(\omega_c t)$

E_m: Peak amplitude
ω_m: Angular frequency

E_c: Peak amplitude
ω_c: Angular frequency

Mathematical Derivation of AM

- Let the AM wave be denoted as e_{AM}.
- The amplitude of the carrier, E_c, is changed by adding the modulating signal e_m.
- New instantaneous amplitude of carrier, $A = E_c + e_m$
- $A = E_c + E_m \sin(\omega_m t)$

Mathematical Derivation of AM (Contd.)

- The instantaneous voltage of the AM wave is:
- $e_{AM} = A\sin(\omega_c t)$
- Substitute A from previous slide:
- $e_{AM} = [E_c + E_m \sin(\omega_m t)] \sin(\omega_c t)$
- This is the standard equation of an AM wave.

$$\boxed{e_{AM} = [E_c + E_m \sin(\omega_m t)] \sin(\omega_c t)}$$

Modulation Index (m)

- **Modulation Index (m)** describes the extent to which the carrier amplitude is varied by the modulating signal.
- Also called depth of modulation or modulation factor.
- $m = \frac{E_m}{E_c}$
- where E_m = peak amplitude of modulating signal, E_c = peak amplitude of carrier.
- Normally, $0 \leq m \leq 1$. Often expressed as a percentage ($\%m = m \times 100$).

$$\boxed{m = \frac{E_m}{E_c}}$$

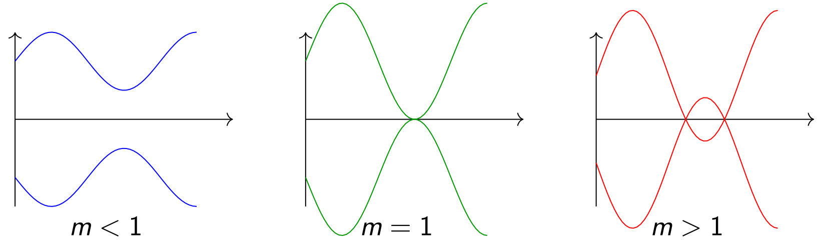
AM Equation using Modulation Index

- We had: $e_{AM} = [E_c + E_m \sin(\omega_m t)] \sin(\omega_c t)$
- Factor out E_c:
- $e_{AM} = E_c \left[1 + \frac{E_m}{E_c} \sin(\omega_m t)\right] \sin(\omega_c t)$
- Substitute $m = \frac{E_m}{E_c}$:
- $e_{AM} = E_c[1 + m \sin(\omega_m t)] \sin(\omega_c t)$

$$\boxed{e_{AM} = E_c[1 + m \sin(\omega_m t)] \sin(\omega_c t)}$$

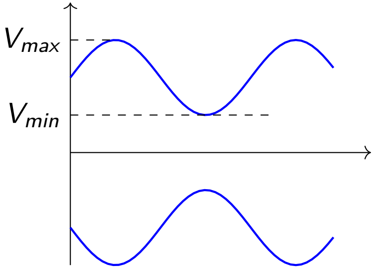
Degrees of Modulation

- **Under Modulation ($m < 1$):** $E_m < E_c$. No distortion. Ideal operating region.
- **Perfect/Critical Modulation ($m = 1$):** $E_m = E_c$. Maximum modulation without distortion.
- **Over Modulation ($m > 1$):** $E_m > E_c$. Severe distortion occurs (envelope gets clipped).



Modulation Index from Envelope Peak/Trough

- Sometimes m is measured directly from the AM waveform.
- Let V_{max} = maximum amplitude of AM wave.
- Let V_{min} = minimum amplitude of AM wave.
- $E_c = \frac{V_{max}+V_{min}}{2}$
- $E_m = \frac{V_{max}-V_{min}}{2}$
- Therefore, $m = \frac{V_{max}-V_{min}}{V_{max}+V_{min}}$



Numerical Example 1: Calculating 'm'

Problem: A carrier wave of peak voltage 10 V is amplitude modulated by a message signal of peak voltage 4 V. Find the modulation index and percentage modulation.

Solution:

- $E_c = 10$ V, $E_m = 4$ V
- $m = \frac{E_m}{E_c}$
- $m = \frac{4}{10} = 0.4$
- Percentage Modulation $= m \times 100\% = 40\%$

Numerical Example 2: V_{max} and V_{min}

Problem: An AM wave displayed on an oscilloscope has a V_{max} of 50 V and a V_{min} of 10 V. Calculate the modulation index and carrier amplitude.

Solution:

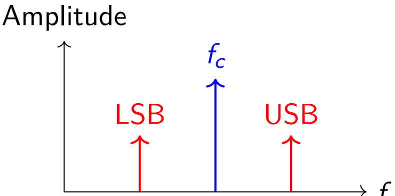
- $V_{max} = 50$ V, $V_{min} = 10$ V
- $m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \frac{50-10}{50+10}$
- $m = \frac{40}{60} = 0.667$ (or 66.7%)
- Carrier Amplitude $E_c = \frac{V_{max} + V_{min}}{2} = \frac{60}{2} = 30$ V

Key Takeaways

- AM varies the carrier's amplitude proportional to the message signal.
- Standard AM Equation: $e_{AM} = E_c[1 + m\sin(\omega_m t)]\sin(\omega_c t)$
- Modulation index $m = \frac{E_m}{E_c}$, or $m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$.
- To avoid distortion, m must be ≤ 1.

Next Lecture Preview

- Lecture 10: AM Representation in Time and Frequency Domain
- Frequency spectrum of AM
- Sidebands (LSB, USB)
- Bandwidth calculation



Thank You

Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

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Lecturer, GP Palanpur

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8. Lecture 2.1: Need of modulation & Types of modulation techniques
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

Recap: Time Domain Equation

- Recall our AM equation from last lecture:

  $$e_{AM} = E_c[1 + m\sin(\omega_m t)]\sin(\omega_c t)$$

- Let's multiply the terms out:

  $$e_{AM} = E_c\sin(\omega_c t) + mE_c\sin(\omega_m t)\sin(\omega_c t)$$

Trigonometric Expansion

- The second term is a product of two sine waves: $\sin(A)\sin(B)$
- Using trig identity: $\sin(A)\sin(B) = \frac{1}{2}[\cos(A-B) - \cos(A+B)]$
- Apply to: $mE_c\sin(\omega_c t)\sin(\omega_m t)$
  $= \frac{mE_c}{2}[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$

The Expanded AM Equation

- Substituting back, the full AM voltage equation is:

Full AM Equation

$$\begin{aligned} e_{AM} = E_c \sin(\omega_c t) & \qquad \text{(Carrier)} \\ + \frac{mE_c}{2}\cos(\omega_c - \omega_m)t & \qquad \text{(LSB)} \\ - \frac{mE_c}{2}\cos(\omega_c + \omega_m)t & \qquad \text{(USB)} \end{aligned}$$

Conclusion: An AM wave consists of THREE distinct frequency components!

The Three Frequency Components

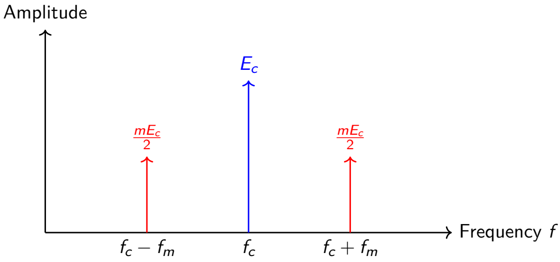
Component	Frequency	Peak Amplitude
Carrier	f_c (from ω_c)	E_c
Lower Sideband (LSB)	$f_c - f_m$	$\frac{mE_c}{2}$
Upper Sideband (USB)	$f_c + f_m$	$\frac{mE_c}{2}$

What are Sidebands?

- Sidebands are the new frequencies generated above and below the carrier frequency during modulation.
- They contain all the message information.
- The carrier itself contains NO message information (it has constant E_c amplitude).
- The amplitudes of both sidebands are equal ($\frac{mE_c}{2}$) and depend on m.

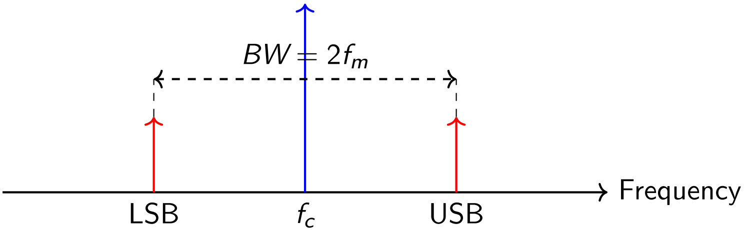
The Frequency Spectrum

- A frequency spectrum plots amplitude versus frequency.
- AM spectrum has three vertical lines:
  - Central line at f_c (tallest, height E_c)
  - Left line at $f_c - f_m$ (height $\frac{mE_c}{2}$)
  - Right line at $f_c + f_m$ (height $\frac{mE_c}{2}$)



Bandwidth of AM

- Bandwidth (BW) is the frequency range occupied by the signal.
- $BW =$ Highest Frequency $-$ Lowest Frequency
- $BW = f_{USB} - f_{LSB}$
- $BW = (f_c + f_m) - (f_c - f_m)$
- $BW = 2f_m$



Practical Implication of Bandwidth

- If a voice signal has frequencies up to 5 kHz ($f_m = 5$ kHz).
- AM Bandwidth required $= 2 \times 5$ kHz $= 10$ kHz.
- This means every AM radio station requires a 10 kHz 'slice' of the frequency spectrum.
- Regulators assign channels based on this 10 kHz spacing to avoid interference.

Numerical Example 1: Spectrum & Bandwidth

Problem: A 1 MHz carrier is AM modulated by a 10 kHz message signal. Determine the sideband frequencies and required bandwidth.

Solution

Given:
Carrier $f_c = 1$ MHz $= 1000$ kHz
Message $f_m = 10$ kHz

Calculations:

$USB = f_c + f_m = 1000 + 10 = 1010$ kHz
$LSB = f_c - f_m = 1000 - 10 = 990$ kHz
$Bandwidth = 2f_m = 2 \times 10$ kHz $= 20$ kHz

Numerical Example 2: Amplitudes

Problem: The unmodulated carrier voltage is 10V. It is modulated to a depth of 60%. Find the amplitudes of the carrier and sidebands.

Solution

Given:
Carrier amplitude $E_c = 10$ V
Modulation index $m = 0.6$

Calculations:

Sideband amplitude (each) $= \frac{mE_c}{2}$
$= \frac{0.6 \times 10}{2} = \frac{6}{2} = 3$ V

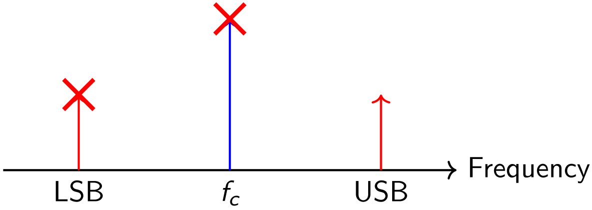
Result: Carrier is 10V, LSB is 3V, USB is 3V.

Key Takeaways

- An AM wave consists of three frequencies: Carrier (f_c), LSB ($f_c - f_m$), and USB ($f_c + f_m$).
- The message information is carried entirely in the sidebands.
- The bandwidth of a standard AM signal is twice the modulating frequency ($2f_m$).
- The frequency spectrum visualizes the amplitudes of these components.

Next Lecture Preview

- Lecture 11: Types of AM Band Spectrum
- DSB-FC (what we studied today)
- DSB-SC (Double Sideband Suppressed Carrier)
- SSB (Single Sideband)



Lecture 2.4: Types of AM band spectrum (DSB, SSB) and their applications

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Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

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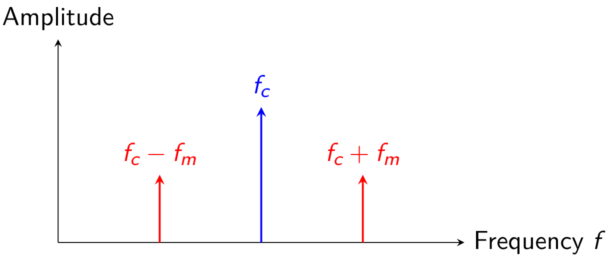
- 2. Redundancy in Standard AM
- 3. Double Sideband Suppressed Carrier (DSB-SC)
- 4. Single Sideband (SSB)
- 5. Vestigial Sideband (VSB) Overview
- 6. Applications of Each Type

Recap: Standard AM (DSB-FC)

- DSB-FC = Double Sideband Full Carrier.
- This is the standard AM we have studied so far.
- Contains Carrier (f_c), LSB (f_c - f_m), and USB ($f_c + f_m$).
- Bandwidth = 2 * f_m.

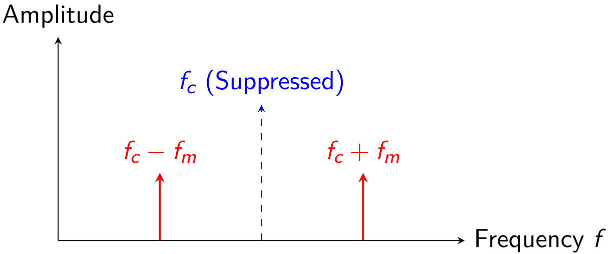


The Problem with DSB-FC

- 1. Power Inefficiency: The carrier contains NO message information, yet consumes most of the transmitter power (up to 66.6%!).
- 2. Bandwidth Inefficiency: Both LSB and USB contain the exact same information. Transmitting both uses twice the necessary bandwidth.

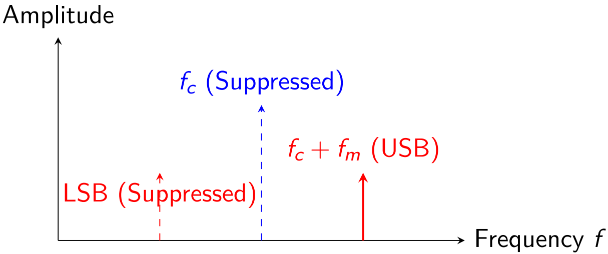
Double Sideband Suppressed Carrier (DSB-SC)

- Concept: Suppress (remove) the carrier, but transmit both sidebands.
- Result: 100% of the transmitted power contains information.
- Bandwidth: Still 2 * f_m (same as DSB-FC).
- Drawback: Complex demodulation required at receiver (needs local carrier generation).



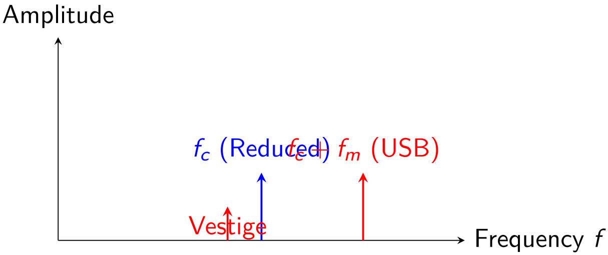
Single Sideband (SSB)

- Concept: Suppress the carrier AND one sideband (either LSB or USB).
- Result: Highly power efficient (only transmitting the necessary information part).
- Bandwidth: Reduced to f_m (half of standard AM!).
- Drawback: Even more complex to generate and receive than DSB-SC.



Vestigial Sideband (VSB)

- Concept: A compromise between DSB and SSB.
- Transmits one full sideband and a 'vestige' (small part) of the other.
- Carrier is transmitted but reduced.
- Bandwidth: slightly larger than f_m, but much less than 2*f_m.
- Easier to generate than pure SSB for low frequencies (like video).



Comparison Summary

- DSB-FC: High Power, BW $= 2f_m$, Simple Rx.
- DSB-SC: Low Power, BW $= 2f_m$, Complex Rx.
- SSB: Lowest Power, BW $= f_m$, Complex Tx/Rx.
- VSB: Medium Power, BW $\quad 1.25f_m$, Simpler Tx than SSB.

Type	**Power**	**Bandwidth**	**Complexity**
DSB-FC	High	$2f_m$	Simple Rx
DSB-SC	Low	$2f_m$	Complex Rx
SSB	Lowest	f_m	Complex Tx/Rx
VSB	Medium	$\approx 1.25f_m$	Simpler Tx than SSB

Applications: DSB-FC

- Used for standard AM Radio Broadcasting.
- Why?
- Because radio receivers must be cheap and simple for the public.
- Transmitting the carrier wastes transmitter power, but makes millions of home receivers very inexpensive.

Applications: DSB-SC

- Used in analog TV color signal transmission (chrominance).
- Used in point-to-point communication.
- Used in FM stereo broadcasting (to transmit the L-R stereo difference signal).

Applications: SSB

- Two-way radio communication (Police, Military, Marine, Aviation).
- Amateur (Ham) radio.
- Why?
- Because power efficiency and bandwidth conservation are critical, and complex receivers are acceptable for professional use.

Applications: VSB

- Analog Television Video Broadcasting.
- Why?
- Video signals contain very low frequencies (near DC). Generating pure SSB for video is practically impossible without severe distortion.
- VSB solves this by leaving a vestige of the other sideband.

Numerical Example: Bandwidth Comparison

- Problem: A voice signal with max frequency 3.4 kHz is transmitted. Find the required bandwidth for DSB-FC and SSB.
- Solution:
- $f_m = 3.4$ kHz
- For DSB-FC: BW $= 2 * f_m = 2 * 3.4$ kHz $= 6.8$ kHz
- For SSB: BW $= f_m = 3.4$ kHz
- Result: SSB saves exactly 50% of the bandwidth.

Key Takeaways

- DSB-FC (Standard AM) wastes power on the carrier and bandwidth on a redundant sideband.
- DSB-SC removes the carrier to save power.
- SSB removes the carrier and one sideband to save power AND halve the bandwidth.
- Choice depends on the tradeoff between transmitter efficiency and receiver simplicity.

Next Lecture Preview

- Lecture 12: Power Relations in AM Wave
- Calculating Total Power (P_t)
- Power in Carrier vs Power in Sidebands
- Mathematical proof of power saving in SSB

Any Questions?

Lecture 2.5: Power relations in AM wave, bandwidth, Power saving in SSB

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

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Basic Power Equation

- In electrical circuits, Power $(P) = V_{\text{rms}}^2/R$
- Where V_{rms} is the Root Mean Square voltage, and R is the resistance (usually antenna resistance).
- For a sine wave $V(t) = V_{\text{peak}} \sin(\omega t)$, $V_{\text{rms}} = V_{\text{peak}}/\sqrt{2}$.
- Therefore, $P = (V_{\text{peak}}/\sqrt{2})^2/R = V_{\text{peak}}^2/(2R)$.

$$P = \frac{V_{\text{peak}}^2}{2R}$$

Carrier Power (P_c)

- Recall the carrier component: $E_c \sin(\omega_c t)$.
- Peak voltage is E_c.
- Carrier Power $P_c = \frac{E_c^2}{2R}$
- This power contains no message information.

$$\boxed{P_c = \frac{E_c^2}{2R}}$$

Sideband Power

- Recall the LSB/USB components have peak voltage: $\frac{mE_c}{2}$.
- Power in one sideband (e.g., $P_{\text{LSB}}) = \frac{\left(\frac{mE_c}{2}\right)^2}{2R}$
- $P_{\text{LSB}} = \frac{m^2 E_c^2}{8R}$
- Since $P_c = \frac{E_c^2}{2R}$, we can rewrite this as:
- $P_{\text{LSB}} = \frac{m^2 P_c}{4}$
- Similarly, $P_{\text{USB}} = \frac{m^2 P_c}{4}$

Total AM Power (P_t)

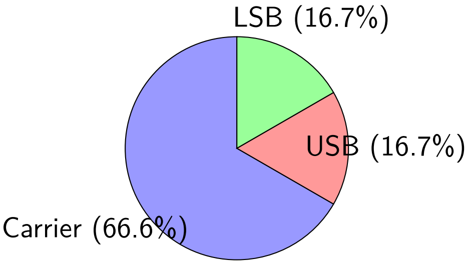
- Total Power P_t = Carrier Power + LSB Power + USB Power
- $P_t = P_c + P_{\text{LSB}} + P_{\text{USB}}$
- $P_t = P_c + \frac{m^2 P_c}{4} + \frac{m^2 P_c}{4}$
- $P_t = P_c + \frac{m^2 P_c}{2}$

Total AM Power

$$P_t = P_c \left[1 + \frac{m^2}{2}\right]$$

Maximum Power for AM

- $P_t = P_c \left[1 + \frac{m^2}{2}\right]$
- Maximum non-distorted AM occurs when $m = 1$ (100% modulation).
- Substitute $m = 1$:
- $P_t = P_c \left[1 + \frac{1^2}{2}\right] = P_c[1 + 0.5] = 1.5P_c$
- At 100% modulation, Total Power is 1.5 times the Carrier Power.



Modulation Efficiency (η)

- Efficiency (η) $= \frac{\text{Useful Power}}{\text{Total Power}}$
- Useful Power = Total Sideband Power (P_{sb}) $= \frac{m^2 P_c}{2}$
- $\eta = \frac{\frac{m^2 P_c}{2}}{P_c\left[1+\frac{m^2}{2}\right]}$
- $\eta = \frac{m^2}{2+m^2}$
- For $m = 1$: $\eta = \frac{1}{2+1} = \frac{1}{3} = 33.3\%$

Max Efficiency for Standard AM is 33.3%

Power Savings in DSB-SC

- In DSB-SC, carrier is suppressed. Saved power $= P_c$.
- % Power Saving $= \left(\frac{\text{Power Suppressed}}{\text{Total Power}}\right) \times 100$
- % Saving (DSB-SC) $= \left[\frac{P_c}{P_t}\right] \times 100$
- Since $P_t = P_c\left[1 + \frac{m^2}{2}\right] = P_c\left[\frac{2+m^2}{2}\right]$
- % Saving $= \left[\frac{2}{2+m^2}\right] \times 100$
- For $m = 1$, Saving $= \frac{2}{3} \times 100 = 66.6\%$

Power Savings in SSB

- In SSB, Carrier AND one Sideband are suppressed.
- Suppressed Power $= P_c + \frac{m^2 P_c}{4}$
- % Saving (SSB) $= \left[\frac{P_c + \frac{m^2 P_c}{4}}{P_t}\right] \times 100$
- After simplifying: % Saving $= \left[\frac{4+m^2}{4+2m^2}\right] \times 100$
- For $m = 1$: Saving $= \frac{4+1}{4+2} = \frac{5}{6} = 83.3\%$

Numerical Example 1: Total Power

Problem: An AM transmitter radiates 9 kW when the unmodulated carrier is 8 kW. Calculate the modulation index.

Solution

$P_t = 9$ kW, $P_c = 8$ kW

$P_t = P_c \left[1 + \frac{m^2}{2}\right]$

$9 = 8 \left[1 + \frac{m^2}{2}\right]$

$\frac{9}{8} = 1 + \frac{m^2}{2} \implies 1.125 = 1 + \frac{m^2}{2}$

$0.125 = \frac{m^2}{2} \implies m^2 = 0.25 \implies m = 0.5$

Problem: A 400 W carrier is modulated to a depth of 75%. Calculate total power and power in the sidebands.

Solution

$P_c = 400$ W, $m = 0.75$

$P_t = 400\left[1 + \frac{(0.75)^2}{2}\right] = 400\left[1 + \frac{0.5625}{2}\right] = 400[1.28125] = 512.5$ W

Total Sideband Power $P_{sb} = P_t - P_c = 512.5 - 400 = 112.5$ W

(Alternatively: $P_{sb} = \frac{m^2 P_c}{2} = \frac{0.5625 \times 400}{2} = 112.5$ W)

Numerical Example 3: Multiple Tones

- **Note on Multiple Modulating Signals:**
- If an AM wave is modulated by multiple signals with indices $m_1, m_2, m_3 \ldots$
- The effective modulation index m_t is:
- $m_t = \sqrt{m_1^2 + m_2^2 + m_3^2 + \ldots}$

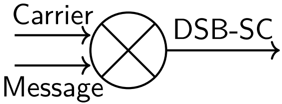
Example: $m_1 = 0.3, m_2 = 0.4$. Find m_t.
$m_t = \sqrt{0.3^2 + 0.4^2} = \sqrt{0.09 + 0.16} = \sqrt{0.25} = 0.5$

Key Takeaways

- Total AM Power: $P_t = P_c\left[1 + \frac{m^2}{2}\right]$
- Maximum efficiency of standard AM is 33.3% (at $m = 1$).
- DSB-SC saves 66.6% power (at $m = 1$).
- SSB saves 83.3% power (at $m = 1$).

Next Lecture Preview

- **Lecture 13: Generation of AM (DSB-SC)**
- How do we actually create these signals?
- Balanced Modulator Circuit
- Circuit diagram and working principle



Lecture 2.6: Generation of AM: DSBSC signal using balanced modulator circuit

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Milav Dabgar

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July 22, 2026

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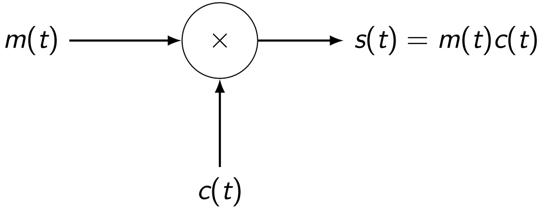
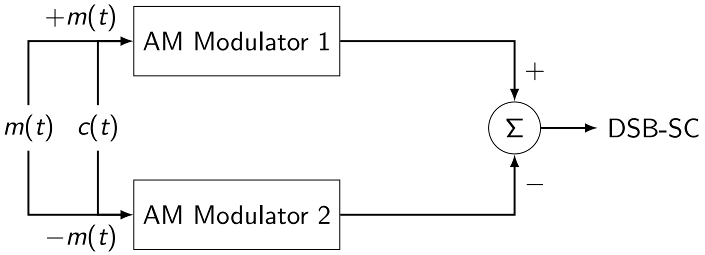
How to Generate DSB-SC?

- Recall DSB-SC equation: $e = \text{message} \times \text{carrier}$
- $e = [E_m \sin(\omega_m t)] \times [E_c \sin(\omega_c t)]$
- Notice there is **NO separate** $+E_c \sin(\omega_c t)$ term.
- Therefore, to generate DSB-SC, we need a circuit that strictly **MULTIPLIES** the two signals.
- Such a circuit is called a Product Modulator or Balanced Modulator.



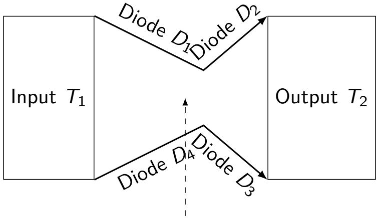
The Balanced Modulator Concept

- A balanced modulator consists of two standard non-linear modulators (like AM modulators).
- They are arranged in a balanced configuration so that the carrier signal is canceled out (suppressed) at the output.
- Only the product terms (the sidebands) appear at the output.



Diode Ring Balanced Modulator

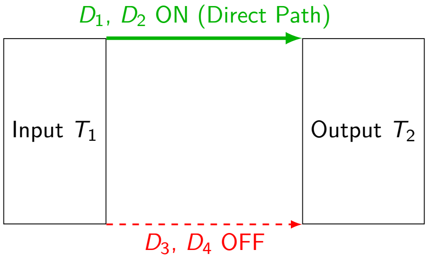
- Most common type of balanced modulator.
- Uses four diodes arranged in a ring.
- Uses two center-tapped transformers (one at input, one at output).
- Carrier acts as a high-speed electronic switch.



