MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 1

ECE

II / III

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J. Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture :**

Topic of Lecture: Generation and Demodulation on AM

Introduction: **Amplitude modulation** (**AM**) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

Genaration of Amplitude modulation:

- ✓ AM was the earliest modulation method used for transmitting audio in radio broadcasting.
- ✓ It was developed during the first quarter of the 20th century.
- ✓ Consider a carrier wave (sine wave) of frequency f_c and amplitude A given by:

$$c(t)=A \sin(2\pi f c t)$$

- ✓ Let $m(t)$ represent the modulation waveform. For this example we shall take the modulation to be simply a sine wave of a frequency f_m, a much lower frequency (such as an audio frequency) than f_c:

$$m(t)=M \cos(2\pi f m t+\phi),$$

where m is the amplitude sensitivity, M is the amplitude of modulation. If $m < 1$, $(1 + m(t)/A)$ is always positive for undermodulation. If $m > 1$ then overmodulation occurs and reconstruction of message signal from the transmitted signal would lead in loss of original signal. Amplitude modulation results when the carrier $c(t)$ is multiplied by the positive quantity $(1 + m(t)/A)$:

$$y(t)=[1+m \cos(2\pi f m t+\phi)]\, A \sin(2\pi f c t)$$

- ✓ In this simple case m is identical to the modulation index, discussed below. With $m = 0.5$ the amplitude modulated signal $y(t)$ thus corresponds to the top graph (labelled "50% Modulation") .
- ✓ Therefore, the modulated signal has three components: the carrier wave $c(t)$ which is unchanged, and two pure sine waves (known as sidebands) with frequencies slightly above and below the carrier frequency f_c.

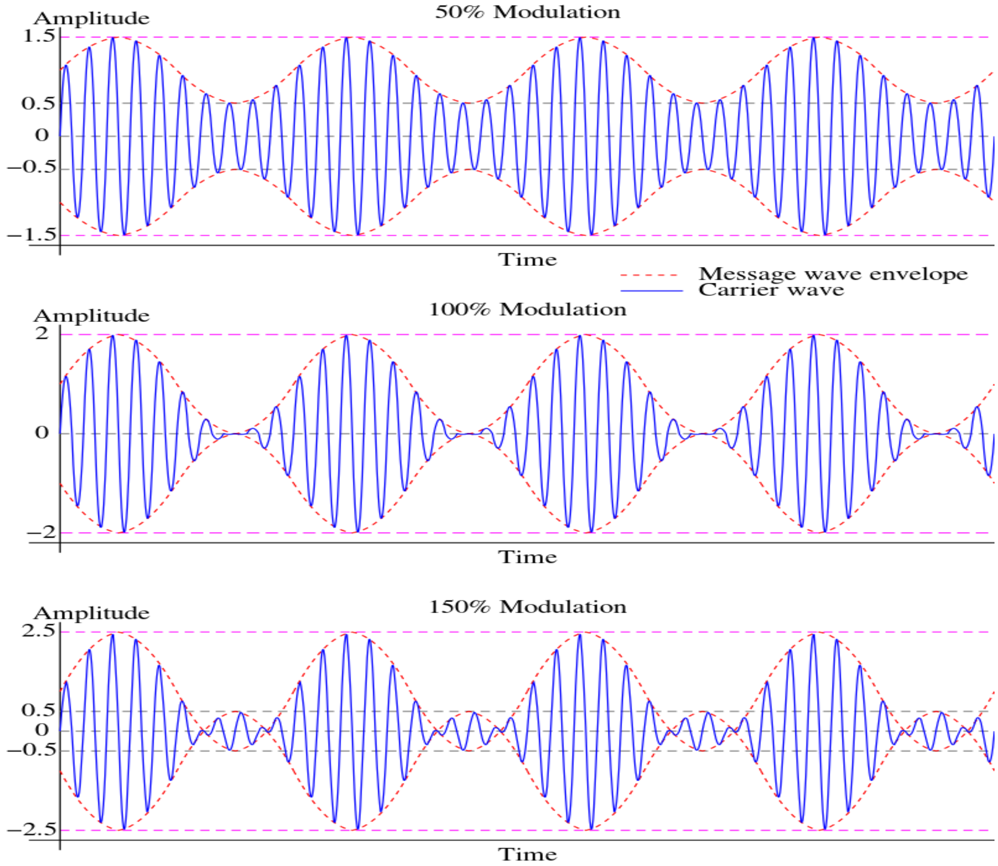
Modulation index and Demodulation of Aplittude modulation

- ✓ The AM modulation index is a measure based on the ratio of the modulation excursions of the RF signal to the level of the unmodulated carrier. It is thus defined as:

  m= Peak value of m(t) / A= M/A

  where M and A are the modulation amplitude and carrier amplitude, respectively; the modulation amplitude is the peak (positive or negative) change in the RF amplitude from its unmodulated value. Modulation index is normally expressed as a percentage, and may be displayed on a meter connected to an AM transmitter.

- ✓ So if m=0.5, carrier amplitude varies by 50% above (and below) its unmodulated level, as is shown in the first waveform, below. For m=1.0, it varies by 100% as shown in the illustration below it.



Modulation methods:

- ✓ Modulation circuit designs may be classified as low- or high-level (depending on whether they modulate in a low-power domain—followed by amplification for transmission—or in the high-power domain of the transmitted signal).

Plate modulation

In plate modulation, the plate voltage of the RF amplifier is modulated with the audio signal. The audio power requirement is 50 percent of the RF-carrier power.

Heising (constant-current) modulation

RF amplifier plate voltage is fed through a choke (high-value inductor). The AM modulation tube plate is fed through the same inductor, so the modulator tube diverts current from the RF amplifier. The choke acts as a constant current source in the audio range. This system has a low power efficiency.

Control grid modulation

The operating bias and gain of the final RF amplifier can be controlled by varying the voltage of the control grid. This method requires little audio power, but care must be taken to reduce distortion.

Clamp tube (screen grid) modulation

The screen-grid bias may be controlled through a *clamp tube*, which reduces voltage according to the modulation signal. It is difficult to approach 100-percent modulation while maintaining low distortion with this system.

Doherty modulation

One tube provides the power under carrier conditions and another operates only for positive modulation peaks. Overall efficiency is good, and distortion is low.

Outphasing modulation

Two tubes are operated in parallel, but partially out of phase with each other. As they are differentially phase modulated their combined amplitude is greater or smaller. Efficiency is good and distortion low when properly adjusted.

Pulse-width modulation (PWM) or pulse-duration modulation (PDM)

A highly efficient high voltage power supply is applied to the tube plate. The output voltage of this supply is varied at an audio rate to follow the program. This system was pioneered by Hilmer Swanson and has a number of variations, all of which achieve high efficiency and sound quality.

Demodulation:

- ✓ The simplest form of AM demodulator consists of a diode which is configured to act as envelope detector. Another type of demodulator, the product detector, can provide better-quality demodulation with additional circuit complexity.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

LECTURE HANDOUTS

L - 2

ECE

II / III

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture:**

Topic of Lecture: DSB SC

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

Generation of DSB SC:

- ✓ In the process of Amplitude Modulation, the modulated wave consists of the carrier wave and two sidebands. The modulated wave has the information only in the sidebands. **Sideband** is nothing but a band of frequencies, containing power, which are the lower and higher frequencies of the carrier frequency.
- ✓ The transmission of a signal, which contains a carrier along with two sidebands can be termed as **Double Sideband Full Carrier** system or simply **DSBFC**.

Mathematical Expression of DSBSC

- ✓ Let us consider the same mathematical expressions for modulating and carrier signals as we have considered in the earlier chapters.

i.e., Modulating signal

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

Carrier signal

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

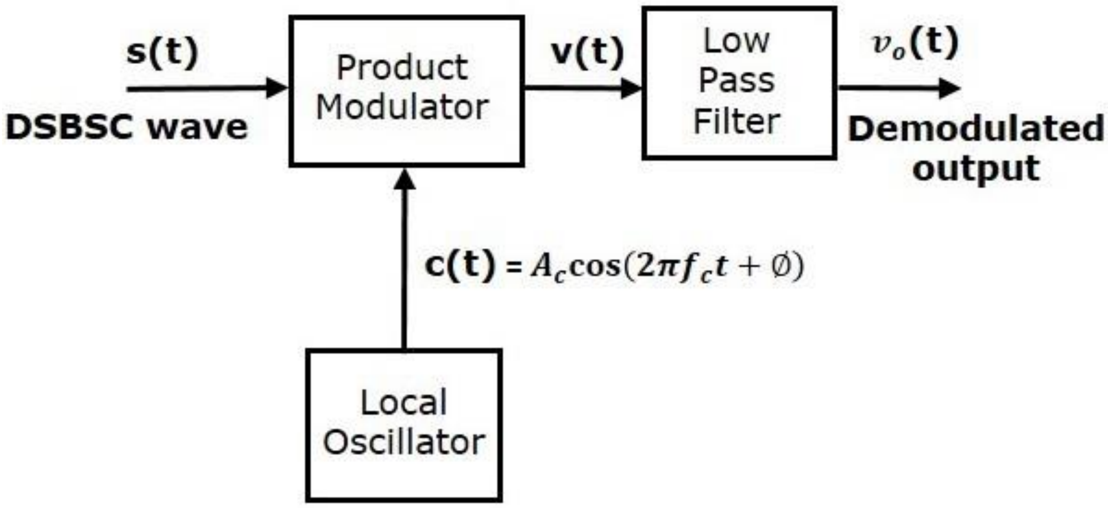
- ✓ Mathematically, we can represent the **equation of DSBSC wave** as the product of modulating and carrier signals.

$$s(t)=m(t)c(t)$$
$$\Rightarrow s(t)=A_mA_c\cos(2\pi f_m t)\cos(2\pi f_c t)$$

Demodulation of DSBSC

- ✓ The process of extracting an original message signal from DSBSC wave is known as detection or demodulation of DSBSC. The following demodulators (detectors) are used for demodulating DSBSC wave.

- Coherent Detector
- Costas Loop



- ✓ In this process, the message signal can be extracted from DSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in DSBSC modulation. The resulting signal is then passed through a Low Pass Filter. Output of this filter is the desired message signal.
- ✓ Let the DSBSC wave be

s(t)=Accos(2πfct)m(t)s(t)=Accos(2πfct)m(t)

- ✓ The output of the local oscillator is

c(t)=Accos(2πfct+ϕ)c(t)=Accos(2πfct+ϕ)

- ✓ Where, ϕϕ is the phase difference between the local oscillator signal and the carrier signal, which is used for DSBSC modulation.
- ✓ From the figure, we can write the output of product modulator as

v(t)=s(t)c(t)v(t)=s(t)c(t)

- ✓ Substitute, s(t)s(t) and c(t)c(t) values in the above equation.

⇒v(t)=Accos(2πfct)m(t)Accos(2πfct+ϕ)⇒v(t)=Accos(2πfct)m(t)Accos(2πfct+ϕ)

=Ac2cos(2πfct)cos(2πfct+ϕ)m(t)=Ac2cos(2πfct)cos(2πfct+ϕ)m(t)

=Ac22[cos(4πfct+ϕ)+cosϕ]m(t)=Ac22[cos(4πfct+ϕ)+cosϕ]m(t)

v(t)=Ac22cosϕm(t)+Ac22cos(4πfct+ϕ)m(t)v(t)=Ac22cosϕm(t)+Ac22cos(4πfct+ϕ)m(t)

- ✓ In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter.
- ✓ Therefore, the output of low pass filter is

v0t=Ac22cosϕm(t)

- ✓ The demodulated signal amplitude will be maximum, when ϕ=00ϕ=00. That's why the local oscillator signal and the carrier signal should be in phase, i.e., there should not be any phase difference between these two signals.

- ✓ The demodulated signal amplitude will be zero, when ϕ=±900ϕ=±900. This effect is called as **quadrature null effect**.

Video Content / Details of website for further learning (if any):

4. https://www.tutorialspoint.com/analog_communication/analog_communicat
5. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf
6. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf

Important Books/Journals for further learning including the page nos.:

3. "Principles of Communication", H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
4. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (136-138)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

Estd. 2000

LECTURE HANDOUTS

L - 3

ECE

II / III

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture:**

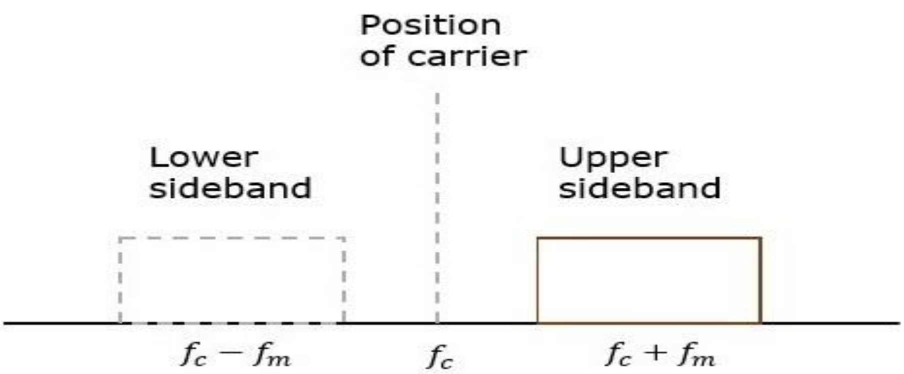
Topic of Lecture: SSB SC, VSB Signals

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

Generation of SSB SC:

- ✓ The process of suppressing one of the sidebands along with the carrier and transmitting a single sideband is called as **Single Sideband Suppressed Carrier** system or simply **SSBSC**. It is plotted as shown in the following figure



Carrier and a sideband are suppressed and a single sideband is allowed for transmission

Mathematical Representation

- ✓ Let us consider the same mathematical expressions for the modulating and the carrier signals as we have considered in the earlier chapters.
- ✓ i.e., Modulating signal

$$m(t)=Amcos(2\pi fmt)m(t)=Amcos(2\pi fmt)$$
$$c(t)=Accos(2\pi fct)c(t)=Accos(2\pi fct)$$

- ✓ Mathematically, we can represent the equation of SSBSC wave as s(t)=AmAc2cos[2π(fc+fm)t]s(t)=AmAc2cos[2π(fc+fm)t]for the upper sideband

Or

s(t)=AmAc2cos[2π(fc−fm)t]s(t)=AmAc2cos[2π(fc−fm)t]for the lower sideband

Demodulation of SSB SC

- ✓ In this process, the message signal can be extracted from SSBSC wave by multiplying it with a carrier, having the same frequency and the phase of the carrier used in SSBSC modulation. The resulting signal is then passed through a Low Pass Filter. The output of this filter is the desired message signal.
- ✓ Consider the following **SSBSC** wave having a **lower sideband**.

s(t)=AmAc2cos[2π(fc−fm)t]s(t)=AmAc2cos[2π(fc−fm)t]

- ✓ The output of the local oscillator is

c(t)=Accos(2πfct)c(t)=Accos(2πfct)

- ✓ From the figure, we can write the output of product modulator as

v(t)=s(t)c(t)v(t)=s(t)c(t)

- ✓ Substitute s(t)s(t) and c(t)c(t) values in the above equation.

v(t)=AmAc2cos[2π(fc−fm)t]Accos(2πfct)v(t)=AmAc2cos[2π(fc−fm)t]Accos(2πfct)
=AmAc22cos[2π(fc−fm)t]cos(2πfct)=AmAc22cos[2π(fc−fm)t]cos(2πfct)
=AmAc24{cos[2π(2fc−fm)]+cos(2πfm)t}=AmAc24{cos[2π(2fc−fm)]+cos(2πfm)t}
v(t)=AmAc24cos(2πfmt)+AmAc24cos[2π(2fc−fm)t]v(t)=AmAc24cos(2πfmt)+AmAc24cos[2π(2fc−fm)t]

- ✓ In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter.
- ✓ Therefore, the output of low pass filter is

v0(t)=AmAc24cos(2πfmt)v0(t)=AmAc24cos(2πfmt)

- ✓ Here, the scaling factor is Ac24Ac24.
- ✓ We can use the same block diagram for demodulating SSBSC wave having an upper sideband. Consider the following **SSBSC** wave having an **upper sideband**.

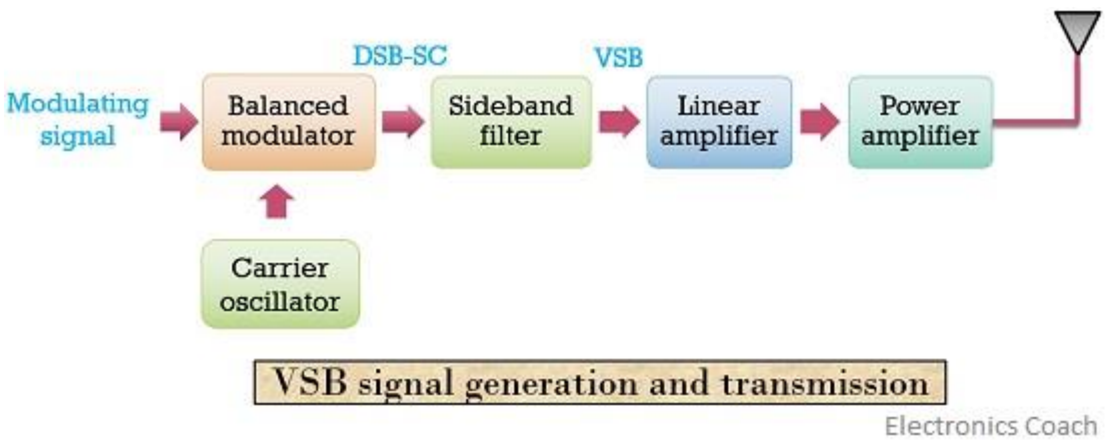
s(t)=AmAc2cos[2π(fc+fm)t]s(t)=AmAc2cos[2π(fc+fm)t]

- ✓ The output of the local oscillator is

c(t)=Accos(2πfct)c(t)=Accos(2πfct)

VSB Signal Generation and Demodulation

Vestigial Sideband (VSB) modulation is a modulation technique which allows **transmission of one sideband** in addition with **a part** or **vestige of the other**. It is basically a compromise between DSB-SC and SSB modulation.



Generation of VSB signal

Video Content / Details of website for further learning (if any):

7. https://web.sonoma.edu/esee/courses/ee442/lectures/sp2017/lect08_angle_mod.pdf

8. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf

9. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf

Important Books/Journals for further learning including the page nos.:

5. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (142-148)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 4

ECE

II / III

Course Name with Code : **16ITD12-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.Kirubakaran**

Unit I : **AMPLITUDE MODULATION** **Date of Lecture:**

Topic of Lecture: Filtering of sidebands
Introduction: Power of **AM** wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components. Where, vrms is the rms value of cos signal. vm is the peak value of cos signal.
Prerequisite knowledge for Complete understanding and learning of Topic: Voltage and Power distribution

Filtering of sidebands:

- ✓ When synthesis is performed using amplitude modulation or frequency modulation, the result, as seen in a spectrum analyzer or a Fourier analysis, is three distinct sets of frequencies: in the center is the carrier frequency (assuming that a sine wave was used for the carrier), with a set of "sidebands" produced by the modulation above the carrier frequency, and a mirror-image set of sidebands below the carrier frequency
- ✓ The "upper" and "lower" sidebands both contain the same information, but the lower sideband is reversed, from a frequency standpoint, from the upper sideband.
- ✓ A sideband filter removes the carrier and/or one set of sidebands. In theory, this can be done with combinations of lowpass, highpass and notch filters, where the lowpass or highpass is used to remove one set of sidebands, and the notch is used to remove the carrier.
- ✓ As a practical matter, it is difficult to get the filters to track properly, and so other circuits that implement other mathematical techniques are usually used. (The process bears some mathematical similarity to that performed by a frequency shifter).
- ✓ In music synthesis, the use of a sideband filter can be a useful technique to "clean up" modulations involving complex signals, particularly in the case of frequency modulation which, because of the large number of sideband frequencies it produces, can tend towards noise when an overtone-rich signal is modulated. It is also useful when modulating against a fixed-

frequency carrier to be able to remove the carrier to get the constant tone out of the output. Sideband filtering can produce some surprising effects; for instance, amplitude modulating a fairly high frequency sine wave with a complex signal such as human voice, and then using a sideband filer to remove all but the lower sidebands, results in the effect known as frequency inversion.

Video Content / Details of website for further learning (if any):

10. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG
11. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution
12. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

6. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (96-98)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 5

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : **16ITD12-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.Kirubakaran**

Unit I : **AMPLITUDE MODULATION** **Date of Lecture:**

Topic of Lecture: Comparison of Amplitude modulation systems

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

Comparison Of Amplitude Modulation Systems:

Continuous ambulatory electrocardiographic monitoring of ST-segment configuration has become a useful technique for evaluation of myocardial ischemia. Concern that direct or amplitude-modulated (AM) recording and playback systems have inherent limitations that cause inaccurate ST-segment recordings has led to preference for frequency-modulated (FM) devices. To determine the accuracy of AM and FM ambulatory electrocardiographic systems, the signal was compared from the same set of 2 bipolar leads simultaneously recorded by standard electrocardiography and AM and FM recorders in 14 patients during treadmill exercise. Also, simultaneous AM and FM recorders were compared in 9 ambulatory patients in 16 monitoring sessions. The AM recording system accurately reproduced ST segments recorded during treadmill exercise (range 4.0 mm of ST-segment depression to 2.0 mm of ST elevation) when measured at the J point ($r = 0.91$, $p < 0.0001$), and 0.08 second after the J point ($r = 0.95$, $p < 0.0001$). FM recording was equally accurate ($r = 0.89$ and 0.95, respectively, $p < 0.0001$). Similarly, during ambulatory recording, the AM technique accurately recorded maximal ST depression in each episode as recorded by the FM device (28 episodes, range 0 to 3 mm of ST depression, $r = 0.85$, $p < 0.0001$). Both AM and FM

ambulatory electrocardiographic systems can accurately reproduce ST-segment deviation associated with ischemia and can be used to monitor transient ST-segment changes in patients with coronary artery disease.
Video Content / Details of website for further learning (if any): 1. https://web.sonoma.edu/esee/courses/ee442/lectures/sp2017/lect08_angle_mod.pdf 2. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf 3. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf
Important Books/Journals for further learning including the page nos.: 7. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (52-58)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 6

LECTURE HANDOUTS

ECE **II / III**

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture:**

Topic of Lecture: Frequency Translation

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

Frequency Translation:

- ✓ Suppose we have a modulated wave s1(t) whose spectrum is centered around frequency f1 and we wish to move it upward in frequency, so that its spectrum is centered around f2.
- ✓ Frequency translation is the process of moving a signal from one part of the frequency axis, to another part of the axis.
- ✓ Frequency translation is often done in wireless communications systems to move a pass band signal to base band before demodulation. Complex multipliers can be used to perform frequency translation, but a more efficient method is to use decimation.

Application of Frequency Translation

- ✓ Applications of Frequency Translation There are two main applications for frequency translation in the context of the QF4A512 and QF1D512 parts.
- ✓ a) Moving the signal of interest closer to DC so that the 512 taps of the filter are more effective.
- ✓ b) Moving the signal of interest below the maximum operating frequency of the parts. The first application is moving the signal of interest closer to DC to more efficiently use the 512 taps available in the filter.
- ✓ In most applications, a 512 tap filter is more than enough to do the required processing, but occasionally a filter with a smaller transition band or more attenuation is needed. Since the filter properties, such as transition band width, are a function of both the number of taps and the sample rate, lowering the sample rate will increase the effectiveness of a given number of taps.
- ✓ Performing frequency translation using decimation lowers the sample rate and increases the effectiveness of the filter taps.

- ✓ The seconds application is decreasing the sample rate of a signal so that it is less than the maximum sample rate supported by the part. This process can be useful when using the QF1D512 part to process a signal with a sample rate over 500 kHz, for example.

Video Content / Details of website for further learning (if any):

13. https://web.sonoma.edu/esee/courses/ee442/lectures/sp2017/lect08_angle_mod.pdf

14. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf

15. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf

Important Books/Journals for further learning including the page nos.:

8. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (176-178)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 7

ECE

II / III

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture :**

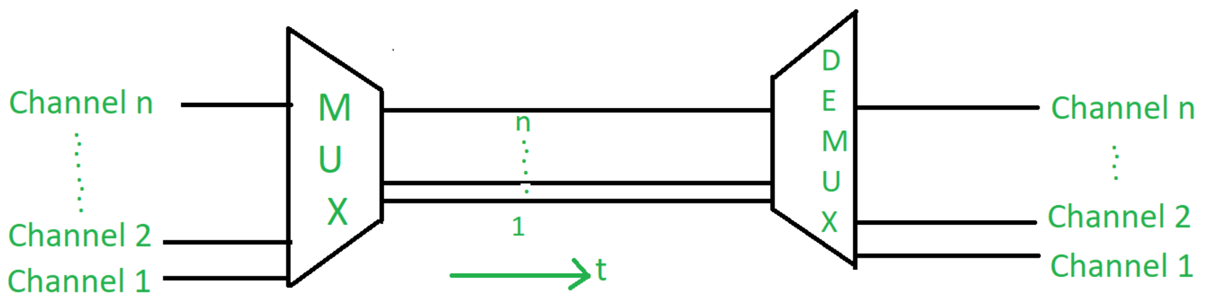
Topic of Lecture: Frequency Division Multiplexing

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

Multiplexing:

- ✓ **Multiplexing** is used in the cases where the signals of lower bandwidth and the transmitting media is having higher bandwidth.
- ✓ In this case, the possibility of sending a number of signals is more.
- ✓ In this the signals are combined into one and are sent over a link which has greater bandwidth of media than the communicating nodes.
- ✓ **Frequency Division Multiplexing (FDM) –**
  In this a number of signals are transmitted at the same time, and each source transfers its signals in the allotted frequency range.
- ✓ There is a suitable frequency gap between the 2 adjacent signals to avoid over-lapping. Since the signals are transmitted in allotted time so this decreases the probability of collision
- ✓ . The frequency spectrum is divided into several logical channels, in which every user feels that they posses a particular bandwidth.
- ✓ A number of signals are sent simultaneously on the same time allocating separate frequency band or channel to each signal.
- ✓ It is used in radio and TV transmission.
- ✓ Therefore to avoid interference between two successive channels **Guard bands** are used.