Carrier $c(t)$ (Switches Diodes ON/OFF)

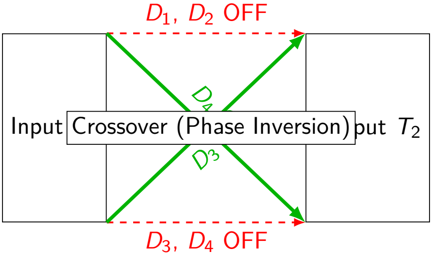
Working Principle: Positive Carrier Half-Cycle

- Carrier signal is much stronger than message signal.
- During positive half of Carrier:
  - Diodes D1 and D2 are Forward Biased (ON).
  - Diodes D3 and D4 are Reverse Biased (OFF).
- Input message passes through T1, through D1/D2, to output T2 without phase inversion.



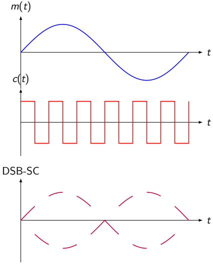
Working Principle: Negative Carrier Half-Cycle

- During negative half of Carrier:
  - Diodes D3 and D4 are Forward Biased (ON).
  - Diodes D1 and D2 are Reverse Biased (OFF).
- Input message passes through T1, through D3/D4 to output T2.
- Crucially, the connections cross over, causing a phase inversion (multiplied by -1).



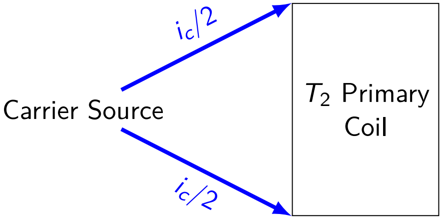
Resulting Waveforms

- The circuit multiplies the message signal by a square wave (+1, -1, +1, -1...) at the carrier frequency.
- Whenever carrier is positive, output = + message.
- Whenever carrier is negative, output = - message.
- Result is a DSB-SC waveform.



Why is the Carrier Suppressed?

- Look at the circuit when NO message is present (input = 0V).
- Carrier flows from center tap to center tap.
- Current splits equally through top and bottom halves of the output transformer primary.
- Magnetic fields cancel out perfectly. Zero voltage induced in secondary.
- Thus, carrier cannot reach the output. It is suppressed!



Opposing Magnetic Fields **Cancel Out!** Output = 0V

Mathematical Analysis (Simplified)

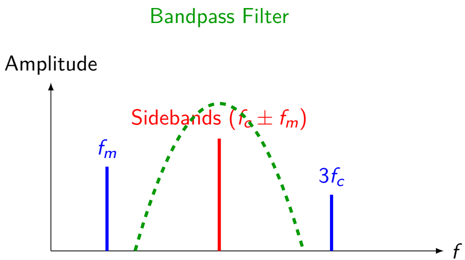
- Let Carrier square wave be $c(t)$.
- Fourier series of square wave:

$$c(t) = A \left[\cos(\omega_c t) - \frac{1}{3} \cos(3\omega_c t) + \dots \right]$$

- Output $v_o(t) = m(t) \times c(t)$
- $v_o(t) = A \left[m(t) \cos(\omega_c t) - \frac{1}{3} m(t) \cos(3\omega_c t) + \dots \right]$
- A Bandpass Filter at the output removes the $3\omega_c, 5\omega_c$ terms.

The Role of the Bandpass Filter

- The output transformer is usually tuned (forming a bandpass filter).
- It is tuned to f_c.
- It allows the desired sidebands $(f_c \pm f_m)$ to pass.
- It blocks the message frequency f_m and higher carrier harmonics $(3f_c, 5f_c)$.
- Result: Clean DSB-SC signal.



Other types of Balanced Modulators

- While Diode Ring is most common, others exist:
  1. FET Balanced Modulator
  2. BJT (Transistor) Balanced Modulator
  3. IC Balanced Modulators (e.g., LM1496)

- All operate on the same principle: **balanced cancellation of the carrier**.

Key Takeaways

- DSB-SC requires a multiplier circuit, not just an adder.
- A balanced modulator suppresses the carrier by splitting its current equally through a transformer to cancel magnetic fields.
- The diode ring modulator uses the carrier as a high-speed switch to multiply the message by $+1$ and -1.
- A bandpass filter removes unwanted harmonics.

Next Lecture Preview

Lecture 14: Demodulation of AM signal

- How do we get the message back at the receiver?
- The Envelope Detector circuit
- Working of diode detector and RC filter

AM Signal → Diode (Rectifier) → RC Filter (Smoother) → Message

Lecture 14: Demodulation of AM signal: Envelope detector using diode

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Previous Lecture Recap

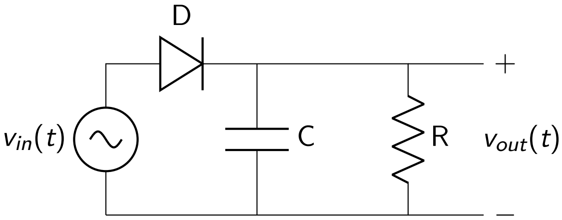
In Lecture 13, we learned how to generate an AM (DSB-SC) signal using a Balanced Modulator at the transmitter.

What is Demodulation?

- Demodulation (or Detection) is the process of extracting the original message signal from the modulated carrier wave.
- It is the exact inverse of modulation.
- Performed at the receiver.
- For standard AM, the simplest and most common demodulator is the Diode Envelope Detector.

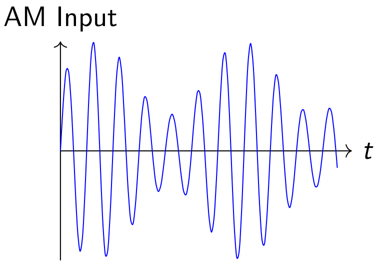
The Envelope Detector Circuit

- Extremely simple and inexpensive circuit.
- Consists of:
  1. A Diode (D) - acts as a rectifier.
  2. A Capacitor (C) - acts as a low-pass filter (smoother).
  3. A Resistor (R) - provides a discharge path for the capacitor.



Working Principle: Step 1 - Rectification

- Assume C is initially removed.
- When the AM wave is positive, the diode is forward biased (ON) and conducts.
- When the AM wave is negative, the diode is reverse biased (OFF) and blocks.
- Output without C is a Half-Wave Rectified AM signal.

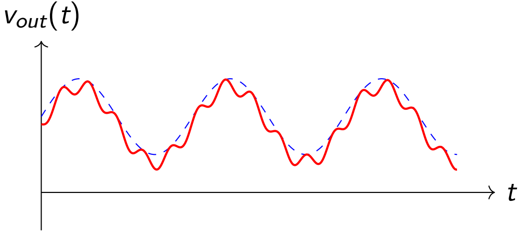


Working Principle: Step 2 - Filtering

- Now add Capacitor C.
- As the input voltage rises, diode conducts, C charges quickly to the peak voltage.
- As input voltage drops from the peak, diode turns OFF (input $<$ capacitor voltage).
- C slowly discharges through resistor R until the next high-frequency peak arrives.
- Diode turns ON again, C recharges to the new peak.

The Resulting Output

- The output voltage across the capacitor roughly follows the 'envelope' of the AM wave.
- The envelope IS the original message signal!
- There is a small high-frequency ripple left, which is easily filtered out later.
- There is also a DC offset that can be blocked by a coupling capacitor.



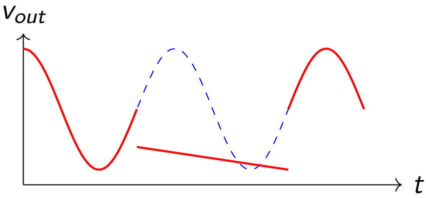
The Importance of RC Time Constant

- The rate at which C discharges is determined by the time constant $\tau = R \times C$.
- Selecting the correct R and C values is critical for accurate demodulation.
- τ must be long enough to maintain voltage between carrier peaks (filters out RF).
- τ must be short enough to allow the capacitor voltage to drop as fast as the audio envelope drops.

$$\frac{1}{f_c} \ll RC < \frac{1}{f_m}$$

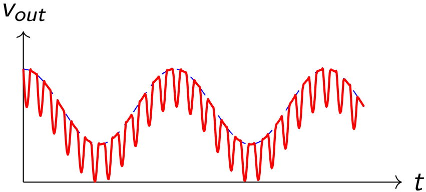
Distortion #1: Diagonal Clipping

- Occurs when RC time constant is TOO LARGE.
- Capacitor discharges too slowly.
- When the audio envelope drops rapidly, the capacitor voltage cannot drop fast enough to follow it.
- Output voltage cuts straight across (diagonally), missing the true envelope shape.



Distortion #2: Excessive Ripple

- Occurs when RC time constant is TOO SMALL.
- Capacitor discharges too quickly between carrier peaks.
- Results in a jagged, high-ripple output.
- Poor filtering of the carrier frequency.
- Can also cause negative peak clipping if m is too high.



Condition for Avoiding Diagonal Clipping

- Mathematically, to ensure the capacitor can discharge fast enough to follow the envelope:

$$RC \leq \frac{1}{\omega_m}\sqrt{\frac{1-m^2}{m}}$$

- Where $\omega_m =$ highest modulating frequency, $m =$ modulation index.
- This formula shows why over-modulation ($m > 1$) or even high modulation (m near 1) is hard to demodulate cleanly with a diode detector.

Why Diode Detectors are Popular

Advantages:

1. Extremely simple and low cost (just a diode, R, and C).
2. Does not require a local carrier generator (unlike DSB-SC demodulators).
3. Works very well for standard AM broadcasting.

Disadvantage:

- Cannot demodulate DSB-SC or SSB (requires the carrier to form the envelope).

Numerical Example: Choosing RC

Problem: An AM radio receives a 1 MHz carrier with 5 kHz audio. Suggest a suitable RC time constant.

Solution:

- Rule: $\frac{1}{f_c} \ll RC \ll \frac{1}{f_m}$
- $\frac{1}{10^6} \ll RC \ll \frac{1}{5000}$
- $1\,\mu\text{s} \ll RC \ll 200\,\mu\text{s}$
- A good choice is the geometric mean: $RC \approx \sqrt{1 \times 200} \approx 14\,\mu\text{s}$.
- If $C = 1\,\text{nF}$, then $R = 14\,\text{k}\Omega$.

Key Takeaways

- The diode envelope detector extracts the message by rectifying and filtering the AM wave.
- The capacitor charges to peak carrier voltage and discharges through the resistor to trace the envelope.
- RC must be carefully chosen: too large causes diagonal clipping, too small causes excessive ripple.
- It only works for standard AM (DSB-FC).

Next Lecture Preview

Lecture 15: Block diagram of superheterodyne receiver

- How a complete AM radio works.
- Why do we need more than just an antenna and a diode detector?
- Introduction to the Superheterodyne principle.

Lecture 2.8: Block diagram of super heterodyne receiver

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Milav Dabgar

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July 22, 2026

Recap of Lecture 14

In Lecture 14, we looked at the Diode Envelope Detector, which is the heart of an AM receiver. Today, we build the entire radio around that heart.

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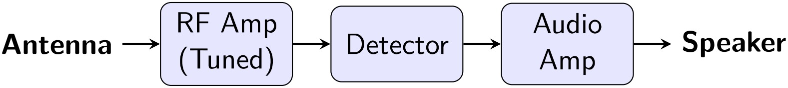
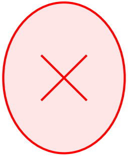
The Old Way: TRF Receiver

- **TRF** = Tuned Radio Frequency receiver.
- **Block diagram:** Antenna → RF Amplifier (tuned) → Detector → Audio Amp → Speaker.
- You had to manually tune the RF amplifier to the exact frequency of the station you wanted (e.g., 900 kHz).
- If you wanted to change stations, you had to re-tune all the RF amplifiers.



Problems with TRF Receivers

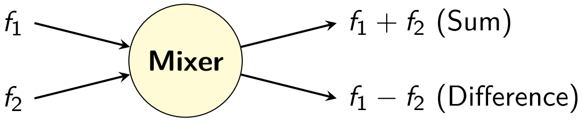
- **1. Instability:** High gain at high RF frequencies causes oscillations (howling).
- **2. Variable Bandwidth:** A tuned circuit's bandwidth changes as you change its center frequency. ($Q = \frac{f_r}{BW}$).
- **3. Poor Selectivity:** Hard to separate two stations that are close together in frequency, especially at the high end of the dial.



Poor Selectivity

The Solution: Heterodyning

- **Heterodyne** = mixing two frequencies together to create new frequencies.
- Specifically, if you mix f_1 and f_2, you get $(f_1 + f_2)$ and $(f_1 - f_2)$.
- This is done using a circuit called a '**Mixer**'.
- We can use this to shift ANY incoming radio frequency to one FIXED lower frequency.



Intermediate Frequency (IF)

- Instead of tuning our main amplifiers to different frequencies every time we change the channel...
- We 'mix' the incoming station frequency with a local oscillator to shift it to a fixed **Intermediate Frequency (IF)**.
- For AM radio, the standard IF is always **455 kHz**.
- Now, all our main amplifiers can be permanently tuned to 455 kHz!

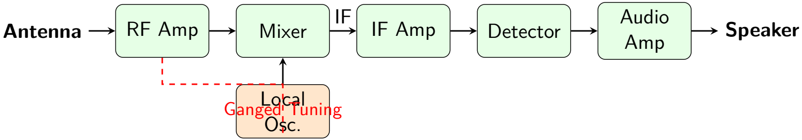
455 kHz

Standard AM IF

The Superheterodyne Block Diagram

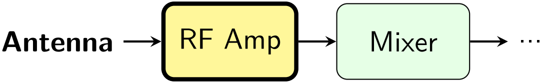
The standard architecture of almost all modern radios.
Consists of 6 main blocks:

- 1. RF Amplifier
- 2. Mixer
- 3. Local Oscillator (LO)
- 4. IF Amplifier
- 5. Detector (Demodulator)
- 6. Audio Amplifier



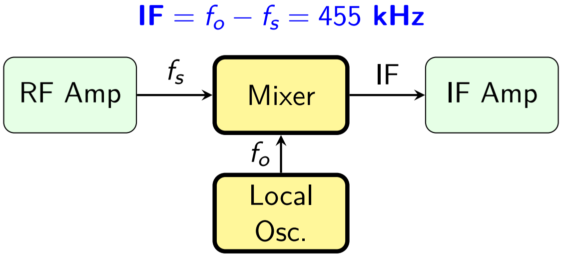
Block 1: RF Amplifier

- Connects to the receiving antenna.
- Selects the desired station frequency (f_s) and rejects others.
- Provides initial amplification to the weak incoming signal.
- Improves the Signal-to-Noise Ratio (SNR).



Blocks 2 & 3: Mixer and Local Oscillator

- **Local Oscillator (LO)** generates a steady sine wave at frequency f_o.
- **Mixer** receives f_s from the RF amp, and f_o from the LO.
- Mixer outputs the difference: $\textbf{IF} = f_o - f_s$.
- The LO and RF Amplifier are 'ganged' (tuned together) so that $f_o - f_s$ ALWAYS equals **455 kHz**.



Numerical Example: Tuning

Problem: We want to listen to an AM station at 1000 kHz. What frequency must the Local Oscillator generate? (Assume standard IF = 455 kHz).

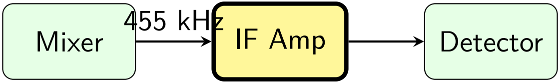
Solution:

- We know: $\text{IF} = f_o - f_s$
- $455 \text{ kHz} = f_o - 1000 \text{ kHz}$
- $f_o = 1000 \text{ kHz} + 455 \text{ kHz}$
- $f_o = 1455 \text{ kHz}$

The LO must be tuned to **1455 kHz**.

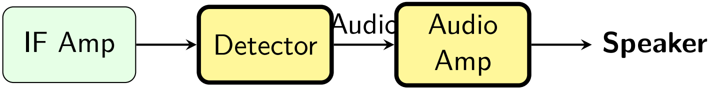
Block 4: IF Amplifier

- Receives the **455 kHz** signal from the mixer.
- This signal still has all the AM envelope information on it!
- Provides the majority of the receiver's **gain** (amplification).
- Provides the receiver's **selectivity** (rejects adjacent stations).
- Is permanently tuned to 455 kHz. Never changes.



Blocks 5 & 6: Detector and Audio Amp

- **Detector (Envelope Detector):** Extracts the original audio message from the 455 kHz IF signal.
- **Audio Amplifier:** Boosts the weak audio signal to a level strong enough to drive a loudspeaker.
- Output goes to the speaker.



Advantages of Superheterodyne

- **1. Constant Selectivity:** Because the IF is fixed, the bandwidth remains constant across the entire tuning range.
- **2. High Gain & Stability:** Easier to amplify a lower, fixed frequency (455 kHz) without oscillation.
- **3. Better Fidelity:** Fixed IF filters can be designed perfectly to pass the required sidebands without cutting off highs.

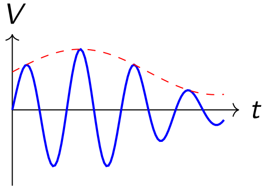
These advantages completely solved the problems of the TRF receiver.

Key Takeaways

- TRF receivers suffered from variable bandwidth and instability.
- Superheterodyne receivers mix the incoming RF (f_s) with a Local Oscillator (f_o) to create a fixed Intermediate Frequency (IF).
- Standard AM IF is **455 kHz**.
- Most amplification and filtering happens at this fixed IF.

Lecture 16: Deep Dive into the Superhet Blocks

- We will trace a signal from the antenna to the speaker.
- Look at the exact waveforms at the output of every single block.
- Understand AGC (Automatic Gain Control).



Waveform Tracing

Any Questions?

Lecture 2.9: Working of each block of super heterodyne receiver with waveforms

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Recap of Lecture 15

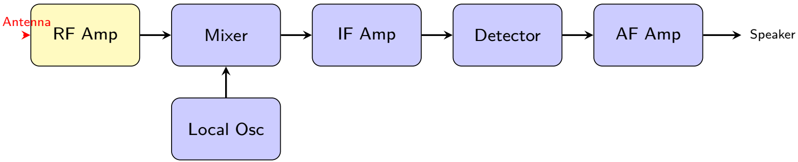
In Lecture 15, we introduced the block diagram of the Superheterodyne receiver and the concept of a fixed Intermediate Frequency (455 kHz).

Course Overview & Today's Agenda

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
3. Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)
4. Lecture 1.4: Block diagram of Analog and Digital communication system
5. Lecture 1.5: Modulation: Definition & its classification
6. Lecture 1.6: Classification based on analog & pulse signal as carrier
7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
8. Lecture 2.1: Need of modulation & Types of modulation techniques

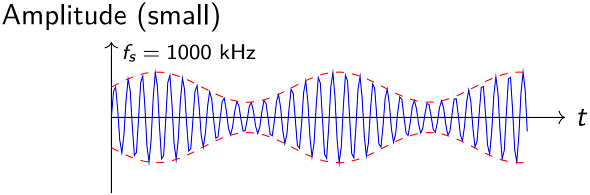
The Journey of the Signal

- Let's assume we are tuning into a station at 1000 kHz.
- The audio message is a simple 1 kHz sine wave.
- We will trace this specific signal through the six blocks.



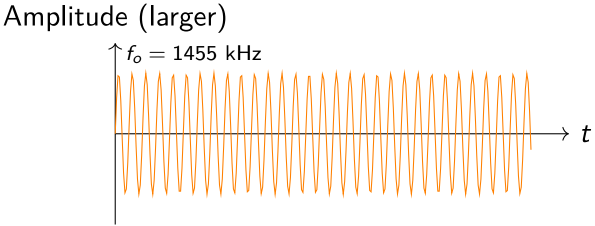
1. Antenna & RF Amplifier Output

- Antenna picks up ALL stations. It's a mess of noise and signals.
- RF Amplifier is tuned to 1000 kHz.
- It filters out other stations and amplifies the 1000 kHz AM signal.
- **Waveform:** A high-frequency (1000 kHz) carrier, modulated by a 1 kHz envelope. Amplitude is very small (microvolts).



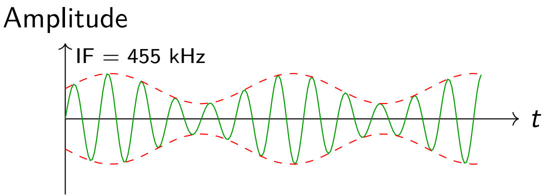
2. Local Oscillator Output

- The LO is tuned to $f_o = f_s + \text{IF}$.
- $f_o = 1000 + 455 = 1455$ kHz.
- It generates a pure, unmodulated sine wave.
- **Waveform:** A constant amplitude sine wave at 1455 kHz. Stronger than the RF signal.



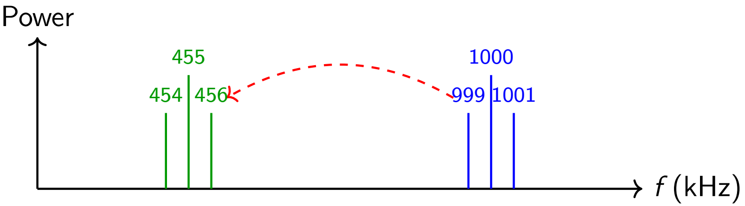
3. Mixer Output

- Mixer takes the RF (1000 kHz AM) and LO (1455 kHz pure) signals and multiplies them.
- Output contains sum (2455 kHz) and difference (455 kHz) frequencies.
- The mixer output has a tuned circuit that selects ONLY the difference frequency.
- **Waveform:** An AM wave! But now the carrier is 455 kHz, not 1000 kHz. The 1 kHz envelope is exactly the same.



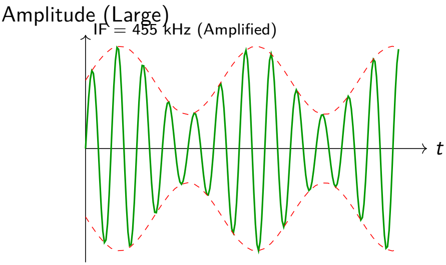
Frequency Spectrum Shift

- Let's look at it in the frequency domain.
- RF Input: Carrier at 1000, LSB at 999, USB at 1001.
- IF Output: Carrier shifted to 455, LSB to 454, USB to 456.
- The entire spectrum 'block' just slid down the frequency axis.



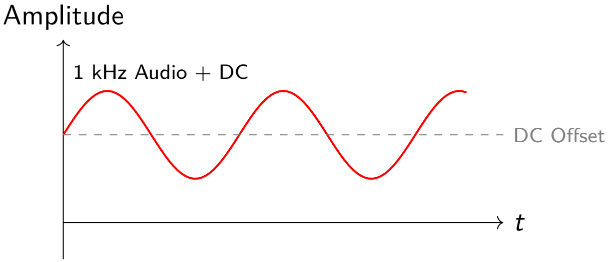
4. IF Amplifier Output

- The 455 kHz signal passes through 2 or 3 stages of IF amplification.
- It is heavily amplified.
- Adjacent channel interference is filtered out completely.
- **Waveform:** Exact same shape as Mixer output, but much LARGER amplitude (volts instead of millivolts).



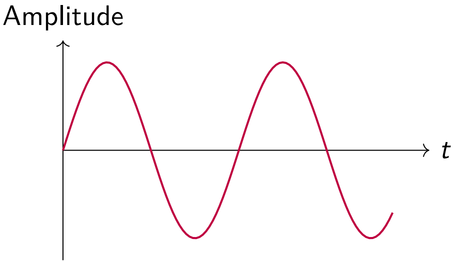
5. Detector Output

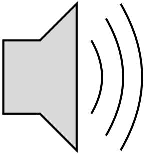
- The large 455 kHz AM signal enters the diode envelope detector.
- The diode rectifies it.
- The capacitor filters out the 455 kHz carrier.
- **Waveform:** The original 1 kHz audio sine wave! (Often with a DC offset).



6. Audio Amplifier & Speaker

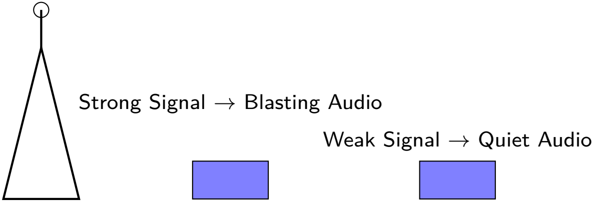
- The DC offset is blocked by a coupling capacitor.
- The 1 kHz audio is amplified to drive the speaker cone.
- **Waveform:** A large, clean 1 kHz sine wave centered on zero volts.
- We successfully transmitted and received a 1 kHz tone!





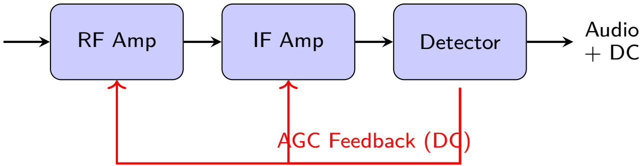
The Need for Automatic Gain Control (AGC)

- **Problem:** Radio signals vary in strength (fading).
- If you drive near a transmitter, the signal gets huge (blasting audio).
- If you drive away, the signal gets weak (quiet audio).
- **Solution:** We need a circuit that automatically turns down the volume for strong signals, and turns it up for weak signals.



How AGC Works

- Remember the DC offset at the Detector output?
- That DC voltage is proportional to the CARRIER STRENGTH.
- Strong station = High DC voltage.
- Weak station = Low DC voltage.
- AGC uses this DC voltage as a feedback signal to control the gain of the RF and IF amplifiers.



AGC Feedback Loop

- If signal increases → DC voltage increases → AGC tells IF/RF amps to reduce gain → Audio stays constant.
- If signal decreases → DC voltage decreases → AGC tells IF/RF amps to increase gain → Audio stays constant.
- **Result:** Consistent listening volume regardless of signal fading.

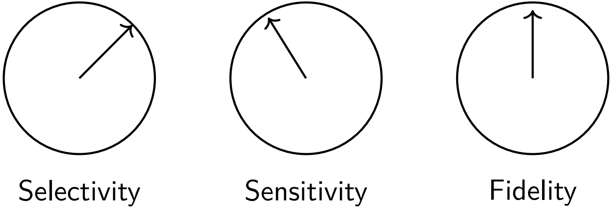
Signal UP → DC UP → Gain DOWN → Audio Maintained

Key Takeaways

- The AM envelope (the message) remains unchanged through the RF, Mixer, and IF blocks.
- The Mixer only shifts the carrier frequency from f_s to 455 kHz.
- The Detector extracts the envelope and discards the 455 kHz carrier.
- Automatic Gain Control (AGC) uses the detector's DC output to automatically adjust receiver gain, combating signal fading.

Next Lecture Preview

- **Lecture 17: Receiver Characteristics**
- How do we measure how 'good' a receiver is?
- Selectivity, Sensitivity, Fidelity.
- Image Frequency Rejection (a specific problem in superhets).



Any Questions?

Lecture 2.10: Characteristics: Selectivity, Sensitivity, Image frequency rejection and Fidelity

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Recap of Lecture 16

In Lecture 16, we traced the signal waveforms through the superheterodyne receiver and learned how AGC combats fading.

Course Overview & Today's Agenda

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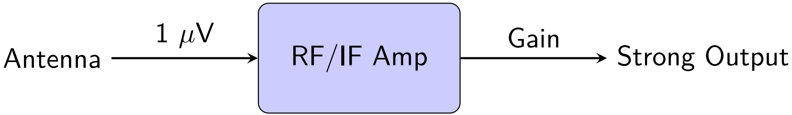
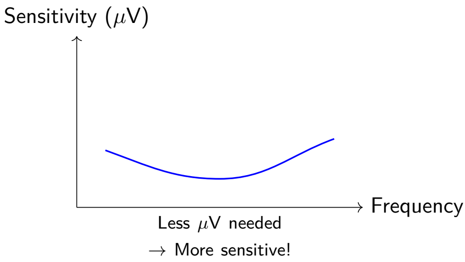
1. Sensitivity

- **Definition:** The ability of a receiver to pick up weak signals and amplify them to a usable level.
- Determined primarily by the RF Amplifier and IF Amplifier (the blocks that provide gain).
- Measured in microvolts (μV). A highly sensitive receiver might pick up a 1 μV signal.
- High sensitivity means you can hear stations that are very far away.



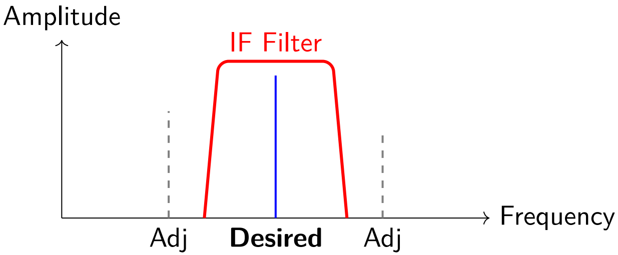
Sensitivity Curve

- Sensitivity is not always constant across the tuning dial.
- Usually plotted as Input Voltage required (to get a standard output) vs Frequency.
- **Lower voltage required = Better sensitivity.**



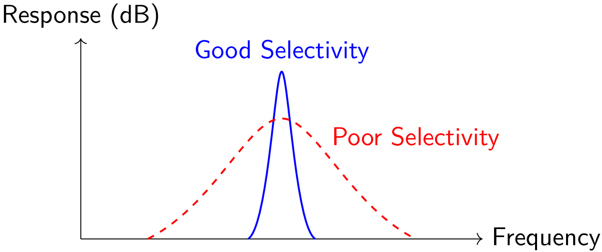
2. Selectivity

- **Definition:** The ability of a receiver to select the desired station frequency and reject all other adjacent frequencies.
- Determined almost entirely by the IF Amplifier filters.
- If selectivity is poor, you will hear two stations playing at the same time.
- Ideal selectivity is a perfect 'rectangular' bandpass filter (10 kHz wide for AM).



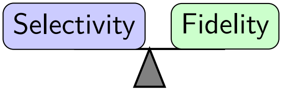
Selectivity Curve

- Plotted as Attenuation (dB) vs Frequency.
- A steep curve (steep 'skirts') means high selectivity.
- A wide, flat curve means poor selectivity.



3. Fidelity

- **Definition:** The ability of a receiver to reproduce the exact original modulating signal without distortion.
- Determined by the Audio Amplifier and the bandwidth of the IF stage.
- If IF bandwidth is too narrow (e.g., 5 kHz), it cuts off the high audio frequencies (treble), making the audio sound muffled.
- **Trade-off:** High Selectivity (narrow filter) often ruins Fidelity (cuts off high frequencies).



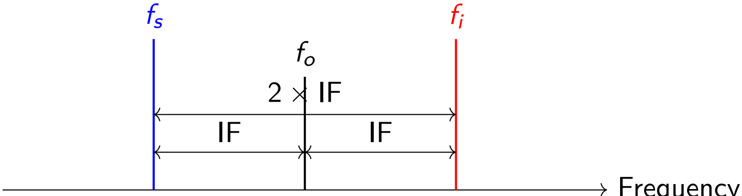
4. The Image Frequency Problem

- This is a specific flaw of the superheterodyne principle.
- Recall: Mixer produces IF when $f_o - f_s = \text{IF}$.
- But what if there is another station at frequency f_i such that: $f_i - f_o = \text{IF}$?
- The mixer will convert BOTH f_s and f_i to 455 kHz!
- Both will pass through the IF amp. You will hear two stations mixed together.

$$\boxed{f_o - f_s = \text{IF} \quad \text{AND} \quad f_i - f_o = \text{IF}}$$

Calculating Image Frequency (f_i)

- Let f_s = desired station.
- $f_o = f_s + \text{IF}$ (Local oscillator is tuned above the station).
- The unwanted image frequency f_i is above the LO by the same amount:
- $f_i = f_o + \text{IF}$
- Substitute f_o:
- $f_i = (f_s + \text{IF}) + \text{IF}$
- $f_i = f_s + 2 \times \textbf{IF}$



Numerical Example: Image Frequency

Example Problem

A receiver is tuned to 900 kHz. The IF is 455 kHz. Calculate the Local Oscillator frequency and the Image Frequency.

Solution:

- $f_s = 900$ kHz, IF $= 455$ kHz
- **1. LO Frequency:**

$$f_o = f_s + \text{IF} = 900 + 455 = 1355 \text{ kHz}$$

- **2. Image Frequency:**

$$f_i = f_s + 2 \times \text{IF}$$

$$f_i = 900 + 2(455) = 900 + 910 = 1810 \text{ kHz}$$

- If a station exists at 1810 kHz, it will cause interference.

How to Reject the Image Frequency

- Once the image frequency enters the Mixer and becomes 455 kHz, it is IMPOSSIBLE to remove it.
- It must be rejected **BEFORE** it reaches the mixer.
- This is the primary job of the **RF Amplifier** block at the front end.
- The RF amp is tuned to f_s, and its filter must be tight enough to block $f_s + 2 \times \text{IF}$.

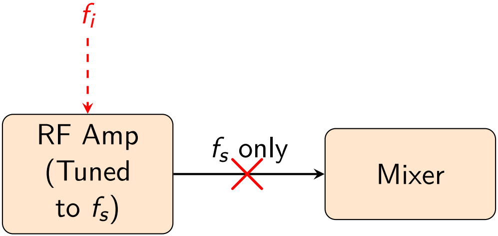


Image Frequency Rejection Ratio (IFRR)

- IFRR measures how well the receiver rejects the image frequency.
- IFRR $= \frac{\text{Gain of RF amp at desired frequency } (f_s)}{\text{Gain of RF amp at image frequency } (f_i)}$
- Usually expressed in decibels (dB).
- Higher IFRR = Better Receiver.
- Higher IF values (e.g., 455 kHz vs 100 kHz) push the image frequency further away, making it easier for the RF amp to reject it.

$$\boxed{\text{IFRR (dB)} = 20\log_{10}\left(\frac{A_{f_s}}{A_{f_i}}\right)}$$

Summary of Receiver Blocks vs Characteristics

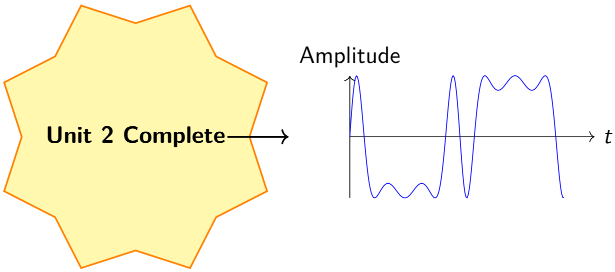
Characteristic	**Determining Block(s)**
Sensitivity	RF Amp & IF Amp (Gain)
Selectivity	IF Amp (Narrow Bandpass Filter)
Fidelity	Audio Amp & IF Amp (Wide Bandpass Filter)
Image Rejection	RF Amp (Pre-mixer filtering)

Key Takeaways

- Sensitivity is the ability to pick up weak signals; Selectivity is the ability to isolate one station.
- Fidelity ensures high-quality audio reproduction without distortion.
- Image Frequency ($f_s + 2 \times \text{IF}$) is an unwanted station that mixes down to the same IF.
- Image rejection must happen in the RF amplifier before the mixer.

End of Unit 2 Preview

- We have completed Unit 2: Amplitude Modulation!
- Next up is Unit 3: Frequency Modulation (FM).
- We will learn why FM provides better audio quality and less noise than AM.



Any Questions?

Lecture 3.1: Frequency modulation: Mathematical representation of FM signal, time & frequency domain

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

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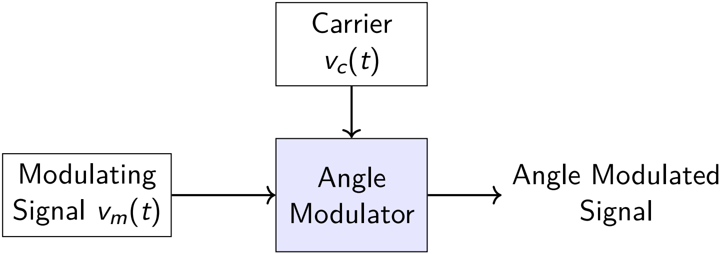
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

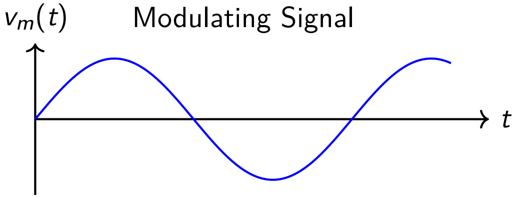
Introduction to Angle Modulation

- Angle Modulation is a class of carrier modulation
- The angle of the carrier wave is varied in accordance with the modulating signal
- Two main types: Frequency Modulation (FM) and Phase Modulation (PM)
- Amplitude remains constant, providing better noise immunity than AM



What is Frequency Modulation?

- Frequency of the carrier signal is varied proportional to the amplitude of the modulating signal
- Higher amplitude in modulating signal = Higher frequency in carrier
- Lower amplitude in modulating signal = Lower frequency in carrier
- Rate of change of frequency depends on the frequency of the modulating signal



Mathematical Representation: General Equation

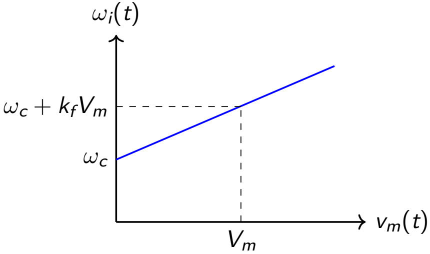
- Unmodulated carrier: $v_c(t) = V_c \cos(\omega_c t + \theta)$
- Let the total instantaneous angle be $\theta_i(t)$
- Then, general FM signal: $v_{FM}(t) = V_c \cos(\theta_i(t))$
- Instantaneous frequency $\omega_i(t) = \frac{d}{dt}\theta_i(t)$

General FM Signal

$$v_{FM}(t) = V_c \cos(\theta_i(t))$$

Instantaneous Frequency in FM

- In FM, instantaneous frequency $\omega_i(t)$ varies with modulating signal $v_m(t)$
- $\omega_i(t) = \omega_c + k_f v_m(t)$
- k_f is the frequency sensitivity constant (Hz/Volt)
- ω_c is the unmodulated carrier frequency



Deriving the FM Equation

- We know $\theta_i(t) = \int \omega_i(t)dt$
- $\theta_i(t) = \int (\omega_c + k_f v_m(t))dt$
- $\theta_i(t) = \omega_c t + k_f \int v_m(t)dt$
- Substitute back: $v_{FM}(t) = V_c \cos\left(\omega_c t + k_f \int v_m(t)dt\right)$

FM Equation with Sinusoidal Modulation

- Let $v_m(t) = V_m \cos(\omega_m t)$
- $\int v_m(t)dt = \frac{V_m}{\omega_m} \sin(\omega_m t)$
- $v_{FM}(t) = V_c \cos\left(\omega_c t + \frac{k_f V_m}{\omega_m} \sin(\omega_m t)\right)$
- Define Modulation Index $m_f = \frac{k_f V_m}{\omega_m} = \frac{\Delta f}{f_m}$

Modulation Index

$$m_f = \frac{k_f V_m}{\omega_m} = \frac{\Delta f}{f_m}$$

Final Standard FM Equation

Standard FM Equation

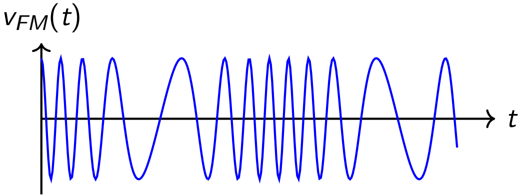
$$v_{FM}(t) = V_c \cos(\omega_c t + m_f \sin(\omega_m t))$$

Where:

- V_c = Carrier amplitude
- ω_c = Carrier angular frequency
- m_f = Modulation index
- ω_m = Modulating angular frequency

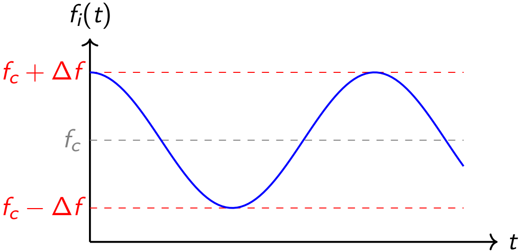
Time Domain Analysis of FM

- Amplitude is strictly constant at V_c
- Zero-crossings are not equally spaced
- Frequency is maximum when modulating signal is at positive peak
- Frequency is minimum when modulating signal is at negative peak



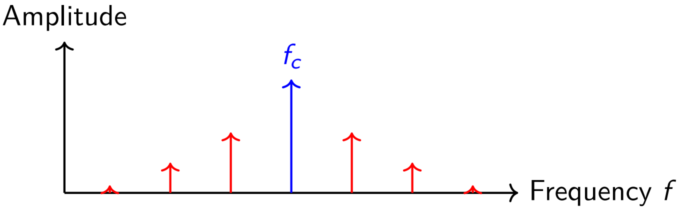
Frequency Deviation

- Frequency deviation (Δf) is the maximum departure of the FM frequency from the carrier frequency
- $\Delta f = k_f V_m$ (in Hz)
- Instantaneous frequency varies between $f_c - \Delta f$ and $f_c + \Delta f$
- Total frequency swing $= 2\Delta f$



Frequency Domain Analysis of FM (Intro)

- Unlike AM which has only two sidebands, FM has an infinite number of sidebands
- The sidebands are spaced at $f_c \pm nf_m$ where n = 1, 2, 3...
- The amplitude of these sidebands depends on the modulation index (m_f)
- This requires Bessel functions to analyze completely (covered next lecture)



Numerical Example

Problem: An FM wave is given by $v(t) = 10\cos(2\pi \times 10^6 t + 5\sin(2\pi \times 10^3 t))$. Find carrier frequency, modulating frequency, and modulation index.

Solution

Compare with: $v_{FM}(t) = V_c \cos(2\pi f_c t + m_f \sin(2\pi f_m t))$

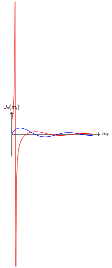
- $2\pi f_c = 2\pi \times 10^6 \implies f_c = 10^6 \text{ Hz} = 1 \text{ MHz}$
- $2\pi f_m = 2\pi \times 10^3 \implies f_m = 10^3 \text{ Hz} = 1 \text{ kHz}$
- $m_f = 5$

Key Takeaways

- Frequency Modulation varies carrier frequency with modulating amplitude
- Standard FM equation includes the modulation index
- FM has a constant amplitude, providing noise immunity
- Standard equation: $v_{FM}(t) = V_c \cos(\omega_c t + m_f \sin(\omega_m t))$
- FM produces multiple sidebands in the frequency domain

Next Lecture Preview

- Bessel's function
- Detailed look at frequency deviation ratio
- More numericals on modulation index and frequency spectrum



Lecture 3.2: Bessel's function, frequency deviation ratio, modulation index (numerical)

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Lecture Agenda

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10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

Recap: FM Signal Expansion

- Standard FM equation: $v(t) = V_c \cos(\omega_c t + m_f \sin(\omega_m t))$
- Expanding this mathematically requires trigonometric identities of the form $\cos(A + B)$
- Result:
  $v(t) = V_c[\cos(\omega_c t)\cos(m_f \sin(\omega_m t)) - \sin(\omega_c t)\sin(m_f \sin(\omega_m t))]$
- This complex form cannot be directly resolved into simple sine waves without advanced mathematics

Introduction to Bessel Functions

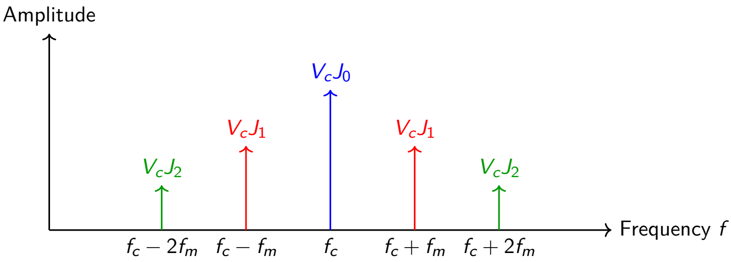
- To express the FM signal as a sum of individual sine waves (carrier + sidebands), we use Bessel functions of the first kind
- The signal becomes:

$$v(t) = V_c \sum_{n=-\infty}^{\infty} J_n(m_f) \cos((\omega_c + n\omega_m)t)$$

- $J_n(m_f)$ is the Bessel function coefficient for the nth sideband
- These coefficients determine the amplitude of each frequency component

FM Frequency Spectrum

- Carrier frequency: f_c with amplitude $V_c J_0(m_f)$
- First sideband pair: $f_c \pm f_m$ with amplitude $V_c J_1(m_f)$
- Second sideband pair: $f_c \pm 2f_m$ with amplitude $V_c J_2(m_f)$
- Infinite number of sidebands exist theoretically

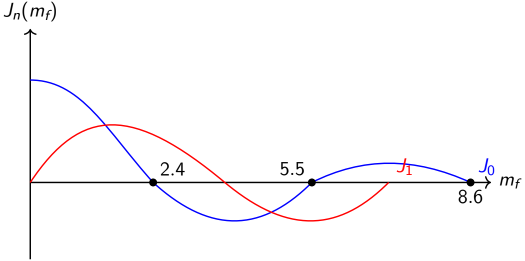


Properties of Bessel Functions

- **Property 1:** $J_{-n}(m_f) = (-1)^n J_n(m_f)$ (odd harmonics are inverted on lower sideband)
- **Property 2:** $\sum_{n=-\infty}^{\infty} J_n^2(m_f) = 1$ (Total power remains constant)
- As m_f increases, the number of significant sidebands increases
- For specific values of m_f, $J_0(m_f) = 0$ (Carrier disappears - carrier null)

Bessel Function Curves

- Plots of $J_n(m_f)$ vs m_f
- Shows how sideband amplitudes oscillate as modulation index increases
- Notice where J_0 crosses zero (approx $m_f = 2.4, 5.5, 8.6$)



Bessel Function Table

- We use tables to find $J_n(m_f)$ values for practical calculations
- Example values for $m_f = 1.0$:

  $J_0 = 0.77, J_1 = 0.44, J_2 = 0.11, J_3 = 0.02$

m_f	J_0	J_1	J_2	J_3	J_4
0.0	1.00	-	-	-	-
0.5	0.94	0.24	0.03	-	-
1.0	0.77	0.44	0.11	0.02	-
2.0	0.22	0.58	0.35	0.13	0.03
2.4	0.00	0.52	0.43	0.20	0.06

Modulation Index in FM

- Modulation index $m_f = \frac{\Delta f}{f_m}$
- Unlike AM ($0 \leq m \leq 1$), FM modulation index can be much greater than 1
- Determines the bandwidth and number of significant sidebands
- Higher m_f = wider bandwidth but better noise performance

AM	**vs**	**FM**
$m \leq 1$		m_f can be $\gg 1$

Deviation Ratio

- Deviation Ratio (D) is the worst-case (maximum) modulation index
- $D = \frac{\Delta f_{max}}{f_{m(max)}}$
- Used in designing commercial FM broadcasting systems
- For standard FM broadcast:
  - $\Delta f_{max} = 75$ kHz
  - $f_{m(max)} = 15$ kHz
  - $\Rightarrow D = \frac{75}{15} = 5$

Carson's Rule for Bandwidth

- Theoretically, FM bandwidth is infinite
- Practically, we use **Carson's Rule** for approximate bandwidth containing 98% of power

Carson's Rule Formula

$$BW = 2(\Delta f + f_m)$$

- Can also be written as: $BW = 2f_m(m_f + 1)$

Numerical Example 1

Problem: An FM carrier of 100 MHz is modulated by a 10 kHz sine wave. The maximum frequency deviation is 50 kHz. Calculate the modulation index and approximate bandwidth.

Solution

Given:

$f_c = 100$ MHz, $f_m = 10$ kHz, $\Delta f = 50$ kHz

1. **Modulation Index:**

$m_f = \frac{\Delta f}{f_m} = \frac{50 \text{ kHz}}{10 \text{ kHz}} = 5$

2. **Approximate Bandwidth (Carson's Rule):**

$BW = 2(\Delta f + f_m) = 2(50 + 10) = 120$ kHz

Numerical Example 2

Problem: If $V_c = 10V$, $m_f = 1$, find amplitudes of the carrier and first two sidebands. (Given: $J_0(1) = 0.77$, $J_1(1) = 0.44$, $J_2(1) = 0.11$)

Solution

Carrier amplitude:
$V_c J_0(1) = 10 \times 0.77 = 7.7$ V
First sideband amplitude:

$V_c J_1(1) = 10 \times 0.44 = 4.4$ V
Second sideband amplitude:

$V_c J_2(1) = 10 \times 0.11 = 1.1$ V

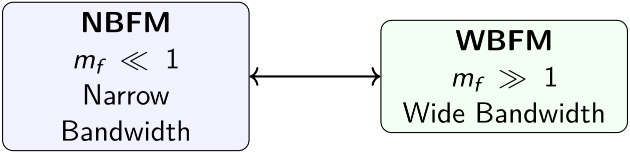
Summary

Key Takeaways:

- Bessel functions determine the amplitudes of FM sidebands
- FM has an infinite number of sidebands, but power is constant
- Deviation Ratio is the max modulation index of the system
- Carson's Rule $BW = 2(\Delta f + f_m)$ provides practical bandwidth

Next Lecture Preview

- Types of Frequency Modulation
- Narrow Band FM (NBFM)
- Wide Band FM (WBFM)
- Differences and Applications



Lecture 3.3: Types of frequency modulation (Narrow Band and Wide Band FM)

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Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

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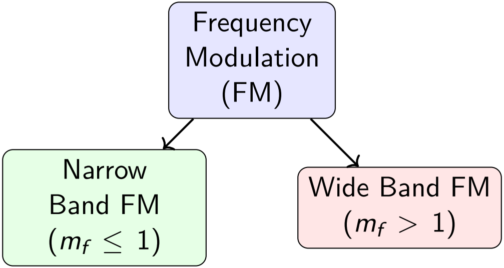
Classification of FM

- FM is classified into two types based on the modulation index (m_f):
- 1. Narrow Band FM (NBFM)
- 2. Wide Band FM (WBFM)
- The classification dictates the bandwidth required and the complexity of the equipment

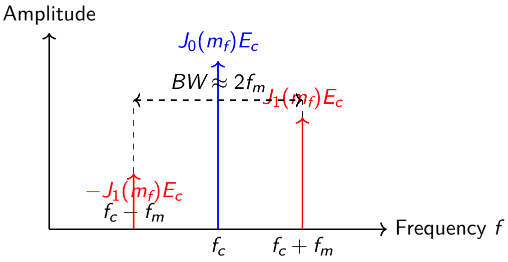


Narrow Band FM (NBFM): Definition

- FM system where modulation index $m_f \leq 1$ (usually $m_f < 0.5$)
- Maximum frequency deviation (Δf) is small
- Only the carrier and the first pair of sidebands are significant
- Higher order sidebands are negligible

NBFM Spectrum and Bandwidth

- Spectrum looks very similar to Amplitude Modulation (AM)
- Contains carrier at f_c and two sidebands at $f_c - f_m$ and $f_c + f_m$
- Bandwidth $BW \approx 2f_m$
- Difference from AM: Lower sideband is phase shifted by 180 degrees

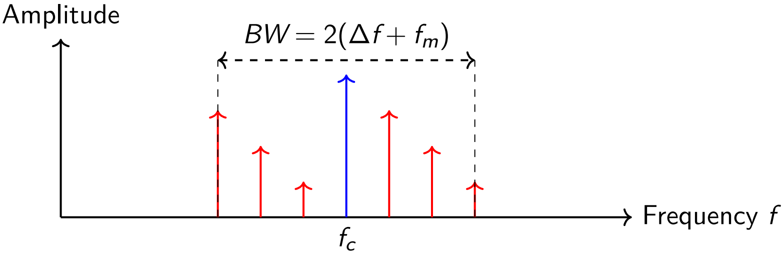


Wide Band FM (WBFM): Definition

- FM system where modulation index $m_f > 1$
- Maximum frequency deviation (Δf) is large (e.g., 75 kHz for broadcast)
- Generates a large number of significant sidebands
- Provides excellent noise immunity at the cost of bandwidth

WBFM Spectrum and Bandwidth

- Spectrum contains carrier and multiple pairs of sidebands
- Bandwidth is calculated using Carson's Rule: $BW = 2(\Delta f + f_m)$
- Much wider bandwidth required compared to NBFM or AM
- Requires higher frequency bands (VHF/UHF) for transmission



Comparison: Modulation Index & Deviation

Parameter	NBFM	WBFM
Modulation Index (m_f)	$m_f \leq 1$	$m_f > 1$ (usually 5 to 2500)
Max Freq. Deviation (Δf)	≈ 5 kHz	≈ 75 kHz

Comparison: Bandwidth & Sidebands

Parameter	**NBFM**	**WBFM**
Bandwidth	$BW \approx 2f_m$	$BW = 2(\Delta f + f_m)$
Sidebands	Only 2 significant sidebands	Many significant sidebands

Comparison: Noise & Quality

Parameter	NBFM	WBFM
Noise Immunity	Less noise immunity	High noise immunity
Audio Quality	Lower audio quality	High audio quality (high fidelity)

Applications of NBFM

- Used where bandwidth conservation is more important than high fidelity
- Mobile communication systems (Police, Fire, Ambulance)
- Marine communication
- Amateur radio (Ham radio)
- Walkie-talkies

Applications of WBFM

- Used where high fidelity and low noise are paramount
- FM Radio Broadcasting (88 MHz - 108 MHz)
- Television audio signals
- Satellite communication links
- High-quality wireless microphones

Numerical Example: NBFM vs WBFM

Problem: An audio signal of 15 kHz modulates a carrier. Determine bandwidth if (a) $\Delta f = 5$ kHz (NBFM case), (b) $\Delta f = 75$ kHz (WBFM case).

Solution (a) NBFM

$m_f = \frac{5}{15} = 0.33 < 1$
$BW \approx 2f_m = 2(15) = 30$ kHz.

Solution (b) WBFM

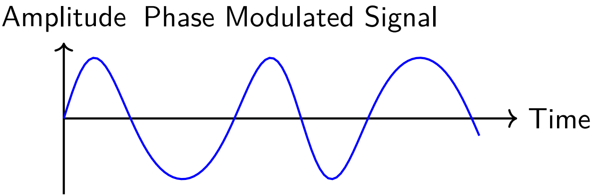
$m_f = \frac{75}{15} = 5 > 1$
$BW = 2(\Delta f + f_m) = 2(75 + 15) = 180$ kHz.

Summary

- NBFM has $m_f \leq 1$, small deviation, and bandwidth like AM ($2f_m$)
- WBFM has $m_f > 1$, large deviation, and uses Carson's Rule for bandwidth
- NBFM is used for voice comms (walkie-talkies); WBFM is used for high-fidelity broadcasting

Next Lecture Preview

- Phase Modulation (PM)
- Definition and waveforms
- Applications of PM
- How PM differs from FM



Lecture 3.4: Phase Modulation: definition, waveforms and applications

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

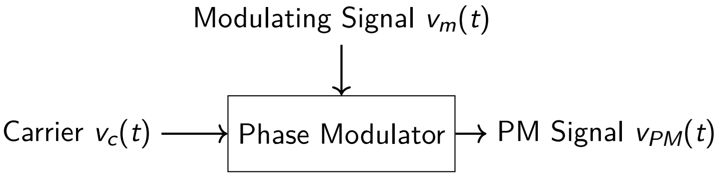
Lecturer, GP Palanpur

July 22, 2026

Lecture Agenda

What is Phase Modulation?