Video Content / Details of website for further learning (if any): 16. https://web.sonoma.edu/esee/courses/ee442/lectures/sp2017/lect08_angle_mod.pdf 17. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf 18. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf
Important Books/Journals for further learning including the page nos.: 9. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (182-184)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 8

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture:**

Topic of Lecture: AM Transmitter, Super heterodyne receiver

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

Basic Super heterodyne receiver theory:

- ✓ The superheterodyne receiver operates by taking the signal on the incoming frequency, mixing it with a variable frequency locally generated signal to convert it down to a frequency where it can pass through a high performance fixed frequency filter before being demodulated to extract the required modulation or signal.
- ✓ It is obviously necessary to look ay this in more detail to understand the principle behind what goes on, but the main process in the superheterodyne radio is that of mixing.

Working

- ✓ In order to look at how a superhet or superheterodyne radio works and the RF circuit design, it is necessary to follow the signal through it.
- ✓ In this way the processes it undergoes can be viewed more closely.
- ✓ The signal that is picked up by the antenna passes into the receiver and enters a mixer.
- ✓ There are three signal ports on the mixer: signal, local oscillator and IF.
- ✓ The signal is obviously applied to the signal port which is designed to accept lower level signals than the LO port.
- ✓ Another locally generated signal, often called the local oscillator, or LO, is fed into the other port

on the mixer and the two signals are mixed.

- ✓ The mixer action is to multiply the instantaneous levels of the two signals together. The non-linear action of the mixer generates signals at frequencies equal to the sum and difference of the incoming signals.

- ✓ Many mixers are what is termed balanced, and this means that the two incoming signals are not present, or at least greatly reduced at the output.

- ✓ The output from the mixer is passed into what is termed the intermediate frequency or IF stages where the signal is amplified and filtered.

- ✓ Any of the converted signals that fall within the passband of the IF filter will be able to pass through the filter and they will also be amplified by the amplifier stages

- ✓ . Any signals that fall outside the passband of the filter will be rejected.

- ✓ Tuning the receiver is simply accomplished by changing the frequency of the local oscillator.

- ✓ This changes the incoming signal frequency for which signals are be converted down and able to pass through the filter.

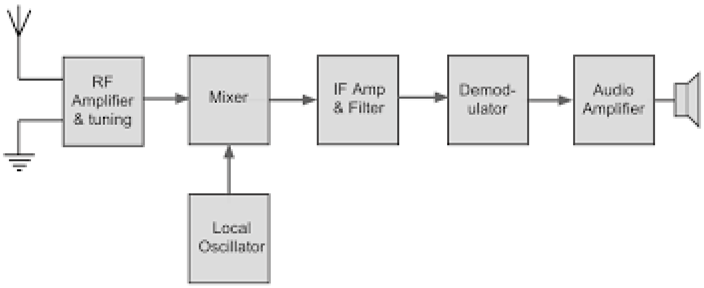
- ✓ It is often helpful to look at a real example to illustrate how the process works.

- ✓ To see how this operates in reality take the example of two signals, one at 1.0 MHz and another at 1.1 MHz.

- ✓ If the IF filter is centred at 0.25 MHz, and the local oscillator is set to 0.75 MHz, then the two signals generated by the mixer as a result of the 1.0 MHz signal fall at 0.25 MHz and 1.75 MHz.

- ✓ Naturally the 1.75 MHz signal is rejected, but the one at 0.25 MHz passes through the IF stages.

- ✓ The signal at 1.1 MHz produces a signal at 0.35 MHz and another at 1.85 MHz. Both of these fall outside bandwidth of the IF filter so the only signal to pass through the IF is that from the signal on 1.0 MHz..



Video Content / Details of website for further learning (if any):

19. https://web.sonoma.edu/esee/courses/ee442/lectures/sp2017/lect08_angle_mod.pdf

20. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf

21. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf

Important Books/Journals for further learning including the page nos.:

10. "Principles of Communication", H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
    (232-234)

Course Faculty

Verified by HOD

Course Name with Code : 16ITD12-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.Kirubakaran

Unit I : AMPLITUDE MODULATION **Date of Lecture:**

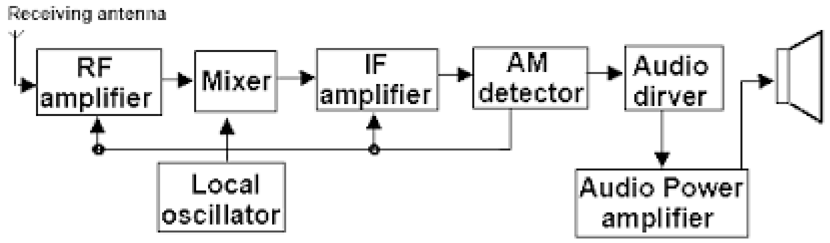
Topic of Lecture: AM Reciever

Introduction: The **frequency** of the **modulated wave** remains constant as the carrier **wave frequency** when the message signal is at zero. The **frequency** increases when the message signal reaches its maximum amplitude.

Prerequisite knowledge for Complete understanding and learning of Topic:
Frequency analysis, mathematical expression, equation

AM Receiver:

- ✓ **AM broadcasting** is a radio broadcasting technology, which employs amplitude modulation (AM) transmissions.
- ✓ It was the first method developed for making audio radio transmissions, and is still used worldwide, primarily for medium wave (also known as "AM band") transmissions, but also on the longwave and shortwave radio bands.
- ✓ The earliest experimental AM transmissions began in the early 1900s.
- ✓ However, widespread AM broadcasting was not established until the 1920s, following the development of vacuum tube receivers and transmitters.
- ✓ AM radio remained the dominant method of broadcasting for the next 30 years, a period called the "Golden Age of Radio", until television broadcasting became widespread in the 1950s and received most of the programming previously carried by radio.
- ✓ Subsequently, AM radio's audiences have also greatly shrunk due to competition from FM (frequency modulation) radio, Digital Audio Broadcasting (DAB), satellite radio, HD (digital) radio and Internet streaming.
- ✓ AM transmissions are much more susceptible than FM or digital signals are to interference, and often have lower audio fidelity.
- ✓ Thus, AM broadcasters tend to specialise in spoken-word formats, such as talk radio, all news and sports, leaving the broadcasting of music mainly to FM and digital stations.



Video Content / Details of website for further learning (if any):

22. https://web.sonoma.edu/esee/courses/ee442/lectures/sp2017/lect08_angle_mod.pdf

23. https://web.sonoma.edu/esee/courses/ee442/archives/sp2019/lectures/lecture07_angle_mod.pdf

24. http://163.152.6.72/contents/vod/21/20121129093533/20121129093533.pdf

Important Books/Journals for further learning including the page nos.:

11. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (116-118)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 10

LECTURE HANDOUTS

ECE | **II / III**

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 2 : ANGLE MODULATION **Date of Lecture:**

Topic of Lecture: Angle modulation , Frequency modulation

Introduction: Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components. Where, vrms is the rms value of cos signal. vm is the peak value of cos signal.

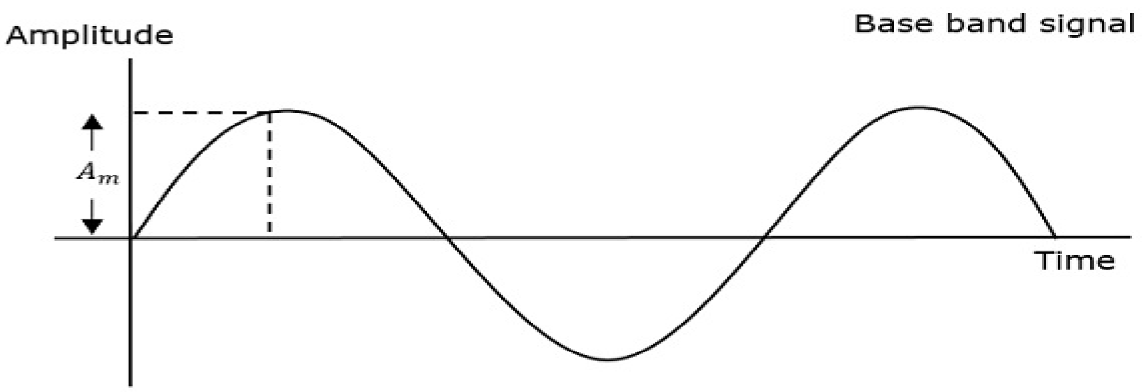
Prerequisite knowledge for Complete understanding and learning of Topic:
Voltage and Power distribution

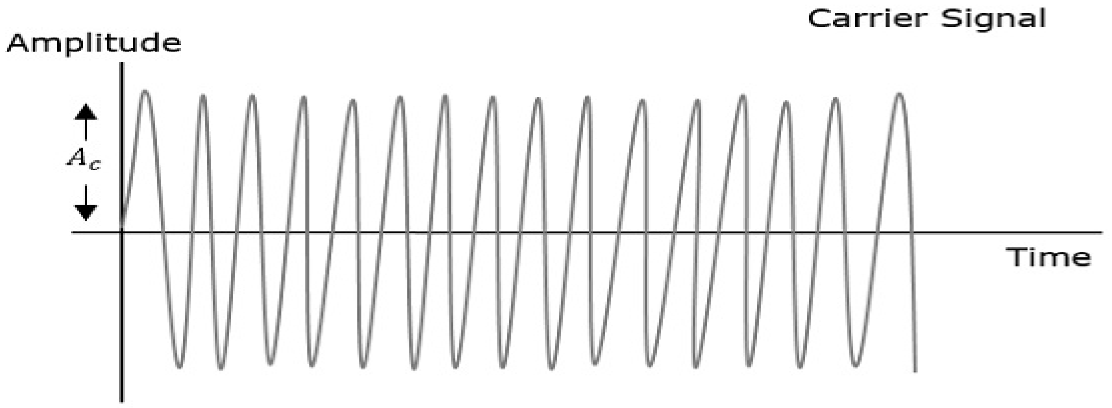
Angle Modulation:

- ✓ The other type of modulation in continuous-wave modulation is the **Angle Modulation**
- ✓ Angle Modulation is the process in which the frequency or the phase of the carrier varies according to the message signal.
- ✓ This is further divided into frequency and phase modulation.
- ✓ Frequency Modulation is the process of varying the frequency of the carrier signal linearly with the message signal.
- ✓ Phase Modulation is the process of varying the phase of the carrier signal linearly with the message signal.

Frequency modulation

- ✓ In amplitude modulation, the amplitude of the carrier varies.
- ✓ But in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- ✓ The amplitude and the phase of the carrier signal remains constant whereas the frequency of the carrier changes.





- ✓ The frequency of the modulated wave remains constant as the carrier wave frequency when the message signal is at zero.
- ✓ The frequency increases when the message signal reaches its maximum amplitude.
- ✓ Which means, with the increase in amplitude of the modulating or message signal, the carrier frequency increases.
- ✓ Likewise, with the decrease in the amplitude of the modulating signal, the frequency also decreases

Mathematical representation

- ✓ Let the carrier frequency be $\mathbf{f_c}$
- ✓ The frequency at maximum amplitude of the message signal = $\mathbf{f_c + \Delta f}$
- ✓ The frequency at minimum amplitude of the message signal = $\mathbf{f_c - \Delta f}$
- ✓ The difference between FM modulated frequency and normal frequency is termed as **Frequency Deviation** and is denoted by **Δf**.
- ✓ The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the **Carrier Swing**.
- ✓ Carrier Swing = 2 × frequency deviation
- ✓ = $\mathbf{2 \times \Delta f}$
- ✓ Equation for FM WAVE
- ✓ The equation for FM wave is −

$$s(t) = A_ccos[W_ct + 2\pi k_fm(t)]$$

Where,

$\mathbf{A_c}$ = the amplitude of the carrier

$\mathbf{w_c}$ = angular frequency of the carrier = $\mathbf{2\pi f_c}$

m(t) = message signal

FM can be divided into **Narrowband FM** and **Wideband FM**.

Video Content / Details of website for further learning (if any):

1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG
2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution
3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (122-124)

Course Faculty

Verified by HOD

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 2 : ANGLE MODULATION **Date of Lecture:**

Topic of Lecture: Narrowband and Wideband FM, Transmission bandwidth of FM Signal

Introduction: Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components. Where, VRMS is the RMS value of COS signal. Vm is the peak value of COS signal.

Prerequisite knowledge for Complete understanding and learning of Topic:
Voltage and Power distribution

Narrowband FM:

- ✓ The features of Narrowband FM are as follows
- ✓ This frequency modulation has a small bandwidth.
- ✓ The modulation index is small
- ✓ Its spectrum consists of carrier, USB, and LSB.
- ✓ This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Wideband FM

- ✓ The features of Wideband FM are as follows –
- ✓ This frequency modulation has infinite bandwidth.
- ✓ The modulation index is large, i.e., higher than **1**.
- ✓ Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
- ✓ This is used in entertainment broadcasting applications such as FM radio, TV, etc.

Transmission bandwidth of FM signals

- ✓ Commercial **FM signals** use a peak frequency deviation of Δf = 75 kHz and a maximum baseband message frequency of **fm** = 15 kHz.
- ✓ Carson's rule estimates the **FM signal bandwidth** as BT = 2(75 + 15) = 180 kHz which is six times the 30 kHz **bandwidth** that would be required for AM **modulation**.

Video Content / Details of website for further learning (if any):
1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG 2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution 3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php
Important Books/Journals for further learning including the page nos.: 1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (96-98)

Course Faculty

Verified by HOD

LECTURE HANDOUTS

L - 12

ECE

II / III

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 2 : **ANGLE MODULATION** **Date of Lecture:**

Topic of Lecture: Generation of FM Signals Direct – indirect FM

Introduction: Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components. Where, vrms is the rms value of cos signal. vm is the peak value of cos signal.

Prerequisite knowledge for Complete understanding and learning of Topic:
Voltage and Power distribution

FM Signal generation

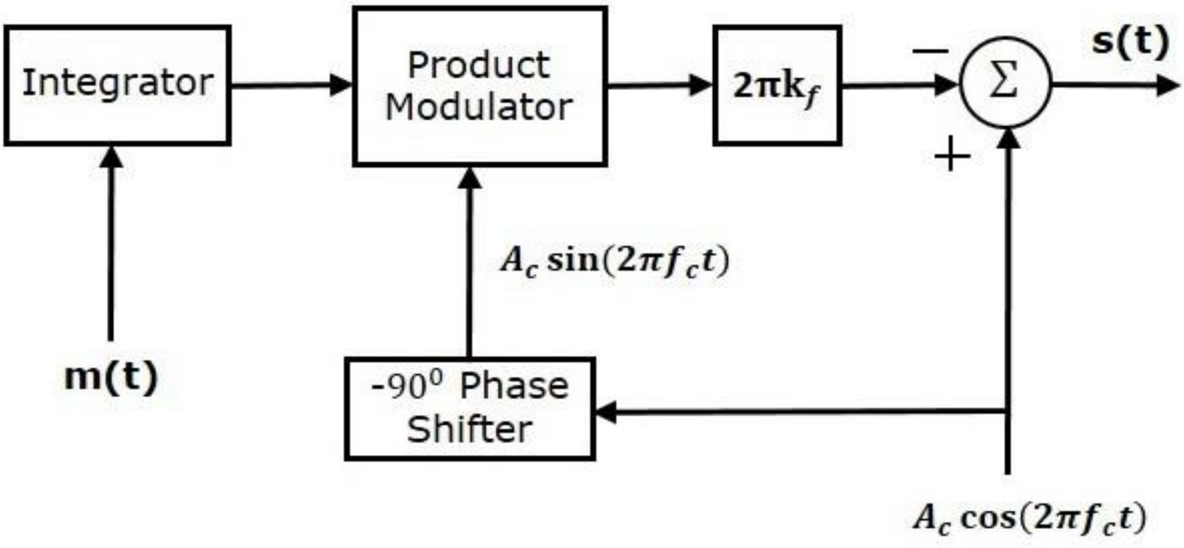
- ✓ FM signals can be generated using either direct or indirect frequency modulation:
- ✓ Direct FM modulation can be achieved by directly feeding the message into the input of a voltage-controlled oscillator.
- ✓ For indirect FM modulation, the message signal is integrated to generate a phase-modulated signal.

Mathematical Representation

- ✓ We know that the standard equation of FM wave is
- ✓ $s(t)=A_c\cos(2\pi f_c t+2\pi k_f\int m(t)dt)s(t)=A_c\cos(2\pi f_c t+2\pi k_f\int m(t)dt)$
- ✓ $\Rightarrow s(t)=A_c\cos(2\pi f_c t)\cos(2\pi k_f\int m(t)dt)-\Rightarrow s(t)=A_c\cos(2\pi f_c t)\cos(2\pi k_f\int m(t)dt)-$
- ✓ $A_c\sin(2\pi f_c t)\sin(2\pi k_f\int m(t)dt)A_c\sin(2\pi f_c t)\sin(2\pi k_f\int m(t)dt)$
- ✓ For NBFM,

$$2\pi k_f\int m(t)dt|||<<1|2\pi k_f\int m(t)dt|<<1$$

- ✓ We know that $\cos\theta\approx 1\cos\theta\approx 1$ and $\sin\theta\approx 1\sin\theta\approx 1$ when $\theta\theta$ is very small.
- ✓ By using the above relations, we will get the **NBFM equation**
- ✓ $s(t)=A_c\cos(2\pi f_c t)-A_c\sin(2\pi f_c t)2\pi k_f\int m(t)dt$



Video Content / Details of website for further learning (if any):

1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG
2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution
3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (196-198)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

Estd. 2000

LECTURE HANDOUTS

L - 13

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 2 : ANGLE MODULATION **Date of Lecture:**

Topic of Lecture: Demodulation of FM signals , FM sterio multiplexing

Introduction: Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components. Where, vrms is the rms value of cos signal. vm is the peak value of cos signal.

Prerequisite knowledge for Complete understanding and learning of Topic:
Voltage and Power distribution

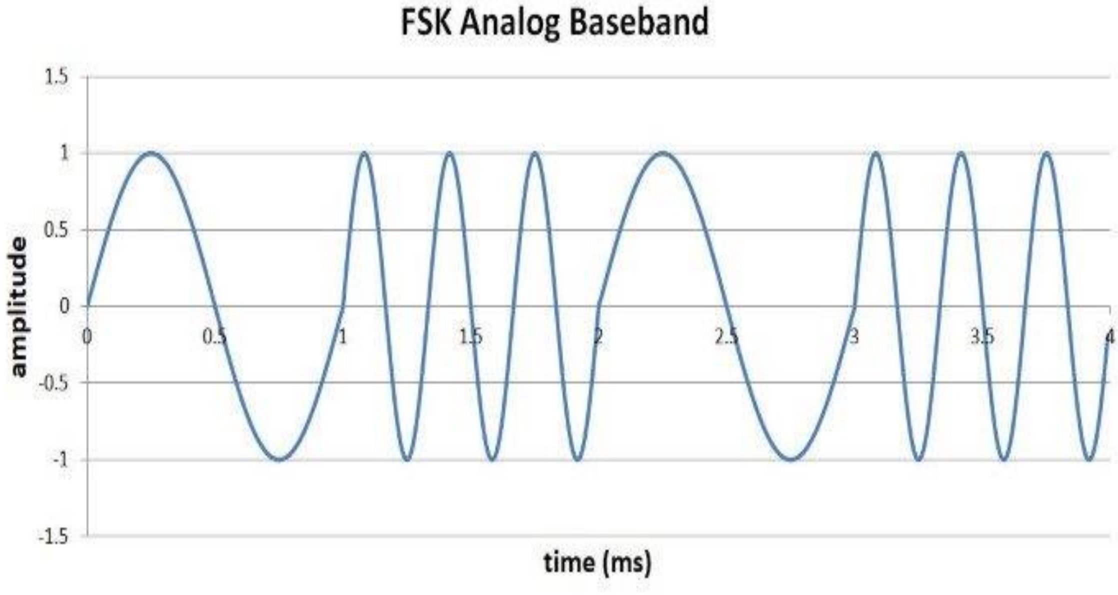
Demodulation of FM signals , FM sterio multiplexing:

Digital Frequency modulation:

- ✓ This type of modulation is called frequency shift keying (FSK). For our purposes it is not necessary to consider a mathematical expression of FSK; rather, we can simply specify that we will have frequency f_1 when the baseband data is logic 0 and frequency f_2 when the baseband data is logic 1.

Time domain:

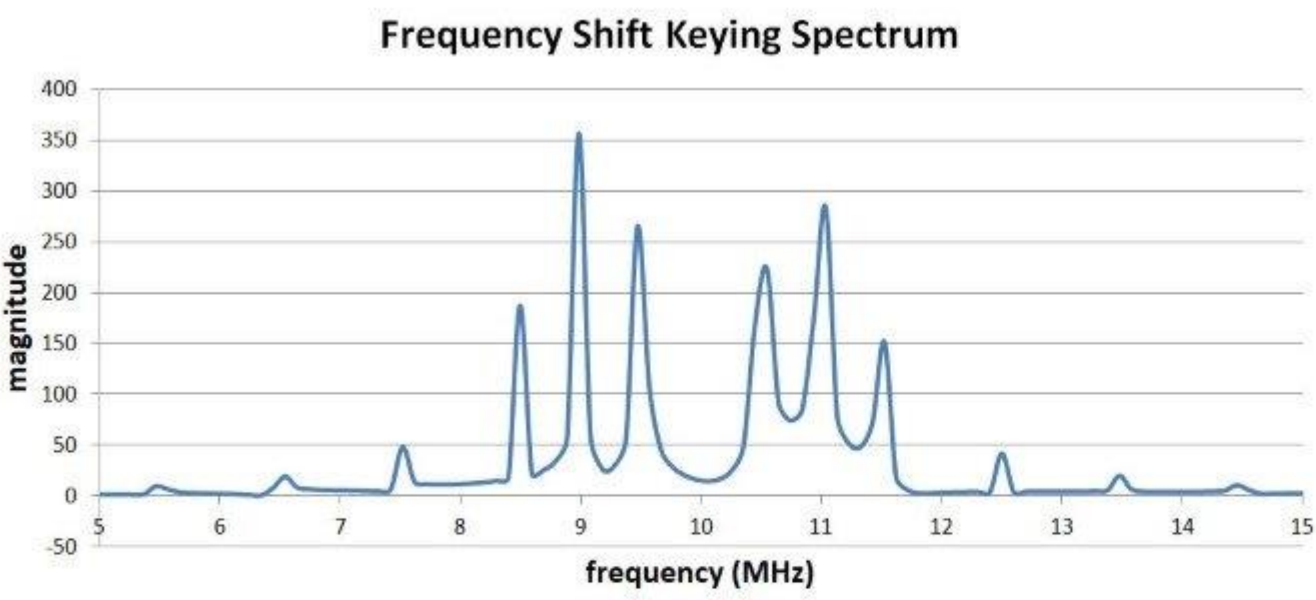
- ✓ One method of generating the ready-for-transmission FSK waveform is to first create an analog baseband signal that switches between f_1 and f_2 according to the digital data.
- ✓ Here is an example of an FSK baseband waveform with f_1 = 1 kHz and f_2 = 3 kHz.
- ✓ To ensure that a symbol is the same duration for logic 0 and logic 1, we use one 1 kHz cycle and three 3 kHz cycles.



- ✓ The baseband waveform is then shifted (using a mixer) up to the carrier frequency and transmitted.
- ✓ This approach is particularly handy in software-defined-radio systems: the analog baseband waveform is a low-frequency signal, and thus it can be generated mathematically then introduced into the analog realm by a DAC.
- ✓ Using a DAC to create the high-frequency transmitted signal would be much more difficult.
- ✓ A more conceptually straightforward way to implement FSK is to simply have two carrier signals with different frequencies (f_1 and f_2); one or the other is routed to the output depending on the logic level of the binary data.
- ✓ This results in a final transmitted waveform that switches abruptly between two frequencies, much like the baseband FSK waveform above except that the difference between the two frequencies is much smaller in relation to the average frequency.
- ✓ In other words, if you were looking at a time-domain plot, it would be difficult to visually differentiate the f_1 sections from the f_2 sections because the difference between f_1 and f_2 is only a tiny fraction of f_1 (or f_2).