- Phase Modulation (PM) is a form of angle modulation
- The phase angle of the carrier signal is varied proportionally to the amplitude of the modulating signal
- Carrier amplitude remains constant
- Carrier frequency also remains constant, but phase changes cause instantaneous frequency changes



Mathematical Representation of PM

- Unmodulated carrier: $v_c(t) = V_c \cos(\omega_c t + \theta)$
- In PM, instantaneous phase angle $\theta_i(t)$ varies with modulating signal $v_m(t)$

  $\theta_i(t) = \omega_c t + k_p v_m(t)$
- Where k_p is the phase sensitivity constant (radians/Volt)

PM Equation with Sinusoidal Modulation

- Let $v_m(t) = V_m \cos(\omega_m t)$
- Substitute into instantaneous phase:
  $\theta_i(t) = \omega_c t + k_p V_m \cos(\omega_m t)$
- The PM signal is $v_{PM}(t) = V_c \cos(\theta_i(t))$
- Result:

Sinusoidal PM Equation

$$v_{PM}(t) = V_c \cos(\omega_c t + k_p V_m \cos(\omega_m t))$$

Modulation Index in PM

- Define PM modulation index $m_p = k_p V_m$
- Unit is radians
- Final standard equation:

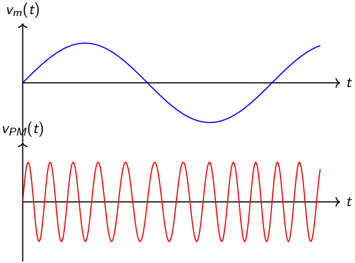
  $v_{PM}(t) = V_c \cos(\omega_c t + m_p \cos(\omega_m t))$
- Notice that m_p depends ONLY on the amplitude of modulating signal, not its frequency

Instantaneous Frequency in PM

- Frequency is the derivative of phase:
  $\omega_i = \frac{d}{dt}\theta_i(t) = \frac{d}{dt}[\omega_c t + m_p \cos(\omega_m t)]$
  $\omega_i = \omega_c - m_p \omega_m \sin(\omega_m t)$
- Frequency deviation in PM is $\Delta f = m_p f_m = k_p V_m f_m$
- **In PM, frequency deviation is proportional to modulating frequency!**

PM Waveforms

- Amplitude is constant
- When modulating signal rises, phase advances
- When modulating signal falls, phase retards
- Zero-crossings shift back and forth



Comparing PM and FM Equations

- FM: $v(t) = V_c \cos(\omega_c t + m_f \sin(\omega_m t))$
- PM: $v(t) = V_c \cos(\omega_c t + m_p \cos(\omega_m t))$
- They are mathematically identical except for a 90-degree phase shift in the modulation term
- An FM modulator can produce PM if the modulating signal is first differentiated
- A PM modulator can produce FM if the modulating signal is first integrated

$$v_m(t) \longrightarrow \boxed{\int dt} \longrightarrow \boxed{\text{PM Modulator}} \rightarrow \text{FM Output}$$

Numerical Example

Problem: A PM signal is $v(t) = 10\cos(2\pi \times 10^6 t + 2\cos(2\pi \times 10^3 t))$. Find modulation index, carrier frequency, and max phase deviation.

Solution

Compare to: $V_c \cos(2\pi f_c t + m_p \cos(2\pi f_m t))$

Values from equation:

$2\pi f_c = 2\pi \times 10^6 \implies f_c = 10^6$ Hz $= 1$ MHz
$m_p = 2$

Results:

Carrier frequency $f_c = 1$ MHz
Modulation index $m_p = 2$ rad
Max phase deviation $= m_p = 2$ rad

Advantages and Disadvantages of PM

Advantages:

- Simple to generate
- Better noise immunity than AM

Disadvantages:

- Phase ambiguity (receiver doesn't know absolute reference phase)
- Requires complex receiving equipment compared to AM

Applications of Phase Modulation

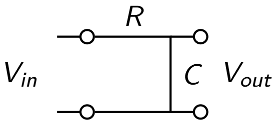
- Rarely used for analog transmission (like voice or music)
- Extensively used in Digital Communication systems (PSK, BPSK, QPSK)
- Used in Wi-Fi, Satellite communications, and mobile networks (4G/5G)
- Indirect generation of FM signals (Armstrong method)

Summary: Key Takeaways

- PM varies the phase of the carrier proportionally to the modulating signal
- Standard equation: $v_{PM}(t) = V_c \cos(\omega_c t + m_p \cos(\omega_m t))$
- Modulation index m_p depends only on modulating amplitude
- PM is highly utilized in digital communications
- PM is mathematically similar to FM but with key differences in modulation index dependencies
- FM can be generated using a PM modulator and an integrator

Next Lecture Preview

- Pre-emphasis and De-emphasis circuits
- Why high frequencies are noisy in FM
- How to improve Signal-to-Noise Ratio (SNR) in FM systems



Lecture 3.5: Pre-emphasis and De-emphasis

Complete Lecture Deck – All 45 Lectures

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Lecturer, GP Palanpur

July 22, 2026

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The Noise Problem in FM

- In FM systems, noise is not uniformly distributed across the audio spectrum
- Noise power spectral density increases quadratically (parabolically) with frequency
- Meaning: High-frequency audio signals suffer from much higher noise than low-frequency signals
- Unfortunately, high-frequency audio components (harmonics, cymbals, 's' sounds) naturally have low amplitudes

The Solution: A Two-Part System

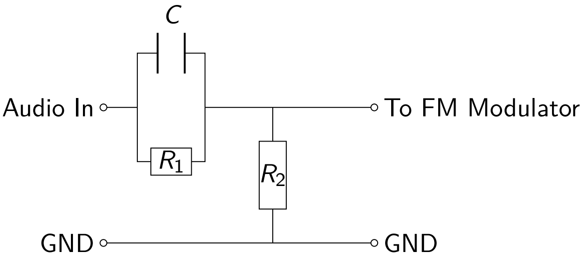
- To improve SNR at high frequencies, we use a two-step filtering process:
- **1. Pre-emphasis at the Transmitter:** Artificially boost the amplitude of high-frequency audio signals before modulation
- **2. De-emphasis at the Receiver:** Attenuate the high-frequency signals after demodulation to restore original levels
- This process significantly reduces the high-frequency noise introduced during transmission

What is Pre-emphasis?

- Pre-emphasis is a high-pass filtering process
- It amplifies high frequencies while leaving low frequencies relatively untouched
- Applied to the modulating (audio) signal *before* it reaches the FM modulator
- Result: High-frequency signals are transmitted with a larger frequency deviation, overpowering the channel noise

Pre-emphasis Circuit

- Typically implemented using a simple High-Pass RC network
- Consists of a resistor (R) and a capacitor (C) in parallel, followed by another resistor
- Cutoff frequency $f_c = \frac{1}{2\pi RC}$
- Frequencies above f_c are boosted

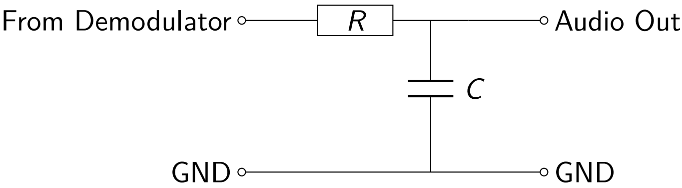


What is De-emphasis?

- De-emphasis is the exact inverse of pre-emphasis
- It is a low-pass filtering process
- Applied to the demodulated audio signal at the receiver
- Attenuates the artificially boosted high frequencies back to their original levels
- Crucially: It also attenuates the high-frequency noise picked up in the channel!

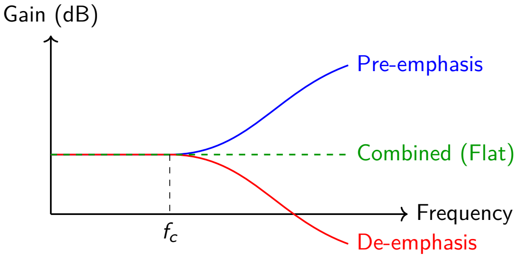
De-emphasis Circuit

- Implemented using a simple Low-Pass RC network
- Consists of a series resistor (R) and a shunt capacitor (C)
- Must have the exact same time constant ($\tau = RC$) as the pre-emphasis circuit
- Frequencies above f_c are attenuated



Combined Frequency Response

- Pre-emphasis curve (Boosts high freq)
- De-emphasis curve (Cuts high freq)
- Combined response is flat (Overall gain = 1 across all audio frequencies)
- Original audio fidelity is preserved while SNR is vastly improved



Standard Time Constants

- The amount of boost/cut is determined by the RC time constant $\tau = R \times C$
- Global standards exist so any radio can receive any station
- In North America (FCC standard): $\tau = 75\mu s$
- In Europe and most of the world: $\tau = 50\mu s$

Numerical Example

Problem: A de-emphasis circuit in an FM receiver has a time constant of $75\mu s$. If the capacitor is $0.01\mu F$, find the required resistance value.

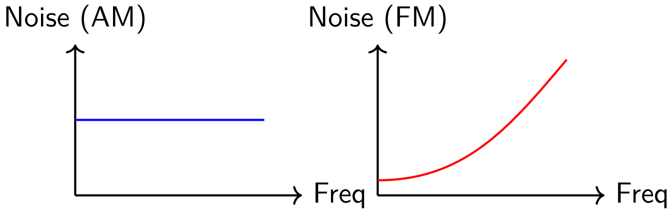
Solution

Time constant $\tau = R \times C$
Given $\tau = 75 \times 10^{-6}$ seconds, $C = 0.01 \times 10^{-6}$ Farads
$R = \frac{\tau}{C} = \frac{75 \times 10^{-6}}{0.01 \times 10^{-6}} = 7500\ \Omega$

Result: $R = 7.5$ kΩ

Why not use this in AM?

- AM noise is evenly distributed across the spectrum
- Pre-emphasizing high frequencies in AM would cause overmodulation (clipping)
- Therefore, pre-emphasis/de-emphasis is exclusively an FM technique



Key Takeaways

- Pre-emphasis and De-emphasis combat high-frequency noise in FM
- Pre-emphasis is applied at the transmitter; De-emphasis at the receiver
- The circuits are simple RC filters governed by a standard time constant

Next Lecture Preview

- Generation of FM waves
- Direct and Indirect methods
- Phase Locked Loop (PLL) as an FM modulator

Lecture 3.6: Generation techniques for FM wave: Phase locked loop FM modulator

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

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7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
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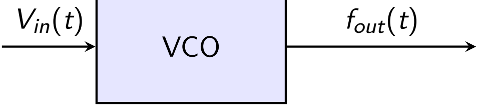
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

Methods of FM Generation

- Two main categories of generating FM signals:
- 1. Direct Method: The carrier frequency is directly varied by the modulating signal
- 2. Indirect Method: A Phase Modulator is used to generate Narrow Band FM, which is then multiplied to Wide Band FM (Armstrong Method)
- Today, most modern transmitters use the Direct Method via a Phase Locked Loop (PLL)

The Core of Direct FM: The VCO

- VCO = Voltage Controlled Oscillator
- An electronic oscillator whose output frequency is directly proportional to its input voltage
- If input voltage increases, output frequency increases
- If we feed an audio (modulating) signal into a VCO, the output is an FM signal!
- Problem: Free-running VCOs drift in frequency due to temperature and aging

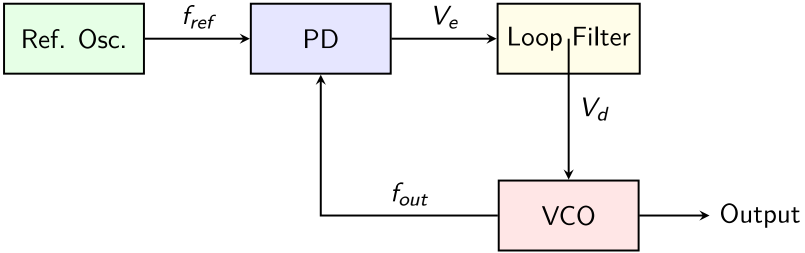


Enter the Phase Locked Loop (PLL)

- A Phase Locked Loop (PLL) is a control system that generates an output signal whose phase is related to the phase of an input reference signal
- It 'locks' the unstable VCO frequency to a highly stable reference frequency (like a Crystal Oscillator)
- Widely used in modern radios, TVs, and cell phones for frequency synthesis and modulation

PLL Block Diagram

- Three main components in a basic PLL:
- 1. Phase Detector (PD): Compares phase of reference input and VCO output
- 2. Loop Filter (LF): Low-pass filter that smooths the PD output to a DC voltage
- 3. Voltage Controlled Oscillator (VCO): Adjusts frequency based on LF voltage

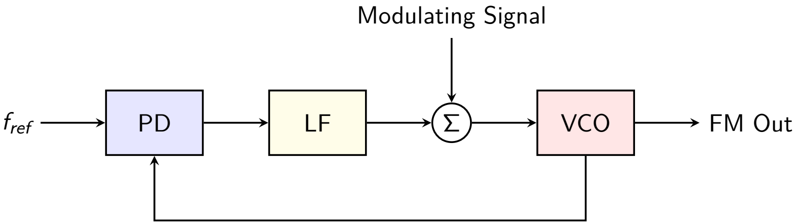


How a PLL Works (The Loop)

- 1. Reference oscillator (very stable) feeds the Phase Detector
- 2. VCO output also feeds the Phase Detector via feedback
- 3. Phase Detector outputs an error voltage proportional to their phase difference
- 4. Loop filter removes high-frequency noise from the error voltage
- 5. Filtered error voltage adjusts the VCO frequency until it exactly matches the reference

PLL as an FM Modulator

- How do we modulate it if it's locked to a fixed frequency?
- We inject the modulating (audio) signal directly into the VCO control voltage, summing it with the loop filter's error voltage
- The Loop Filter is designed with a very low cutoff frequency
- It corrects slow drifts but ignores the fast changes caused by the audio signal



Working of the PLL FM Modulator

- Slow drifts (e.g., temperature changes) pass through the loop filter and are corrected by the Phase Detector
- Fast audio frequencies (e.g., 1 kHz sine wave) are blocked by the loop filter from the feedback correction
- Therefore, the VCO frequency changes rapidly according to the audio (FM modulation) while maintaining an average center frequency locked to the crystal

Advantages of PLL FM Modulator

- ✓ Excellent frequency stability (center frequency is crystal controlled)
- ✓ Direct FM generation (no complex multiplier chains like the Armstrong method)
- ✓ Easily implemented as an Integrated Circuit (IC)
- ✓ Linear modulation over a wide frequency deviation

Summary

Key Takeaways

- Direct FM generation is typically achieved using a Voltage Controlled Oscillator (VCO)
- A Phase Locked Loop (PLL) is used to stabilize the center frequency of the VCO
- Audio is injected into the VCO, and the loop filter prevents the PLL from cancelling out the modulation

Next Lecture Preview

- How do we extract the audio back out?
- Basics of FM Demodulation
- The Ratio Detector circuit

Looking Ahead

Next lecture covers the basics of FM demodulators, specifically the Ratio detector.

Lecture 3.7: Basics and types of FM demodulators: Ratio detector

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

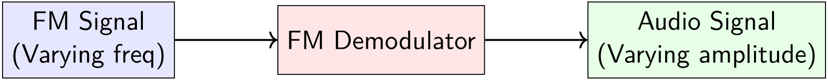
July 22, 2026

Lecture Agenda

- What is FM Demodulation?
- Requirements of an FM Demodulator
- Types of FM Demodulators
- The Ratio Detector Circuit
- How the Ratio Detector Works
- Advantages of the Ratio Detector

What is FM Demodulation?

- FM demodulation is the process of extracting the original modulating (audio) signal from the frequency-modulated carrier
- Also known as Frequency Discrimination
- The circuit must produce an output voltage whose amplitude is directly proportional to the instantaneous frequency of the input signal
- Input: Varying Frequency $\rightarrow$ Output: Varying Voltage



Requirements of an FM Demodulator

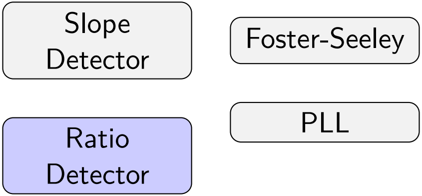
- 1. Linearity: Output voltage must be exactly proportional to frequency deviation
- 2. **Amplitude Independence**: Output should NOT change if the input signal's amplitude changes (should reject AM noise)
- 3. High conversion efficiency

Note

If a demodulator is sensitive to amplitude changes, a separate 'Limiter' circuit must be used before it.

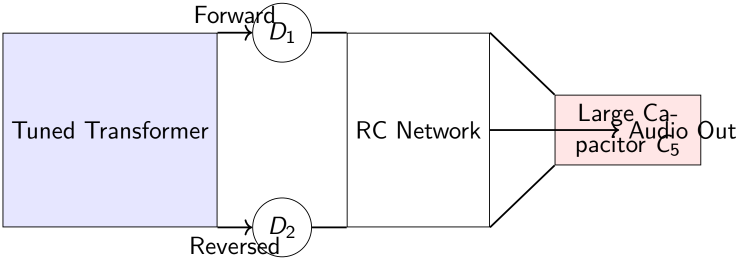
Types of FM Demodulators

- Slope Detector (Simple but non-linear)
- Foster-Seeley Discriminator (Linear, but requires a pre-limiter)
- Ratio Detector (Linear, and has built-in amplitude limiting)
- Phase Locked Loop (PLL) Demodulator (Modern, highly linear, IC based)



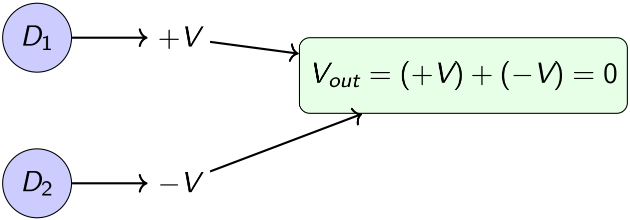
The Ratio Detector Circuit

- Consists of a tuned transformer, two diodes, capacitors, and resistors
- Key distinguishing feature: The two diodes (D_1 and D_2) face in **OPPOSITE** directions relative to the circuit loop
- Also features a large stabilizing capacitor (C_5) across the output resistors



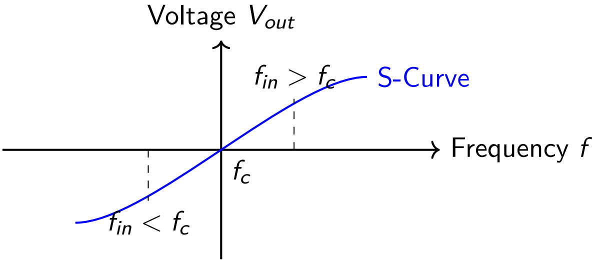
Working Principle: Center Frequency

- When input frequency $f_{in} = f_c$ (carrier frequency, no modulation):
- The phase relationships in the tuned transformer cause equal voltages to be applied to both diodes
- Both diodes conduct equally
- The voltage drops across the output resistors are equal and opposite
- Net output voltage $V_{out} = 0$



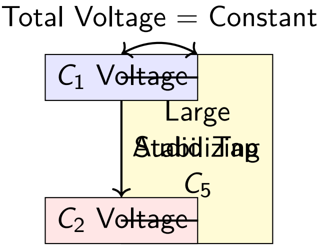
Working Principle: Frequency Deviation

- **When $f_{in} > f_c$:**
  - Phase shift in transformer changes. Voltage at D_1 increases, D_2 decreases
  - D_1 conducts more heavily. Output voltage swings **Positive**
- **When $f_{in} < f_c$:**
  - Voltage at D_2 increases, D_1 decreases
  - D_2 conducts more heavily. Output voltage swings **Negative**



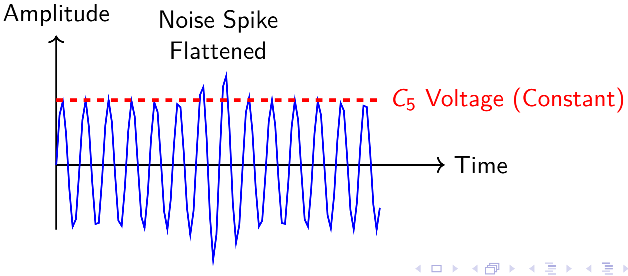
The 'Ratio' in Ratio Detector

- The total voltage across the two output capacitors remains constant, fixed by the large stabilizing capacitor (C_5)
- However, the **ratio** of the voltages across the individual capacitors changes with frequency
- The output audio is tapped from the center point, measuring this changing ratio
- Hence the name: Ratio Detector



Built-in Amplitude Limiting

- **Why is it better than Foster-Seeley?**
- The large capacitor C_5 charges to the peak average voltage of the carrier
- Because it's a large capacitor, its voltage cannot change rapidly
- If a sudden noise spike (amplitude change) hits, C_5 absorbs it, preventing it from affecting the output
- **Therefore, the Ratio Detector does not need a preceding Limiter circuit!**



Advantages and Disadvantages

Advantages

- Excellent AM rejection (built-in limiting)
- Good linearity
- Lower cost (no limiter stage needed)

Disadvantages

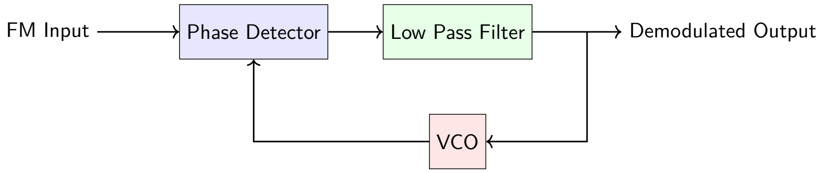
- Complex transformer tuning required
- Difficult to integrate into modern ICs

Summary

- An FM demodulator converts frequency variations back into voltage variations (audio)
- The Ratio Detector is a classic analog demodulator
- It uses a specialized transformer and oppositely-facing diodes
- Its major advantage is built-in amplitude limiting due to a large stabilizing capacitor

Next Lecture Preview

- PLL as an FM Demodulator
- How the Phase Locked Loop works in reverse
- Why PLL is the modern standard for demodulation



Lecture 3.8: PLL as FM demodulator

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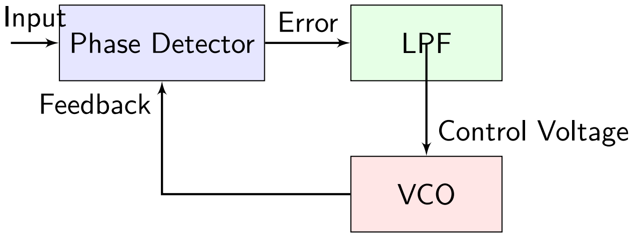
July 22, 2026

Lecture Agenda

Review: PLL Block Diagram

- Three main components:
- 1. Phase Detector (PD)
- 2. Low-Pass Filter (LPF) / Loop Filter
- 3. Voltage Controlled Oscillator (VCO)



The 'Locking' Concept (Review)

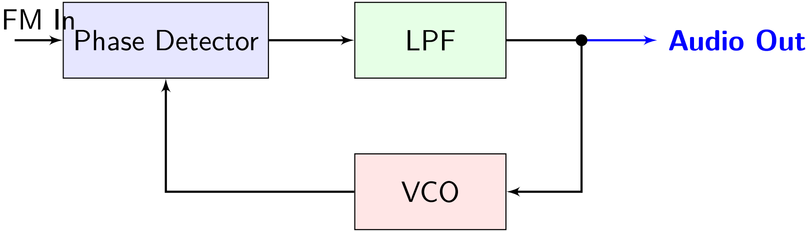
- The Phase Detector compares the Input signal frequency with the VCO frequency
- It generates an error voltage proportional to the phase/frequency difference
- The LPF smooths this error voltage
- The smoothed voltage forces the VCO to change its frequency until it exactly matches the Input frequency
- When frequencies match, the PLL is 'Locked'

PLL as an FM Demodulator

- Imagine the Input signal is an FM wave (its frequency is constantly changing with the audio)
- To stay 'Locked', the VCO must constantly change its frequency to match the incoming FM wave
- What tells the VCO to change its frequency? The error voltage from the LPF!
- Therefore, the error voltage must be an exact replica of the original audio signal that caused the FM wave to change in the first place!

Extracting the Audio

- In a PLL demodulator, the output is NOT taken from the VCO
- The demodulated Audio Output is tapped directly after the Low-Pass Filter (the VCO control voltage line)
- As the FM signal deviates up in frequency, the control voltage goes up
- As the FM signal deviates down, the control voltage goes down



Role of the Low-Pass Filter

- In the demodulator, the LPF must be designed carefully
- Its cutoff frequency must be high enough to pass the highest audio frequency (e.g., 15 kHz)
- But low enough to filter out high-frequency noise and the RF carrier components from the Phase Detector
- This is different from the PLL modulator, where the LPF had to be very slow

Amplitude Rejection in PLL

- Does a PLL need a limiter?
- Generally, Phase Detectors (like the XOR gate or Gilbert cell types used in ICs) are somewhat sensitive to amplitude variations
- However, most PLL ICs include a built-in Limiter amplifier stage right before the Phase Detector
- Therefore, a complete PLL demodulator system has excellent AM rejection

Advantages of PLL Demodulators

- **1. Highly Linear:** VCO voltage-to-frequency conversion is very linear, resulting in distortion-free audio
- **2. No Coils/Transformers:** Unlike the Ratio detector, it uses no inductors, making it perfect for Integrated Circuits (ICs)
- **3. High Noise Immunity:** The loop naturally ignores noise that doesn't fit the expected frequency tracking profile
- **4. Easy to Tune:** Center frequency is set by a simple resistor or capacitor, not by tuning fragile transformers

Real-World Application: The NE565 IC

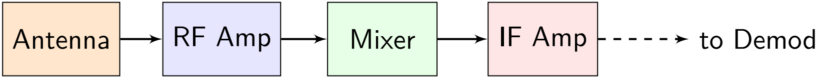
- The NE565 (or LM565) is a classic, widely used PLL IC
- Contains the Phase Detector, Amplifier, and VCO on a single chip
- External resistor and capacitor set the free-running frequency
- Used for FM demodulation, FSK (frequency shift keying) decoding in modems, and tone decoding

Key Takeaways

- A PLL demodulates FM by using its error control voltage as the audio output
- The VCO in the PLL tracks the instantaneous frequency of the incoming FM signal
- PLLs are preferred in modern electronics because they do not require tuning coils

Next Lecture Preview

- The Complete FM Receiver
- Superheterodyne architecture for FM
- Block-by-block working and waveforms



Lecture 3.9: FM Receiver: Block diagram and working with waveforms

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July 22, 2026

Lecture Agenda

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
3. Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)
4. Lecture 1.4: Block diagram of Analog and Digital communication system
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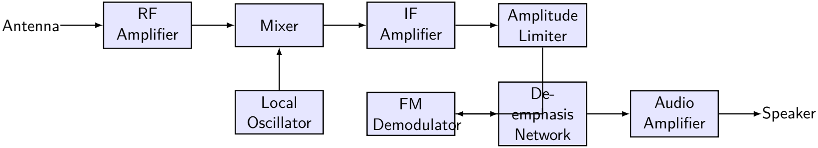
The Superheterodyne Architecture

- Just like AM receivers, modern FM receivers use the Superheterodyne architecture
- The incoming RF signal is mixed down to a fixed **Intermediate Frequency (IF)**
- Why? It is much easier to design high-gain, highly selective amplifiers for a fixed frequency than for a tunable one
- The architecture is similar to AM, but operates at higher frequencies and includes specific FM stages (Limiter, De-emphasis)

FM Receiver Block Diagram



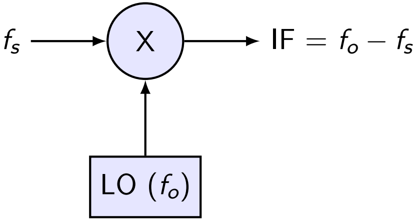
- Follows Superheterodyne principle

1. RF Amplifier

- Receives weak FM signals (88 MHz - 108 MHz) from the antenna
- Provides initial amplification to improve the Signal-to-Noise Ratio (SNR)
- Tunes to the desired station frequency (f_s)
- Rejects image frequencies before they reach the mixer

2. Mixer and Local Oscillator

- The Local Oscillator (LO) generates a steady sine wave at frequency f_o
- Usually, $f_o > f_s$ (super-heterodyne)
- The Mixer combines the incoming FM signal (f_s) and the LO signal (f_o)
- Produces sum and difference frequencies: $f_o + f_s$ and $f_o - f_s$
- The difference frequency ($f_o - f_s$) is the Intermediate Frequency (IF)



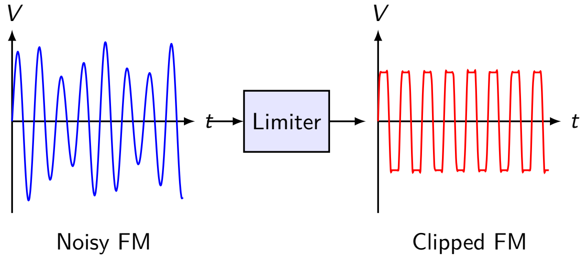
3. The Standard FM IF

$$\boxed{\textbf{IF} = \textbf{10.7 MHz}}$$

- For standard FM broadcasting, the IF is universally set to 10.7 MHz
- This is much higher than the AM IF (455 kHz) because FM bandwidth is much wider (200 kHz per channel)
- The IF Amplifier consists of multiple stages tuned exactly to 10.7 MHz
- It provides the majority of the receiver's gain and selectivity

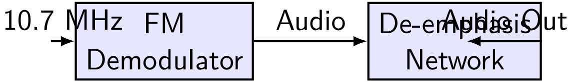
4. Amplitude Limiter

- By the time the signal leaves the IF amp, it has picked up amplitude noise (static, interference)
- The Amplitude Limiter 'clips' the tops and bottoms of the FM wave
- This removes all amplitude variations, leaving a pure, clean FM signal of constant amplitude
- Essential before a Foster-Seeley discriminator; optional if using a Ratio Detector or modern PLL



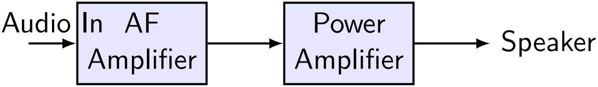
5. FM Demodulator & De-emphasis

- **Demodulator (Discriminator/PLL):**
  - Converts the 10.7 MHz varying-frequency signal back into the original audio voltage variations
- **De-emphasis Network:**
  - A low-pass RC filter ($75\mu s$ or $50\mu s$)
  - Attenuates the artificially boosted high frequencies and cuts high-frequency channel noise



6. Audio Amplifier and Speaker

- The output of the de-emphasis network is the original, clean audio signal
- However, it is a very weak voltage (millivolts)
- The Audio Frequency (AF) Amplifier boosts the voltage
- The Power Amplifier boosts the current to drive the speaker cone
- Result: High-fidelity sound!



Signal Waveforms Through the Receiver

- **Antenna:** Weak FM signal at RF (e.g., 98.3 MHz) with AM noise
- **Mixer Out:** Same FM variations, but centered at IF (10.7 MHz), still noisy
- **Limiter Out:** Clean FM signal at 10.7 MHz, constant amplitude
- **Demod Out:** Audio signal with emphasized high frequencies
- **De-emphasis Out:** Original studio audio signal

Key Takeaways

- The standard Intermediate Frequency (IF) for FM broadcasting is 10.7 MHz
- The amplitude limiter is the key stage that gives FM its noise-free characteristics
- The FM superheterodyne architecture is very similar to AM, but with the addition of limiting and de-emphasis

Next Lecture Preview

- Comparison of AM, FM, and PM
- Summary of all modulation techniques
- Which to choose for which application?

Feature	**AM**	**FM**	**PM**
Amplitude	Varies	Constant	Constant
Frequency	Constant	Varies (prop. to $m(t)$)	Varies (prop. to $m'(t)$)
Noise Immunity	Poor	Excellent	Excellent

Lecture 3.10: Compare AM, FM and PM

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Recap: What is Modulated?

- **AM (Amplitude Modulation):**
  - Carrier amplitude varies proportionally to modulating signal amplitude
  - Frequency and Phase remain constant
- **FM (Frequency Modulation):**
  - Carrier frequency varies proportionally to modulating signal amplitude
  - Amplitude remains constant
- **PM (Phase Modulation):**
  - Carrier phase varies proportionally to modulating signal amplitude
  - Amplitude remains constant

Comparison: The Waveforms

- **AM Waveform:** Shows an 'envelope' that matches the audio signal. Zero-crossings are constant.
- **FM Waveform:** Constant amplitude. Zero-crossings bunch up (high frequency) and spread out (low frequency) based on audio amplitude.
- **PM Waveform:** Visually identical to FM for a sine wave input, but zero-crossings shift based on the *derivative* of the audio signal.

Comparison: Bandwidth

- **AM:**
  - Requires very small bandwidth: $BW = 2f_m$
  - E.g., for 5 kHz audio, $BW = 10$ kHz
- **FM (Wideband):**
  - Requires very large bandwidth: $BW = 2(\Delta f + f_m)$ (Carson's Rule)
  - E.g., standard FM broadcast $BW = 200$ kHz
- **PM:**
  - Requires very large bandwidth, similar to FM, as it produces infinite sidebands

Comparison: Noise Immunity

- **AM:**
  - Poor noise immunity. Most atmospheric and electrical noise is amplitude-based (static).
  - Receiver cannot remove it without removing the signal.
- **FM:**
  - Excellent noise immunity. Amplitude noise can be clipped off by limiters.
  - Uses pre-emphasis/de-emphasis to further kill high-frequency noise.
- **PM:**
  - Excellent noise immunity, similar to FM.

Comparison: Transmitted Power

- **AM:**
  - Transmitted power varies with the modulation index ($P_t = P_c(1 + m^2/2)$).
  - Carrier contains most of the power but no information (inefficient).
- **FM & PM:**
  - Transmitted power is constant, regardless of modulation index.
  - All transmitter power is utilized all the time, making the power amplifiers more efficient (can operate in Class C).

Comparison: Receiver Complexity

- **AM:**
  - Receiver is very simple and cheap (Envelope detector is just a diode and capacitor).
- **FM:**
  - Receiver is complex and more expensive.
  - Requires limiters, discriminators (or PLLs), and de-emphasis networks.
- **PM:**
  - Receiver is highly complex.
  - Requires a coherent reference signal for demodulation (phase ambiguity problem).

Modulation Index Dependency

- **AM:** Modulation index m depends ONLY on the amplitude of the modulating signal.
- **FM:** Modulation index $m_f = \frac{k_f V_m}{f_m}$ depends on BOTH amplitude and frequency of the modulating signal.
- **PM:** Modulation index $m_p = k_p V_m$ depends ONLY on the amplitude of the modulating signal.

Summary Table (Exam Prep)

Parameter	**AM**	**FM**	**PM**
Constant	Freq & Phase	Amp & Phase	Amp & Freq
Bandwidth	$2f_m$ (Small)	$2(\Delta f + f_m)$ (Large)	Large
Noise Immunity	Poor	Excellent	Excellent
Tx Power	Varies	Constant	Constant
Complexity	Simple	Complex	Very Complex

Applications: Where do we use them?

- **AM:** AM Radio Broadcasting (MW/SW), Aviation communication (Air-traffic control), video signal in TV.
- **FM:** FM Radio Broadcasting (88-108 MHz), Audio signal in TV, Police/Emergency radios (NBFM), Satellite downlinks.
- **PM:** Rarely used for analog. It is the foundation for Digital Communications (PSK, QPSK, Wi-Fi, Cellular networks).

Why Aviation uses AM

- **Interesting Fact:** Why does air-traffic control still use AM?
- **Capture Effect in FM:** If two FM signals collide, the receiver only locks onto the stronger one; the weaker one is completely lost.
- In AM, if two signals collide, you hear a heterodyne 'whistle', but you can still hear both voices.
- For safety, it's better for a pilot to hear that two people are trying to talk at once, rather than one being silently overridden.

Unit 3 Summary

- We covered the mathematics of Angle Modulation (FM and PM)
- We learned about bandwidth (Carson's Rule) and Bessel functions
- We explored generation (PLL) and demodulation (Ratio Detector, PLL)
- We compared AM, FM, and PM

Key Takeaways

- AM is simple and bandwidth-efficient but suffers from poor noise immunity
- FM provides high-fidelity, noise-free communication at the cost of large bandwidth
- PM is similar to FM in analog form, but is primarily used for modern digital data transmission

Next Lecture Preview

- Unit 4: Sampling theory and waveform coding
- Moving from Analog to Digital Communication
- Statement and proof of the Sampling Theorem

Lecture 4.1: Statement and proof of sampling theorem

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Course Overview & Today's Agenda

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Learning Objectives

- Define what sampling means in communication.
- State the sampling theorem for band-limited signals.
- Understand the mathematical proof in time and frequency domains.
- Calculate sampling frequencies for given analog signals.

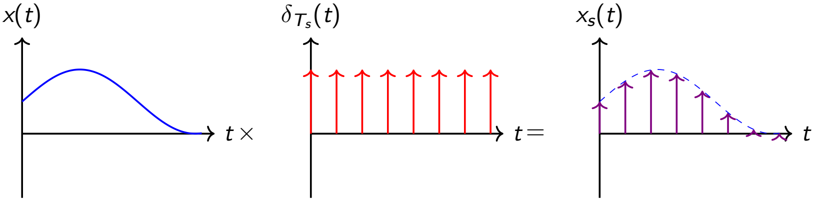
Introduction: Why Sample?

- Analog signals are continuous in time and amplitude.
- Digital systems require discrete values.
- Sampling is the first step in Analog-to-Digital Conversion (ADC).
- It converts a continuous-time signal into a discrete-time signal.

$$\text{Analog Signal } x(t) \longrightarrow \boxed{\text{Sampler}} \longrightarrow \text{Discrete Signal } x_s(t)$$

The Sampling Process

- A continuous signal $x(t)$ is multiplied by a periodic impulse train.
- The resulting signal is a sampled signal $x_s(t)$.
- The time between consecutive samples is the sampling interval, T_s.
- The sampling frequency is $f_s = 1/T_s$.



Statement of Sampling Theorem

- A continuous-time signal may be completely represented in its samples and recovered back if the sampling frequency is greater than or equal to twice the highest frequency component present in the signal.

Mathematical Condition

$$f_s \geq 2f_m$$

Where:
f_s = sampling frequency
f_m = maximum frequency of the baseband signal

Understanding the Theorem

- If $f_s \geq 2f_m$, no information is lost during sampling.
- The original analog signal can be exactly reconstructed from these samples.
- The condition $f_s = 2f_m$ is known as the **Nyquist rate**.
- If $f_s < 2f_m$, information is lost (a condition known as aliasing).

Proof of Sampling Theorem: Time Domain

- Let $x(t)$ be a band-limited signal with max frequency f_m.
- Let $\delta_{T_s}(t)$ be an ideal impulse train with period T_s.
- The sampled signal is the multiplication in time domain:

Sampled Signal Equation

$$x_s(t) = x(t) \cdot \delta_{T_s}(t)$$

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

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

Proof of Sampling Theorem: Frequency Domain

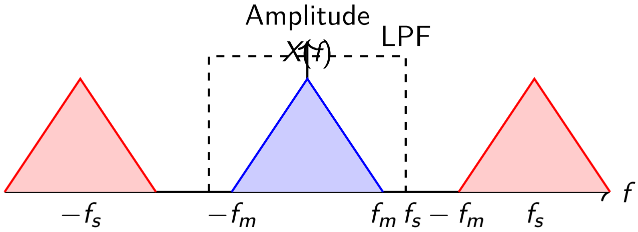
- Applying Fourier Transform to $x_s(t)$:

$$\boxed{X_s(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)}$$

- The spectrum of the sampled signal consists of infinite replicas of the original spectrum $X(f)$.
- These replicas are shifted by integer multiples of f_s.

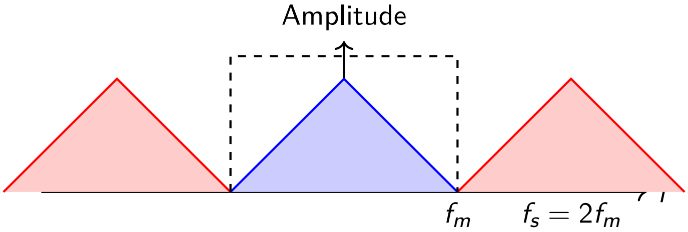
Spectrum Analysis for $f_s > 2f_m$

- When $f_s > 2f_m$, the replicas do not overlap.
- There is a gap between f_m and $(f_s - f_m)$.
- A low-pass filter with cutoff frequency f_m can easily extract the original spectrum $X(f)$.
- Hence, the signal is successfully reconstructed.



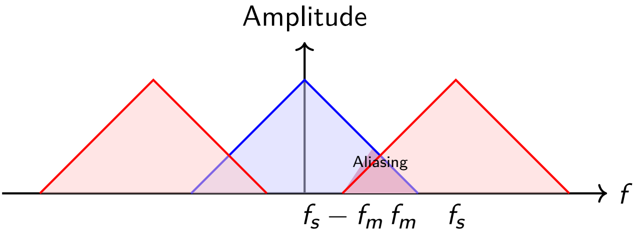
Spectrum Analysis for $f_s = 2f_m$

- When $f_s = 2f_m$, the replicas just touch each other.
- There is no gap (guard band) between them.
- An ideal low-pass filter is required to extract the original spectrum $X(f)$.
- This boundary condition represents the **Nyquist rate**.



Spectrum Analysis for $f_s < 2f_m$

- When $f_s < 2f_m$, the replicas overlap.
- High-frequency components take on the identity of low frequencies (**Aliasing**).
- A low-pass filter cannot separate the original spectrum.
- The original signal is distorted and cannot be recovered.



Numerical Example: Sampling Frequency

Problem: An analog signal is given by
$x(t) = 5\cos(2\pi \cdot 2000t) + 3\sin(2\pi \cdot 4000t)$. Find the Nyquist rate.

- **Step 1:** Identify frequency components from the signal equation.
    - The signal is a sum of two sinusoids.
    - General form is $\cos(2\pi ft)$ or $\sin(2\pi ft)$.
    - Therefore, $f_1 = 2000$ Hz and $f_2 = 4000$ Hz.
- **Step 2:** Find the maximum frequency f_m.
    - $f_m = \max(f_1, f_2)$
    - $f_m = \max(2000, 4000) = 4000$ Hz.

Numerical Example: Solution

Solution Continued:

Calculation of Nyquist Rate

- We found the maximum frequency $f_m = 4000$ Hz.
- Nyquist rate $= 2 \times f_m$
- Nyquist rate $= 2 \times 4000$ Hz $= 8000$ Hz.

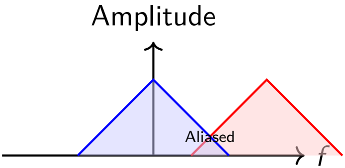
Conclusion:
To avoid aliasing, the sampling frequency f_s must be ≥ 8000 Hz (or 8 kHz).

Key Takeaways

- Sampling converts continuous-time signals to discrete-time signals.
- Sampling theorem states $f_s \geq 2f_m$ for lossless recovery.
- The spectrum of a sampled signal consists of infinite replicas.
- Reconstruction requires a low-pass filter.

Next Lecture Preview

- **Lecture 29: Nyquist rate and interval, Aliasing error**
- We will formally define Nyquist parameters.
- Deep dive into Aliasing error and anti-aliasing filters.
- Explore under, over, and critical sampling in detail.



Lecture 4.2: Nyquist rate and interval, Aliasing error, under/over/critical sampling

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Course Overview & Today's Agenda

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
3. Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)
4. Lecture 1.4: Block diagram of Analog and Digital communication system
5. Lecture 1.5: Modulation: Definition & its classification
6. Lecture 1.6: Classification based on analog & pulse signal as carrier
7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
8. Lecture 2.1: Need of modulation & Types of modulation techniques
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

Nyquist Rate

- The minimum sampling rate required to avoid aliasing is called the Nyquist rate.
- Formula: Nyquist Rate (f_N) $= 2 \times f_m$
- Where f_m is the highest frequency component of the message signal.
- It acts as a theoretical lower bound for the sampling frequency f_s.

Nyquist Rate Formula

$$f_N = 2 \times f_m$$

Nyquist Interval

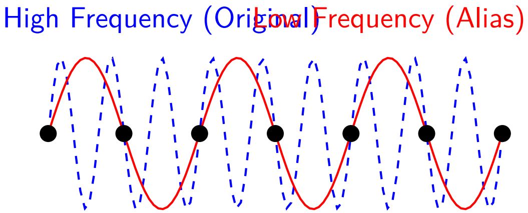
- The Nyquist interval is the maximum time interval allowed between consecutive samples to avoid aliasing.
- It is the reciprocal of the Nyquist rate.
- Formula: Nyquist Interval (T_N) $= \frac{1}{f_N} = \frac{1}{2 \times f_m}$
- Sampling interval T_s must be $\leq T_N$.

Nyquist Interval Formula

$$T_N = \frac{1}{2f_m}$$

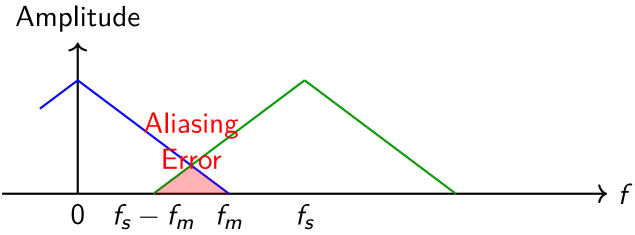
What is Aliasing?

- Aliasing occurs when a signal is sampled at a rate less than the Nyquist rate ($f_s < 2f_m$).
- High-frequency components take on the 'alias' or identity of low-frequency components.
- The original signal cannot be uniquely recovered.
- It introduces permanent distortion in the reconstructed signal.



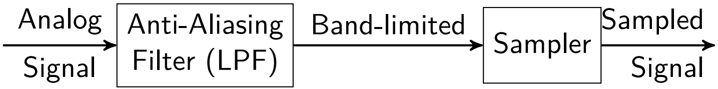
Aliasing Error in Frequency Domain

- In the frequency domain, aliasing is seen as the overlapping of adjacent spectrum replicas.
- The tail of the higher frequency replica folds into the lower frequency replica.
- This folded portion is called aliasing error or fold-over distortion.
- A low-pass filter cannot separate the overlapped regions.



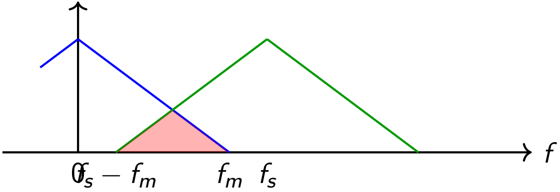
How to Avoid Aliasing?

- To prevent aliasing, we must ensure $f_s \geq 2f_m$.
- Use an **Anti-Aliasing Filter** before the sampler.
- An anti-aliasing filter is a low-pass filter that strictly limits the maximum frequency f_m of the input signal.
- It removes any stray high-frequency noise that could cause aliasing.



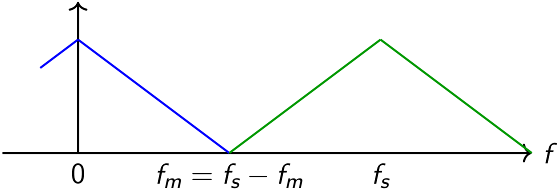
Under Sampling ($f_s < 2f_m$)

- **Condition:** Sampling frequency is less than the Nyquist rate.
- **Spectrum:** Replicas overlap.
- **Result:** Aliasing occurs, signal is distorted.
- **Reconstruction:** Impossible to recover original signal perfectly.



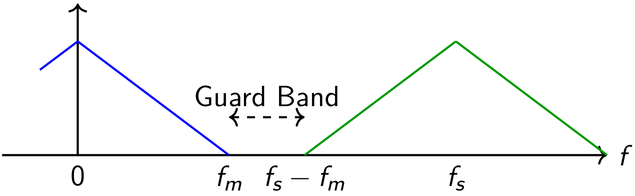
Critical Sampling ($f_s = 2f_m$)

- **Condition:** Sampling frequency equals the Nyquist rate.
- **Spectrum:** Replicas touch but do not overlap.
- **Result:** No aliasing, but reconstruction is very difficult.
- **Reconstruction:** Requires an ideal 'brick-wall' low-pass filter, which is impossible to build.



Over Sampling ($f_s > 2f_m$)

- **Condition:** Sampling frequency is greater than the Nyquist rate.
- **Spectrum:** Replicas are separated by a guard band.
- **Result:** No aliasing, signal is cleanly preserved.
- **Reconstruction:** Easy, can be done using a practical, non-ideal low-pass filter.



Numerical Example: Sinc Function

Problem: Find Nyquist rate and interval for $x(t) = 10\ \text{sinc}(1000t)$.

- Recall $\text{sinc}(at)$ has a bandwidth of $B = \frac{a}{2}$ Hz.
- Here, $a = 1000$, so Bandwidth $f_m = \frac{1000}{2} = 500$ Hz.
- Highest frequency component $f_m = 500$ Hz.

$$\text{Bandwidth } f_m = \frac{a}{2} \implies f_m = \frac{1000}{2} = 500 \text{ Hz}$$

Numerical Example: Solution

Given $f_m = 500$ Hz, let's calculate the Nyquist rate and interval.

- **Nyquist Rate (f_N):**
  $f_N = 2 \times f_m = 2 \times 500 = 1000$ Hz.
- **Nyquist Interval (T_N):**
  $T_N = \frac{1}{f_N} = \frac{1}{1000} = 1$ millisecond (1 ms).
- **Conclusion:** The signal must be sampled at least every 1 ms.

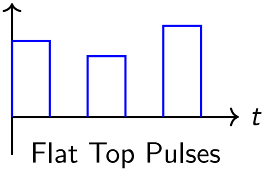
$$\mathbf{f_N = 1000\ Hz, \quad T_N = 1\ ms}$$

Key Takeaways

- Nyquist rate ($2f_m$) is the absolute minimum sampling frequency.
- Aliasing is the overlapping of frequency spectra due to under sampling.
- An anti-aliasing filter limits the signal bandwidth before sampling.
- Over sampling provides a guard band, making practical filtering possible.

Next Lecture Preview

- **Lecture 30: Ideal, Natural and flat top sampling**
- Moving from theoretical sampling to practical circuits.
- Exploring different pulse shapes used in sampling.
- Understanding aperture effect in flat top sampling.



Lecture 4.3: Ideal, Natural and flat top sampling

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

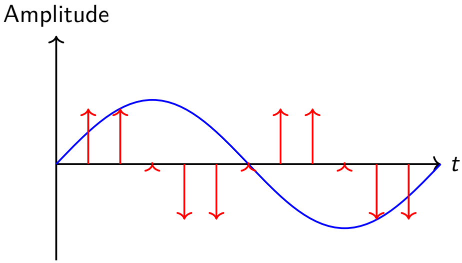
Lecture Overview

Classification of Sampling

- Sampling techniques are classified based on the type of sampling pulse used.
- 1. Ideal Sampling (Instantaneous sampling using impulses)
- 2. Natural Sampling (Using practical rectangular pulses)
- 3. Flat Top Sampling (Using Sample and Hold circuits)
- Theoretical models use Ideal sampling, while real systems use Flat Top.

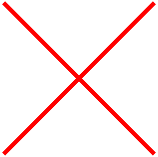
1. Ideal Sampling

- Also known as impulse sampling or instantaneous sampling.
- The sampling signal is a train of ideal Dirac delta impulses.
- Width of each pulse approaches zero, amplitude approaches infinity.
- Mathematical model used to prove the sampling theorem.



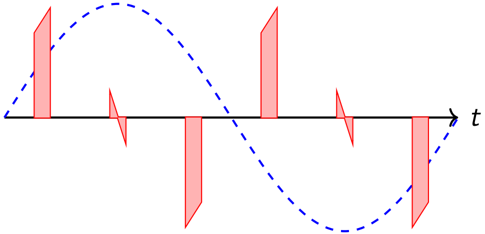
Limitations of Ideal Sampling

- Practical impossibility: We cannot generate zero-width pulses.
- Infinite bandwidth required to transmit ideal impulses.
- Infinite power is theoretically required to generate Dirac impulses.
- Conclusion: Ideal sampling is purely theoretical.



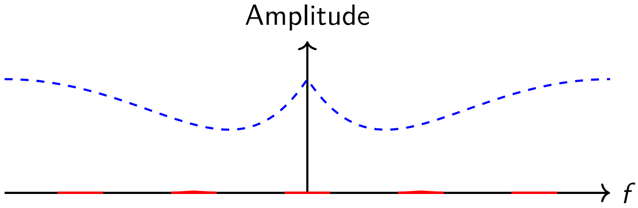
2. Natural Sampling

- Uses a train of practical rectangular pulses of finite width τ.
- The analog signal x(t) is multiplied by the pulse train.
- The top of the sampled pulse follows the contour (shape) of the analog signal.
- Since the top is not flat, it's called 'Natural' sampling.



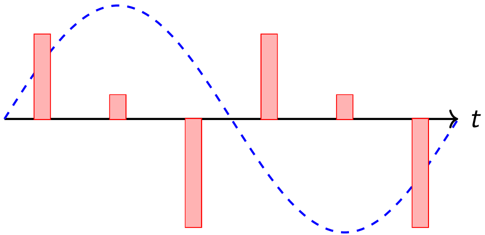
Spectrum of Natural Sampling

- The frequency spectrum consists of the baseband spectrum and replicas.
- However, the replicas are weighted by a sinc function envelope.
- The shape of the original spectrum is preserved without distortion.
- Reconstruction is easily achieved using a standard low-pass filter.



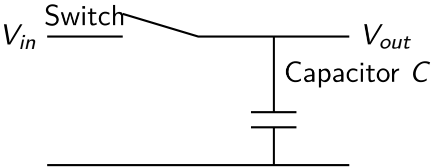
3. Flat Top Sampling

- The most widely used sampling technique in practical ADCs.
- The top of the sampled pulse remains constant (flat) for the entire pulse duration τ.
- The amplitude is determined by the analog signal value at the instant sampling begins.
- Produces rectangular pulses of varying heights but flat tops.



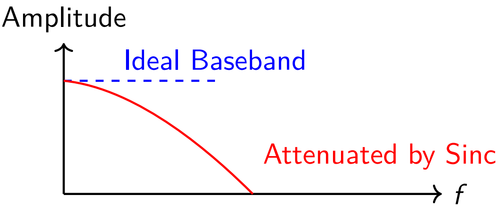
Sample and Hold Circuit

- Flat top sampling is implemented using a Sample-and-Hold (S/H) circuit.
- Consists of an FET switch and a holding capacitor (C).
- During 'Sample', switch closes, capacitor charges to input voltage quickly.
- During 'Hold', switch opens, capacitor holds the voltage steady for conversion.



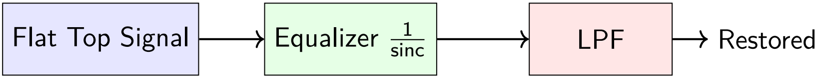
The Aperture Effect

- In flat top sampling, the flat pulse acts as a filter that distorts the spectrum.
- The spectrum is multiplied by a sinc function $H(f)$.
- This causes attenuation of higher frequencies within the baseband.
- This distortion is known as the Aperture Effect.