Frequency domain:

- ✓ Let's look at the effects of FSK in the frequency domain. We'll use our same 10 MHz carrier frequency (or average carrier frequency in this case), and we'll use ±1 MHz as the deviation. (This is unrealistic, but convenient for our current purposes.)
- ✓ So the transmitted signal will be 9 MHz for logic 0 and 11 MHz for logic 1. Here is the spectrum:



- ✓ Note that there is no energy at the "carrier frequency." This is not surprising, considering that the modulated signal is never at 10 MHz.
- ✓ It is always at either 10 MHz minus 1 MHz or 10 MHz plus 1 MHz, and this is precisely where we see the two dominant spikes: 9 MHz and 11 MHz.
- ✓ But what about the other frequencies present in this spectrum? Well, FSK spectral analysis is not particularly straightforward.
- ✓ We know that there will be additional Fourier energy associated with the abrupt transitions between frequencies.
- ✓ It turns out that FSK results in a sinc-function type of spectrum for each frequency, i.e., one is centered on f_1 and the other is centered on f_2.
- ✓ These account for the additional frequency spikes seen on either side of the two dominant spikes.

Video Content / Details of website for further learning (if any):

4. https://www.youtube.com/watch?v=rrgon8Qne_E

5. https://www.youtube.com/watch?v=ogJB5fiQ9kM

6. https://www.youtube.com/watch?v=eJ5m0Sbr2qw

Important Books/Journals for further learning including the page nos.:

2. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (126-128)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 14

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 2 : **ANGLE MODULATION** **Date of Lecture:**

Topic of Lecture: PLL Nonlinear model and linear model of PLL , Non linear effects in FM Systems

Introduction: A phase-locked loop or phase lock loop (PLL) is a control system that generates an output signal whose phase is related to the phase of an input signal.

Prerequisite knowledge for Complete understanding and learning of Topic:

Voltage and Power distribution

Linear and Non Linear PLL , Non linear effects in FM System:

Analog or linear PLL (APLL):

Phase detector is an analog multiplier. Loop filter is active or passive. Uses a voltage-controlled oscillator (VCO). APLL is said to be a type II if its loop filter has transfer function with exactly one pole at the origin (see also Egan's conjecture on the pull-in range of type II APLL).

Digital PLL (DPLL):

An analog PLL with a digital phase detector (such as XOR, edge-trigger JK, phase frequency detector). May have digital divider in the loop.

All digital PLL (ADPLL):

Phase detector, filter and oscillator are digital. Uses a numerically controlled oscillator (NCO).

Software PLL (SPLL):

Functional blocks are implemented by software rather than specialized hardware.

Neuronal PLL (NPLL):

Phase detector, filter and oscillator are neurons or small neuronal pools. Uses a rate controlled oscillator (RCO). Used for tracking and decoding low frequency modulations (< 1 kHz), such as those occurring during mammalian-like active sensing.

Charge-pump PLL (CP-PLL):

CP-PLL is a modification of phase-locked loops with phase-frequency detector and square waveform signals. See also Gardner's conjecture on CP-PLL.

Video Content / Details of website for further learning (if any): 1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG 2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution 3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php
Important Books/Journals for further learning including the page nos.: 1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (226-228)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 15

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 2 : **ANGLE MODULATION** **Date of Lecture:**

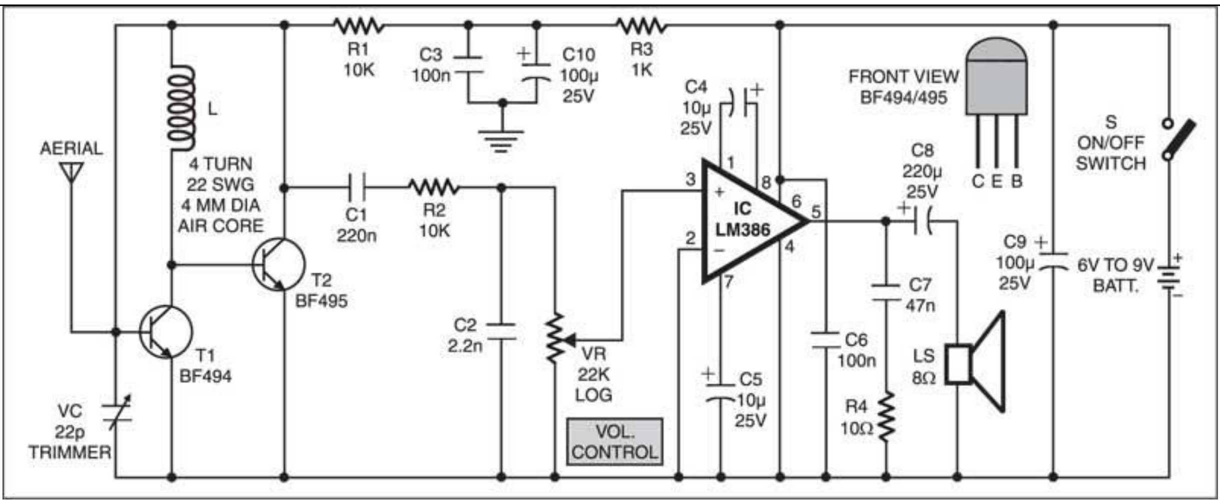
Topic of Lecture: FM Broadcast receivers , FM Stereo receivers

Introduction: A radio receiver, also known as a receiver, a wireless or simply a radio, is an electronic device that receives radio waves and converts the information carried by them to a usable form. It is used with an antenna.

Prerequisite knowledge for Complete understanding and learning of Topic:
Voltage and Power distribution

FM Broadcast Receiver:

- ✓ Frequency modulation is used in a radio broadcast in the 88-108MHz VHF band.
- ✓ This bandwidth range is marked as FM on the band scales of radio receivers, and the devices that are able to receive such signals are called FM receivers.
- ✓ The FM radio transmitter has a 200kHz wide channel.
- ✓ The maximum audio frequency transmitted in FM is 15 kHz as compared to 4.5 kHz in AM. This allows a much larger range of frequencies to be transferred in FM and thus the quality of FM transmission is significantly higher than of AM transmission. Presented below is an electronics circuit for FM receiver along with its full explanation



FM Stereo Receiver

- ✓ With the digital age, **Radio** Data System (RDS) enables **FM** to carry text information such as traffic, weather, and **radio** station information which can be displayed on the end-user's device interface.

Video Content / Details of website for further learning (if any):

1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG
2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution
3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (246-248)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 16

ECE

II / III

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 2 : **ANGLE MODULATION** **Date of Lecture:**

Topic of Lecture: Pre-emphasis in FM

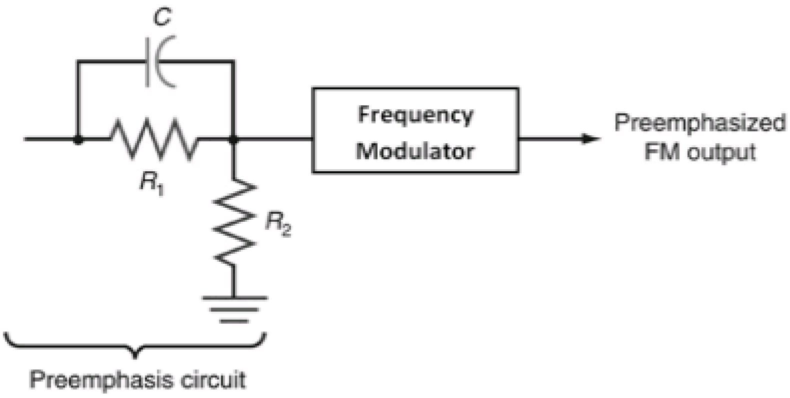
Introduction: Prior to some process, such as transmission over cable, or recording to phonograph record or tape, the input frequency range most susceptible to noise is boosted. This is referred to as "pre-emphasis" – before the process the signal will undergo. Later, when the signal is received, or retrieved from recording, the reverse transformation is applied ("de-emphasis") so that the output accurately reproduces the original input. Any noise added by transmission or record/playback, to the frequency range previously boosted, is now attenuated in the de-emphasis stage.

Prerequisite knowledge for Complete understanding and learning of Topic:

Pre-emphasis in FM

Pre-emphasis in FM

- ✓ **Pre-emphasis:** The noise suppression ability of FM decreases with the increase in the frequencies.
- ✓ Thus increasing the relative strength or amplitude of the high frequency components of the message signal before modulation is termed as Pre-emphasis.



- ✓ The pre-emphasis process is done at the transmitter side, while the de-emphasis process is done at the receiver side.
- ✓ Thus a high frequency modulating signal is emphasized or boosted in amplitude in transmitter before modulation.

- ✓ To compensate for this boost, the high frequencies are attenuated or de-emphasized in the receiver after the demodulation has been performed.

- ✓ Due to pre-emphasis and de-emphasis, the S/N ratio at the output of receiver is maintained constant.

- ✓ The de-emphasis process ensures that the high frequencies are returned to their original relative level before amplification.
- ✓ Pre-emphasis circuit is a high pass filter or differentiator which allows high frequencies to pass, whereas de-emphasis circuit is a low pass filter or integrator which allows only low frequencies to pass.

Video Content / Details of website for further learning (if any):

1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG

2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution

3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (296-298)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 17

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 2 : ANGLE MODULATION **Date of Lecture:**

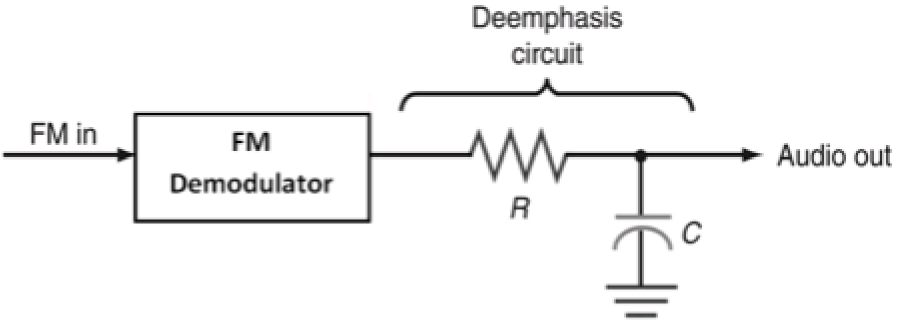
Topic of Lecture: De emphasis on FM

Introduction: De-emphasis means attenuating those frequencies by the amount by which they are boosted. However pre-emphasis is done at the transmitter and the de-emphasis is done in the receiver. The purpose is to improve the signal-to-noise ratio for FM reception.

Prerequisite knowledge for Complete understanding and learning of Topic:
Voltage and Power distribution

De emphasis on FM:

- ✓ In the de-emphasis circuit, by reducing the amplitude level of the received high frequency signal by the same amount as the increase in pre-emphasis is termed as De-emphasis



- ✓ The pre-emphasis process is done at the transmitter side, while the de-emphasis process is done at the receiver side.
- ✓ Thus a high frequency modulating signal is emphasized or boosted in amplitude in transmitter before modulation.
- ✓ To compensate for this boost, the high frequencies are attenuated or de-emphasized in the receiver after the demodulation has been performed. Due to pre-emphasis and de-emphasis, the S/N ratio at the output of receiver is maintained constant.
- ✓ The de-emphasis process ensures that the high frequencies are returned to their original relative level before amplification.
- ✓ Pre-emphasis circuit is a high pass filter or differentiator which allows high frequencies to pass, whereas de-emphasis circuit is a low pass filter or integrator which allows only low frequencies to pass.

Video Content / Details of website for further learning (if any):

1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG
2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution
3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (196-198)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 18

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 2 : ANGLE MODULATION Date of Lecture:

Topic of Lecture: Comparison of performance of AM and FM Systems

Introduction: Modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal, with a separate signal that typically contains information to be transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Comparison of performance of AM and FM Systems

Comparison of performance of AM and FM Systems:
AM OR FM :

- ✓ AM (or Amplitude Modulation) and FM (or Frequency Modulation) are ways of broadcasting radio signals.
- ✓ Both transmit the information in the form of electromagnetic waves.
- ✓ AM works by modulating (varying) the amplitude of the signal or carrier transmitted according to the information being sent, while the frequency remains constant.
- ✓ This differs from FM technology in which information (sound) is encoded by varying the frequency of the wave and the amplitude is kept constant.

PROS AND CONS AM VS FN:

- ✓ The advantages of AM radio are that it is relatively easy to detect with simple equipment, even if the signal is not very strong.
- ✓ The other advantage is that it has a narrower bandwidth than FM, and wider coverage compared with FM radio.
- ✓ The major disadvantage of AM is that the signal is affected by electrical storms and other radio frequency interference.
- ✓ Also, although the radio transmitters can transmit sound waves of frequency up to 15 kHz, most receivers are able to reproduce frequencies only up to 5kHz or less
- ✓ . Wideband FM was invented to specifically overcome the interference disadvantage of AM radio.
- ✓ A distinct advantage that FM has over AM is that FM radio has better sound quality than AM radio.
- ✓ The disadvantage of FM signal is that it is more local and cannot be transmitted over

long distance.

- ✓ Thus, it may take more FM radio stations to cover a large area. Moreover, the presence of tall buildings or land masses may limit the coverage and quality of FM.
- ✓ Thirdly, FM requires a fairly more complicated receiver and transmitter than an AM signal does

Video Content / Details of website for further learning (if any):

1. https://unacademy.com/lesson/power-distribution-in-am-signal/MUUG39VG
2. https://www.scribd.com/doc/63855216/Amplitude-Modulation-Power-Distribution
3. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am-theory-equations-formulas.php

Important Books/Journals for further learning including the page nos.:

1. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (226-227)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 19

LECTURE HANDOUTS

ECE | **II / III**

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 3 : **PULSE MODULATION** **Date of Lecture:**

Topic of Lecture: Time Division Multiplexing

Introduction: Amplitude modulation (AM) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

Time Division Multiplexing:

- ✓ Time division multiplexing is a technique where several optical signals are combined, transmitted together, and separated again based on different arrival times.
- ✓ In an optical fiber communication system, interleaving pulse trains can carry different data channels in a single fiber [1, 3].
- ✓ The use of multiple channels allows increased overall data transmission capacities without increasing the data rates of the single channels, or transmission of data of different users simultaneously.
- ✓ However, the time slot per bit must be reduced.
- ✓ Even if the bandwidth of the data modulator is limited, this can be done by using a train of ultra-short pulses (rather than a continuous optical wave) as the input of the modulator.
- ✓ Schematic of optical time division multiplexing.
- ✓ Two interleaving pulse sequences are combined to a single pulse train.
- ✓ In a communications system, each pulse may represent a “1” bit (if present) or a “0” (if suppressed).
- ✓ Special requirements of data transmitters for optical time division multiplexing are a short pulse duration and a low timing jitter.
- ✓ Also, the extinction ratio should be high, i.e. each combined channel should exhibit a very low power level between the bit slots, because such a background could otherwise interfere with other channels.
- ✓ For combining the signals, one typically requires some kind of optical delay lines.
- ✓ An alternative to time division multiplexing is wavelength division multiplexing, where the channels are distinguished by wavelength rather than by arrival time.

✓ In the context of distributed fiber-optic sensors [2], optical time division multiplexing means that signals are assigned to certain locations in the sensor via their arrival times. Such systems usually operate with ultra-short pulses.
Video Content / Details of website for further learning (if any): 1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php 2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/ 3. https://byjus.com/jee/amplitude-modulation/
Important Books/Journals for further learning including the page nos.: 1. "Principles of Communication", H.Taub, D L Schilling, G Saha, Pearson Education, 2008. 2. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 20

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION Date of Lecture:

Topic of Lecture: Types of Pulse Modulation

Introduction: **Amplitude modulation** (**AM**) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

Pulse Modulation

- ✓ Pulse-amplitude modulation (PAM), is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses.
- ✓ It is an analog pulse modulation scheme in which the amplitudes of a train of carrier pulses are varied according to the sample value of the message signal.
- ✓ Demodulation is performed by detecting the amplitude level of the carrier at every single period
- ✓ Types of Pulse modulation
- ✓ Lets us look at some of the different types of pulse modulation.
- ✓ Pulse Amplitude Modulation (PAM)
- ✓ It is the simplest form of Pulse Modulation. In this type of modulation, each sample is made proportional to the amplitude of the signal at the instant of sampling. The PAM signal follows the amplitude of the original signal, as the signal traces out the path of the whole wave. Here a signal which is sampled at Nyquist rate can be reconstructed by passing it through an efficient Low Pass Filter (LPF) with exact cutoff frequency. It is very easy to generate and demodulate PAM. This technique transmits the data by encoding in the amplitude of a series of signal pulses.
- ✓ There are two types of PAM.
- ✓ Single Polarity PAM: A fixed DC level is added to the signal so that the signal is always

positive.

- ✓ Double Polarity PAM: Here the pulses are both positive and negative.
- ✓ PAM is illustrated in the figure below.
- ✓ Pulse Amplitude Modulation (PAM)
- ✓ From the figure, it is clear that the pulse amplitude modulated signal is following the amplitude of the message signal.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

LECTURE HANDOUTS

L - 21

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION Date of Lecture:

Topic of Lecture: PAM single polarity

Introduction: Amplitude modulation (AM) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

PULSE AMPLITUDE MODULATION

- ✓ Pulse-amplitude modulation (PAM), is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses.
- ✓ It is an analog pulse modulation scheme in which the amplitudes of a train of carrier pulses are varied according to the sample value of the message signal.
- ✓ Demodulation is performed by detecting the amplitude level of the carrier at every single period.

ADVANTAGES OF PAM

- ✓ Advantages of PAM
- ✓ Both Modulation and demodulation are simple.
- ✓ Easy construction of transmitter and receiver circuits.
- ✓ Disadvantages of PAM
- ✓ Large bandwidth is required for transmission.
- ✓ More noise.
- ✓ Here the amplitude is varying. Therefore, the power required will be more.
- ✓ Applications of PAM
- ✓ Mainly used in Ethernet communication.
- ✓ Many microcontrollers use this technique in order to generate control signals.

- ✓ It is used in Photo-biology.
- ✓ It acts as an electronic driver for LED circuits.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 22

LECTURE HANDOUTS

ECE **II / III**

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION Date of Lecture:

Topic of Lecture: PAM double polarity

Introduction: **Amplitude modulation** (**AM**) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

GENERATION OF PWM

- ✓ PWM signal can be generated by using a comparator, where modulating signal and sawtooth signal form the input of the comparator. It is the simplest method for PWM generation.
- ✓ Considering both ±ve sides, the maximum of the input signal should be less than that of sawtooth signal.
- ✓ The comparator will compare the two signals together to generate the PWM signal at its output as shown in the third waveform
- ✓ The rising edges of the PWM signal coincides with the falling edge of the sawtooth signal.
- ✓ When the sawtooth signal is at the minimum value which is less than the minimum of the input signal, then the positive input of the comparator is at higher potential which gives the comparator output as positive.
- ✓ When the sawtooth signal rises and is at the maximum value, the negative input of the comparator is at higher potential, which will produce the comparator output to be negative.
- ✓ Thus the input signal magnitude determines the comparator output and its potential, which then decides the width of the pulse generated at the output.
- ✓ In other words we can say that the width of the pulse generated signal is directly proportional to the amplitude of the modulating signal.

- ✓ The PWM signal generated above is sent to an inverter which reverses the polarity of the pulses.
- ✓ This is then followed by a differentiator which generates +ve spikes for PWM signal going from High to Low and -ve spikes for Low to High transistion. The spikes generated are shown in the fourth waveform
- ✓ These spikes are then fed to the positive edge triggered pulse generator which generates fixed width pulses when a +ve spike appears, coinciding with the falling edge of the PWM signal.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 23

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION Date of Lecture:

Topic of Lecture: PWM Generation
Introduction: Amplitude modulation (**AM**) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.
Prerequisite knowledge for Complete understanding and learning of Topic: Modulation Index, Modulation Methods, Demodulation

GENERATION OF PWM

- ✓ PWM signal can be generated by using a comparator, where modulating signal and sawtooth signal form the input of the comparator. It is the simplest method for PWM generation.
- ✓ Considering both ±ve sides, the maximum of the input signal should be less than that of sawtooth signal.
- ✓ The comparator will compare the two signals together to generate the PWM signal at its output as shown in the third waveform
- ✓ The rising edges of the PWM signal coincides with the falling edge of the sawtooth signal.
- ✓ When the sawtooth signal is at the minimum value which is less than the minimum of the input signal, then the positive input of the comparator is at higher potential which gives the comparator output as positive.
- ✓ When the sawtooth signal rises and is at the maximum value, the negative input of the comparator is at higher potential, which will produce the comparator output to be negative.
- ✓ Thus the input signal magnitude determines the comparator output and its potential, which then decides the width of the pulse generated at the output.
- ✓ In other words we can say that the width of the pulse generated signal is directly proportional to the amplitude of the modulating signal.

- ✓ The PWM signal generated above is sent to an inverter which reverses the polarity of the pulses.
- ✓ This is then followed by a differentiator which generates +ve spikes for PWM signal going from High to Low and -ve spikes for Low to High transistion. The spikes generated are shown in the fourth waveform
- ✓ These spikes are then fed to the positive edge triggered pulse generator which generates fixed width pulses when a +ve spike appears, coinciding with the falling edge of the PWM signal.

Video Content / Details of website for further learning (if any):

4. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
5. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
6. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

3. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
4. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 24

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION **Date of Lecture:**

Topic of Lecture: Demodulation of PWM

Introduction: **Amplitude modulation** (**AM**) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

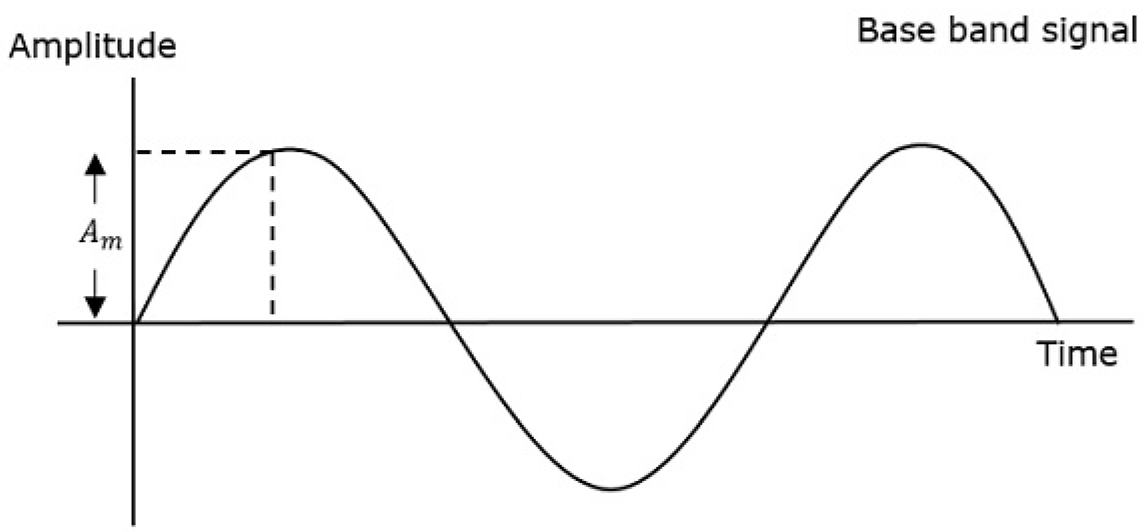
Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

Demodulation of PWM :

- ✓ The PWM pulses obtained at the comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered.
- ✓ Hence, corresponding to each trailing edge of PWM signal, the monostable output goes high.
- ✓ It remains high for a fixed time decided by its own RC components.
- ✓ Thus, as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM
- ✓ **Pulse Position Modulation (PPM)** is an analog modulating scheme in which the amplitude and width of the pulses are kept constant,
- ✓ while the position of each pulse, with reference to the position of a reference pulse varies

according to the instantaneous sampled value of the message signal.

- ✓ The transmitter has to send synchronizing pulses (or simply sync pulses) to keep the transmitter and receiver in synchronism.
- ✓ These sync pulses help maintain the position of the pulses.
- ✓ The following figures explain the Pulse Position Modulation.



Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 25

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION **Date of Lecture:**

Topic of Lecture: PPM Generation

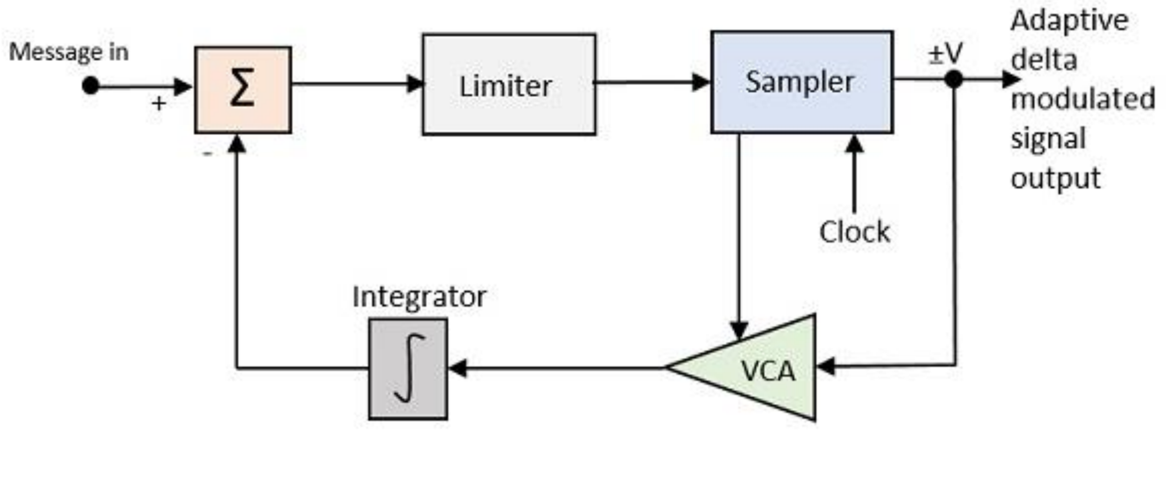
Introduction: PPM Generation modulation or Continuously variable slope delta modulation (CVSD) is a modification of DM in which the step size is not fixed.

Prerequisite knowledge for Complete understanding and learning of Topic:
PPM Generation , Adaptive delta modulation for PCM

PPM Generation:

- ✓ In digital modulation, we have come across certain problem of determining the step-size, which influences the quality of the output wave.
- ✓ A larger step-size is needed in the steep slope of modulating signal and a smaller stepsize is needed where the message has a small slope. The minute details get missed in the process. So, it would be better if we can control the adjustment of step-size, according to our requirement in order to obtain the sampling in a desired fashion. This is the concept of **Adaptive Delta Modulation**.

Following is the block diagram of Adaptive delta modulator.

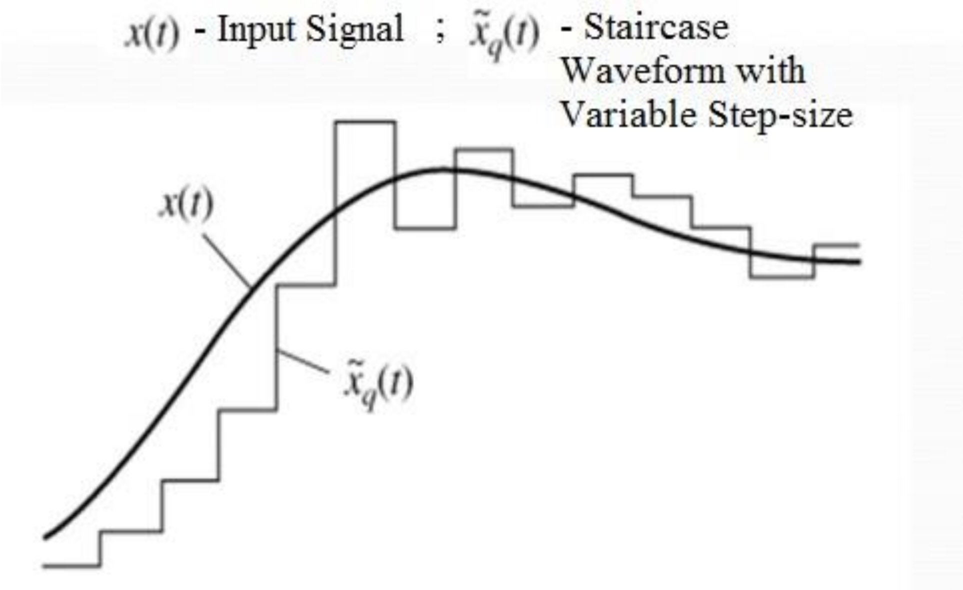


Adaptive delta modulator

- ✓ The gain of the voltage controlled amplifier is adjusted by the output signal from the sampler. The amplifier gain determines the step-size and both are proportional.
- ✓ ADM quantizes the difference between the value of the current sample and the predicted value of the next sample. It uses a variable step height to predict the next values, for the faithful reproduction of the fast varying values.
- ✓ There are a few techniques which have paved the basic path to digital communication processes. For the signals to get digitized, we have the sampling and quantizing techniques.
- ✓ For them to be represented mathematically, we have LPC and digital multiplexing techniques. These digital modulation techniques are further discussed.

Theory:

- ✓ In Adaptive Delta Modulation, the step size of the staircase signal is not fixed and changes depending upon the input signal. Here first the difference between the present sample value and previous approximation is calculated.
- ✓ This error is quantized i.e. if the present sample is smaller than the previous approximation, quantized value is high or else it is low. The output of the one-bit quantizer is given to the Logic step size control circuit where the step size is decided.



- ✓ At the logic step size control circuit, the output is decided based on the quantizer output. If the quantizer output is high, then the step size is doubled for the next sample. If the quantizer output is low, the step size is reduced by one step for the next sample.

Advantages

Some of the advantages of this modulation method are listed below-

- ✓ Adaptive delta modulation decreases slope error present in delta modulation.
- ✓ During demodulation, it uses a low pass filter which removes the quantized noise.
- ✓ The slope overload error and granular error present in delta modulation are solved using this modulation. Because of this, the signal to noise ratio of this modulation is better than delta modulation.
- ✓ In the presence of bit errors, this modulation provides robust performance. This reduces the need for error detection and correction circuits in radio design.
- ✓ The dynamic range of Adaptive delta modulation is large as the variable step size covers large range of values.

Multiplexing:

- ✓ Multiplexing is the process of combining multiple signals into one signal, over a shared medium. These signals, if analog in nature, the process is called as analog multiplexing. If digital signals are multiplexed, it is called as digital multiplexing.
- ✓ Multiplexing was first developed in telephony. A number of signals were combined to send through a single cable. The process of multiplexing divides a communication channel into several number of logical channels, allotting each one for a different message signal or a data stream to be transferred. The device that does multiplexing, can be called as a MUX. The reverse process, i.e., extracting the number of channels from one, which is done at the receiver is called as de-multiplexing. The device which does de-multiplexing is called as DEMUX.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=lCnc8rG1BPc
2. https://www.youtube.com/watch?v=w48JLyg-sRo
3. https://www.youtube.com/watch?v=iU-gePiPhec

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub,D L Schilling G Saha, Pearson Education, 2008. (171-173)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 26

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION **Date of Lecture:**

Topic of Lecture: PPM Modulation

Introduction: This type of digital pulse modulation technique is called differential pulse code modulation. The DPCM works on the principle of prediction. The value of the present sample is predicted from the previous samples. The prediction may not be exact, but it is very close to the actual sample value.

Prerequisite knowledge for Complete understanding and learning of Topic:
PPM Modulation , Transmitter and Receiver

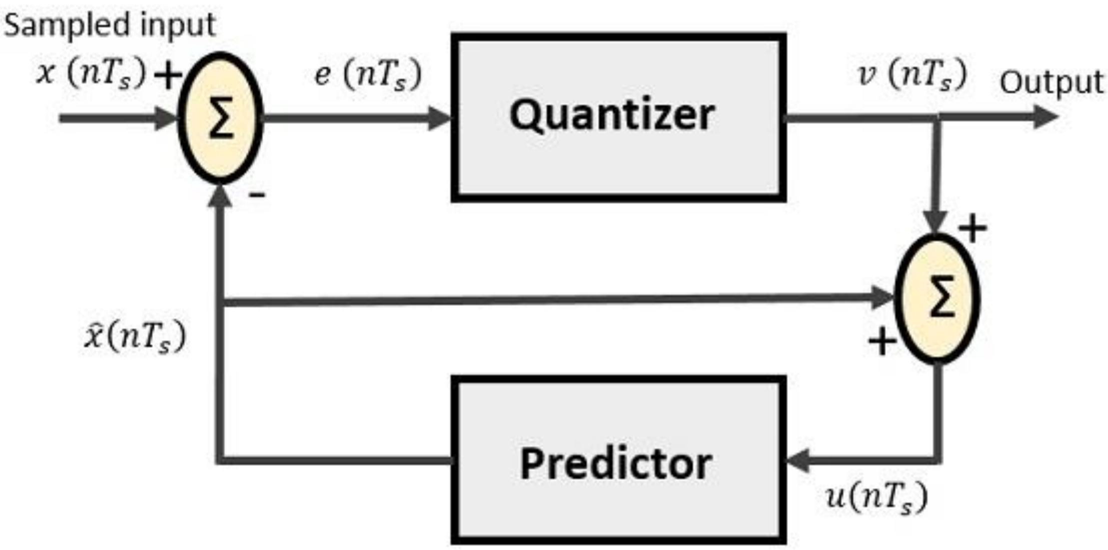
PPM Modulation:

- ✓ Differential pulse-code modulation (DPCM) is a signal encoder that uses the baseline of pulse-code modulation (PCM) but adds some functionalities based on the prediction of the samples of the signal. The input can be an analog signal or a digital signal.
- ✓ If the input is a continuous-time analog signal, it needs to be sampled first so that a discrete-time signal is the input to the DPCM encoder.

- Option 1: take the values of two consecutive samples; if they are analog samples, quantize them; calculate the difference between the first one and the next; the output is the difference.
- Option 2: instead of taking a difference relative to a previous input sample, take the difference relative to the output of a local model of the decoder process; in this option, the difference can be quantized, which allows a good way to incorporate a controlled loss in the encoding.

DPCM Transmitter

- ✓ The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter.
- ✓ The predictor produces the assumed samples from the previous outputs of the transmitter circuit.
- ✓ The same predictor circuit is used in the decoder to reconstruct the original input.



DPCM RECEIVER:

- ✓ The notation of the signals is the same as the previous ones. In the absence of noise, the encoded receiver input will be the same as the encoded transmitter output.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=UjkrvGc3lCo
2. https://www.youtube.com/watch?v=PFbm-jsTIpA
3. https://nptel.ac.in/courses/117103114/

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (181-183)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 27

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 3 : PULSE MODULATION **Date of Lecture:**

Topic of Lecture: Demodulation of PPM

Introduction: In telecommunication, intersymbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable.

Prerequisite knowledge for Complete understanding and learning of Topic:

Demodulation of PPM

Demodulation of PPM:

PPM:

- ✓ In telecommunication, inter symbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols.
- ✓ This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. The spreading of the pulse beyond its allotted time interval causes it to interfere with neighboring pulses. ISI is usually caused by multipath propagation or the inherent linear or non-linear frequency response of a communication channel causing successive symbols to "blur" together.
- ✓ The presence of ISI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible.

Causes

Multipath propagation

- ✓ One of the causes of intersymbol interference is multipath propagation in which a wireless signal from a transmitter reaches the receiver via multiple paths. The causes of this include reflection (for instance, the signal may bounce off buildings), refraction (such as through the foliage of a tree) and atmospheric effects such as atmospheric ducting and ionospheric reflection.
- ✓ Since the various paths can be of different lengths, this results in the different versions of the signal arriving at the receiver at different times. These delays mean that part or all of a given symbol will be spread into the subsequent symbols, thereby interfering with the correct detection of those symbols.

- ✓ Additionally, the various paths often distort the amplitude and/or phase of the signal, thereby causing further interference with the received signal.

Bandlimited channels

- ✓ Another cause of intersymbol interference is the transmission of a signal through a bandlimited channel, i.e., one where the frequency response is zero above a certain frequency (the cutoff frequency). Passing a signal through such a channel results in the removal of frequency components above this cutoff frequency.
- ✓ In addition, components of the frequency below the cutoff frequency may also be attenuated by the channel.
- ✓ This filtering of the transmitted signal affects the shape of the pulse that arrives at the receiver. The effects of filtering a rectangular pulse not only change the shape of the pulse within the first symbol period, but it is also spread out over the subsequent symbol periods.
- ✓ When a message is transmitted through such a channel, the spread pulse of each individual symbol will interfere with following symbols.
- ✓ Bandlimited channels are present in both wired and wireless communications. The limitation is often imposed by the desire to operate multiple independent signals through the same area/cable; due to this, each system is typically allocated a piece of the total bandwidth available.
- ✓ For wireless systems, they may be allocated a slice of the electromagnetic spectrum to transmit in (for example, FM radio is often broadcast in the 87.5–108 MHz range). This allocation is usually administered by a government agency; in the case of the United States this is the Federal Communications Commission (FCC). In a wired system, such as an optical fiber cable, the allocation will be decided by the owner of the cable.
- ✓ The bandlimiting can also be due to the physical properties of the medium - for instance, the cable being used in a wired system may have a cutoff frequency above which practically none of the transmitted signal will propagate.
- ✓ Communication systems that transmit data over bandlimited channels usually implement pulse shaping to avoid interference caused by the bandwidth limitation.
- ✓ If the channel frequency response is flat and the shaping filter has a finite bandwidth, it is possible to communicate with no ISI at all. Often the channel response is not known beforehand, and an adaptive equalizer is used to compensate the frequency response.

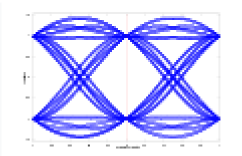
Effects on eye patterns:

- ✓ One way to study ISI in a PCM or data transmission system experimentally is to apply the received wave to the vertical deflection plates of an oscilloscope and to apply a sawtooth wave at the transmitted symbol rate R (R = 1/T) to the horizontal deflection plates.
- ✓ The resulting display is called an eye pattern because of its resemblance to the human eye for binary waves. The interior region of the eye pattern is called the eye opening. An eye pattern provides a great deal of information about the performance of the pertinent system.

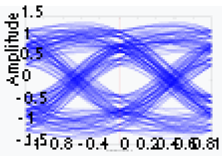
1. The width of the eye opening defines the time interval over which the received wave can be sampled without error from ISI. It is apparent that the preferred time for sampling is the instant of time at which the eye is open widest.
2. The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.
3. The height of the eye opening, at a specified sampling time, defines the margin over noise.

- ✓ An eye pattern, which overlays many samples of a signal, can give a graphical representation of the signal characteristics.

- ✓ The first image below is the eye pattern for a binary phase-shift keying (PSK) system in which a one is represented by an amplitude of -1 and a zero by an amplitude of $+1$. The current sampling time is at the center of the image and the previous and next sampling times are at the edges of the image. The various transitions from one sampling time to another (such as one-to-zero, one-to-one and so forth) can clearly be seen on the diagram.
- ✓ The noise margin - the amount of noise required to cause the receiver to get an error - is given by the distance between the signal and the zero amplitude point at the sampling time; in other words, the further from zero at the sampling time the signal is the better.
- ✓ For the signal to be correctly interpreted, it must be sampled somewhere between the two points where the zero-to-one and one-to-zero transitions cross. Again, the further apart these points are the better, as this means the signal will be less sensitive to errors in the timing of the samples at the receiver.
- ✓ The effects of ISI are shown in the second image which is an eye pattern of the same system when operating over a multipath channel. The effects of receiving delayed and distorted versions of the signal can be seen in the loss of definition of the signal transitions. It also reduces both the noise margin and the window in which the signal can be sampled, which shows that the performance of the system will be worse (i.e. it will have a greater bit error ratio).



The eye diagram of a binary PSK system



The eye diagram of the same system with multipath effects added

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=DqXbdvE7tQg
2. https://www.youtube.com/watch?v=owJqcARASPY
3. https://nptel.ac.in/courses/117103125/

Important Books/Journals for further learning including the page nos.:

1. "Electronic Communication Systems Fundamentals through Advanced", Wayne Tomasi, Pearson Education, 2008. (191-193)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 28

LECTURE HANDOUTS

ECE | **II / III**

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 4 : **PULSE DIGITAL MODULATION** **Date of Lecture:**

Topic of Lecture: Elements of digital communication systems

Introduction: Elements of digital communication systems modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Modulation Index, Modulation Methods, Demodulation

Elements of digital communication systems:

The elements which form a digital communication system is represented by the following block diagram for the ease of understanding.

Digital Communication

- ✓ Following are the sections of the digital communication system.

Source

- ✓ The source can be an analog signal. Example: A Sound signal

Input Transducer

- ✓ This is a transducer which takes a physical input and converts it to an electrical signal (Example: microphone).
- ✓ This block also consists of an analog to digital converter where a digital signal is needed for further processes.
- ✓ A digital signal is generally represented by a binary sequence.

Source Encoder

- ✓ The source encoder compresses the data into minimum number of bits.
- ✓ This process helps in effective utilization of the bandwidth.
- ✓ It removes the redundant bits unnecessary excess

Output Transducer

- ✓ This is the last block which converts the signal into the original physical form, which was at the input of the transmitter. It converts the electrical signal into physical output (Example: loud speaker).

Output Signal

- ✓ This is the output which is produced after the whole process. Example − The sound signal received.
- ✓ This unit has dealt with the introduction, the digitization of signals, the advantages and the elements of digital communications. In the coming chapters, we will learn about the concepts of Digital communications, in detail.bits,i.e.,zeroes.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

DESIGNING YOUR FUTURE
Estd. 2000

IQAC

L - 29

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 4 : PULSE DIGITAL MODULATION **Date of Lecture:**

Topic of Lecture: Advantages of Digital Communication systems
Introduction: Amplitude modulation (AM) is a modulation technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.
Prerequisite knowledge for Complete understanding and learning of Topic: Advantages of Digital Communication, Modulation Methods, Demodulation

Advantages of Digital Communication Systems:

As the signals are digitized, there are many advantages of digital communication over analog communication, such as −

- ✓ The effect of distortion, noise, and interference is much less in digital signals as they are less affected.
- ✓ Digital circuits are more reliable.
- ✓ Digital circuits are easy to design and cheaper than analog circuits.
- ✓ The hardware implementation in digital circuits, is more flexible than analog.
- ✓ The occurrence of cross-talk is very rare in digital communication.
- ✓ The signal is un-altered as the pulse needs a high disturbance to alter its properties, which is very difficult.
- ✓ Signal processing functions such as encryption and compression are employed in digital circuits to maintain the secrecy of the information.

Elements of PCM

- Sampler. ...
- Quantizer. ...
- Encoder. ...

- Regenerative Repeater. ...
- Decoder. ...
- Reconstruction Filter.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. "Communication Systems", Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 30

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 4 : PULSE DIGITAL MODULATION **Date of Lecture:**

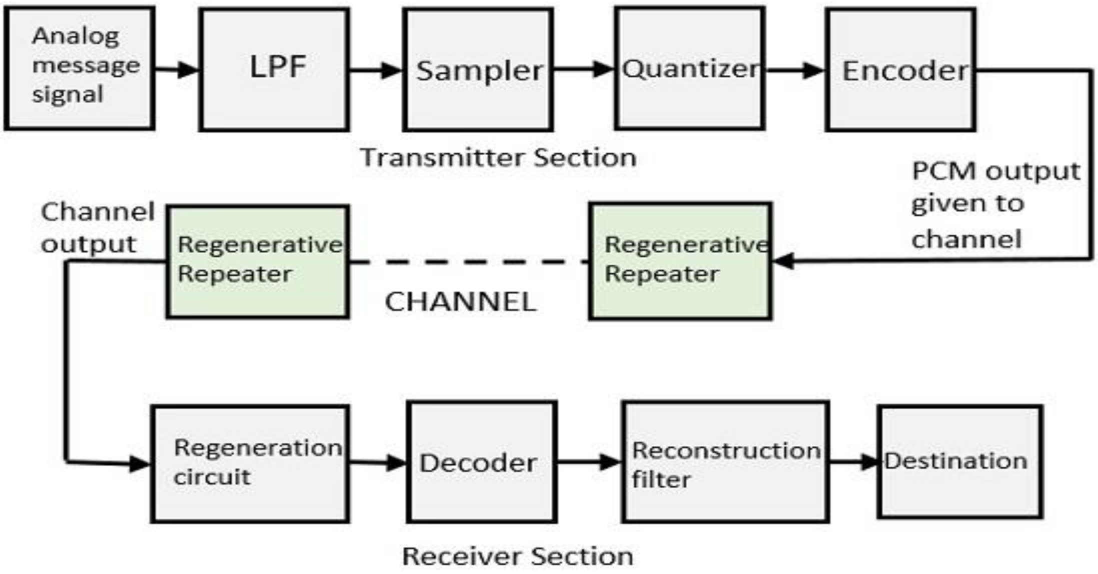
Topic of Lecture: Elements of PCM

Introduction: Elements of PCM in electronic communication, most commonly for transmitting information via a radio carrier wave. In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal being transmitted.

Prerequisite knowledge for Complete understanding and learning of Topic:
Elements of PCM, Modulation Methods, Demodulation

Elements of PCM:

- ✓ The transmitter section of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding, which are performed in the analog-to-digital converter section.
- ✓ The low pass filter prior to sampling prevents aliasing of the message signal.
- ✓ The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train.
- ✓ Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver section of the PCM elements



.r

Low Pass Filter

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the technique which helps to collect the sample data at instantaneous values of message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component W of the message signal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Video Content / Details of website for further learning (if any):

1. https://www.electronics-notes.com/articles/radio/modulation/amplitude-modulation-am.php
2. https://www.elprocus.com/what-is-amplitude-modulation-derivations-typesand-applications/
3. https://byjus.com/jee/amplitude-modulation/

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub, D L Schilling, G Saha, Pearson Education, 2008.
2. “Communication Systems”, Simon Haykin, John Wiley & Sons, 2010. (34-38)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 31

LECTURE HANDOUTS

ECE | **II / III**

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 4 : **PULSE DIGITAL MODULATION** **Date of Lecture:**

Topic of Lecture: Sampling, Quantization & Coding, Quantization error

Introduction: Sampling Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components. Where, vrms is the rms value of cos signal. vm is the peak value of cos signal.

Prerequisite knowledge for Complete understanding and learning of Topic:
Sampling, Quantization & Coding, Quantization error

Sampling, Quantization & Coding, Quantization error:
Sampling:

- ✓ Unmodulated carrier can be described mathematically as
  $V_c(t) = E_c\sin(2\pi f_c t)$
- ✓ Instantaneous amplitude of the modulated wave can be expressed as
  $V_{am}(t) = [E_c + E_m\sin(2\pi f_m t)][\sin(2\pi f_c t)]$
  $E_m = m E_c$, $V_{am}(t) = [E_c + m E_c\sin(2\pi f_m t)][\sin(2\pi f_c t)]$
- ✓ Factoring Ec, $V_{am}(t) = [1 + m \sin(2\pi f_m t)][E_c\sin(2\pi f_c t)][1 + m \sin(2\pi f_m t)]$ – constant + modulating signal$[E_c\sin(2\pi f_c t)]$ – unmodulated carrier
- ✓ Multiplying, $V_{am}(t) = [E_c\sin(2\pi f_c t)] + [m \sin(2\pi f_m t)][E_c\sin(2\pi f_c t)]$
- ✓ Therefore, $V_{am}(t) = [E_c\sin(2\pi f_c t)] – (m E_c/2)\cos[2\pi(f_c+f_m)t] + (m E_c/2)\cos[2\pi(f_c-f_m)t]$

From the above equation,
$[E_c\sin(2\pi f_c t)]$ – carrier signal(volts)– $(m E_c/2)\cos[2\pi(f_c+f_m)t]$ - upper side freq (volts)

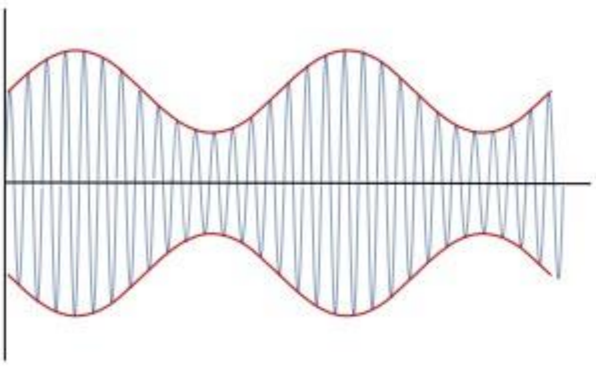


Fig. Amplitude modulated wave

Carrier signal equations

Looking at the theory, it is possible to describe the carrier in terms of a sine wave as follows:

C(t) = C sin(ωc + φ)C(t) = C sin(ωc + φ)

Where:

- ✓ carrier frequency in Hertz is equal to ωc / 2 π
- ✓ C is the carrier amplitude
- ✓ φ is the phase of the signal at the start of the reference time

Both C and φ can be omitted to simplify the equation by changing C to "1" and φ to "0".

Modulating signal equations

- ✓ The modulating waveform can either be a single tone. This can be represented by a cosine waveform, or the modulating waveform could be a wide variety of frequencies - these can be represented by a series of cosine waveforms added together in a linear fashion.

- ✓ For the initial look at how the signal is formed, it is easiest to look at the equation for a simple single tone waveform and then expand the concept to cover the more normal case. Take a single tone waveform:

m(t) = M sin(ωm + φ)

Where:

modulating signal frequency in Hertz is equal to ωm / 2 π
M is the carrier amplitude. φ is the phase of the signal at the start of the reference time
Both C and φ can be omitted to simplify the equation by changing C to "1" and φ to "0".

Power distribution:

Consider the following equation of amplitude modulated wave.

s(t)=Accos(2πfct)+Acμ2cos[2π(fc+fm)t]+Acμ2cos[2π(fc−fm)t] s(t)
=Accos(2πfct)+Acμ2cos[2π(fc+fm)t]+Acμ2cos[2π(fc−fm)t]

Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components.

Pt=Pc+PUSB+PLSBPt=Pc+PUSB+PLSB

We know that the standard formula for power of cos signal is

P=vrms2R=(vm/2–√)22P=vrms2R=(vm/2)22

Where,

vrms is the rms value of cos signal.
vm is the peak value of cos signal.

First, let us find the powers of the carrier, the upper and lower sideband one by one.

Carrier power

Pc=(Ac/2–√)2R=Ac22RPc=(Ac/2)2R=Ac22R

Upper sideband power

PUSB=(Acμ/22–√)2R=Ac2μ28RPUSB=(Acμ/22)2R=Ac2μ28R

Similarly, we will get the lower sideband power same as that of the upper side band power.