Aperture Equalization

- To correct the aperture effect, we use an Equalizer at the receiver.
- The equalizer has a transfer function $E(f) = 1/H(f)$.
- It amplifies the higher frequencies that were attenuated.
- Result: The original flat spectrum is restored before final low-pass filtering.



Comparison of Sampling Techniques

Technique	**Pulse Characteristics**	**Practicality**	**Aperture Effect**
Ideal	Impulses, top is undefined	Not practical	None
Natural	Finite pulses, top follows signal	Difficult for ADCs	None
Flat Top	Finite pulses, top is flat/constant	Highly practical (S/H circuit)	Yes (needs equalizer)

Problem: A flat top sampling system operates at $f_s = 8$ kHz with a pulse width $\tau = 25\ \mu$s. Calculate the duty cycle.

Solution

Step 1: Find sampling interval T_s

$$T_s = \frac{1}{f_s} = \frac{1}{8000} = 125\,\mu\text{s}$$

Step 2: Calculate Duty Cycle

$$\text{Duty Cycle} = \left(\frac{\tau}{T_s}\right) \times 100\%$$

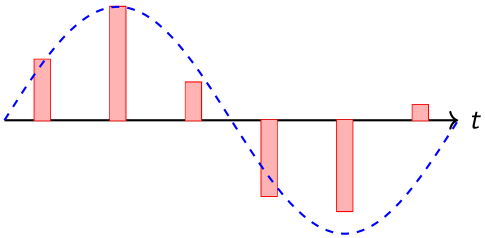
$$\text{Duty Cycle} = \left(\frac{25\,\mu\text{s}}{125\,\mu\text{s}}\right) \times 100\% = 20\%$$

Key Takeaways

- Ideal sampling uses impulses and is purely theoretical.
- Natural sampling preserves spectrum shape but has varying pulse heights.
- Flat top sampling is practical, uses a sample-and-hold circuit, but causes aperture effect.
- Aperture equalization corrects high-frequency roll-off at the receiver.

Next Lecture Preview

- Lecture 31: Pulse Modulation techniques: PAM
- How sampled signals are transmitted as Pulse Amplitude Modulation.
- Generation and demodulation of PAM.
- Bandwidth requirements for PAM signals.



Lecture 4.4: Pulse Modulation techniques: PAM

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

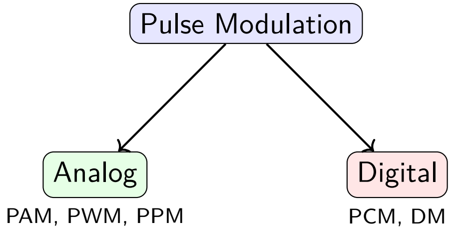
Lecturer, GP Palanpur

July 22, 2026

Lecture Overview

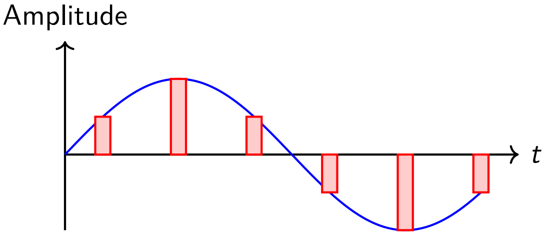
Introduction to Pulse Modulation

- In Continuous Wave (CW) modulation (AM, FM), the carrier is a continuous sine wave.
- In Pulse Modulation, the carrier is a periodic train of pulses.
- A parameter of the pulse train is varied in accordance with the message signal.
- Analog Pulse Modulation types: PAM, PWM, PPM.



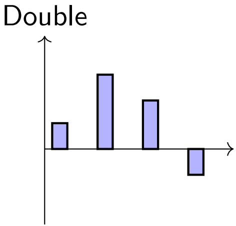
What is PAM?

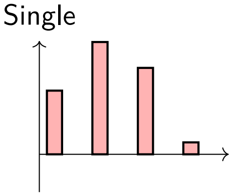
- PAM = Pulse Amplitude Modulation.
- The amplitude of the carrier pulses is varied in proportion to the instantaneous amplitude of the modulating signal.
- The width and position of the pulses remain constant.
- PAM is essentially the direct result of flat top sampling.



Types of PAM

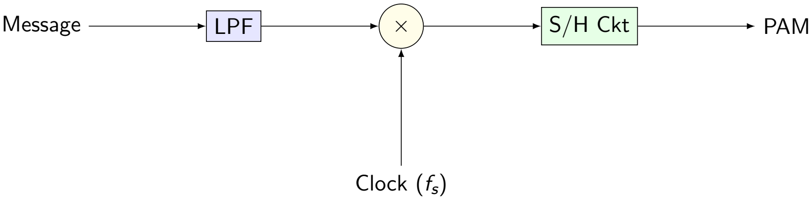
- **1. Single Polarity PAM:**
  - A fixed DC bias is added to the signal so all pulses are positive.
  - Simplifies circuitry but wastes power.
- **2. Double Polarity PAM:**
  - Pulses can be both positive and negative depending on the signal.
  - More power-efficient but slightly more complex.





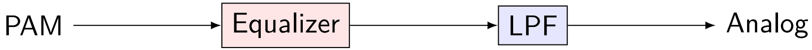
Generation of PAM Signal

- PAM is generated using a Sample-and-Hold circuit (flat top sampling).
- Components: LPF (Anti-aliasing) $\rightarrow$ Multiplier (Sampler) $\rightarrow$ Pulse Shaping.
- Clock provides the sampling frequency f_s.
- The output is a series of flat-topped pulses with heights proportional to the input.



Demodulation of PAM Signal

- Demodulation is the process of recovering the original analog signal.
- PAM demodulation is simple: Pass the PAM signal through a Low-Pass Filter (LPF).
- The LPF cutoff frequency is set to the maximum signal frequency f_m.
- An equalizer is often added to correct the aperture effect.



Bandwidth of PAM Signal

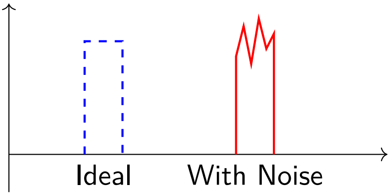
- Unlike continuous signals, pulses have very wide frequency spectra.
- Let τ be the duration of the PAM pulse.
- The transmission bandwidth BW required is $\geq \frac{1}{2\tau}$.
- Since τ is very small, the required bandwidth for PAM is very large compared to AM or FM.

Bandwidth Requirement

$$BW \geq \frac{1}{2\tau}$$

Disadvantages of PAM

- **1. High bandwidth requirement** due to narrow pulses.
- **2. Highly susceptible to noise.**
  - Noise adds directly to the amplitude of the pulses, corrupting the information.
- **3. Transmission power varies** because pulse amplitude varies.



Applications of PAM

- Rarely used for direct transmission over long distances due to noise.
- Primarily used as an intermediate step in generating other signals (like PCM).
- Used in instrumentation and measurement systems.
- Basis for Time Division Multiplexing (TDM).

Numerical Example: Bandwidth

Problem: An audio signal with $f_m = 3$ kHz is sampled at 8 kHz to generate a PAM signal. The pulse width is 10 μs. Find minimum transmission bandwidth.

- **Step 1:** Identify given data.
  - $\tau = 10\ \mu\text{s} = 10 \times 10^{-6}$ s
- **Step 2:** Note that f_s and f_m are distractors for transmission bandwidth of the pulse.

Numerical Example: Solution

Solution

Formula: $BW \geq \frac{1}{2\tau}$

$BW = \frac{1}{2 \times (10 \times 10^{-6})}$

$BW = \frac{1}{20 \times 10^{-6}} = \frac{10^6}{20}$

$BW = 50,000$ Hz or 50 kHz

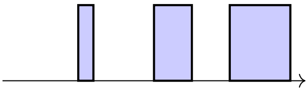
Compare this to AM bandwidth (2×3 kHz $= 6$ kHz). PAM requires much more!

Key Takeaways

- PAM varies pulse amplitude according to message signal.
- It is generated using Sample-and-Hold and demodulated using a LPF.
- PAM suffers from high noise susceptibility and large bandwidth requirements.
- It is mainly an intermediate step for digital systems like PCM.

Next Lecture Preview

- Lecture 32: Pulse Modulation techniques: PWM, PPM
- If PAM is noisy, can we vary other pulse parameters?
- Exploring Pulse Width Modulation (PWM) and Pulse Position Modulation (PPM).
- Comparing all three analog pulse modulation techniques.



PWM (Width varies)

Lecture 4.5: Pulse Modulation techniques: PWM, PPM

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

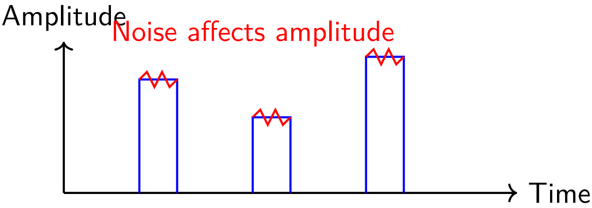
Lecturer, GP Palanpur

July 22, 2026

Lecture Overview

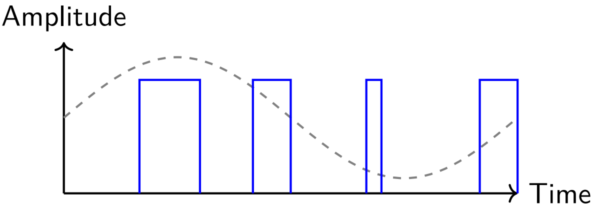
Why look beyond PAM?

- PAM stores information in the amplitude of the pulse.
- Noise in transmission channels primarily affects amplitude.
- To improve noise immunity, we need a method where amplitude remains constant.
- Solution: Vary the timing characteristics of the pulse (Width or Position).



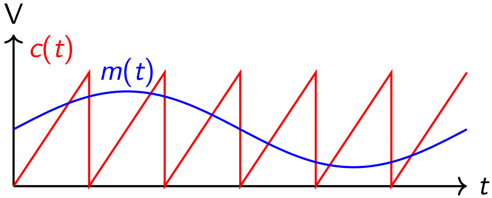
What is PWM?

- PWM = Pulse Width Modulation (also known as PDM or PLM).
- The width (duration) of the carrier pulses is varied proportionally to the message signal.
- Amplitude and position of the leading edge (usually) remain constant.
- Higher signal amplitude = Wider pulse.



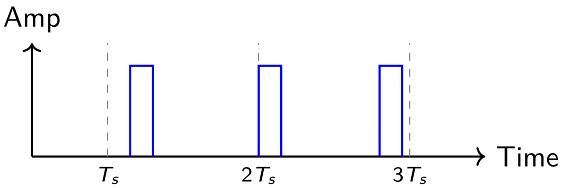
Generation of PWM

- PWM is often generated using a comparator circuit.
- Inputs to comparator:
  1. Message signal ($m(t)$)
  2. Sawtooth or triangular carrier wave ($c(t)$)
- Output is HIGH when $m(t) > c(t)$, and LOW otherwise.
- This naturally creates pulses whose widths vary with $m(t)$.



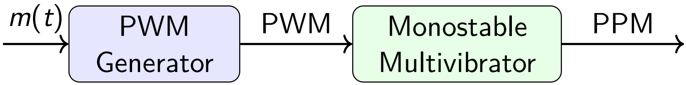
What is PPM?

- PPM = Pulse Position Modulation.
- The position of the pulse relative to its unmodulated time slot is varied proportionally to the message signal.
- Amplitude and width of the pulses remain absolutely constant.
- Provides the best noise immunity among analog pulse modulations.



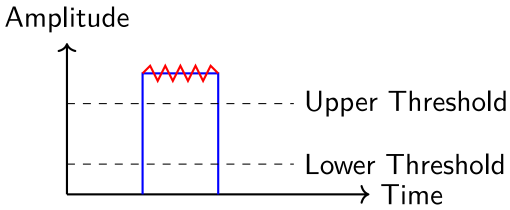
Generation of PPM

- PPM is typically generated directly from a PWM signal.
- Step 1: Generate PWM signal.
- Step 2: Pass PWM through a Monostable Multivibrator (One-shot).
- The multivibrator triggers on the trailing edge of the PWM pulse.
- Result: A constant-width pulse whose position shifts as the PWM trailing edge shifts.



Demodulation (Receiver Concepts)

- Since amplitude is constant, receivers use a 'slicer' or limiter to remove noise spikes from the top and bottom.
- **PWM Demodulation:** Pass through an LPF (since average value of PWM is proportional to message).
- **PPM Demodulation:** Convert PPM back to PWM using a flip-flop and a reference clock, then pass through LPF.



Comparison: Amplitude & Noise

- **PAM:** Amplitude varies. Poor noise immunity.
- **PWM:** Amplitude constant. Good noise immunity.
- **PPM:** Amplitude constant. Excellent noise immunity.
- In PWM and PPM, limiters can remove amplitude noise completely.

Feature	**PAM**	**PWM**	**PPM**
Amplitude	Varies	Constant	Constant
Noise Immunity	Poor	Good	Best

Comparison: Power & Bandwidth

- **PAM:** Transmitted power varies. Bandwidth is high.
- **PWM:** Transmitted power varies (wider pulses carry more energy). Bandwidth is high.
- **PPM:** Transmitted power is CONSTANT (pulses are identical). Bandwidth is very high (needs sharp edges to define position).

Feature	**PAM**	**PWM**	**PPM**
Transmitted Power	Varies	Varies	Constant
Bandwidth	High	High	Very High

Summary Comparison Table

Parameter	**PAM**	**PWM**	**PPM**
Variable	Amplitude	Width	Position
Bandwidth	High	High	Very High
Tx Power	Varies	Varies	Constant
Noise Immunity	Poor	Good	Best
Complexity	Simple	Moderate	Complex

Numerical Concept: Duty Cycle in PWM

Problem: A PWM system operates at $f_s = 10$ kHz. The pulse width varies from $20\,\mu$s to $80\,\mu$s based on signal amplitude. Find max and min duty cycle.

Solution

Step 1: Calculate sampling time period (T_s)
$T_s = \frac{1}{f_s} = \frac{1}{10000} = 100\,\mu\text{s}$
Step 2: Calculate Minimum Duty Cycle

$$\text{Min Duty Cycle} = \left(\frac{20}{100}\right) \times 100 = 20\%$$

Step 3: Calculate Maximum Duty Cycle

$$\text{Max Duty Cycle} = \left(\frac{80}{100}\right) \times 100 = 80\%$$

Applications

- **PWM Applications:** Motor speed control, LED dimming, Class-D audio amplifiers, power supplies.
- **PPM Applications:** Optical communication, Radio control (RC) planes/drones, deep space telemetry.
- **Why?** PPM is chosen when power efficiency is critical and bandwidth is plentiful.

Key Takeaways

- PWM varies pulse width; PPM varies pulse position.
- Both provide superior noise immunity compared to PAM.
- PPM offers constant transmission power but requires higher bandwidth.
- PWM is often used as an intermediate step to generate PPM.

Next Lecture Preview

- **Lecture 33: Concept of Quantization**
- So far, time is discrete, but amplitude is continuous.
- How do we make amplitude discrete to get true digital data?
- Introduction to quantization and quantization error.

Lecture 4.6: Concept of Quantization

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

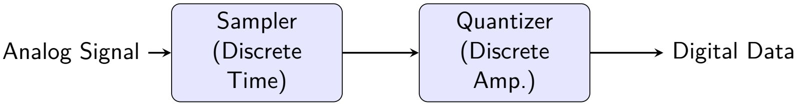
Lecturer, GP Palanpur

July 22, 2026

Lecture Overview

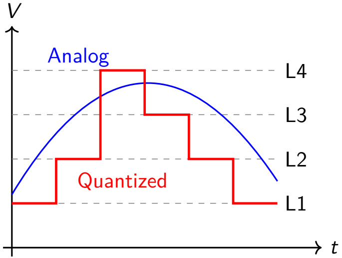
The Missing Step

- Sampling makes the signal discrete in **TIME**.
- However, the sampled amplitude can still be any infinite number of values (e.g., 2.13456... V).
- Digital systems (computers) can only store a finite number of binary values.
- Therefore, we must make the amplitude discrete as well. This is **Quantization**.



What is Quantization?

- Quantization is the process of rounding off sampled continuous values to the nearest predetermined discrete level.
- The total voltage range is divided into a finite number of 'Quantization Levels' (L).
- Any sample falling within a certain zone is assigned the specific value of that level.
- It acts like a mathematical 'staircase'.



Quantization Levels (L)

- The number of levels (L) depends on the number of bits (n) used to encode each sample.

Formula

$$L = 2^n$$

- **Example:** If we use 3 bits per sample, $L = 2^3 = 8$ levels.
- **Example:** CD audio uses 16 bits, so $L = 2^{16} = 65,536$ levels.

Step Size (Δ)

- Step size (Δ) is the voltage difference between two adjacent quantization levels.

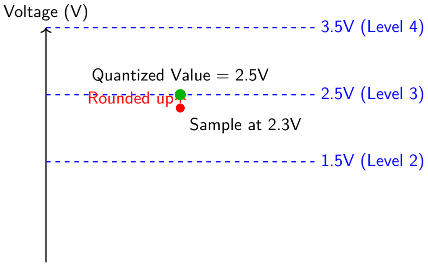
Formula

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

- Where $V_{max} - V_{min}$ is the total dynamic range of the signal.
- A smaller step size means higher resolution and better accuracy.

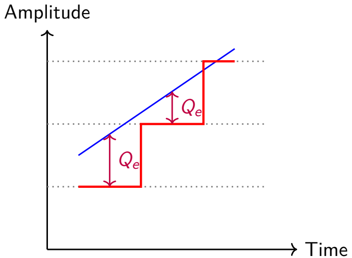
The Quantization Process

- Assume a signal ranges from 0 to 8V, and we have 8 levels. Step size $\Delta = 1V$.
- Levels are at 0.5V, 1.5V, 2.5V, etc.
- If a sample is 2.3V, it is rounded to the nearest level: 2.5V.
- If a sample is 2.8V, it is also rounded to 2.5V.



Quantization Error (Noise)

- Because we round off values, the quantized signal is slightly different from the original signal.
- **Quantization Error (Q_e)** = Quantized Value - Actual Sampled Value.
- This error sounds like a low-level hiss in audio, hence called 'Quantization Noise'.
- Unlike channel noise, this noise is generated internally by the system.



Limits of Quantization Error

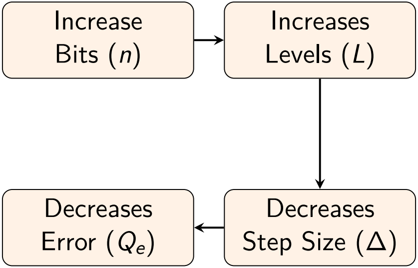
- What is the maximum possible error?
- The worst-case scenario is when a sample falls exactly halfway between two levels.
- Therefore, Maximum Quantization Error $= \Delta/2$.

$$-\frac{\Delta}{2} \leq Q_e \leq +\frac{\Delta}{2}$$

- The error Q_e always lies in this range.

How to Reduce Quantization Noise?

- To reduce error, we must reduce the step size Δ.
- To reduce Δ, we must increase the number of levels L.
- To increase L, we must use more bits (n).
- **Trade-off:** More bits = better quality, but requires higher bandwidth and more storage.



Numerical Example: Step Size

Problem: An analog signal varies from -5V to +5V. It is digitized using a 4-bit quantizer. Calculate the number of levels, step size, and max quantization error.

- **Step 1:** Bits $n = 4$.
- **Step 2:** Number of levels $L = 2^n = 2^4 = 16$ levels.
- **Step 3:** Signal range $= V_{max} - V_{min} = 5 - (-5) = 10$V.

Numerical Example: Solution

- **Step 4:** Step Size $\Delta = \frac{\text{Range}}{L} = \frac{10\text{V}}{16} = 0.625\text{V}$.
- **Step 5:** Max Quantization Error $= \frac{\Delta}{2} = \frac{0.625}{2} = 0.3125\text{V}$.

Final Answer

- $\Delta = 0.625$**V**
- **Max Error = 0.3125V**

Conclusion: Any sample will be inaccurate by at most 0.3125 volts.

Key Takeaways

- Quantization makes signal amplitude discrete by rounding to fixed levels.
- Levels $(L) = 2^n$, where n is bits per sample.
- Step Size (Δ) is the voltage gap between levels.
- Rounding creates unavoidable Quantization Error, limited to $\pm\Delta/2$.
- Increasing bits reduces error but increases bandwidth.

Next Lecture Preview

- **Lecture 34: Classification of Quantization**
- Are all steps the same size?
- Uniform vs Non-uniform quantization.
- How to handle very quiet sounds efficiently using Companding.

Lecture 4.7: Classification of quantization

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

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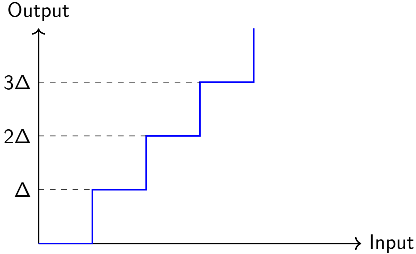
July 22, 2026

Lecture Overview

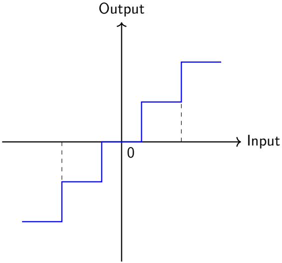
Uniform Quantization

- In uniform quantization, the step size (Δ) remains constant across the entire dynamic range.
- The distance between any two adjacent levels is identical.
- This is the simplest form of quantization to design and build.
- It is classified into two types based on how it handles the zero-voltage crossing.



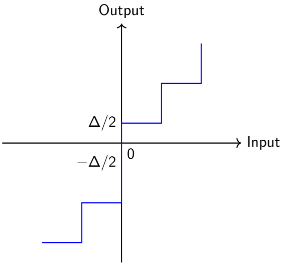
1. Mid-Tread Quantizer

- A type of uniform quantizer where the origin (0 Volts) lies in the middle of a step (a 'tread').
- **Result:** There is an explicit quantization level exactly at zero volts.
- **Advantage:** Perfectly silent (0V) analog signals are digitized as exactly zero, causing no noise during silence.
- Most common type used for audio.



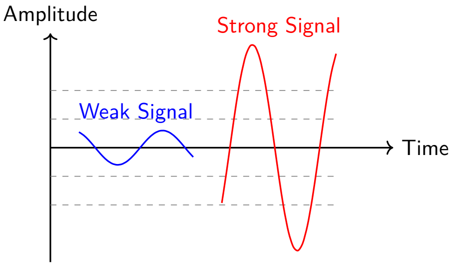
2. Mid-Riser Quantizer

- A type of uniform quantizer where the origin (0 Volts) lies exactly on the rising edge of a step (a 'riser').
- **Result:** There is NO quantization level at zero volts. The closest levels are $+\Delta/2$ and $-\Delta/2$.
- **Disadvantage:** A silent 0V input will toggle between positive and negative levels, creating a constant hiss (idle channel noise).



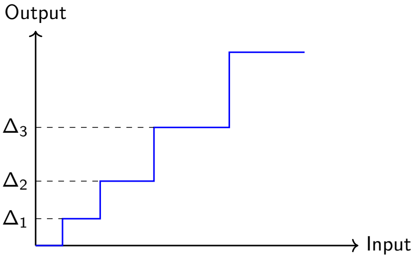
The Problem with Uniform Quantization

- Many natural signals, especially human voice, have mostly low amplitudes with occasional high peaks.
- In a uniform quantizer, a small signal $\Delta/2$ and a large signal $\Delta/2$ have the SAME maximum absolute error.
- However, the Signal-to-Noise Ratio (SNR) is much worse for weak signals.
- **Example:** A 0.5V error on a 1V signal is huge (50%). A 0.5V error on a 10V signal is small (5%).



Non-uniform Quantization

- **Solution:** Make the step sizes variable!
- Small step sizes (Δ) for small signal amplitudes (high resolution for weak signals).
- Large step sizes (Δ) for large signal amplitudes (low resolution for strong signals).
- This maintains a constant Signal-to-Noise Ratio across the entire dynamic range.



How to Implement Non-Uniform Quantization?

- Building an ADC with non-uniform physical steps is electronically very difficult and expensive.
- Instead, we use a clever mathematical trick called **COMPANDING**.

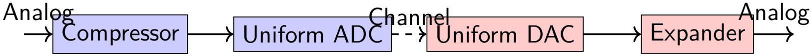
Companding Principle

Companding = COMpressing (at transmitter) + **exPANDING** (at receiver).

- We use a standard Uniform ADC, but we distort the analog signal before it goes in.

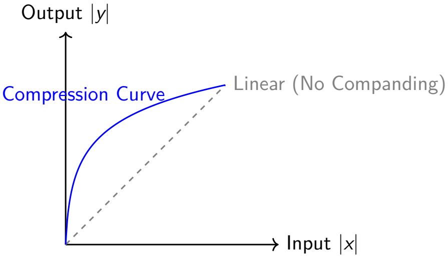
The Companding Process

- **Transmitter (Compressor):** Amplifies weak signals greatly, and amplifies strong signals very little (logarithmic curve).
- The compressed signal is then fed into a standard Uniform Quantizer.
- **Receiver (Expander):** Performs the exact inverse mathematical operation.
- **Result:** Effectively achieves Non-uniform quantization easily.



Companding Curves

- The compression characteristic is roughly logarithmic.
- There are two standard curves used globally in digital telephony:
  1. **μ-law (Mu-law):** Used in North America and Japan.
  2. **A-law:** Used in Europe and the rest of the world (including India).



μ-law and A-law Equations

- μ-**law** uses the parameter μ to define the amount of compression (standard $\mu = 255$).
- **A-law** uses the parameter A (standard $A = 87.6$).
- Both provide an SNR of about 38 dB using only 8 bits per sample.
- Without companding, achieving 38 dB SNR for weak signals would require 12 or 13 bits!

Bit Savings

Companding saves 4-5 bits per sample while maintaining equivalent voice quality!

Numerical Insight: Bit Rate Savings

Consider a voice channel sampled at $f_s = 8$ kHz.

Bandwidth Comparison

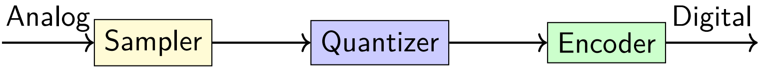
- **With Uniform Quantization (13 bits):**
  Bit rate $= 8000 \times 13 = 104$ kbps.
- **With Companded Non-uniform Quantization (8 bits):**
  Bit rate $= 8000 \times 8 = 64$ kbps.
- **Savings** $= 40$ kbps per channel.
  In a massive network, this saves huge amounts of bandwidth.

Key Takeaways

- Uniform quantizers have constant step sizes; non-uniform have variable step sizes.
- Mid-tread quantizers have a level at zero volts, handling silence (0V) better than mid-riser.
- Non-uniform quantization provides better SNR for weak signals like human voice.
- Companding (A-law or μ-law) is the practical way to implement non-uniform quantization.

Next Lecture Preview

- **Lecture 35: PCM Transmitter**
- We have sampled, we have quantized.
- Now we encode it into binary.
- Putting it all together to form a Pulse Code Modulation (PCM) transmitter.



Lecture 4.8: PCM Transmitter

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

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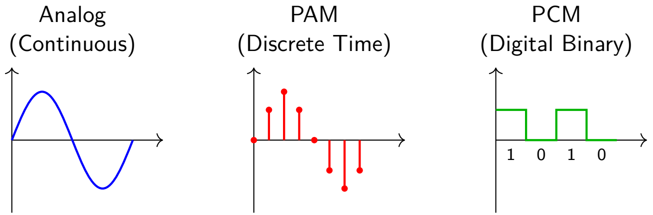
July 22, 2026

Lecture Overview

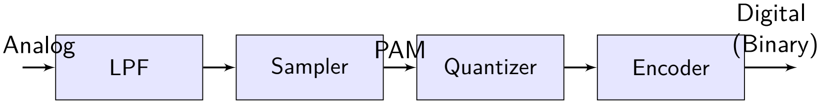
What is PCM?