PLSB=Ac2μ28RPLSB=Ac2μ28R

Now, let us add these three powers in order to get the power of AM wave.

Pt=Ac22R+Ac2μ28R+Ac2μ28RPt=Ac22R+Ac2μ28R+Ac2μ28R

$$\Rightarrow Pt=(Ac22R)(1+\mu 24+\mu 24)\Rightarrow Pt=(Ac22R)(1+\mu 24+\mu 24)$$
$$\Rightarrow Pt=Pc(1+\mu 22)\Rightarrow Pt=Pc(1+\mu 22)$$

We can use the above formula to calculate the power of AM wave, when the carrier power and the modulation index are known.

If the modulation index $\mu=1\mu=1$ then the power of AM wave is equal to 1.5 times the carrier power. So, the power required for transmitting an AM wave is 1.5 times the carrier power for a perfect modulation.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=W3syAFjmaFw

2. https://www.youtube.com/watch?v=cnN2YGkmlGk

3. https://www.youtube.com/watch?v=19q2X6Prr7s

Important Books/Journals for further learning including the page nos.:

1. “Principles of Communication”, H.Taub,D L Schilling G Saha, Pearson Education, 2008. (32 -34)

Course Faculty

Verified by HOD

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 4 : **PULSE DIGITAL MODULATION** **Date of Lecture:**

Topic of Lecture: PAM and Other forms of pulse modulations Differential PCM system (DPCM)

Introduction: Differential PCM system is a class of carrier modulation that is used transmission systems. The class comprises frequency modulation (FM) and phase modulation (PM), and is based on altering the frequency or the phase, respectively, of a carrier signal to encode the message signal. This contrasts with varying the amplitude of the carrier, practiced in amplitude modulation (AM) transmission, the earliest of the major modulation methods used widely in early radio broadcasting.

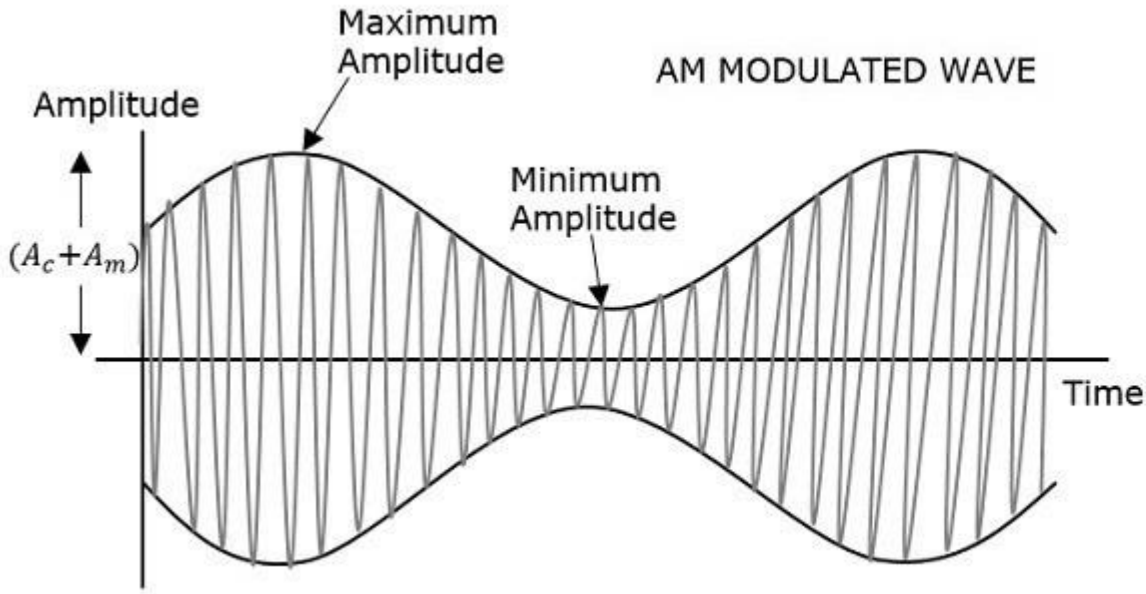
Prerequisite knowledge for Complete understanding and learning of Topic:

PAM and Other forms of pulse modulations Differential PCM system

PAM and Other forms of pulse modulations Differential PCM system:

PAM:

- ✓ In amplitude modulation, the amplitude of the carrier varies. But in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- ✓ The amplitude and the phase of the carrier signal remains constant whereas the frequency of the carrier changes. This can be better understood by observing the following figures.



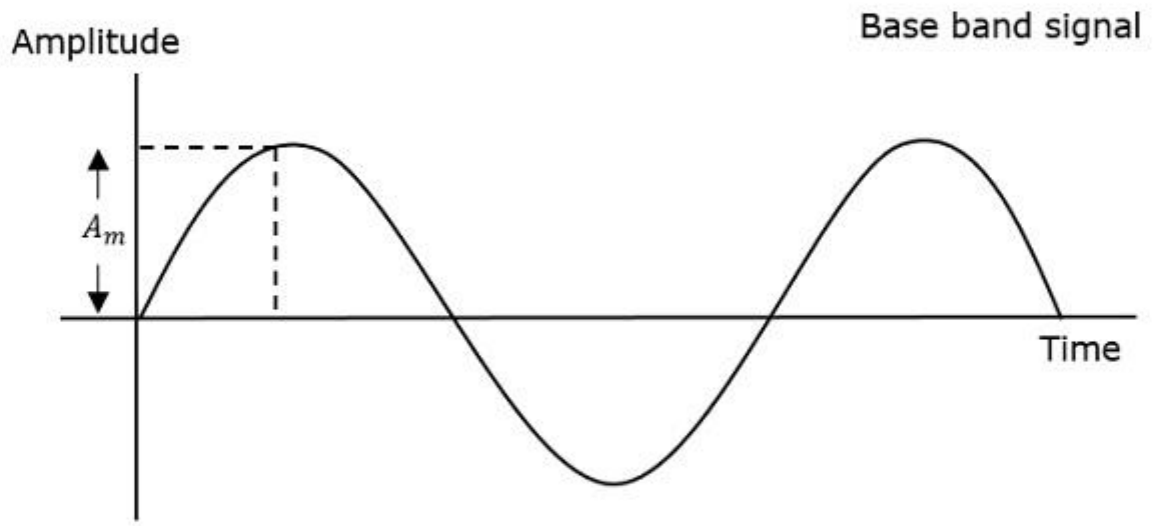
- ✓ The frequency of the modulated wave remains constant as the carrier wave frequency when the message signal is at zero. The frequency increases when the message signal reaches its maximum amplitude.
- ✓ Which means, with the increase in amplitude of the modulating or message signal, the carrier frequency increases. Likewise, with the decrease in the amplitude of the modulating signal, the frequency also decreases.

Mathematical Representation

- ✓ Let the carrier frequency be $\mathbf{f_c}$
- ✓ The frequency at maximum amplitude of the message signal = $\mathbf{f_c + \Delta f}$
- ✓ The frequency at minimum amplitude of the message signal = $\mathbf{f_c - \Delta f}$
- ✓ The difference between FM modulated frequency and normal frequency is termed as **Frequency Deviation** and is denoted by **Δf**.
- ✓ The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the **Carrier Swing**.
- ✓ Carrier Swing = 2 × frequency deviation = **2 × Δf**

Phase modulation:

- ✓ In frequency modulation, the frequency of the carrier varies. But in **Phase Modulation (PM)**, the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- ✓ The amplitude and the frequency of the carrier signal remains constant whereas the phase of the carrier changes. This can be better understood by observing the following figures.



- ✓ The equation for PM wave is

$$s(t) = A_c\cos[W_c t + k_p m(t)]$$

Where,

A_c = the amplitude of the carrier

w_c = angular frequency of the carrier = $2\pi f_c$

m(t) = message signal

✓ Phase modulation is used in mobile communication systems, while frequency modulation is used mainly for FM broadcasting. ✓ The frequency of the wave also changes the phase of the wave. Though they are related, their relationship is not linear. Phase modulation is an indirect method of producing FM. The amount of frequency shift, produced by a phase modulator increases with the modulating frequency.
Video Content / Details of website for further learning (if any): 1. https://www.tutorialspoint.com/electronic_circuits 2. https://www.physics-and-radio-electronics.com/electronic-devices-and-circuits 3. https://nptel.ac.in/courses/117103063/
Important Books/Journals for further learning including the page nos.: 1. "Electronic Communication Systems Fundamentals through Advanced", Wayne Tomasi, Pearson Education, 2008. (54-56)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 33

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 4 : PULSE DIGITAL MODULATION **Date of Lecture:**

Topic of Lecture: TDM

Introduction:TDM phase modulation, the maximum difference between the instantaneous phase angle of the modulated wave and the phase angle of the unmodulated carrier.

Prerequisite knowledge for Complete understanding and learning of Topic:
TDM Phase deviation and modulation index

Time division Multiplexing:
Phase deviation:

- ✓ Phase modulation (PM) is a modulation pattern for conditioning communication signals for transmission. It encodes a message signal as variations in the instantaneous phase of a carrier wave. Phase modulation is one of the two principal forms of angle modulation, together with frequency modulation.
- ✓ The phase of a carrier signal is modulated to follow the changing signal level (amplitude) of the message signal. The peak amplitude and the frequency of the carrier signal are maintained constant, but as the amplitude of the message signal changes, the phase of the carrier changes correspondingly.
- ✓ Phase modulation is widely used for transmitting radio waves and is an integral part of many digital transmission coding schemes that underlie a wide range of technologies like Wi-Fi, GSM and satellite television.
- ✓ PM is used for signal and waveform generation in digital synthesizers, such as the Yamaha DX7, to implement FM synthesis. A related type of sound synthesis called phase distortion is used in the Casio CZ synthesizers.

Theory:

- ✓ PM changes the phase angle of the complex envelope in direct proportion to the message signal.
- ✓ If *m(t)* is the message signal to be transmitted and the carrier onto which the signal is modulated is,

$$c(t)=A_c \sin(\omega_c t+\phi_c)$$

then the modulated signal is,

$$y(t)= Ac \sin(\acute{\omega}ct+m(t)+\phi c),$$

- ✓ This shows how m(t) modulates the phase - the greater m(t) is at a point in time, the greater the phase shift of the modulated signal at that point. It can also be viewed as a change of the frequency of the carrier signal, and phase modulation can thus be considered a special case of FM in which the carrier frequency modulation is given by the time derivative of the phase modulation.
- ✓ The modulation signal could here be

$$m(t)=\cos(\acute{\omega}ct+h\acute{\omega}m(t))$$

The mathematics of the spectral behavior reveals that there are two regions of particular interest:

- ✓ For small amplitude signals, PM is similar to amplitude modulation (AM) and exhibits its unfortunate doubling of baseband bandwidth and poor efficiency.
- ✓ For a single large sinusoidal signal, PM is similar to FM, and its bandwidth is approximately

$$2(h+1)fm$$

where $fm=\acute{\omega}m/2\pi$ and h is the modulation index defined below. This is also known as Carson's Rule for PM.

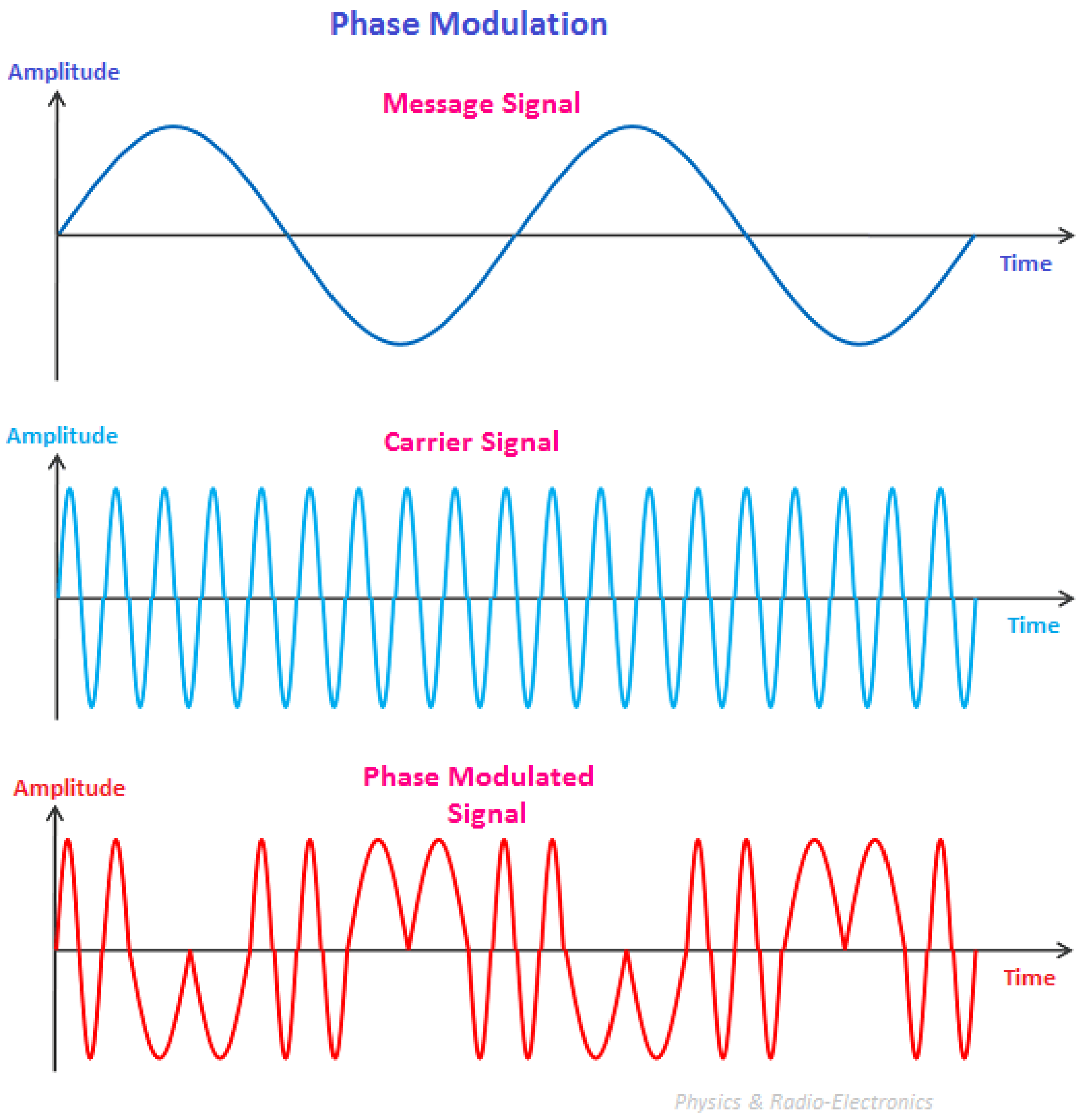


Fig. Phase modulation

Modulation index:

- ✓ As with other modulation indices, this quantity indicates by how much the modulated variable varies around its unmodulated level. It relates to the variations in the phase of the carrier signal:

$$h=\Delta\phi,$$

where $\Delta\phi$ is the peak phase deviation. Compare to the modulation index for frequency modulation.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=ePWAv08O99c
2. https://www.youtube.com/watch?v=lTLGTvdHHA8
3. https://www.youtube.com/watch?v=NslPxOu9M5I

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (74-75)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

Estd. 2000

L - 34

LECTURE HANDOUTS

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 4 : PULSE DIGITAL MODULATION **Date of Lecture:**

Topic of Lecture: Delta modulation

Introduction: Delta modulation is used in FM radio to describe the maximum difference between an FM modulated frequency and the nominal carrier frequency. The term is sometimes mistakenly used as synonymous with frequency drift, which is an unintended offset of an oscillator from its nominal frequency.

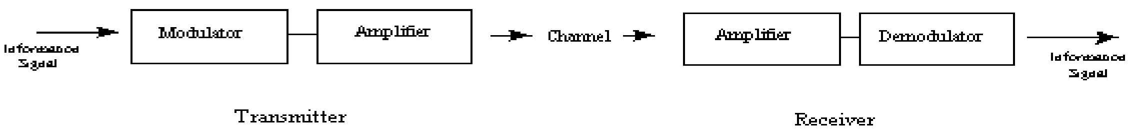
Prerequisite knowledge for Complete understanding and learning of Topic:
Delta modulation, Percent modulation

Delta modulation:

Frequency deviation:
Basic System:
Transmitter:
- ✓ The sub-system that takes the information signal and processes it prior to transmission. The transmitter modulates the information onto a carrier signal, amplifies the signal and broadcasts it over the channel

Channel:
- ✓ The medium which transports the modulated signal to the receiver. Air acts as the channel for broadcasts like radio. May also be a wiring system like cable TV or the Internet.

Receiver:
- ✓ The sub-system that takes in the transmitted signal from the channel and processes it to retrieve the information signal. The receiver must be able to discriminate the signal from other signals which may using the same channel (called tuning), amplify the signal for processing and demodulate (remove the carrier) to retrieve the information. It also then processes the information for reception (for example, broadcast on a loudspeaker).



Modulation:

- ✓ The information signal can rarely be transmitted as is, it must be processed. In order to use electromagnetic transmission, it must first be converted from audio into an electric signal.

- ✓ The conversion is accomplished by a transducer. After conversion it is used to modulate a carrier signal.

FM:

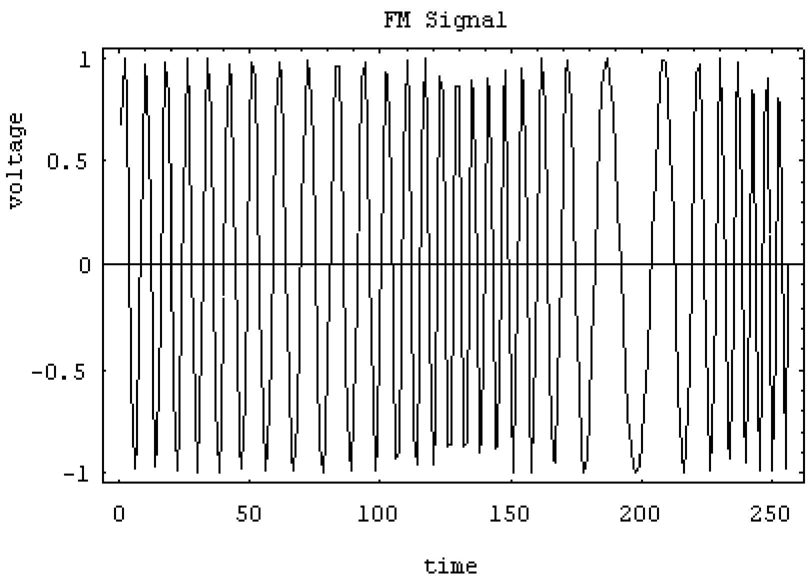
- ✓ Frequency modulation uses the information signal, $V_m(t)$ to vary the carrier frequency within some small range about its original value. Here are the three signals in mathematical form:

$$\text{Information: } V_m(t)$$

$$\text{Carrier: } V_c(t) = V_{co} \sin (2\ p\ f_c\ t + f)$$

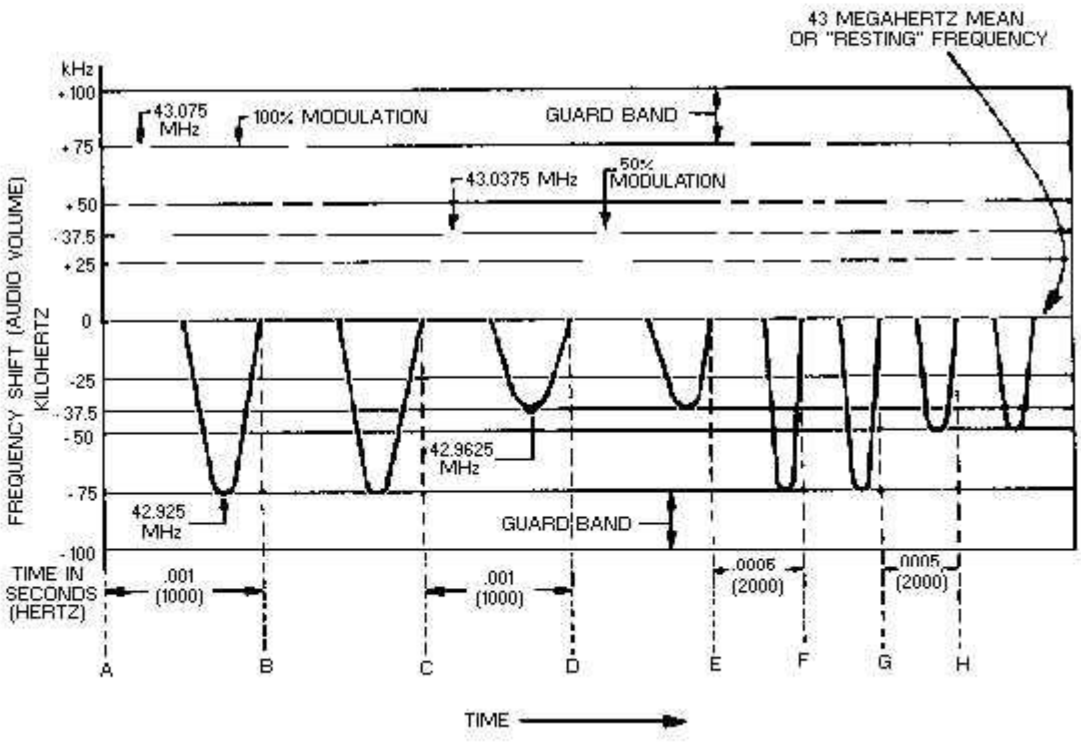
$$\text{FM: } V_{FM}\ (t) = V_{co} \sin (2\ p\ [f_c + (Df/V_{mo})\ V_m\ (t)\ ]\ t + f)$$

- ✓ We have replaced the carrier frequency term, with a time-varying frequency. We have also introduced a new term: Df, the peak frequency deviation. In this form, you should be able to see that the carrier frequency term: $f_c + (Df/V_{mo})\ V_m\ (t)$ now varies between the extremes of f_c - Df and f_c + Df. The interpretation of Df becomes clear: it is the farthest away from the original frequency that the FM signal can be. Sometimes it is referred to as the "swing" in the frequency.
- ✓ We can also define a modulation index for FM, analogous to AM
- ✓ $b = Df/f_m$, where f_m is the maximum modulating frequency used.
- ✓ The simplest interpretation of the modulation index, b, is as a measure of the peak frequency deviation, Df. In other words, b represents a way to express the peak deviation frequency as a multiple of the maximum modulating frequency, f_m, i.e. $Df = b\ f_m$.



Percent modulation:

- ✓ Modulation for AM exists when the amplitude of the modulation envelope varies between 0 volts and twice its normal unmodulated value.
- ✓ At 100-percent modulation there is a power increase of 50 percent. Because the modulating wave is not constant in voice signals, the degree of modulation constantly varies.
- ✓ In this case the vacuum tubes in an AM system cannot be operated at maximum efficiency because of varying power requirements.
- ✓ In frequency modulation, 100-percent modulation has a meaning different from that of AM. The modulating signal varies only the frequency of the carrier.
- ✓ Therefore, tubes do not have varying power requirements and can be operated at maximum efficiency and the fm signal has a constant power output.



- ✓ In fm a modulation of 100 percent simply means that the carrier is deviated in frequency by the full permissible amount. For example, an 88.5-megahertz fm station operates at 100-percent modulation when the modulating signal deviation frequency band is from 75 kilohertz above to 75 kilohertz below the carrier (the maximum allowable limits).
- ✓ This maximum deviation frequency is set arbitrarily and will vary according to the applications of a given fm transmitter. In the case given above, 50-percentmodulation would mean that the carrier was deviated 37.5 kilohertz above and below the resting frequency (50 percent of the 150-kilohertz band divided by 2).
- ✓ Other assignments for fm service may limit the allowable deviation to 50 kilohertz, or even 10 kilohertz. Since there is no fixed value for comparison, the term "percent of modulation" has little meaning for fm.
- ✓ The term MODULATIONINDEX is more useful in fm modulation discussions. Modulation index is frequency deviation divided by the frequency of the modulating signal.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=c11i32lGmQA

2. https://www.youtube.com/watch?v=-njI4zKP2RI

3. https://www.youtube.com/watch?v=beFoCZ7oMyY

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (44-46)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 35

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 4 : PULSE DIGITAL MODULATION **Date of Lecture:**

Topic of Lecture: Adaptive delta modulation

Introduction: Adaptive delta modulation of the frequency of the modulated wave remains constant as the carrier wave frequency when the message signal is at zero. The frequency increases when the message signal reaches its maximum amplitude.