- PCM = Pulse Code Modulation.
- It is not a modulation in the traditional analog sense.
- It is a waveform coding technique to convert an analog signal into a standard digital binary sequence.
- Unlike PAM/PWM/PPM which are analog pulse techniques, PCM is purely digital.



PCM Transmitter Block Diagram

- The transmitter consists of four main blocks:
  1. Low Pass Filter (Anti-Aliasing)
  2. Sampler
  3. Quantizer
  4. Encoder
- Output is a stream of binary bits.



Block 1: Low Pass Filter

- Also known as the Anti-Aliasing Filter.
- Function: Band-limits the incoming analog signal to a maximum frequency (f_m).
- Removes high-frequency noise that could cause aliasing during sampling.
- Ensures the Nyquist criterion can be strictly met.

Block 2: Sampler

- Function: Converts continuous-time signal to discrete-time signal.
- Operates at a sampling frequency $f_s \geq 2f_m$.
- Usually implemented as a Flat Top Sampler (Sample-and-Hold circuit).
- Output is a PAM signal.

Block 3: Quantizer

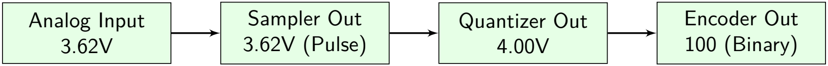
- Function: Converts continuous-amplitude samples to discrete-amplitude levels.
- Rounds the flat-topped PAM pulses to the nearest standard voltage level.
- May include a compressor circuit (companding) for non-uniform quantization.
- Output is a Quantized PAM signal.

Block 4: Encoder

- Function: Converts the discrete voltage levels into a binary code.
- Each level is assigned a unique *n*-bit binary sequence.
- Example: If level is 3, and we use 3 bits, encoder outputs '011'.
- The output is a continuous serial stream of 1s and 0s (PCM data).

PCM Waveform Evolution

- Let's trace a single sample:
  1. Analog input: 3.62 Volts
  2. Sampler output: 3.62 Volts (Pulse)
  3. Quantizer output: 4.00 Volts (Rounded to nearest level)
  4. Encoder output: 100 (Binary representation of level 4)



Bit Rate in PCM

- Bit Rate (R_b) is the speed at which bits are transmitted, measured in bits per second (bps).
- Formula: $R_b = n \times f_s$
- Where 'n' is the number of bits per sample.
- Where 'f_s' is the sampling frequency (samples per second).
- Bits/sample $\times$ samples/second $=$ bits/second.

Bit Rate Formula

$$R_b = n \times f_s$$

Numerical Example: PCM Bit Rate

Problem: A voice signal with max frequency $f_m = 3.4$ kHz is sampled at 8 kHz. The quantizer has 256 levels. Calculate the bit rate.

Step-by-Step Solution

Step 1: Identify Given Data
Sampling Frequency $f_s = 8000$ Hz
Levels $L = 256$

Step 2: Find Number of Bits (n)

We know Levels $L = 2^n$
$256 = 2^n$
Therefore, $n = 8$ bits per sample.

Numerical Example: Solution

Step-by-Step Solution (Continued)

Step 3: Calculate Bit Rate R_b

$R_b = n \times f_s$

$R_b = 8 \text{ bits/sample} \times 8000 \text{ samples/sec}$

$R_b = 64,000 \text{ bits/sec}$

Final Answer: $R_b = 64$ kbps

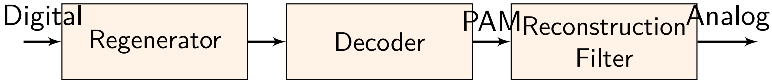
- This is the worldwide standard for a digital voice channel (DS0).

Key Takeaways

- PCM converts analog waveforms to digital binary data.
- Transmitter sequence: Anti-Aliasing LPF $\rightarrow$ Sampler $\rightarrow$ Quantizer $\rightarrow$ Encoder.
- The Sampler creates intermediate PAM; Quantizer rounds it; Encoder turns it to binary.
- System Bit Rate $(R_b) = n \times f_s$.

Next Lecture Preview

- Lecture 36: PCM Receiver
- How do we turn those binary bits back into music or voice?
- Understanding regeneration, decoding, and reconstruction.
- Advantages of PCM over analog systems.



Lecture 4.9: PCM Receiver & System Review

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

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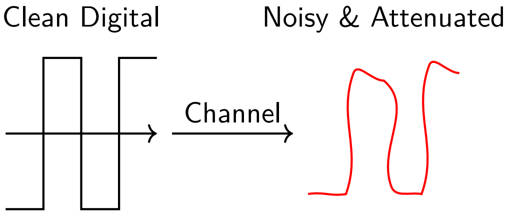
July 22, 2026

Lecture Overview

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
2. Lecture 1.2: Signals and its representation: analog and digital Signal, Pulse, Impulse
3. Lecture 1.3: Saw-tooth, sinusoidal and rectangular signals (In Time & frequency domain)
4. Lecture 1.4: Block diagram of Analog and Digital communication system
5. Lecture 1.5: Modulation: Definition & its classification
6. Lecture 1.6: Classification based on analog & pulse signal as carrier
7. Lecture 1.7: Noise in communication system, classification, S/N, noise figure
8. Lecture 2.1: Need of modulation & Types of modulation techniques
9. Lecture 2.2: Amplitude Modulation: Modulating & Carrier signal, modulation Index, mathematical representation
10. Lecture 2.3: AM representation in time and frequency domain, Frequency Spectrum

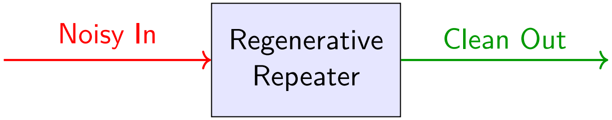
The Transmission Channel

- The PCM bit stream travels through a channel (cable, fiber, wireless).
- Along the way, it suffers from Attenuation (loss of signal strength) and Distortion (noise).
- Clean binary pulses (square waves) become rounded, weak, and noisy.
- If they degrade too much, the receiver might confuse a '1' for a '0'.



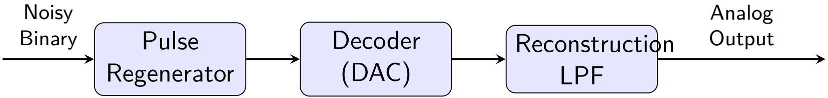
Regenerative Repeaters

- Unlike analog amplifiers which amplify both signal AND noise, digital systems use **Regenerators**.
- Placed at intervals along the channel.
- **Function:** They 'read' the noisy digital signal, decide if it's a 1 or 0, and generate a brand-new, perfectly clean pulse.
- **Result:** Noise does not accumulate over long distances!



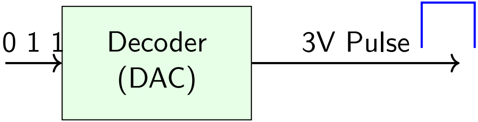
PCM Receiver Block Diagram

- The receiver reverses the transmitter's process.
- **Main blocks:**
  1. Pulse Regenerator (at the input)
  2. Decoder (Digital-to-Analog Converter)
  3. Reconstruction Filter (Low-Pass Filter)



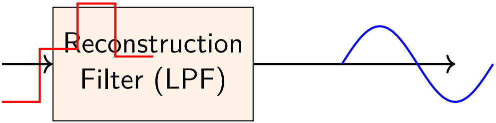
Block 1: Decoder

- **Function:** Converts the incoming binary bits back into discrete voltage levels.
- It groups the bits into *n*-bit words.
- It acts as a Digital-to-Analog Converter (DAC).
- If companding was used at the transmitter, an **Expander** is included here to restore the original dynamic range.

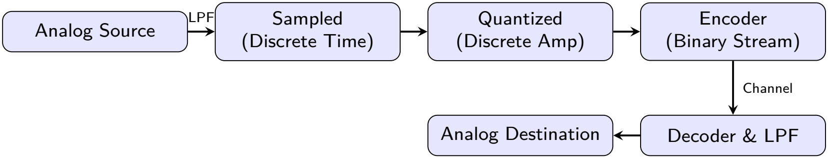


Block 2: Reconstruction Filter

- The output of the decoder is a staircase-like signal (quantized PAM).
- **Function:** A Low-Pass Filter (LPF) smooths out the staircase to recover the continuous analog waveform.
- Cutoff frequency is set to f_m.
- Removes the high-frequency harmonics created by the abrupt 'steps'.



Overall PCM Process Summary



Advantages of PCM

1. **Incredible Noise Immunity:** Regenerators prevent noise accumulation.
2. **Easy Multiplexing:** Multiple digital signals can be interleaved easily (TDM).
3. **Secure:** Digital data can be easily encrypted (scrambled).
4. **Storage:** Digital data is easily stored in memory, hard drives, CDs.
5. **Error Coding:** Extra bits can be added to detect and correct errors.

Disadvantages of PCM

1. **Huge Bandwidth Requirement:** A 4 kHz voice signal requires 64 kbps of digital bandwidth.
2. **High Complexity:** Requires complex ADCs, DACs, and precise synchronization.
3. **Quantization Noise:** Inherent and permanent distortion introduced during digitization.

Numerical Example: PCM Bandwidth

Problem: What is the transmission bandwidth required for the PCM signal in our previous example ($R_b = 64$ kbps)?

Solution

Rule of thumb: Minimum Baseband Bandwidth $(B_T) \geq \frac{R_b}{2}$.

Step 1: $B_T \geq \frac{64{,}000}{2}$

Step 2: $B_T = 32$ kHz

System Comparison: Analog vs Digital

Analog (AM/FM)	**Digital (PCM)**
Low bandwidth	High bandwidth
Noise accumulates	Noise is rejected (regenerators)
Hard to encrypt	Easy to encrypt
Hard to multiplex	Easy to multiplex (TDM)

Verdict: In modern times, bandwidth is cheap (fiber optics), so digital wins.

Key Takeaways (Unit 4)

- Sampling Theorem ($f_s \geq 2f_m$) is the foundation of digital comms.
- Quantization discretizes amplitude but adds noise.
- PCM transmitters sample, quantize, and encode analog into binary.
- PCM receivers decode and filter binary back into analog.
- Regenerative repeaters give PCM unparalleled noise immunity.

Next Unit Preview

Unit 5: Electromagnetic Wave and Antenna

- How do we actually transmit these signals wirelessly?
- Properties of EM waves.
- How antennas radiate and capture energy.
- Radiation patterns and antenna parameters.

Lecture 5.1: Properties of EM (electromagnetic) wave

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Course Overview & Today's Agenda

1. Lecture 1.1: Electromagnetic (EM) wave spectrum, frequency bands and their applications domain
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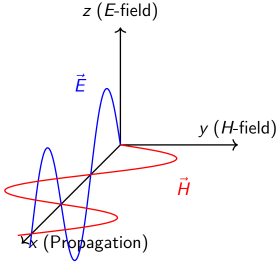
Learning Objectives

- Define Electromagnetic (EM) waves
- Understand the properties of EM waves in free space
- Relate Electric (E) and Magnetic (H) fields
- Perform basic calculations for frequency and wavelength

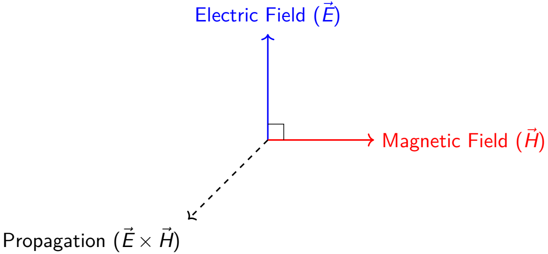
What is an Electromagnetic Wave?

- Created by accelerating electric charges
- Consists of oscillating electric and magnetic fields
- Does not require a physical medium to travel (can travel through a vacuum)
- Carries energy and momentum across space



Electric (E) and Magnetic (H) Fields

- E-field and H-field are mutually perpendicular
- Both fields are also perpendicular to the direction of propagation (Transverse nature)
- The cross product of E and H gives the direction of wave travel (Poynting vector)
- In free space, they are in phase with each other



Properties of EM Waves: Speed

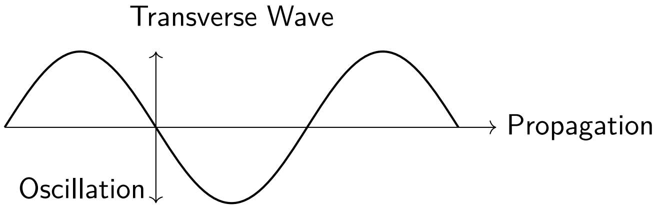
- In a vacuum, EM waves travel at the speed of light (c)
- $c \approx 3 \times 10^8$ meters per second
- Speed depends on the medium's permittivity (ε) and permeability (μ)

Speed of EM Waves

$$v = \frac{1}{\sqrt{\mu\varepsilon}}$$

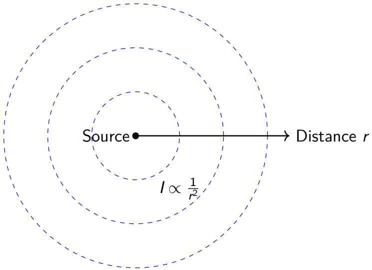
Properties of EM Waves: Transverse Nature

- They are transverse waves, not longitudinal
- Can be polarized (restricted to a particular plane of oscillation)
- Subject to reflection, refraction, diffraction, and interference
- Carry no net electrical charge



Energy and Power in EM Waves

- Energy is shared equally between the Electric and Magnetic fields
- Power density is described by the Poynting Vector
- $P = E \times H$
- Intensity decreases with the square of the distance (Inverse Square Law)



The Wave Equation

- Relates speed (c), frequency (f), and wavelength (λ)
- f is measured in Hertz (Hz)
- λ is measured in meters (m)

The Wave Equation

$$c = f \times \lambda$$

$$c = 3 \times 10^8 \text{ m/s}$$

Numerical Example: Finding Wavelength

Problem: An FM radio station broadcasts at a frequency of 100 MHz. Calculate the wavelength of the EM wave.

Solution

Given:

Frequency $f = 100$ MHz $= 100 \times 10^6$ Hz

Speed of light $c = 3 \times 10^8$ m/s

Formula:

$\lambda = \frac{c}{f}$

Calculation:

$\lambda = \frac{3\times10^8}{100\times10^6} = \frac{3\times10^8}{10^8} = 3$ meters

Numerical Example: Finding Frequency

Problem: A microwave oven uses EM waves with a wavelength of 12.2 cm. What is the frequency?

Solution

Given:
Wavelength $\lambda = 12.2$ cm $= 0.122$ m
Speed of light $c = 3 \times 10^8$ m/s
Formula:

$f = \frac{c}{\lambda}$
Calculation:

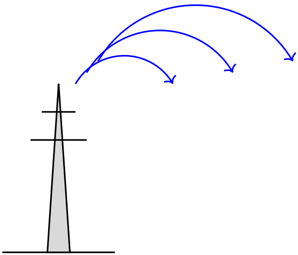
$f = \frac{3\times10^8}{0.122} \approx 2.45 \times 10^9$ Hz $= 2.45$ GHz

Summary of EM Wave Properties

- ✓ Transverse waves consisting of E and H fields
- ✓ Travel at 3×10^8 m/s in vacuum
- ✓ Obey the wave equation $c = f \times \lambda$
- ✓ Undergo reflection, refraction, and exhibit polarization

Next Lecture Preview

- Lecture 38: Antenna Fundamentals
- What is an antenna?
- Resonant vs. Non-resonant antennas
- Principle of Transmitting and Receiving antennas



Lecture 5.2: Antenna fundamentals: Resonant/Nonresonant, ideal antenna, principle of Tx/Rx antenna

Complete Lecture Deck – All 45 Lectures

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Learning Objectives

- Explain the function of an antenna as a transducer
- Understand how a transmission line couples to an antenna
- Compare resonant and non-resonant antennas
- Identify the theoretical baseline for antennas (Isotropic)

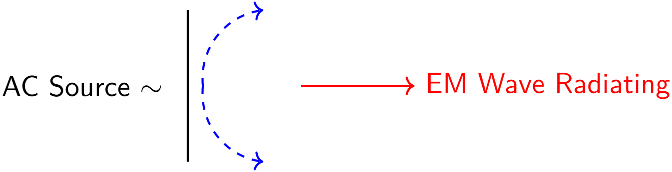
What is an Antenna?

- A metallic conductor system capable of radiating and capturing EM waves
- Acts as a transducer in communication systems
- Tx Antenna: Converts electrical signals (current/voltage) into EM waves
- Rx Antenna: Converts EM waves back into electrical signals

Electrical Signal → Tx Antenna ⇢ EM Waves ⇢ Rx Antenna → Electrical Signal

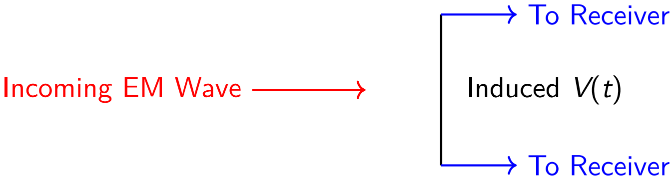
Principle of Transmitting (Tx) Antenna

- High-frequency AC current is fed into the antenna
- Accelerating charges in the conductor create changing magnetic fields
- Changing magnetic fields induce changing electric fields
- These coupled fields detach from the antenna and propagate into space



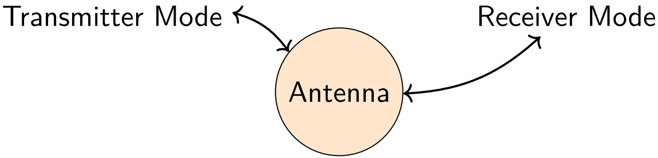
Principle of Receiving (Rx) Antenna

- Passing EM waves induce a small voltage across the antenna terminals
- The electric field of the wave pushes electrons in the conductor
- This creates an alternating current matching the frequency of the wave
- The signal is then fed into a receiver circuit for amplification



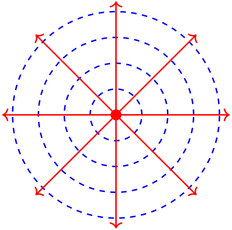
Principle of Reciprocity

- An antenna works equally well as a transmitter or a receiver
- Its properties (gain, radiation pattern, impedance) are identical in both modes
- Simplifies analysis: we only need to study an antenna in one mode
- Most communication systems use the same antenna for both Tx and Rx



The Ideal Antenna: Isotropic Radiator

- A theoretical, hypothetical point source antenna
- Radiates EM energy equally in all directions (3D sphere)
- Does not exist in reality, but used as a mathematical reference
- Used to define 'Gain' of practical antennas (dBi)

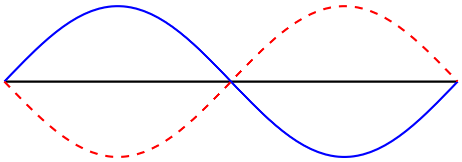


Spherical Radiation Pattern

Resonant Antennas

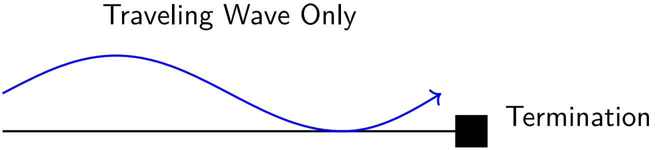
- Open-ended transmission lines
- Support standing waves (incident and reflected waves)
- Designed for a specific narrow frequency band
- Examples: Half-wave dipole, Quarter-wave monopole

Standing Wave Pattern



Non-Resonant Antennas

- Properly terminated antennas (no reflection)
- Support traveling waves only
- Operate over a very wide range of frequencies (Broadband)
- Examples: Rhombic antenna, Beverage antenna



Resonant vs Non-Resonant: Summary

Feature	**Resonant Antenna**	**Non-Resonant Antenna**
Wave Type	Standing waves	Traveling waves
Bandwidth	Narrow bandwidth	Wide bandwidth (Broadband)
Radiation Pattern	Bidirectional/ Omnidirectional	Unidirectional

Choice depends on the application's bandwidth requirements.

Numerical Example: Antenna Length

Problem: Calculate the physical length of a half-wave resonant dipole for a 300 MHz signal.

Solution

Given:
Frequency $f = 300$ MHz $= 300 \times 10^6$ Hz
Speed of light $c = 3 \times 10^8$ m/s

Step 1: Wavelength λ

$\lambda = \frac{c}{f} = \frac{3\times10^8}{300\times10^6} = 1$ meter

Step 2: Antenna length L

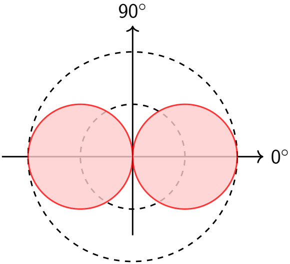
$L = \frac{\lambda}{2} = \frac{1}{2} = 0.5$ meters

Key Takeaways

- Antennas are transducers between electrical signals and EM waves
- Reciprocity means Tx and Rx properties are identical
- Isotropic radiator is a theoretical 3D spherical radiator
- Antennas are classified as Resonant (standing waves) or Non-resonant (traveling waves)

Next Lecture Preview

- Lecture 39: Antenna Parameters Part 1
- Radiation pattern
- Polarization
- Bandwidth and Beam width



Lecture 5.3: Antenna parameters: Radiation pattern, polarization, bandwidth, beam width

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Milav Dabgar

Lecturer, GP Palanpur

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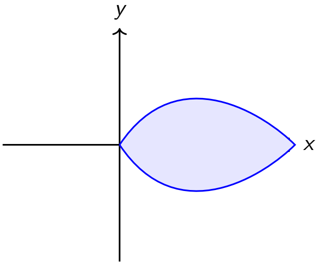
Recap: Previous Lecture

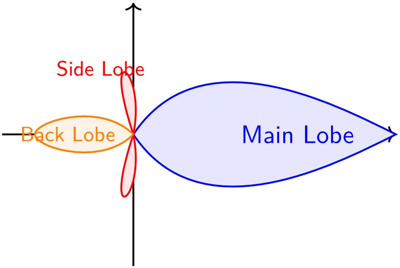
- In Lecture 38, we covered antenna fundamentals, resonant/non-resonant types, and the isotropic radiator.

- A graphical representation of the radiation properties as a function of space coordinates
- Shows where the antenna directs most of its energy
- Usually plotted in 2D (polar coordinates) for E-plane and H-plane
- Can be plotted in terms of Field strength or Power density



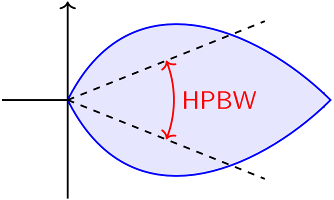
Parts of a Radiation Pattern

- Major Lobe (Main Lobe): Direction of maximum radiation
- Minor Lobes: Any lobe other than the main lobe
- Side Lobes: Adjacent to the main lobe (usually unwanted)
- Back Lobe: Directly opposite the main lobe



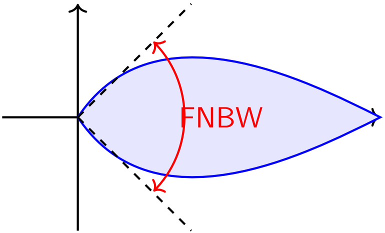
Beam Width

- Measures how highly directional an antenna is
- Half-Power Beam Width (HPBW): The angular separation between two points on the main lobe where power is half of the maximum
- Half power corresponds to -3 dB
- Narrow beamwidth = Highly directional



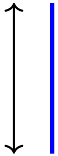
First Null Beam Width (FNBW)

- Angular separation between the first nulls (zero power points) on either side of the main lobe
- FNBW is roughly twice the HPBW
- Provides a measure of the total width of the main lobe
- Important for determining interference with adjacent targets

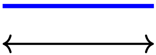


Polarization

- The orientation of the Electric field (E-field) vector of the radiated wave
- Determined by the physical orientation of the antenna
- Linear Polarization: Vertical or Horizontal
- Circular Polarization: Right-hand or Left-hand



Vertical Polarization



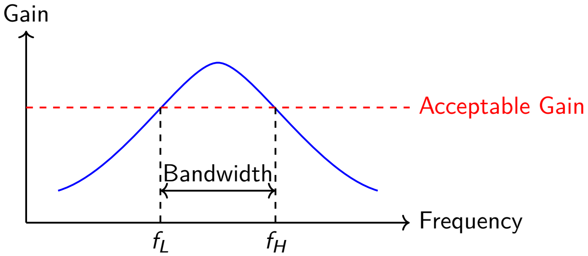
Horizontal Polarization

Importance of Polarization Match

- Maximum signal transfer occurs when Tx and Rx polarization match
- Cross-polarization (e.g., Tx Vertical, Rx Horizontal) results in severe signal loss (theoretically zero reception)
- Circular polarization is used in satellites to overcome orientation issues
- TV broadcasting usually uses horizontal polarization

Antenna Bandwidth

- The range of frequencies over which the antenna operates optimally
- Defined by specific criteria like impedance match (VSWR < 2) or gain drop
- Narrowband antennas: Bandwidth is a small percentage of center frequency (e.g., Dipole)
- Broadband antennas: Bandwidth spans octaves (e.g., Log-periodic)



Problem: An antenna has a main lobe spanning from $\theta = 30^\circ$ to $\theta = 60^\circ$ at its -3dB points. Its operating frequency range is 900 MHz to 960 MHz.