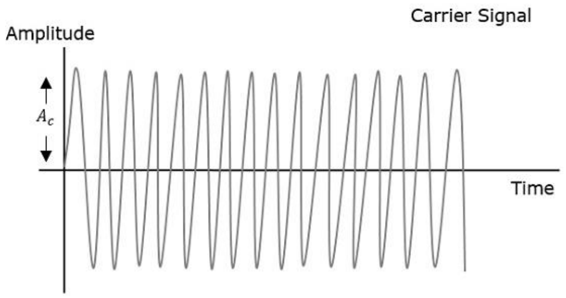
Prerequisite knowledge for Complete understanding and learning of Topic:
Adaptive delta modulation, mathematical expression, equation

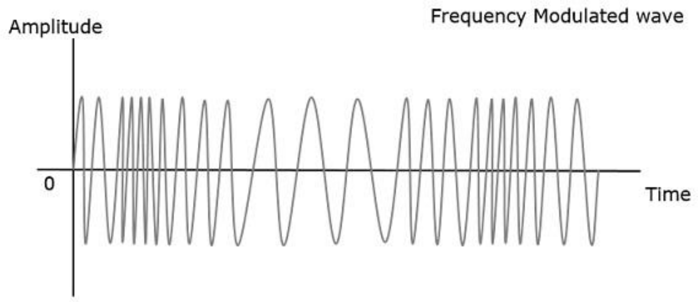
Adaptive delta modulation:

delta modulation analysis:

Angle Modulation is the process in which the frequency or the phase of the carrier varies according to the message signal. This is further divided into frequency and phase modulation.

- ✓ Frequency Modulation is the process of varying the frequency of the carrier signal linearly with the message signal.
- ✓ Phase Modulation is the process of varying the phase of the carrier signal linearly with the message signal.
- ✓ In amplitude modulation, the amplitude of the carrier varies. But in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- ✓ The amplitude and the phase of the carrier signal remains constant whereas the frequency of the carrier changes. This can be better understood by observing the following figures.





- ✓ The frequency of the modulated wave remains constant as the carrier wave frequency when the message signal is at zero. The frequency increases when the message signal reaches its maximum amplitude.
- ✓ Which means, with the increase in amplitude of the modulating or message signal, the carrier frequency increases. Likewise, with the decrease in the amplitude of the modulating signal, the frequency also decreases.

Mathematical Representation

- ✓ Let the carrier frequency be $\mathbf{f_c}$
- ✓ The frequency at maximum amplitude of the message signal = $\mathbf{f_c + \Delta f}$
- ✓ The frequency at minimum amplitude of the message signal = $\mathbf{f_c - \Delta f}$
- ✓ The difference between FM modulated frequency and normal frequency is termed as **Frequency Deviation** and is denoted by $\mathbf{\Delta f}$.
- ✓ The deviation of the frequency of the carrier signal from high to low or low to high can be termed as the **Carrier Swing**.
- ✓ Carrier Swing = 2 × frequency deviation

  = $\mathbf{2 \times \Delta f}$

Equation for FM WAVE:

- ✓ The equation for FM wave is −

$$s(t) = A_c\cos[W_c t + 2\pi k_f m(t)]$$

Where,

$\mathbf{A_c}$ = the amplitude of the carrier

$\mathbf{w_c}$ = angular frequency of the carrier = $\mathbf{2\pi f_c}$

m(t) = message signal

- ✓ FM can be divided into Narrowband FM and Wideband FM.

Narrowband FM

The features of Narrowband FM are as follows −

- This frequency modulation has a small bandwidth.
- The modulation index is small.
- Its spectrum consists of carrier, USB, and LSB.
- This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Wideband FM

The features of Wideband FM are as follows

- This frequency modulation has infinite bandwidth.
- The modulation index is large, i.e., higher than **1**.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
- This is used in entertainment broadcasting applications such as FM radio, TV, etc.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=gsUaHawPy-w
2. https://www.youtube.com/watch?v=PmuZnJfheK4
3. https://www.youtube.com/watch?v=6Y9n8dMYL-o

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (97-99)

Course Faculty

Verified by HOD

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 4 : **PULSE DIGITAL MODULATION** **Date of Lecture:**

Topic of Lecture: Comparison of PCM and DM systems, noise in PCM and DM systems

Introduction: Comparison of PCM and DM systems, the bandwidth requirement of angle modulated waveforms can be obtained depending upon modulation index. The modulation index can be classified as low (less than 1), medium (1 to 10) and high (greater than 10). The low index systems are called narrowband FM. For such system the frequency spectrum resembles AM.

Prerequisite knowledge for Complete understanding and learning of Topic:
Comparison of PCM and DM systems, noise in PCM and DM systems

Comparison of PCM and DM systems, noise in PCM and DM systems:

- ✓ We know that AM contains only two sidebands per modulating frequency. But angle modulated signal contains large number of sidebands depending upon the modulation index. Since FM and PM have identical modulated waveforms, their frequency content is same. Consider the PM equation or spectrum analysis,

$$e(t) = E_c \sin[\omega_c t + m \cos\omega_m t]$$

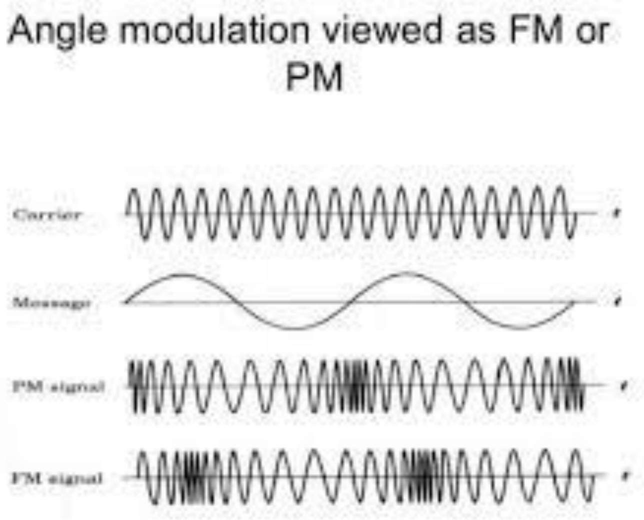
Using Bessel functions, this equation can be expanded as,

$$\begin{aligned} e(t) = E_c \{ & J_0 \sin\omega_c t \\ & + J_1[\sin(\omega_c + \omega_m)t - \sin(\omega_c - \omega_m)t] \\ & + J_2[\sin(\omega_c + 2\omega_m)t + \sin(\omega_c - 2\omega_m)t] \\ & + J_3[\sin(\omega_c + 3\omega_m)t + \sin(\omega_c - 3\omega_m)t] \\ & + J_4[\sin(\omega_c + 4\omega_m)t - \sin(\omega_c - 4\omega_m)t] + \ldots \} \end{aligned}$$

Here $J_0, J_1, J_2 \ldots$ are the Bessel functions.

Bandwidth Requirement:

- ✓ The bandwidth requirement of angle modulated waveforms can be obtained depending upon modulation index. The modulation index can be classified as low (less than 1), medium (1 to 10) and high (greater than 10).
- ✓ The low index systems are called narrowband FM.



- ✓ For such system the frequency spectrum resembles AM. Hence minimum bandwidth is given as,

$$BW=2f_m \text{ Hz}$$

For high index modulation,

$$BW=2\partial$$

The bandwidth obtained is,

$$BW=2nf_m$$

Here ‘n’ is the number of significant sidebands obtained from Bessel table.

Carson’s Rule:

- ✓ Carson’s Rule gives approximate minimum bandwidth of angle modulated signal as

$$BW=2[f_{max}+\partial]\text{Hz}$$

- ✓ Here $f_{m(max)}$ is the maximum modulating frequency. As per Carson’s rule, the bandwidth accommodates almost 98% of the total transmitted power.

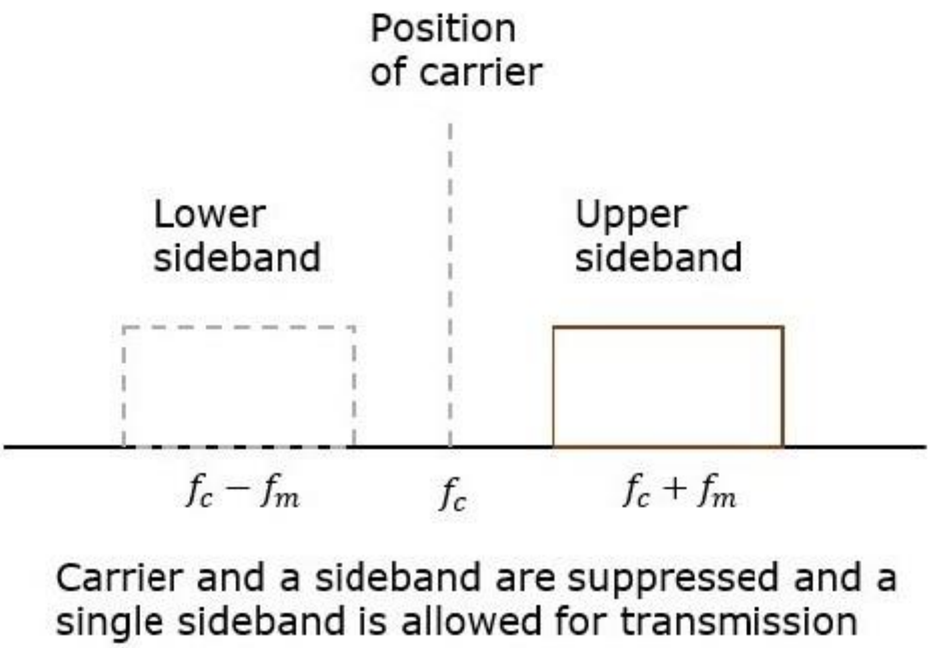


Fig. SSBSC in angle modulation

Video Content / Details of website for further learning (if any): 1. https://www.youtube.com/watch?v=gsUaHawPy-w 2. https://www.youtube.com/watch?v=vdPHNrSN0gc 3. https://www.youtube.com/watch?v=Mv6bzMQF0Lc
Important Books/Journals for further learning including the page nos.: 1. “Electronic Communication Systems Fundamentals through Advanced”, Wayne Tomasi, Pearson Education, 2008. (85-87)

Course Faculty

Verified by HOD

LECTURE HANDOUTS

L - 37

ECE

II /III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES **Date of Lecture:**

Topic of Lecture: Introduction of digital modulation techniques

Introduction: Introduction of digital modulation techniques of a communication channel refers to the maximum rate of error-free data that can theoretically be transferred over the channel if the link is subject to random data **transmission** errors, for a particular noise level.

Prerequisite knowledge for Complete understanding and learning of Topic:
Introduction of digital modulation techniques

Introduction of digital modulation techniques:
Shannon Hartley theorem:

- ✓ In information theory, the Shannon–Hartley theorem tells the maximum rate at which information can be transmitted over a communications channel of a specified bandwidth in the presence of noise.
- ✓ It is an application of the noisy-channel coding theorem to the archetypal case of a continuous-time analog communications channel subject to Gaussian noise.

Statement:

- ✓ The Shannon–Hartley theorem states the channel capacity C, meaning the theoretical tightest upper bound on the information rate of data that can be communicated at an arbitrarily low error rate using average received signal power S through an analog communication channel subject to additive white Gaussian noise (AWGN) of power N :
  $C=B\log_2(1+S/N)$

Nyquist rate:

- ✓ In 1927, Nyquist determined that the number of independent pulses that could be put through a telegraph channel per unit time is limited to twice the bandwidth of the channel. In symbolic notation,

$f_p \leq 2B$

where, f_p is the pulse frequency (in pulses per second) and B is the bandwidth (in hertz).

- ✓ The quantity 2B later came to be called the Nyquist rate, and transmitting at the limiting pulse rate of 2B pulses per second as signaling at the Nyquist rate.
- ✓ Nyquist published his results in 1928 as part of his paper "Certain topics in Telegraph Transmission Theory".

Hartley's law:

- ✓ During 1928, Hartley formulated a way to quantify information and its line rate (also known as data signaling rate R bits per second). This method, later known as Hartley's law, became an important precursor for Shannon's more sophisticated notion of channel capacity.
- ✓ Hartley argued that the maximum number of distinguishable pulse levels that can be transmitted and received reliably over a communications channel is limited by the dynamic range of the signal amplitude and the precision with which the receiver can distinguish amplitude levels.
- ✓ Specifically, if the amplitude of the transmitted signal is restricted to the range of [−A ... +A] volts, and the precision of the receiver is ±ΔV volts, then the maximum number of distinct pulses M is given by

  $M=1+A/\Delta V$

- ✓ By taking information per pulse in bit/pulse to be the base-2-logarithm of the number of distinct messages M that could be sent, Hartley constructed a measure of the line rate R as:

  $R=fplog2(M)$

Where, fp is the pulse rate, also known as the symbol rate, in symbols/second or baud.

Noisy channel coding theorem and capacity:

- ✓ Claude Shannon's development of information theory during World War II provided the next big step in understanding how much information could be reliably communicated through noisy channels.
- ✓ Building on Hartley's foundation, Shannon's noisy channel coding theorem (1948) describes the maximum possible efficiency of error-correcting methods versus levels of noise interference and data corruption.
- ✓ The proof of the theorem shows that a randomly constructed error-correcting code is essentially as good as the best possible code; the theorem is proved through the statistics of such random codes.

Shannon's theorem shows how to compute a channel capacity from a statistical description of a channel, and establishes that given a noisy channel with capacity C and information transmitted at a line rate R, then if

$$R<C$$

- ✓ There exists a coding technique which allows the probability of error at the receiver to be made arbitrarily small.
- ✓ This means that theoretically, it is possible to transmit information nearly without error up to nearly a limit of C bits per second.
- ✓ The converse is also important. If

$$R>C$$

- ✓ The probability of error at the receiver increases without bound as the rate is increased. So no useful information can be transmitted beyond the channel capacity.
- ✓ The theorem does not address the rare situation in which rate and capacity are equal.
- ✓ The Shannon–Hartley theorem establishes what that channel capacity is for a finite-bandwidth continuous-time channel subject to Gaussian noise.
- ✓ It connects Hartley's result with Shannon's channel capacity theorem in a form that is equivalent to specifying the M in Hartley's line rate formula in terms of a signal-to-noise ratio, but achieving reliability through error-correction coding rather than through reliably distinguishable pulse levels.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=3ekWsXeZ8TM

2. https://www.youtube.com/watch?v=7nrIkh0B3Xo

3. https://www.youtube.com/watch?v=ePNHPQYT4DY

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (102-104)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 38

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES **Date of Lecture:**

Topic of Lecture: Generation, Detection, Signal space diagram

Introduction: Digital amplitude modulation involves varying the amplitude of a carrier wave in discrete sections according to binary data. With digital frequency modulation, the frequency of a carrier or a baseband signal is varied in discrete sections according to binary data.

Prerequisite knowledge for Complete understanding and learning of Topic:
Generation of space, Detection, Signal space diagram

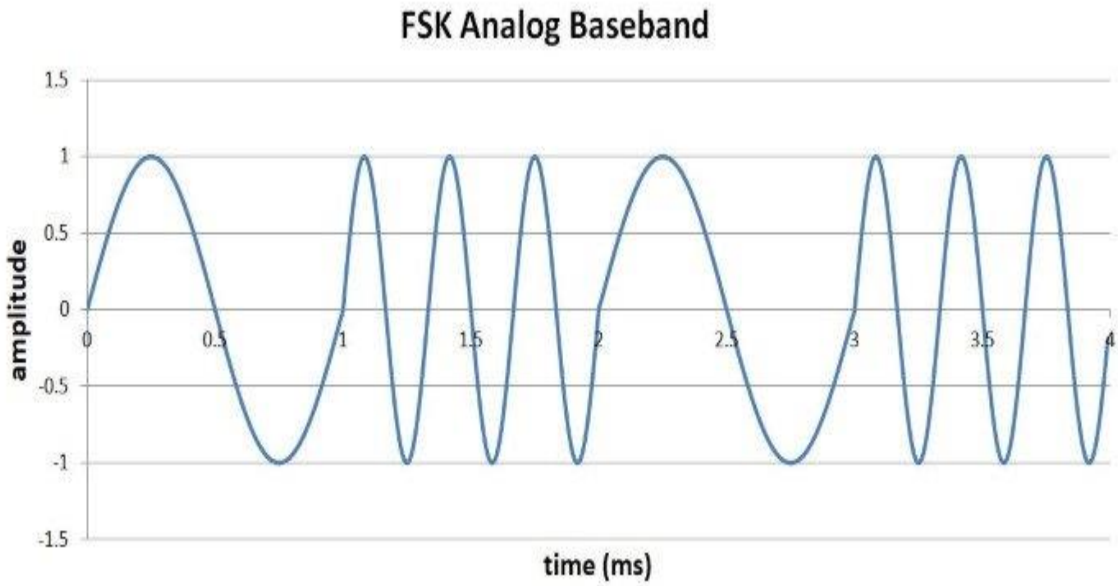
Generation of space, Detection, Signal space diagram:

Digital Frequency modulation:

- ✓ This type of modulation is called frequency shift keying (FSK). For our purposes it is not necessary to consider a mathematical expression of FSK; rather, we can simply specify that we will have frequency f_1 when the baseband data is logic 0 and frequency f_2 when the baseband data is logic 1.

Time domain:

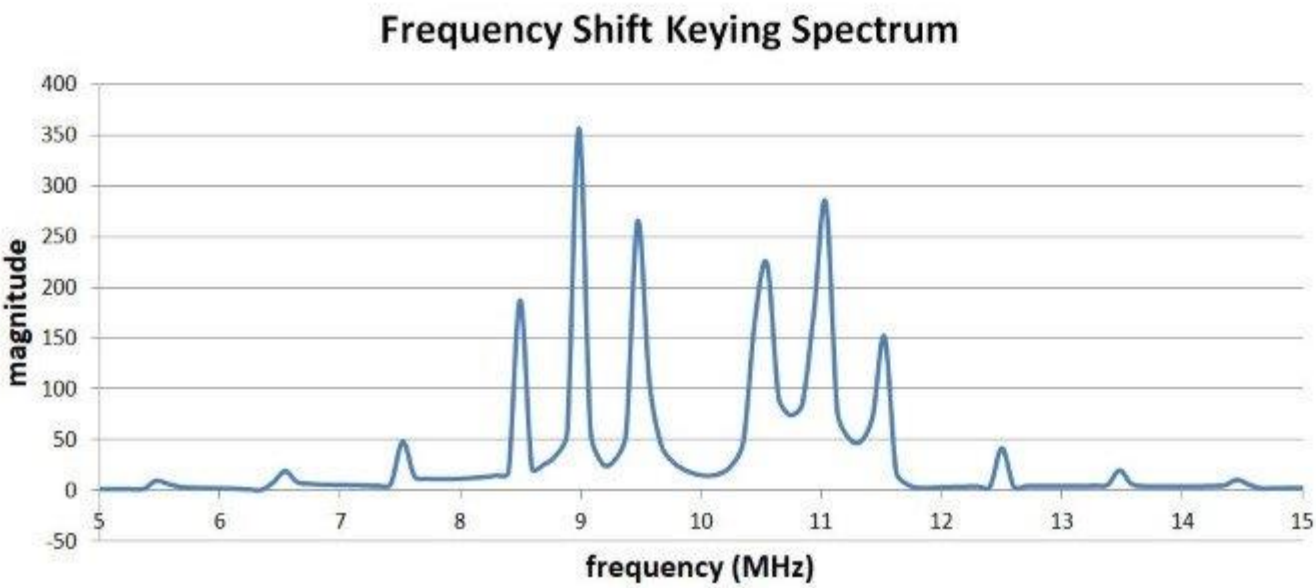
- ✓ One method of generating the ready-for-transmission FSK waveform is to first create an analog baseband signal that switches between f_1 and f_2 according to the digital data.
- ✓ Here is an example of an FSK baseband waveform with f_1 = 1 kHz and f_2 = 3 kHz.
- ✓ To ensure that a symbol is the same duration for logic 0 and logic 1, we use one 1 kHz cycle and three 3 kHz cycles.



- ✓ The baseband waveform is then shifted (using a mixer) up to the carrier frequency and transmitted.
- ✓ This approach is particularly handy in software-defined-radio systems: the analog baseband waveform is a low-frequency signal, and thus it can be generated mathematically then introduced into the analog realm by a DAC.
- ✓ Using a DAC to create the high-frequency transmitted signal would be much more difficult.
- ✓ A more conceptually straightforward way to implement FSK is to simply have two carrier signals with different frequencies (f_1 and f_2); one or the other is routed to the output depending on the logic level of the binary data.
- ✓ This results in a final transmitted waveform that switches abruptly between two frequencies, much like the baseband FSK waveform above except that the difference between the two frequencies is much smaller in relation to the average frequency.
- ✓ In other words, if you were looking at a time-domain plot, it would be difficult to visually differentiate the f_1 sections from the f_2 sections because the difference between f_1 and f_2 is only a tiny fraction of f_1 (or f_2).

Frequency domain:

- ✓ Let's look at the effects of FSK in the frequency domain. We'll use our same 10 MHz carrier frequency (or average carrier frequency in this case), and we'll use ±1 MHz as the deviation. (This is unrealistic, but convenient for our current purposes.)
- ✓ So the transmitted signal will be 9 MHz for logic 0 and 11 MHz for logic 1. Here is the spectrum:



- ✓ Note that there is no energy at the "carrier frequency." This is not surprising, considering that the modulated signal is never at 10 MHz.
- ✓ It is always at either 10 MHz minus 1 MHz or 10 MHz plus 1 MHz, and this is precisely where we see the two dominant spikes: 9 MHz and 11 MHz.
- ✓ But what about the other frequencies present in this spectrum? Well, FSK spectral analysis is not particularly straightforward.
- ✓ We know that there will be additional Fourier energy associated with the abrupt transitions between frequencies.
- ✓ It turns out that FSK results in a sinc-function type of spectrum for each frequency, i.e., one is centered on f_1 and the other is centered on f_2.
- ✓ These account for the additional frequency spikes seen on either side of the two dominant spikes.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=rrgon8Qne_E

2. https://www.youtube.com/watch?v=ogJB5fiQ9kM

3. https://www.youtube.com/watch?v=eJ5m0Sbr2qw

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (122-124)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

L - 39

LECTURE HANDOUTS

ECE | **II / III**

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 5 : **DIGITAL MODULATION SCHEMES** **Date of Lecture:**

Topic of Lecture: Calculation of bit error probability and Power spectra of ASK

Introduction: Calculation of bit error probability Serial-data speed is usually stated in terms of bit rate. However, another oft-quoted measure of speed is baud rate. Though the two aren't the same, similarities exist under some circumstances.

Prerequisite knowledge for Complete understanding and learning of Topic:
Calculation of bit error probability and Power spectra of ASK

Calculation of bit error probability and Power spectra of ASK:

Background:

- ✓ Most data communications over networks occurs via serial-data transmission.
- ✓ Data bits transmit one at a time over some communications channel, such as a cable or a wireless path. Figure 1 typifies the digital-bit pattern from a computer or some other digital circuit. This data signal is often called the baseband signal.
- ✓ The data switches between two voltage levels, such as +3 V for a binary 1 and +0.2 V for a binary 0. Other binary levels are also used. In the non-return-to-zero (NRZ) format (Fig. 1, again), the signal never goes to zero as like that of returnto-zero (RZ) formatted signals.

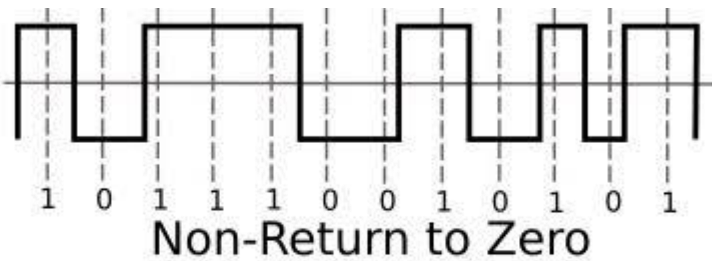


Fig. 1 Non-return to zero (NRZ) is the most common binary data format. Data rate is indicated in bits per second (bits/s).

Bit Rate:

- ✓ Bit Rate The speed of the data is expressed in bits per second (bits/s or bps).
- ✓ The data rate R is a function of the duration of the bit or bit time (TB) (Fig. 1, again): R = 1/TB Rate is also called channel capacity C. If the bit time is 10 ns, the data rate equals: R = 1/10 x 10–9 = 100 million bits/s This is usually expressed as 100 Mbits/s.
- ✓ Overhead Bit rate is typically seen in terms of the actual data rate.
- ✓ Yet for most serial transmissions, the data represents part of a more complex protocol frame or packet format, which includes bits representing source address, destination address, error detection and correction codes, and other information or control bits.

- ✓ In the protocol frame, the data is called the "payload." Non-data bits are known as the "overhead." At times, the overhead may be substantial—up to 20% to 50% depending on the total payload bits sent over the channel.

For example, an Ethernet frame can have as many as 1542 bytes or octets, depending on the data payload. Payload can range from 42 to 1500 octets. With a maximum payload, the overhead is only 42/1542 = 0.027, or about 2.7%.

- ✓ It would be even greater if the payload was anything smaller. This relationship is usually expressed as a percentage of the payload size to the maximum frame size, otherwise known as the protocol efficiency: Protocol efficiency = payload/frame size = 1500/1542 = 0.9727 or 97.3% Typically, the actual line rate is stepped up by a factor influenced by the overhead to achieve an actual target net data rate.
- ✓ In One Gigabit Ethernet, the actual line rate is 1.25 Gbits/s to achieve a net payload throughput of 1 Gbit/s. In a 10-Gbit/s Ethernet system, gross data rate equals 10.3125 Gbits/s to achieve a true data rate of 10 Gbits/s.
- ✓ The net data rate also is referred to as the throughput, or payload rate, of effective data rate.

Baud Rate:

- ✓ The term "baud" originates from the French engineer Emile Baudot, who invented the 5-bit teletype code. Baud rate refers to the number of signal or symbol changes that occur per second. A symbol is one of several voltage, frequency, or phase changes. NRZ binary has two symbols, one for each bit 0 or 1, that represent voltage levels.
- ✓ In this case, the baud or symbol rate is the same as the bit rate. However, it's possible to have more than two symbols per transmission interval, whereby each symbol represents multiple bits. With more than two symbols, data is transmitted using modulation techniques.
- ✓ When the transmission medium can't handle the baseband data, modulation enters the picture. Of course, this is true of wireless. Baseband binary signals can't be transmitted directly; rather, the data is modulated on to a radio carrier for transmission. Some cable connections even use modulation to increase the data rate, which is referred to as "broadband transmission."
- ✓ By using multiple symbols, multiple bits can be transmitted per symbol. For example, if the symbol rate is 4800 baud and each symbol represents two bits, that translates into an overall bit rate of 9600 bits/s. Normally the number of symbols is some power of two.
- ✓ If N is the number of bits per symbol, then the number of required symbols is S = 2N. Thus, the gross bit rate is:

$$R = \text{baud rate} \times \log2S = \text{baud rate} \times 3.32 \log10S$$

- ✓ If the baud rate is 4800 and there are two bits per symbol, the number of symbols is 22 = 4. The bit rate is:

$$R = 4800 \times 3.32 \log(4) = 4800 \times 2 = 9600 \text{ bits/s}$$

- ✓ If there's only one bit per symbol, as is the case with binary NRZ, the bit and baud rates remain the same.

Multilevel Modulation:

- ✓ Many different modulation schemes can implement high bit rates. For example, frequency-shift keying (FSK) typically uses two different frequencies in each symbol interval to represent binary 0 and 1. Therefore, the bit rate is equal to the baud rate.
- ✓ However, if each symbol represents two bits, it requires the four frequencies (4FSK). In 4FSK, the bit rate is two times the baud rate. Phase-shift keying (PSK) is another popular example. When employing binary PSK, each symbol represents a 0 or 1.
- ✓ A binary 0 equals 0°, while a binary 1 is 180°. With one bit per symbol, the baud and bit rates are the same.
- ✓ However, multiple bits per symbol can be easily implemented.

Video Content / Details of website for further learning (if any): 1. https://www.youtube.com/watch?v=K4_rKWFpqXg 2. https://www.youtube.com/watch?v=ucrZlde8vtk 3. https://www.youtube.com/watch?v=YDL9N1dOdVs
Important Books/Journals for further learning including the page nos.: 1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (134-135)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 40

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES **Date of Lecture:**

Topic of Lecture: FSK

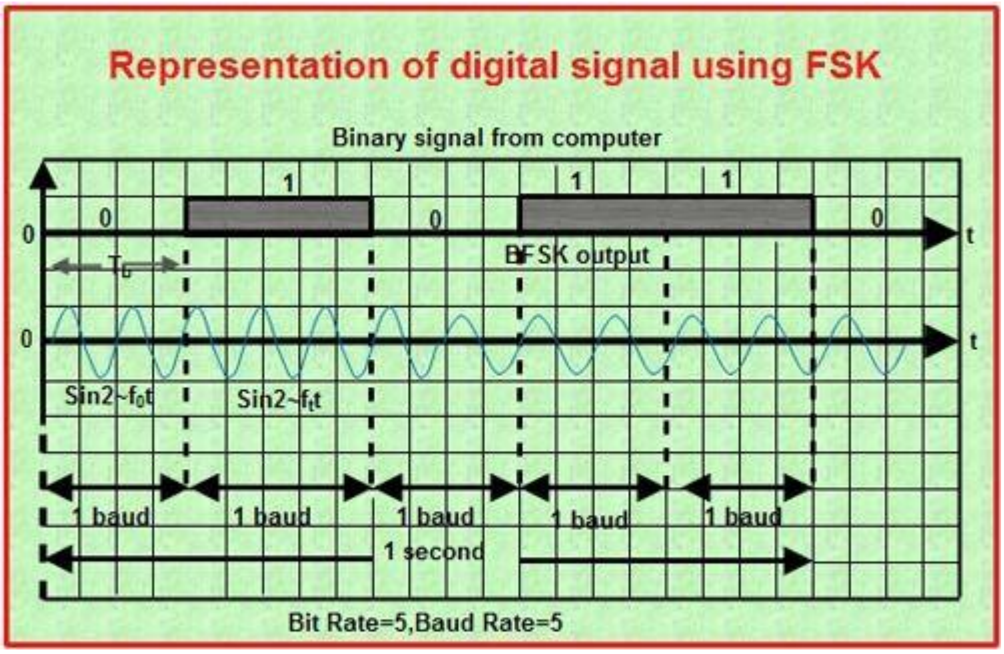
Introduction: A rectangular-pulse polar baseband signal is used to modulate an RF carrier in FSK. If the baseband signal has a data rate of 200 kbit/sec and the two RF frequencies are 150 kHz apart, determine the bandwidth. 2f + 2B = 150 kHz + 200 kHz = 350 kHz.

Prerequisite knowledge for Complete understanding and learning of Topic:
FSK, Bandwidth consideration

FSK – FSK receiver:

FSK:

- ✓ In "frequency shift keying (FSK)", the frequency of a sinusoidal carrier is shifted between two discrete values. One of these frequencies (f_1) represents a binary "1" and the other value (f_0) represents a binary "0".
- ✓ The representation of digital data using FSK is as shown in Fig. Note that there is no change in the amplitude of the carrier.



Bandwidth for FSK in terms of baud rate

- ✓ For FSK also bit rate is equal to baud rate.
- ✓ We can imagine the FSK spectrum to be a combination of two ASK spectrums centered at frequencies f_H and f_L.

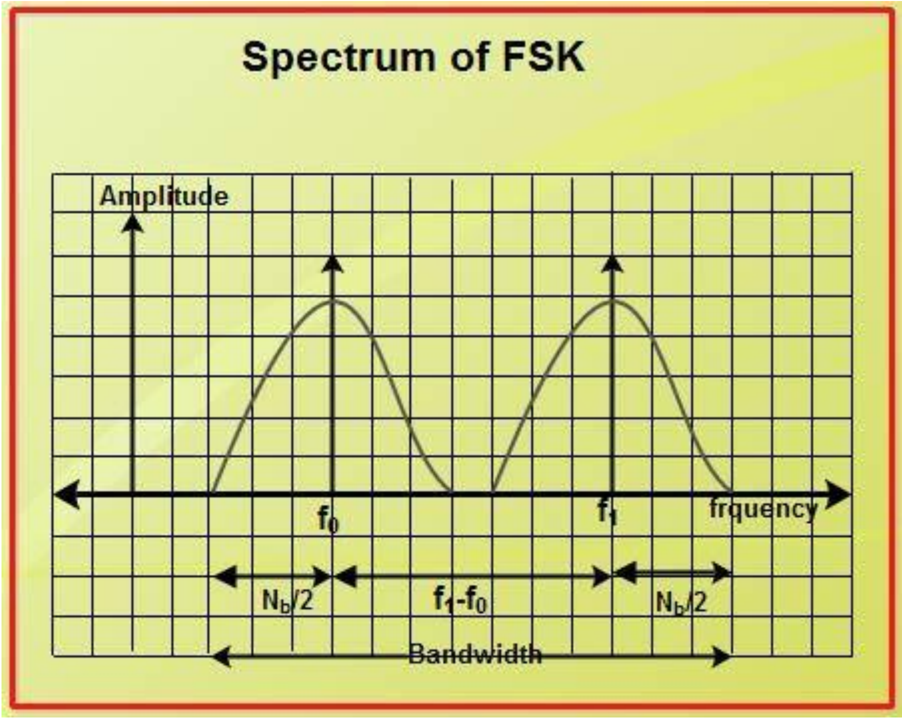


Fig. Bandwidth

$$\mathbf{BW = \underline{N_b} + (f1 - f_0) + \underline{N_b} = (f1 - f_0) + N_b}$$

Where N_b = baud rate = bit rate = f_b

- ✓ Minimum bandwidth will correspond to the situation in which $(f1 - f_0) = N_b$

$$\mathbf{BW\ (min) = N_b + N_b = 2\ N_b}$$

Advantages of FSK

- ✓ FSK is relatively easy to implement.
- ✓ It has better noise immunity than ASK. Therefore the probability of error free reception of data is high.

Disadvantages of FSK

- ✓ The major disadvantage is its high bandwidth requirement as' discussed earlier.
- ✓ Therefore FSK is extensively used in low speed modems having bit rates below 1200 bits/sec.
- ✓ The FSK is not preferred for the high speed modems because with increase in speed, the bit rate increases.
- ✓ This increases the channel bandwidth required to transmit the FSK signal.
- ✓ As the telephone lines have a very low bandwidth, it is not possible to satisfy the bandwidth requirement of FSK at higher speed. Therefore FSK is preferred only for the low speed modems.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=k7UvtvW8uck
2. https://www.youtube.com/watch?v=rrgon8Qne_E
3. https://www.youtube.com/watch?v=zpSB3DmGqko

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (134-136)

Course Faculty

Verified by HOD

Course Name with Code : **19GES23-ANALOG AND DIGITAL COMMUNICATION**

Course Teacher : **Dr. J.KIRUBAKARAN, ASP/ECE**

Unit 5 : **DIGITAL MODULATION SCHEMES** **Date of Lecture:**

Topic of Lecture: PSK

Introduction: Binary Phase-shift keying (BPSK) is a digital modulation scheme that conveys data by changing, or modulating, two different phase s of a reference signal (the carrier wave). The constellation points chosen are usually positioned with uniform angular spacing around a circle.

Prerequisite knowledge for Complete understanding and learning of Topic:
BPSK modulator, Block diagram, BPSK demodulator

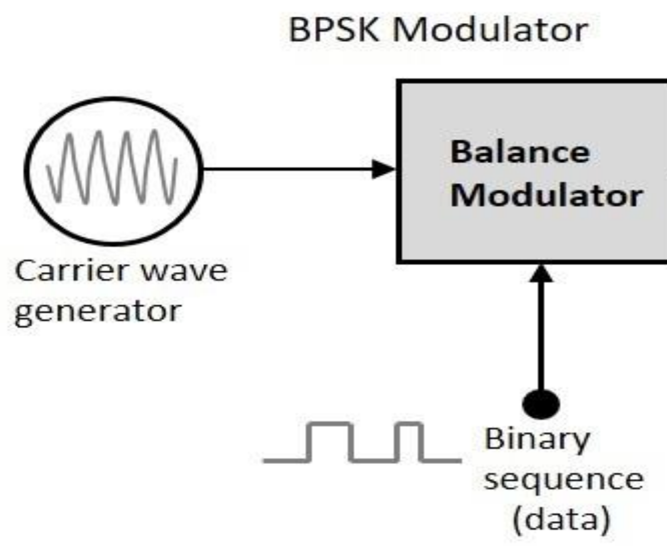
Phase shift keying:

BPSK:

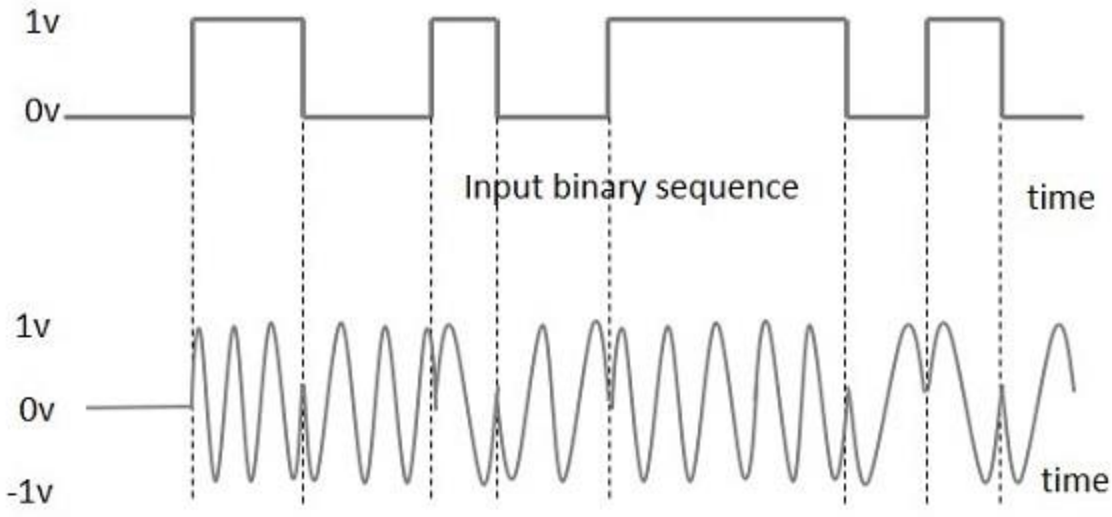
- ✓ BPSK (also sometimes called PRK, phase reversal keying, or 2PSK) is the simplest form of phase shift keying (PSK).
- ✓ It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180°.
- ✓ Therefore, it handles the highest noise level or distortion before the demodulator reaches an incorrect decision. That makes it the most robust of all the PSKs.
- ✓ It is, however, only able to modulate at 1 bit/symbol (as seen in the figure) and so is unsuitable for high data-rate applications.
- ✓ In the presence of an arbitrary phase-shift introduced by the communications channel, the demodulator (see, e.g. Costas loop) is unable to tell which constellation point is which. As a result, the data is often differentially encoded prior to modulation.
- ✓ BPSK is functionally equivalent to 2-QAM modulation.

Block diagram:

- ✓ The block diagram of Binary Phase Shift Keying consists of the balance modulator which has the carrier sine wave as one input and the binary sequence as the other input.
- ✓ Following is the diagrammatic representation.



- ✓ The modulation of BPSK is done using a balance modulator, which multiplies the two signals applied at the input. For a zero binary input, the phase will be **0°** and for a high input, the phase reversal is of **180°**.
- ✓ Following is the diagrammatic representation of BPSK Modulated output wave along with its given input.

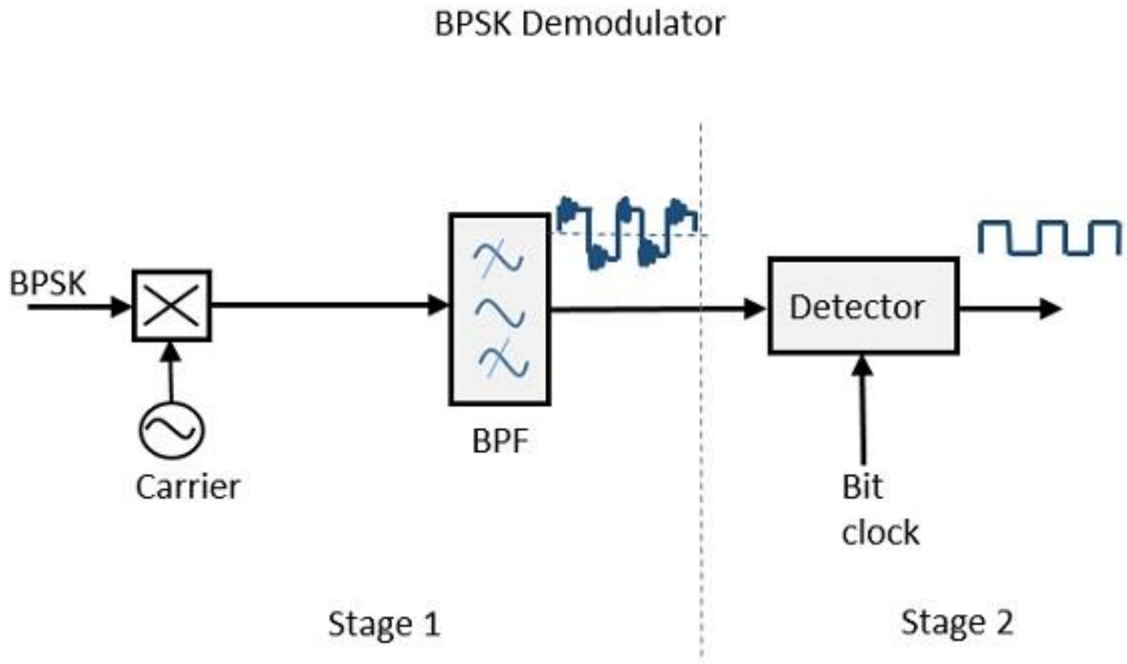


BPSK Modulated output wave

- ✓ The output sine wave of the modulator will be the direct input carrier or the inverted 180°phaseshifted180°phaseshifted input carrier, which is a function of the data signal.

BPSK Demodulator:

- ✓ The block diagram of BPSK demodulator consists of a mixer with local oscillator circuit, a bandpass filter, a two-input detector circuit.
- ✓ The diagram is as follows.



- ✓ By recovering the band-limited message signal, with the help of the mixer circuit and the band pass filter, the first stage of demodulation gets completed. The base band signal which is band limited is obtained and this signal is used to regenerate the binary message bit stream.
- ✓ In the next stage of demodulation, the bit clock rate is needed at the detector circuit to produce the original binary message signal. If the bit rate is a sub-multiple of the carrier frequency, then the bit clock regeneration is simplified. To make the circuit easily understandable, a decision-making circuit may also be inserted at the 2^{nd} stage of detection.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=c1z9LiRpxLk
2. https://www.youtube.com/watch?v=qw8w8mfhVlc
3. https://www.youtube.com/watch?v=Sbo5JLBv_3k

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (133-135)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 42

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES **Date of Lecture:**

Topic of Lecture: DPSK, DEPSK

Introduction: Differential Phase Shift Keying (DPSK) is a form of Phase Shift Keying in which two bits are modulated at once, selecting one of four possible carrier phase shifts (0, 90, 180, or 270 degrees). DPSK allows the signal to carry twice as much information as ordinary PSK using the same bandwidth.

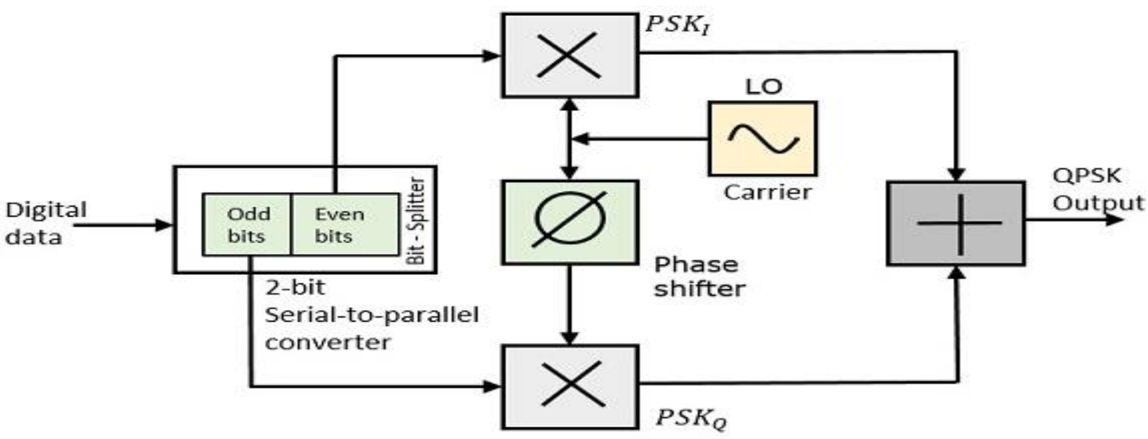
Prerequisite knowledge for Complete understanding and learning of Topic:
DPSK Modulator, Demodulator and 8 PSK

DPSK,:

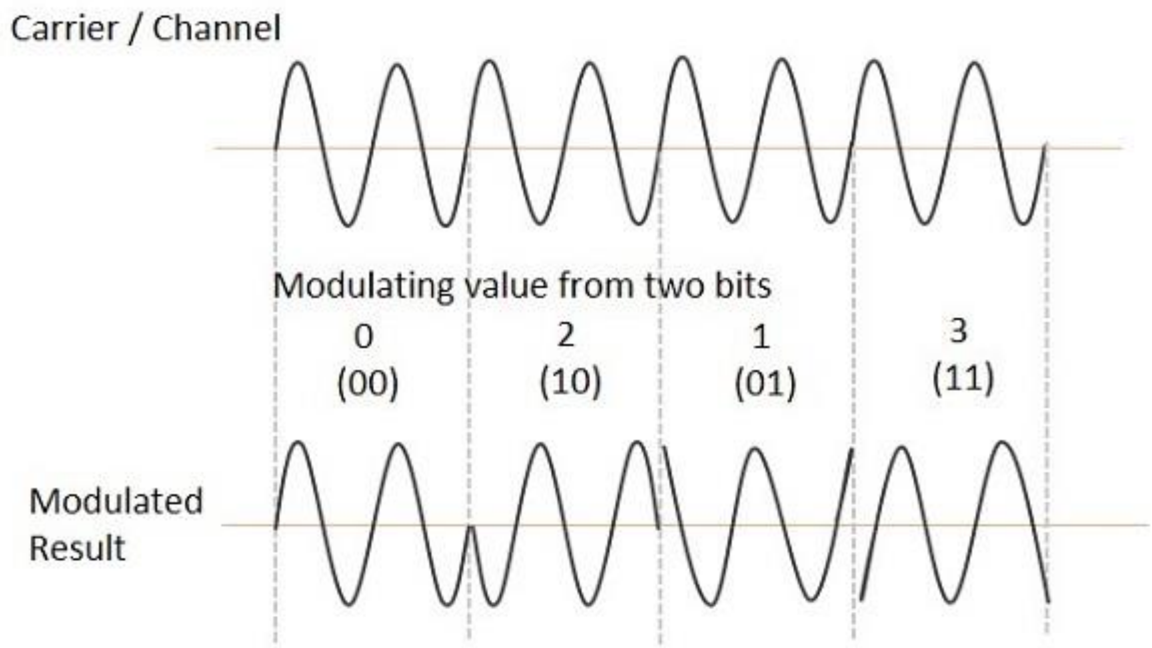
- ✓ The Quadrature Phase Shift Keying QPSKQPSK is a variation of BPSK, and it is also a Double Side Band Suppressed Carrier DSBSCDSBSC modulation scheme, which sends two bits of digital information at a time, called as bigits.
- ✓ Instead of the conversion of digital bits into a series of digital stream, it converts them into bit pairs. This decreases the data bit rate to half, which allows space for the other users.

DPSK Modulator:

- ✓ The DPSK Modulator uses a bit-splitter, two multipliers with local oscillator, a 2-bit serial to parallel converter, and a summer circuit. Following is the block diagram for the same.

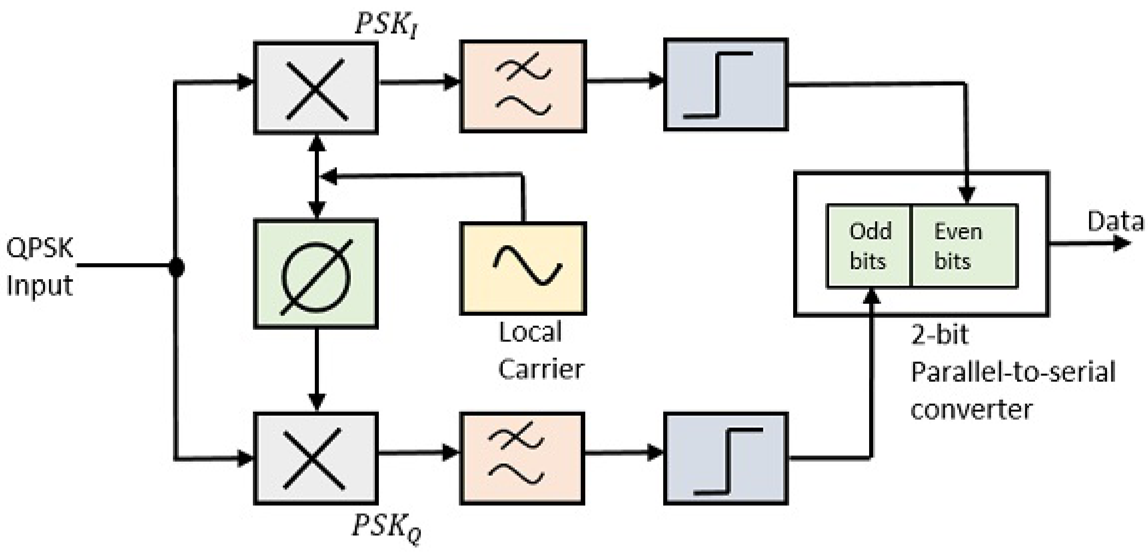


- ✓ At the modulator's input, the message signal's even bits (i.e., 2nd bit, 4th bit, 6th bit, etc.) and odd bits (i.e., 1st bit, 3rd bit, 5th bit, etc.) are separated by the bits splitter and are multiplied with the same carrier to generate odd BPSK (called as PSK_I) and even BPSK (called as PSK_Q). The PSK_Q signal is anyhow phase shifted by 90° before being modulated.
- ✓ The QPSK waveform for two-bits input is as follows, which shows the modulated result for different instances of binary inputs.



DPSK Demodulator:

- ✓ The QPSK Demodulator uses two product demodulator circuits with local oscillator, two band pass filters, two integrator circuits, and a 2-bit parallel to serial converter.
- ✓ Following is the diagram for the same.



- ✓ The two product detectors at the input of demodulator simultaneously demodulate the two BPSK signals. The pair of bits are recovered here from the original data. These signals after processing, are passed to the parallel to serial converter.