Solution

Calculate HPBW:

$\text{HPBW} = 60^\circ - 30^\circ = 30^\circ$

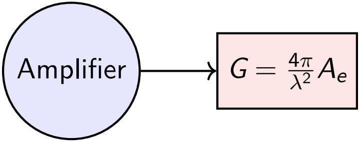
Calculate Bandwidth:

$\text{BW} = 960\ \text{MHz} - 900\ \text{MHz} = 60\ \text{MHz}$

Key Takeaways

- Radiation patterns visualize where an antenna focuses its energy
- Beam width (HPBW) quantifies antenna directivity
- Matching polarization between Tx and Rx is vital to minimize signal loss
- Bandwidth specifies the functional frequency range of the antenna

- Lecture 40: Antenna Parameters Part 2
- Antenna resistance
- Directivity
- Power gain and Antenna gain



Lecture 5.4: Antenna parameters: antenna resistance, directivity, power gain, antenna gain

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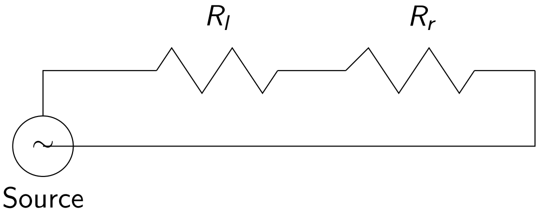
Antenna Resistance

- Total antenna resistance (R_t) consists of two parts:
- 1. Radiation Resistance (R_r)
- 2. Ohmic Loss Resistance (R_l)

$R_t = R_r + R_l$



Radiation Resistance (R_r)

- The fictitious resistance that would dissipate the same amount of power as the antenna actually radiates into space
  $P_{rad} = I^2 \times R_r$
- Desired to be as high as possible compared to loss resistance
- For a half-wave dipole, $R_r \approx 73\Omega$

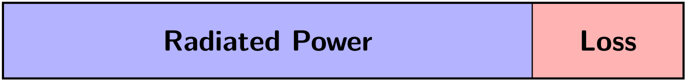
Key Formula

$P_{rad} = I^2 R_r$

Antenna Efficiency (η)

- Ratio of the power radiated to the total power supplied to the antenna

  $\eta = \frac{P_{rad}}{P_{total}}$
- In terms of resistance: $\eta = \frac{R_r}{R_r + R_l}$
- High efficiency means very little power is lost as heat in the antenna wires



Total Input Power

Directivity (D)

- The ratio of the radiation intensity in a given direction from the antenna to the radiation intensity averaged over all directions
- Compares the antenna's peak intensity to an Isotropic radiator
- Focuses purely on the shape of the radiation pattern
- Does NOT take antenna efficiency into account

Power Gain / Antenna Gain (G)

- Takes both Directivity and Efficiency into account
  $G = \eta \times D$
- Where η is efficiency and D is directivity
- It's the actual, realized gain of the antenna in the real world

Gain Formula

$G = \text{Efficiency} \times \text{Directivity}$

Understanding Gain Units (dBi vs dBd)

- Gain is usually expressed in decibels (dB)
- **dBi**: Gain relative to an Isotropic radiator
- **dBd**: Gain relative to a half-wave Dipole radiator
- 0 dBd = 2.15 dBi (because a dipole inherently has 2.15 dB gain over an isotropic radiator)

Antenna Type	Gain in dBi	Gain in dBd
Isotropic	0 dBi	−2.15 dBd
Half-wave Dipole	2.15 dBi	0 dBd

Effective Isotropic Radiated Power (EIRP)

- The total power that would have to be radiated by an isotropic antenna to achieve the same signal strength as the actual antenna in the main beam direction

  EIRP $=$ Transmitter Power $\times$ Antenna Gain (linear)
- In dB:

  $\text{EIRP(dBm)} = P_{tx}\text{(dBm)} + G\text{(dBi)} - \text{Cable Losses(dB)}$

Transmitter → Cable Loss → Antenna Gain → EIRP

Numerical Example: Efficiency

Problem: An antenna has a radiation resistance of 72Ω and a loss resistance of 8Ω. Calculate its efficiency.

Solution

Given:

$R_r = 72\Omega,\ R_l = 8\Omega$

Formula:

$\eta = \frac{R_r}{R_r + R_l}$

Calculation:

$\eta = \frac{72}{72+8} = \frac{72}{80} = 0.9 \text{ or } 90\%$

Numerical Example: Gain Calculation

Problem: The antenna from the previous example has a directivity of 20. What is its Power Gain?

Solution

Given:
$D = 20$, $\eta = 0.9$

Formula:

$$G = \eta \times D$$

Calculation:

$$G = 0.9 \times 20 = 18$$

In decibels: $G(dB) = 10 \log_{10}(18) \approx 12.55$ dB

Summary of Power Parameters

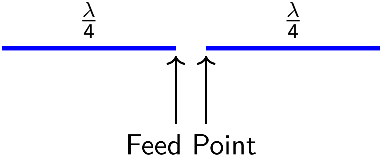
- **Radiation Resistance** represents the power radiated into space
- **Efficiency** measures how much power is radiated vs lost as heat
- **Directivity** describes how focused the beam pattern is
- **Gain** combines Directivity and Efficiency for practical measurement

Key Takeaways

- Antenna efficiency is governed by the ratio of radiation resistance to total resistance
- Directivity is a theoretical measure of how focused an antenna's beam is
- Gain is the practical measure of performance, equal to Directivity multiplied by Efficiency
- EIRP is a critical system parameter for regulatory compliance

Next Lecture Preview

- Lecture 41: Dipole antenna and radiation pattern
- What is a half-wave dipole?
- Current and voltage distribution
- Plotting the dipole's radiation pattern



Lecture 5.5: Dipole antenna and radiation pattern

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Lecturer, GP Palanpur

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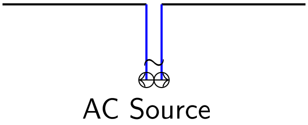
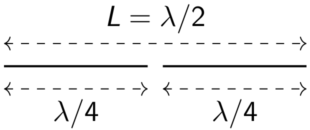
What is a Dipole Antenna?

- A simple wire antenna consisting of two identical conductive elements
- Usually fed by a coaxial cable or twin-lead feed line in the center
- Called 'dipole' because it has two poles (two halves)
- The most widely used class of antenna



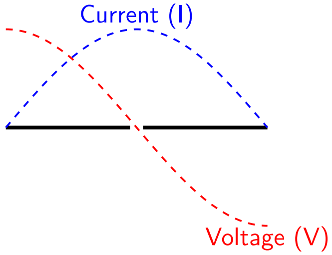
The Half-Wave Dipole ($\lambda/2$)

- A dipole whose total physical length is exactly half the wavelength of the operating frequency
- $L = \lambda/2$
- It is a resonant antenna (creates standing waves)
- Offers a good balance of physical size and radiation efficiency



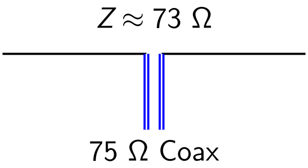
Voltage and Current Distribution

- Since the ends are open circuits, current at the ends must be zero
- Voltage at the ends is maximum
- At the center (feed point), current is maximum and voltage is minimum
- This creates an impedance (V/I) that is lowest at the center (approx 73 Ω)



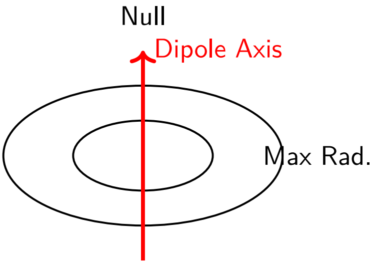
Impedance of a Half-Wave Dipole

- The radiation resistance is approximately 73 Ω
- Since it is resonant, the reactive part (capacitance/inductance) is 0 Ω
- $Z = 73 + j0\ \Omega$
- Matches very well with standard 75 Ω coaxial cables (like TV antennas)



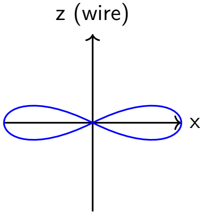
Radiation Pattern of a Half-Wave Dipole

- Radiates strongest perpendicular to the axis of the wire
- Radiates zero energy directly off the ends of the wire (nulls)
- The 3D radiation pattern resembles a doughnut (toroid)
- It is an Omni-directional antenna in the H-plane (if mounted vertically)

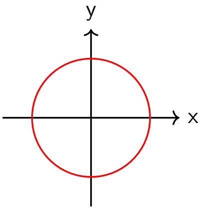


2D Cross-Sections (E-Plane and H-Plane)

- E-Plane (Elevation if vertical): Figure-8 pattern
- H-Plane (Azimuth if vertical): Perfect circle
- Gain: 2.15 dBi (It concentrates power slightly better than an isotropic radiator)



E-Plane (Figure-8)



H-Plane (Circle)

End-Effect (Fringing)

- In reality, EM fields bulge slightly at the ends of the wire
- This makes the antenna act electrically longer than its physical length
- To compensate, real dipoles are cut about 5% shorter than $\lambda/2$
- Practical Length $L \approx 0.95 \times (\lambda/2) \approx 142.5/f(\text{MHz})$ meters

Numerical Example: Sizing a Dipole

Problem: Design a half-wave dipole for an FM station at 100 MHz. Calculate the practical physical length.

Solution

Given:
$f = 100$ MHz $= 100 \times 10^6$ Hz

Step 1: Wavelength
$\lambda = \frac{c}{f} = \frac{3\times10^8}{100\times10^6} = 3$ m

Step 2: Theoretical Length
$L_{\text{theoretical}} = \frac{\lambda}{2} = 1.5$ m

Step 3: Practical Length
$L_{\text{practical}} = 0.95 \times 1.5 = 1.425$ meters

Advantages and Applications

- **Advantages:** Simple, cheap to build, natural 73 Ω match, predictable radiation
- **Disadvantages:** Physically large at low frequencies (e.g., AM radio)
- **Applications:** TV reception, FM radio, reference antenna for testing, driven element in Yagi antennas

Key Takeaways

- A half-wave dipole is a resonant antenna with a length of $\lambda/2$.
- Current is max at center, Voltage is max at ends ($Z \approx 73\ \Omega$).
- It has a characteristic doughnut-shaped 3D radiation pattern (Figure-8 in a 2D cross-section).
- Practical dipole construction requires shortening the antenna by roughly 5% due to end effects.

Next Lecture Preview

- **Lecture 42:** Radiation Pattern for Unidirectional, bidirectional and Omni directional antenna
- Categorizing antennas by their radiation shape
- When to use which pattern

Lecture 5.6: Radiation Pattern for Unidirectional, bidirectional and Omni directional antenna

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

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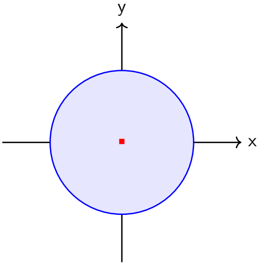
Omni-Directional Antennas

- Radiates RF energy equally in all horizontal directions (360 degrees)
- The H-plane pattern is a perfect circle
- Does NOT radiate equally in the vertical plane (that would be Isotropic)
- Usually mounted vertically

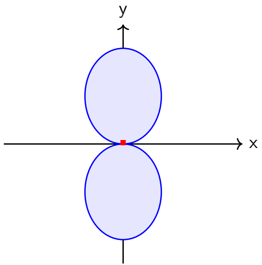


Examples and Applications

- **Examples:** Vertical monopole (car antenna), vertical dipole, collinear array
- **Applications:** FM/AM Radio broadcasting, Wi-Fi routers, Cell phone towers, mobile handsets
- **Use Case:** When the transmitter doesn't know where the receivers are, or receivers are moving

Bi-Directional Antennas

- Radiates strongly in two opposite directions
- Has deep nulls (zero radiation) in the directions perpendicular to the main lobes
- Radiation pattern looks like a Figure-8

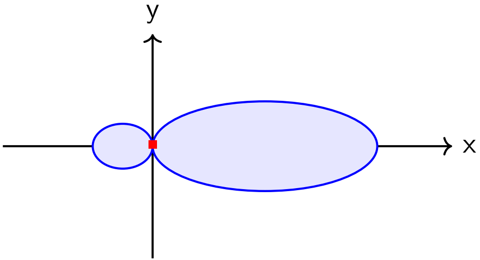


Examples and Applications

- **Examples:** Horizontal half-wave dipole, basic loop antenna
- **Applications:** Communication along long corridors, highways, or railway lines
- **Use Case:** When you want to cover a long strip of area but reject noise from the sides

Uni-Directional Antennas

- Radiates RF energy heavily in one specific direction
- Suppresses radiation in all other directions (high Front-to-Back ratio)
- Pattern has one large Major Lobe and very small minor/back lobes
- Provides high Gain



Examples and Applications

- **Examples:** Yagi-Uda antenna (TV antenna), Parabolic Dish, Horn antenna
- **Applications:** Point-to-Point links, Satellite TV, Radar, Microwave backhaul
- **Use Case:** When transmitter and receiver are fixed, and you need maximum range and minimum interference

Comparing the Three Types

- **Omni:** 360° coverage, Low Gain, Short Range (Broadcast)
- **Bi-directional:** 2 opposite directions, Medium Gain, Strip coverage
- **Uni-directional:** 1 narrow direction, High Gain, Long Range (Point-to-Point)

Type	Coverage	Gain	Application
Omni	360°	Low	Mobile/Broadcast
Bi-directional	2 Directions	Medium	Corridors
Uni-directional	1 Direction	High	Point-to-Point

Beamwidth and Gain Relationship

- As an antenna becomes more Uni-directional, its beamwidth decreases
- As beamwidth decreases, Gain increases
- Gain is achieved by taking energy away from unwanted directions and adding it to the desired direction
- Conservation of Energy: Antennas don't create power, they just focus it

Quick Check: Scenario Matching

- **Scenario 1:** Communicating with a space probe on Mars. (Answer: Uni-directional / Dish)
- **Scenario 2:** Broadcasting a local radio station to a whole city. (Answer: Omni-directional)
- **Scenario 3:** Providing Wi-Fi up and down a long hotel hallway. (Answer: Bi-directional)

Key Takeaways

- Omnidirectional antennas provide 360-degree coverage, ideal for broadcasting and mobile applications
- Bidirectional antennas (like dipoles) radiate in a Figure-8 pattern, useful for corridor coverage
- Unidirectional antennas focus energy tightly in one direction, yielding high gain for long-distance point-to-point links
- Antenna gain is achieved by narrowing the beamwidth, not by amplifying power

Next Lecture Preview

- Lecture 43: Antenna: Loop antenna, folded dipole
- Structure and properties of Loop antennas
- Why we use Folded dipoles instead of standard dipoles

Lecture 5.7: Antenna: Loop antenna, folded dipole

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

Course Overview & Today's Agenda

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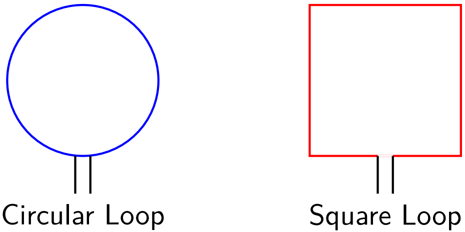
What is a Loop Antenna?

- A continuous wire formed into a closed shape (circle, square, rectangle)
- The ends of the wire are connected to the transmission line
- Functions primarily by responding to the Magnetic Field (H-field) of the EM wave
- Can be constructed as a single turn or multi-turn coil



Small Loop Antennas

- Perimeter is much less than a wavelength (Length $< \lambda/10$)
- Has a very low radiation resistance
- Radiation pattern is a Figure-8 (similar to a short dipole)
- Highly inefficient for transmitting, but excellent for receiving due to low noise pickup

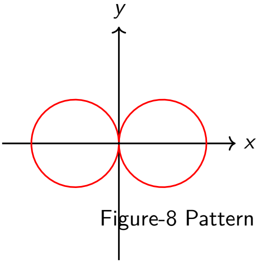
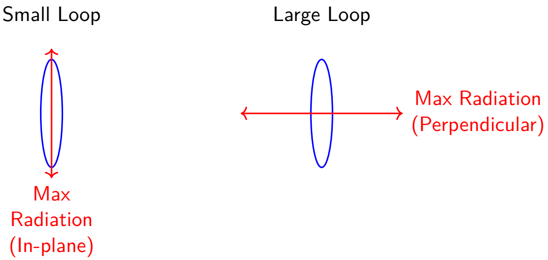


Figure-8 Pattern

Large Loop Antennas

- Perimeter is roughly equal to one wavelength (Length $\approx \lambda$)
- Resonant antenna with high radiation resistance
- Radiates maximally perpendicular to the plane of the loop
- Good efficiency for both transmitting and receiving

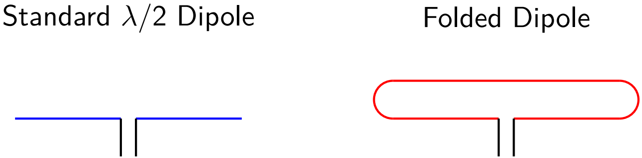


Applications of Loop Antennas

- AM Radio receivers (Ferrite rod antennas)
- Direction finding and navigation (due to the sharp nulls in the Figure-8 pattern)
- UHF Television reception (often a circular loop on set-top antennas)
- RFID tags and NFC (Near Field Communication) on smartphones

The Folded Dipole

- A half-wave dipole where the ends are folded back and connected together
- Consists of two parallel $\lambda/2$ conductors connected at the ends
- One conductor is split in the center to attach the feed line
- Radiation pattern is identical to a standard half-wave dipole (Figure-8)



Impedance of a Folded Dipole

- Standard dipole impedance $= 73\ \Omega$
- Folded dipole impedance is scaled by the square of the number of conductors
- For a 2-wire folded dipole: $73\ \Omega \times 2^2 = 73 \times 4 \approx 300\ \Omega$
- Matches perfectly with $300\ \Omega$ twin-lead cable (common in older TV systems)

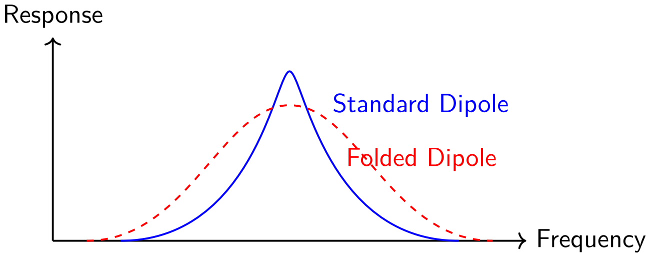
Impedance Equation

$$Z_{folded} = Z_{standard} \times N^2$$

Where N is the number of parallel conductors. For $N = 2$, $Z = 73 \times 4 = 292\ \Omega \approx 300\ \Omega$.

Bandwidth of a Folded Dipole

- A folded dipole has a wider bandwidth than a standard dipole
- The parallel conductors act like a thicker wire, which lowers the Q-factor
- Lower Q-factor means broader frequency response
- Ideal for wideband applications like Television broadcasting (which requires 6-8 MHz per channel)



Numerical Example: Folded Dipole Impedance

Problem: A special folded dipole is constructed using 3 parallel wires of equal diameter. Calculate its approximate feed-point impedance.

Solution

Given:
Standard dipole $Z = 73\ \Omega$
Number of wires $N = 3$

Formula:

$$Z_{folded} = Z_{standard} \times N^2$$

Calculation:

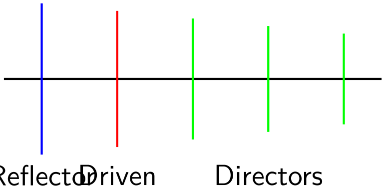
$Z = 73 \times 3^2$
$Z = 73 \times 9$
$Z = 657\ \Omega$

Key Takeaways

- Loop antennas respond primarily to magnetic fields and are great for direction finding
- Folded dipoles offer 4x the impedance ($\approx 300\ \Omega$) of a standard dipole
- Folded dipoles have wider bandwidth, making them ideal for TV reception
- Both antennas solve specific impedance and noise issues

Next Lecture Preview

- Lecture 44: Antenna: Yagi-Uda antenna and smart antenna
- How a TV antenna (Yagi) gets its high gain
- Parasitic elements: Reflectors and Directors
- Introduction to modern Smart Antennas



Lecture 5.8: Antenna: Yagi-Uda antenna and smart antenna

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

July 22, 2026

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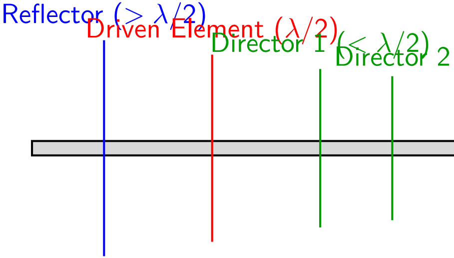
What is a Yagi-Uda Antenna?

- Invented by Hidetsugu Yagi and Shintaro Uda in Japan (1926)
- A highly directional (uni-directional) antenna array
- Consists of multiple parallel dipole elements mounted on a boom
- Widely used for TV reception and amateur radio

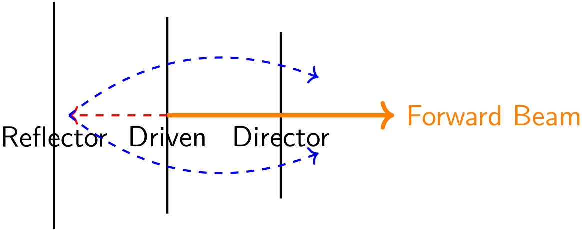


Elements of a Yagi

- **Driven Element:** Usually a folded dipole (connects to the transmitter/receiver)
- **Parasitic Elements:** Not connected to any wires; they re-radiate energy
- **Reflector:** Placed behind the driven element. Slightly longer than $\lambda/2$. Reflects energy forward.
- **Director(s):** Placed in front of the driven element. Slightly shorter than $\lambda/2$. Pulls energy forward.

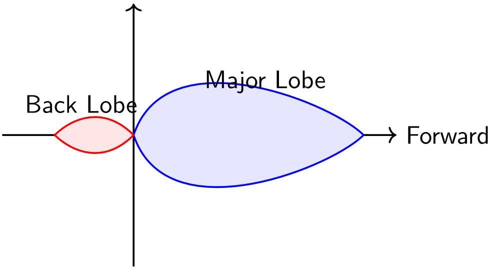
How the Yagi Works

- Driven element radiates an EM wave
- Wave hits the Reflector, induces a current, and re-radiates out of phase (cancels backward wave, adds to forward wave)
- Wave hits the Directors, which focus the wave further forward
- **Result:** High gain in the forward direction, deep null in the backward direction



Yagi Performance

- **Gain:** Typically 7 to 15 dBd, depending on the number of directors
- More directors = Narrower beamwidth and Higher Gain
- **Front-to-Back (F/B) Ratio:** High (rejects interference from behind)
- **Bandwidth:** Relatively narrow (usually tuned for a specific frequency band)



Introduction to Smart Antennas

- Traditional antennas (like Yagi) have a fixed radiation pattern
- **Smart Antennas** are arrays of antenna elements with digital signal processing
- They can dynamically change their radiation pattern in real-time
- Used extensively in 4G LTE, 5G, and modern Wi-Fi routers

- **Switched Beam:** Has a set of pre-defined fixed beams and switches between them to find the best signal
- **Adaptive Array (Beamforming):** Continuously steers a dynamic beam directly at the user while placing 'nulls' on sources of interference

Beamforming and MIMO

- **Beamforming:** Adjusting the phase of signals at each element to focus energy in one direction
- **MIMO (Multiple Input Multiple Output):** Using multiple antennas to send multiple data streams simultaneously
- Increases data capacity without needing more bandwidth
- The backbone of 5G cellular technology

Numerical Example: Array Gain

Problem: If a smart antenna array combines 4 identical elements, what is the theoretical maximum increase in power gain?

Solution

Principle: Power gain increases linearly with the number of elements (N).

Formula:

$$\text{Gain Increase (dB)} = 10 \cdot \log_{10}(N)$$

Calculation:

$$10 \cdot \log_{10}(4) = 10 \cdot 0.602 = 6.02 \text{ dB increase}$$

Summary: Key Takeaways

- The Yagi-Uda antenna uses parasitic elements (reflectors and directors) to shape a unidirectional beam
- Adding more directors to a Yagi increases gain and narrows the beamwidth
- Smart antennas use digital signal processing to dynamically steer radiation patterns (beamforming)
- MIMO technology uses multiple antennas to increase data capacity, heavily used in 5G and modern Wi-Fi

Next Lecture Preview

- **Lecture 45: Full Course Recap**
- Reviewing Unit 1 to Unit 5
- Connecting the dots from basic signals to smart antennas

Lecture 5.9: Full Course Recap (Units 1 to 5)

Complete Lecture Deck – All 45 Lectures

Milav Dabgar

Lecturer, GP Palanpur

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Recap: Previous Lecture

- In Lecture 44, we learned about highly directional Yagi antennas and modern Smart Antennas.

The Big Picture: A Communication System

- Information Source $\rightarrow$ Transmitter $\rightarrow$ Channel $\rightarrow$ Receiver $\rightarrow$ Destination
- Unit 1: The Basics (Signals, Noise, EM Spectrum)
- Units 2 & 3: Analog Modulation (AM & FM Transmitter/Receiver)
- Unit 4: Digital Modulation (Sampling, PCM)
- Unit 5: The Channel Interface (EM Waves & Antennas)

Unit 1 Recap: Basics of Communication

- EM Spectrum and Frequency Bands (VHF, UHF, Microwave)
- Analog vs Digital signals (Time and Frequency domains)
- Need for Modulation (Antenna size, multiplexing)
- Noise in Communication Systems (Signal-to-Noise Ratio)

Unit 2 Recap: Amplitude Modulation (AM)

- Changing the amplitude of a high-frequency carrier
- Modulation Index (m) and sidebands (DSB-FC, DSB-SC, SSB)
- Power distribution: Carrier takes 67%, Sidebands take 33%
- Superheterodyne Receiver: Mixing to an Intermediate Frequency (IF)

Unit 3 Recap: Frequency Modulation (FM)

- Changing the frequency of the carrier; amplitude remains constant
- Modulation Index (m_f) and Bessel functions (infinite sidebands)
- Pre-emphasis and De-emphasis to reduce high-frequency noise
- Phase Locked Loop (PLL) as an FM demodulator

Comparing AM and FM

Amplitude Modulation (AM)	**Frequency Modulation (FM)**
Constant frequency, noisy	Constant amplitude, noise-resistant
Narrow bandwidth, long range	Wide bandwidth, shorter range (VHF)
Used for long-distance voice	Used for local high-fidelity audio

Unit 4 Recap: Sampling and Waveform Coding

- Nyquist Theorem: $f_s \geq 2f_m$ (to avoid aliasing)
- Pulse Modulation: PAM, PWM, PPM
- Quantization: Converting continuous analog values to discrete levels
- Pulse Code Modulation (PCM): The foundation of digital audio

Numerical Example: PCM Bit Rate

Problem: An audio signal (bandwidth 4 kHz) is sampled at the Nyquist rate and quantized into 256 levels. Calculate the bit rate.

Solution

Step 1: Sampling rate

$f_s = 2 \times 4 \text{ kHz} = 8 \text{ kHz} = 8000 \text{ samples/sec}$

Step 2: Bits per sample (n)

$2^n = 256 \implies n = 8 \text{ bits}$

Step 3: Bit rate

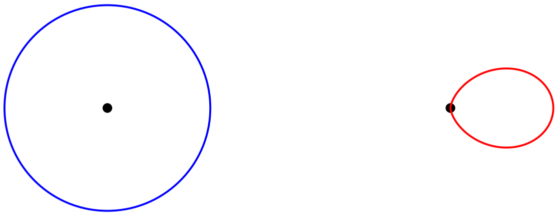
$R = f_s \times n = 8000 \times 8 = 64,000 \text{ bps} = 64 \text{ kbps}$

Unit 5 Recap: EM Waves and Antennas

- EM Waves travel at the speed of light: $c = f\lambda$
- Antennas are transducers between wires and free space
- Dipole ($\lambda/2$) is the fundamental resonant antenna (73 ohms, figure-8 pattern)
- Antenna Parameters: Gain, Directivity, Beamwidth, Polarization

Directional vs Omni Antennas

- Omni-directional: 360-degree coverage, good for mobile/broadcast
- Directional (Yagi, Dish): High gain, narrow beam, good for point-to-point links
- Smart Antennas: Dynamic beamforming using arrays and MIMO



Omni-directional

Directional Lobe

The Journey from Voice to Space

- 1. Your voice is converted to an analog signal (Mic)
- 2. It is sampled and digitized into PCM (Unit 4)
- 3. It is modulated onto a high-frequency carrier (Units 2 & 3)
- 4. It is fed to an antenna and radiated as an EM wave (Unit 5)
- 5. The receiver captures it and reverses the process

Full Course Summary

- Communication relies on modulation to transmit low-frequency data over high-frequency carriers
- Digital techniques (PCM) offer superior noise immunity compared to analog
- Antennas must be properly sized and matched to ensure efficient energy transfer
- Modern systems (5G, Wi-Fi) combine these principles with advanced Smart Antennas

Key Takeaways

- Modulation is essential for effective transmission and multiplexing
- Digital communication (Unit 4) provides resilience against noise compared to AM/FM
- The physical transmission of data relies on the properties of EM waves and the geometric design of antennas
- A complete communication system requires synergy between the signal processing blocks and the RF hardware

Full Course Recap

- Unit 1: Basics of Communication System
- Unit 2: Amplitude Modulation (AM) Communication
- Unit 3: Frequency Modulation (FM) Communication
- Unit 4: Sampling theory and waveform coding
- Unit 5: Electromagnetic Wave and antenna