DPSK:

- ✓ It describes **8-PSK modulation** basics or multilevel PSK modulation which is a type of digital modulation based on carrier phase change.
- ✓ In Phase Shift keying modulation or PSK modulation phase of carrier is changed according to the digital data. It is digital modulation technique. It is used in broadcast video systems, aircraft and satellite systems.
- ✓ We can achieve bandwidth efficiency when we represent each signal element to map more than one bit.In BPSK modulation digital data of 1 and 0 is represented by 180 degree phase change. In QPSK by phase shift of 90 degree, here 2 bits are mapped on each signal.
- ✓ In Multilevel PSK more than 2 bits are mapped using different phase angles. In 8-PSK eight different phase angles are used to represent bits, here 3 bits. Figure below shows constellation of 8-PSK signal.

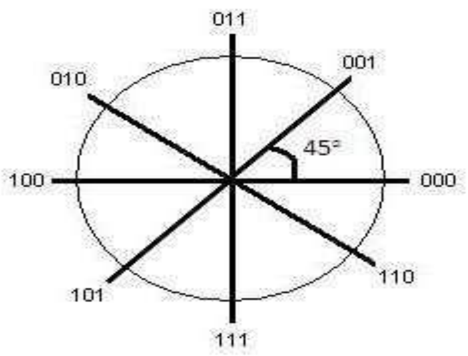


Fig. 8 PSK module

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=GT6_h7yhST4
2. https://www.youtube.com/watch?v=ij760lCUtfw
3. https://www.youtube.com/watch?v=O3-L9tvTvGg /

Important Books/Journals for further learning including the page nos.:

1. “Electronic Communication Systems Fundamentals through Advanced”, Wayne Tomasi, Pearson Education, 2008. (123-125)

Course Faculty

Verified by HOD

MUTHAYAMMAL ENGINEERING COLLEGE
(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

LECTURE HANDOUTS

L - 43

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES **Date of Lecture:**

Topic of Lecture: QPSK, MSK and GMSK

Introduction: Quadrature Phase Shift Keying (QPSK) is a form of Phase Shift Keying in which two bits are modulated at once, selecting one of four possible carrier phase shifts (0, 90, 180, or 270 degrees). QPSK allows the signal to carry twice as much information as ordinary PSK using the same bandwidth.

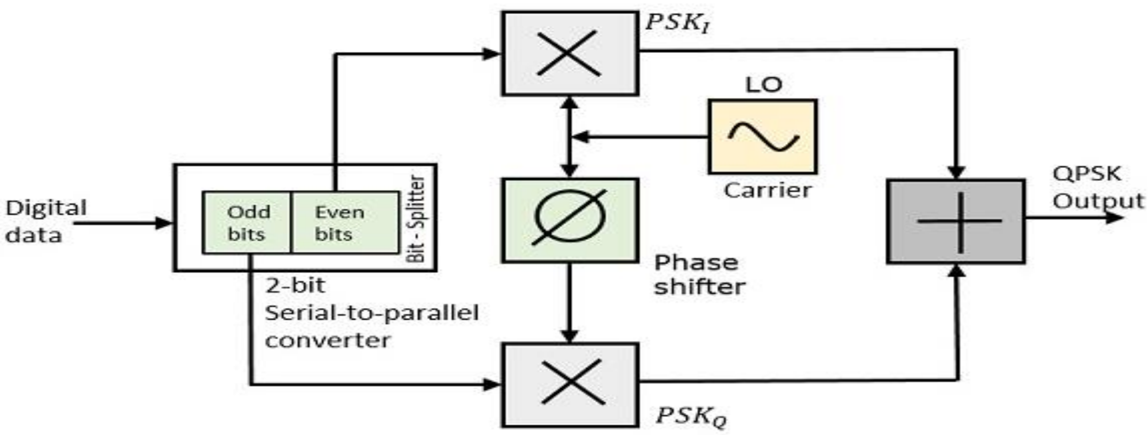
Prerequisite knowledge for Complete understanding and learning of Topic:
QPSK Modulator, Demodulator and 8 PSK

QPSK:

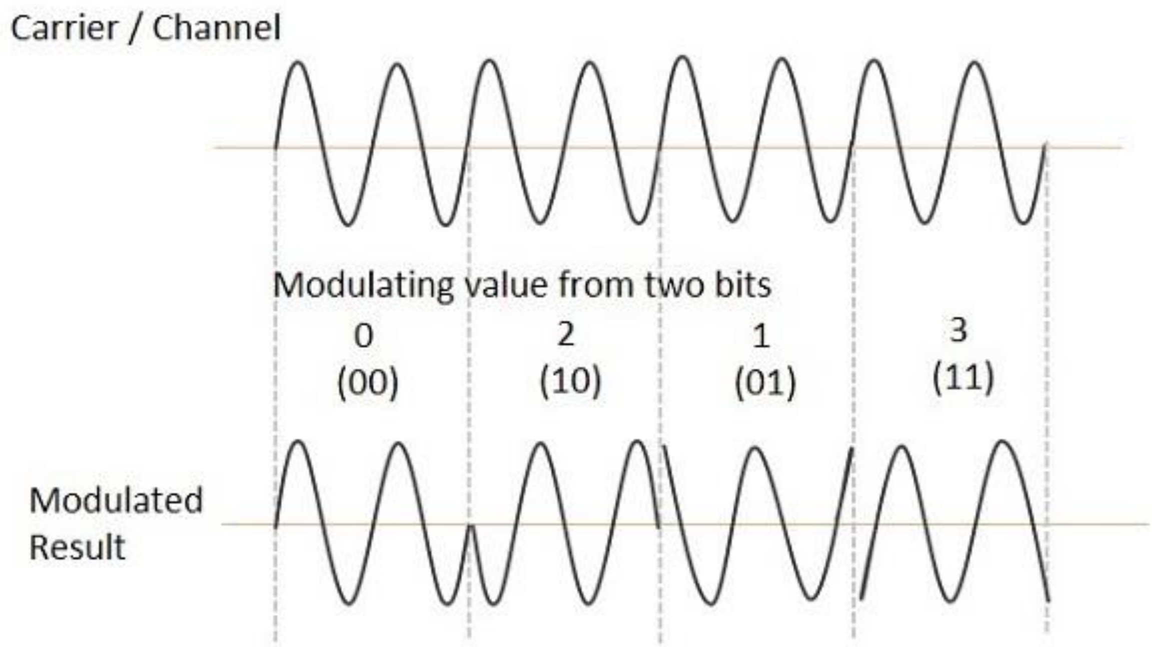
- ✓ The Quadrature Phase Shift Keying QPSKQPSK is a variation of BPSK, and it is also a Double Side Band Suppressed Carrier DSBSCDSBSC modulation scheme, which sends two bits of digital information at a time, called as bigits.
- ✓ Instead of the conversion of digital bits into a series of digital stream, it converts them into bit pairs. This decreases the data bit rate to half, which allows space for the other users.

QPSK Modulator:

- ✓ The QPSK Modulator uses a bit-splitter, two multipliers with local oscillator, a 2-bit serial to parallel converter, and a summer circuit. Following is the block diagram for the same.

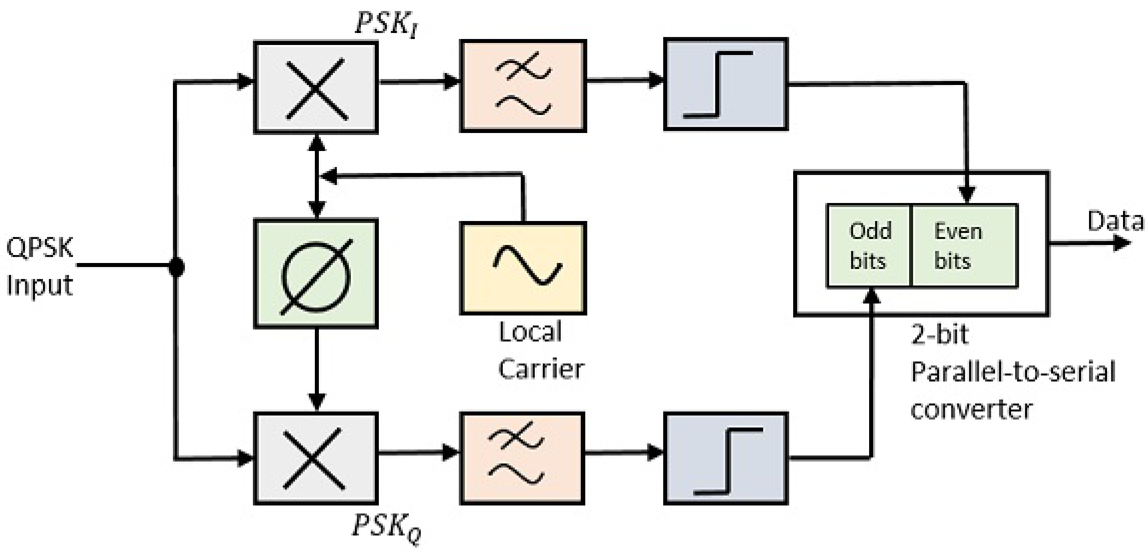


- ✓ At the modulator's input, the message signal's even bits (i.e., 2nd bit, 4th bit, 6th bit, etc.) and odd bits (i.e., 1st bit, 3rd bit, 5th bit, etc.) are separated by the bits splitter and are multiplied with the same carrier to generate odd BPSK (called as PSK_I) and even BPSK (called as PSK_Q). The PSK_Q signal is anyhow phase shifted by 90° before being modulated.
- ✓ The QPSK waveform for two-bits input is as follows, which shows the modulated result for different instances of binary inputs.



MSK Demodulator:

- ✓ The MSK Demodulator uses two product demodulator circuits with local oscillator, two band pass filters, two integrator circuits, and a 2-bit parallel to serial converter.
- ✓ Following is the diagram for the same.



- ✓ The two product detectors at the input of demodulator simultaneously demodulate the two BPSK signals. The pair of bits are recovered here from the original data. These signals after processing, are passed to the parallel to serial converter.

MSK:

- ✓ It describes **8-PSK modulation** basics or multilevel PSK modulation which is a type of digital modulation based on carrier phase change.
- ✓ In Phase Shift keying modulation or PSK modulation phase of carrier is changed according to the digital data. It is digital modulation technique. It is used in broadcast video systems, aircraft and satellite systems.
- ✓ We can achieve bandwidth efficiency when we represent each signal element to map more than one bit.In BPSK modulation digital data of 1 and 0 is represented by 180 degree phase change. In QPSK by phase shift of 90 degree, here 2 bits are mapped on each signal.
- ✓ In Multilevel PSK more than 2 bits are mapped using different phase angles. In 8-PSK eight different phase angles are used to represent bits, here 3 bits. Figure below shows constellation of 8-PSK signal.

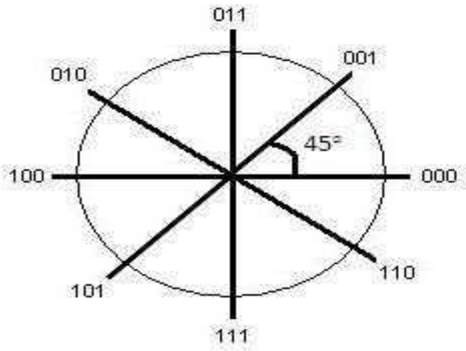


Fig. 8 PSK module

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=GT6_h7yhST4

2. https://www.youtube.com/watch?v=ij760lCUtfw

3. https://www.youtube.com/watch?v=O3-L9tvTvGg /

Important Books/Journals for further learning including the page nos.:

1. “Electronic Communication Systems Fundamentals through Advanced”, Wayne Tomasi, Pearson Education, 2008. (123-125)

Course Faculty

Verified by HOD

LECTURE HANDOUTS

L - 44

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES Date of Lecture:

Topic of Lecture: Similarity of BFSK and BPSK

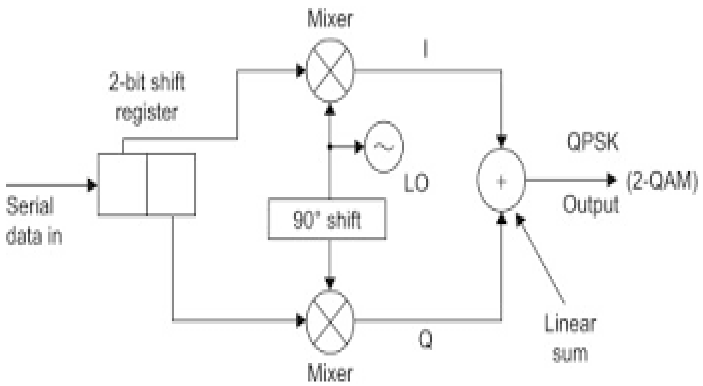
Introduction: BFSK(quadrature amplitude modulation) is a method of combining two amplitude-modulated (AM) signals into a single channel, thereby doubling the effective bandwidth. **BFSK** is used with pulse amplitude modulation (FSK in digital systems, especially in wireless applications.

Prerequisite knowledge for Complete understanding and learning of Topic:
FSK, interference and noise and BFSK

Similarity of BFSK and BPSK:

BFSK:

- ✓ Quadrature amplitude modulation (QAM) is the name of a family of digital modulation methods and a related family of analog modulation methods widely used in modern telecommunications to transmit information.
- ✓ It conveys two analog message signals, or two digital bit streams, by changing (modulating) the amplitudes of two carrier waves, using the amplitude-shift keying (ASK) digital modulation scheme or amplitude modulation (AM) analog modulation scheme.
- ✓ The two carrier waves of the same frequency are out of phase with each other by 90°, a condition known as orthogonality or quadrature.
- ✓ The transmitted signal is created by adding the two carrier waves together. At the receiver, the two waves can be coherently separated (demodulated) because of their orthogonality property.
- ✓ Another key property is that the modulations are low-frequency/low-bandwidth waveforms compared to the carrier frequency, which is known as the narrowband assumption.



BFSK Demodulation:

- ✓ In a QAM signal, one carrier lags the other by 90°, and its amplitude modulation is customarily referred to as the in-phase component, denoted by $I(t)$.
- ✓ The other modulating function is the quadrature component, $Q(t)$. So the composite waveform is mathematically modeled as:

$$Ss(t)=2\pi(\sin fct)I(t)$$

$$Sc(t)= 2\pi(\cos fct)I(t)$$

Where, f_c is the carrier frequency.

- ✓ At the receiver, a coherent demodulator multiplies the received signal separately with both a cosine and sine signal to produce the received estimates of $I(t)$ and $Q(t)$.

Analog QAM is used in:

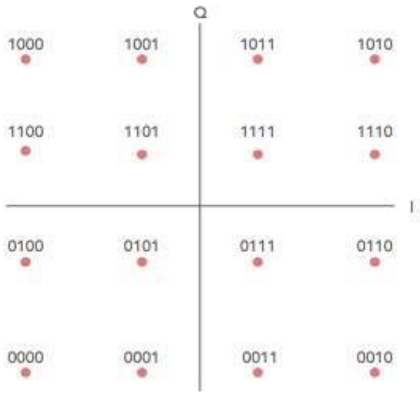
- ✓ NTSC and PAL analog color television systems, where the I- and Q-signals carry the components of chroma (colour) information. The QAM carrier phase is recovered from a special colorburst transmitted at the beginning of each scan line.
- ✓ C-QUAM ("Compatible QAM") is used in AM stereo radio to carry the stereo difference information.

Interference and noise:

- ✓ In moving to a higher order QAM constellation (higher data rate and mode) in hostile RF/microwave QAM application environments, such as in broadcasting or telecommunications, multipath interference typically increases.
- ✓ There is a spreading of the spots in the constellation, decreasing the separation between adjacent states, making it difficult for the receiver to decode the signal appropriately. In other words, there is reduced noise immunity.
- ✓ There are several test parameter measurements which help determine an optimal QAM mode for a specific operating environment. The following three are most significant:
  - ✓ Carrier/interference ratio
  - ✓ Carrier-to-noise ratio
  - ✓ Threshold-to-noise ratio

BFSK:

- ✓ AM, quadrature amplitude modulation provides some significant benefits for data transmission. As 16QAM transitions to 64QAM, 64QAM to 256 QAM and so forth, higher data rates can be achieved, but at the cost of the noise margin.
- ✓ Accordingly there is a balance to be made between the data rate and QAM modulation order, power and the acceptable bit error rate. Whilst further error correction can be introduced to mitigate any deterioration in link quality, this will also decrease the data throughput.
- ✓ As the QAM order increases, so the distance between the different points on the constellation diagram decreases and there is a higher possibility of data errors being introduced.
- ✓ To utilise the high order QAM formats, the link must have a very good E_b/N_o otherwise data errors will be present.
- ✓ When the E_b/N_o deteriorates, then other the power level must be increased, or the QAM order reduced if the bit error rate is to be preserved.



Format and application:

- ✓ QAM is in many radio communications and data delivery applications. However some specific variants of QAM are used in some specific applications and standards.
- ✓ There is a balance between data throughput and signal to noise ratio required. As the order of the QAM signal is increased, i.e. progressing from 16QAM to 64QAM, etc. the data throughput achievable under ideal conditions increases.
- ✓ However the downside is that a better signal to noise ratio is required to achieve this.

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=MAZHck0oWZA
2. https://www.youtube.com/watch?v=d7l5NbFfBiU
3. https://www.youtube.com/watch?v=rcr6epRzbp4

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (134-136)

Course Faculty

Verified by HOD

LECTURE HANDOUTS

L - 45

ECE

II / III

Course Name with Code : 19GES23-ANALOG AND DIGITAL COMMUNICATION

Course Teacher : Dr. J.KIRUBAKARAN, ASP/ECE

Unit 5 : DIGITAL MODULATION SCHEMES **Date of Lecture:**

Topic of Lecture: Comparison of Digital modulation systems using bit error probability

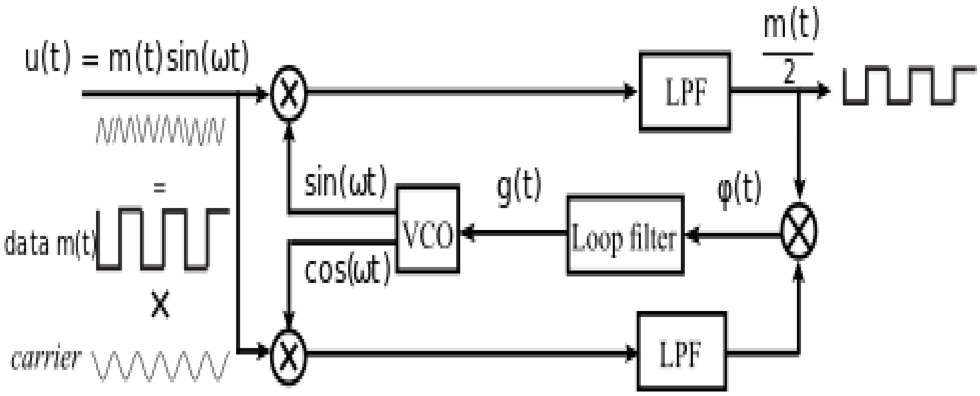
Introduction: Digital modulation systems using bit error probability a Costas loop is a phase-locked **loop** (PLL) based circuit which is used for carrier frequency recovery from suppressed-carrier modulation signals (e.g. double-sideband suppressed carrier signals) and phase modulation signals (e.g. BPSK, QPSK).

Prerequisite knowledge for Complete understanding and learning of Topic:
Classical implementation, Digital modulation systems using bit error probability.

Comparison of Digital modulation systems using bit error probability:

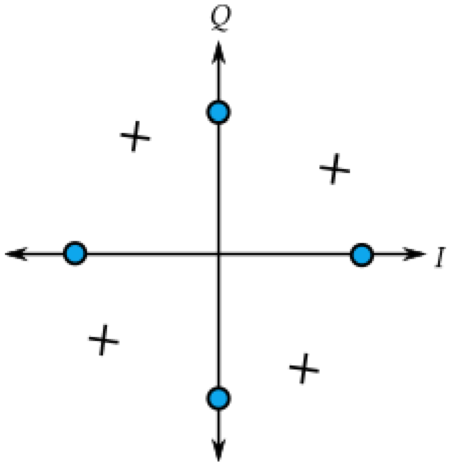
Digital modulation systems using bit error probability:

- ✓ In the classical implementation of a Costas loop, a local voltage-controlled oscillator (VCO) provides quadrature outputs, one to each of two phase detectors, *e.g.*, product detectors.
- ✓ The same phase of the input signal is also applied to both phase detectors and the output of each phase detector is passed through a low-pass filter.
- ✓ The outputs of these low-pass filters are inputs to another phase detector, the output of which passes through noise-reduction filter before being used to control the voltage-controlled oscillator.
- ✓ The overall loop response is controlled by the two individual low-pass filters that precede the third phase detector while the third low-pass filter serves a trivial role in terms of gain and phase margin.



Carrier recovery:

- A **carrier recovery** system is a circuit used to estimate and compensate for frequency and phase differences between a received signal's carrier wave and the receiver's local oscillator for the purpose of coherent demodulation.



- In the transmitter of a communications carrier system, a carrier wave is modulated by a baseband signal. At the receiver the baseband information is extracted from the incoming modulated waveform.
- In an ideal communications system, the carrier signal oscillators of the transmitter and receiver would be perfectly matched in frequency and phase thereby permitting perfect coherent demodulation of the modulated baseband signal.
- However, transmitters and receivers rarely share the same carrier oscillator. Communications receiver systems are usually independent of transmitting systems and contain their own oscillators with frequency and phase offsets and instabilities. Doppler shift may also contribute to frequency differences in mobile radio frequency communications systems.

Methods:

- For a quiet carrier or a signal containing a dominant carrier spectral line, carrier recovery can be accomplished with a simple band-pass filter at the carrier frequency or with a phase-locked loop, or both.
- However, many modulation schemes make this simple approach impractical because most signal power is devoted to modulation—where the information is present—and not to the carrier frequency. Reducing the carrier power results in greater transmitter efficiency. Different methods must be employed to recover the carrier in these conditions.

Squaring loop:

- In squaring loop modulated signal is squared, Band Pass Filtered and then multiplied with the carrier that is recovered using PLL to get back base band signal.
- Although the circuit is simple to implement, it explicits the presence of phase error. Figure 2 shows the block diagram of square loop BPSK demodulator. m(t) be the message signal transmitted.
- The BPSK modulated signal is given by A m(t) cos(ωc t + θc). - 1 The squared output is given by A2m 2 (t) cos2 (ωc t + θc). The BPF output is given by A0cos2(ωc t + θc).
- PLL output is given by A0cos2(ωc t + θe). Frequency divider output is given by A0cos(ωc t + θe) The recovered carrier output is given by cos(ωc t + θe) - 2 The recovered carrier has phase error of θe The output of LPF is the demodulated output and is given by A0A m(t)

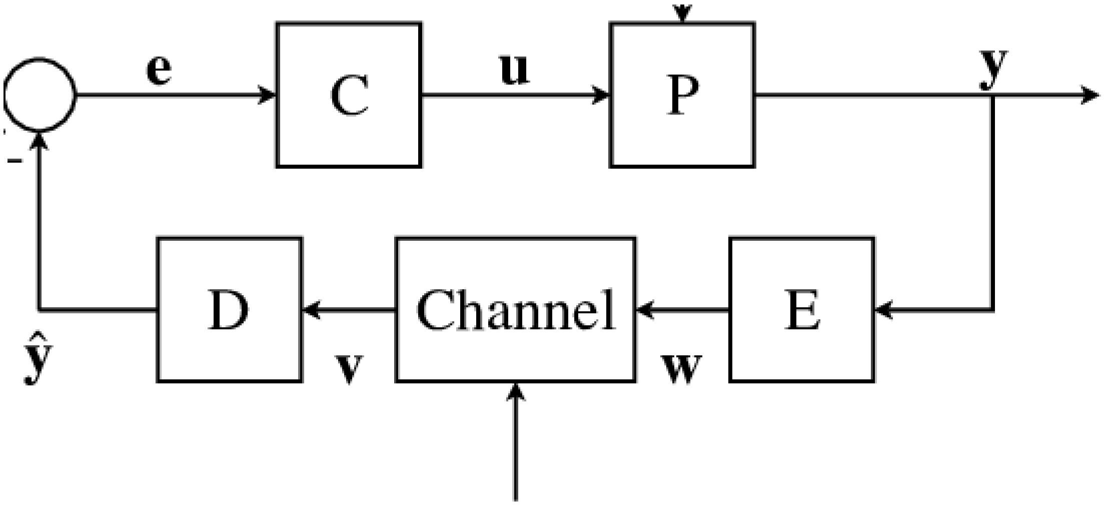


Fig. Closed Square loop

✓ Costas loop demodulator is an optimal method to attain data and carrier recovery for BPSK signal. It comprises of mixer, low pass filter, phase detector, loop filter and numerically controlled oscillator (NCO). The arm connected to in phase signal is called I channel and one that is connected to quadrature phase signal is called Q channel

Video Content / Details of website for further learning (if any):

1. https://www.youtube.com/watch?v=vJRI5FqAQFI

2. https://www.youtube.com/watch?v=nBUs5V8adj4

3. https://www.youtube.com/watch?v=-kxvrE89wU0

Important Books/Journals for further learning including the page nos.:

1. "Principles of Communication", H.Taub,D L Schilling G Saha, Pearson Education, 2008. (139-141)

Course Faculty

Verified by HOD